



USER'S MANUAL

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Telas

THE TELOS ALLIANCE®

TelosAlliance.com

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

Telos iQ6 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.

We support you...

By Phone/Fax

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

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



For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Register your product

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

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Telos Alliance Warranty

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

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

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

The Telos Alliance

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1 Introduction

1.1 The iQ6

The Telos iQ6 is the Telco gateway for Axia's iQ console. iQ6 combines a six line selector and two third-generation Telos hybrids with Digital Dynamic EQ. The two hybrids are designed to be used within a single studio for airing simultaneous callers and achieving optimal audio quality.

Control the iQ6 from the Axia iQ console equipped with the Telco expansion frame via a producer's PC running the VX Producer software. Take and screen calls directly via the producer's PC that uses the VX Producer software. All controllers function with a TCP connection and the iQ6 provides support for twelve connections.

The iQ6 comes with a POTS interface. This provides support for the connection of six lines for the two integrated hybrids.

Audio input and output is accomplished through Livewire for seamless integration to the Axia iQ console. There are no dedicated connectors on the back panel for audio. All control and audio passes through the Ethernet Adapter and the CAT-5e cable which connects to it.

1.2 Features

Telos introduced the first digital hybrid which increased the quality of caller audio. Over the years, each new Telos product continues to raise the bar. The iQ6 is no exception.

- Third generation advanced hybrid algorithms.
- New symmetrical wide-range AGC and noise gate by Omnia, with adjustable gain settings.
- Adjustable caller override improves performance and allows you to customize the degree to which the announcer ducks the caller audio.
- ◆ DDEQ-Digital dynamic EQ[™] keeps audio spectrally consistent from call to call.
- Separate Send level and Receive level meters for each hybrid.
- Worldwide disconnect signal detection (loop drop, dial tone, or reorder tone).
- ♦ Livewire Audio.
- ◆ Axia iQ console integration.
- Conference linking lets you set up high-quality conferencing between callers with no external equipment needed.
- ◆ Caller ID support on POTS lines.
- Support for up to twelve controllers.

VX control protocol for compatibility with Telos VSet, VX Producer, and third-party call control software applications.

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2 Installation and Configuration

2.1 Getting Started

The iQ6 mounts in a 1RU high space in a standard 19" rack. There are ventilation holes to the side and top. Do not restrict the ventilation holes. Find a suitable location that will provide air movement and access to the front and rear of the iQ6.

Connect the Telco circuits (section 2.2)

Connect power, and once the Telos iQ6 flash screen appears, configure the IP address of the iQ6. Press any of the three keys to access the configuration options.

Press the down arrow key [‡] to highlight System, press (OK)

Press
[↓] to highlight Networking, press (OK)

Press to highlight IP address, press (OK)

Use \mathcal{F} to delete and move the curser back. Once you press $\hat{\mathcal{U}}$, you begin to enter a value at curser position, starting with 0. Use \mathcal{F} and $\hat{\mathcal{U}}$ to select the desired value, then press (OK). Once the value is complete, press (OK) once more.

The remainder of the necessary configuration will occur with a PC's web interface. Use a direct connection between a PC and the iQ6 with the use of a cross over cable or a normal patch cable to your LAN (local area network). Connect the required cable to the Livewire port. The PC should have an IP address that is within the same subnet as the iQ6. (section 2.3)

Configure VSets, VX Producer software, and Axia iQ console to function with the iQ6. (section 2.4)

2.2 TELCO

The iQ6 has one slot for a POTS Telco interface module, supporting up to six lines. There are three subsections to the Telco installation. They are:

- ◆ 2.2.1 Plain old telephone service POTS
- ◆ 2.2.2 Integration with a PBX (private telephone system)

2.2.1 Plain old telephone service – POTS (Analog lines)

The iQ6 uses a plug-in interface module to attach to POTS (regular, analog loop start, telephone lines). The country setting is accessed from the front display. The default country is USA.

From the Status screen, press any of the three keys to access the configuration options.

Press the down arrow key $\sqrt[n]{}$ to highlight Telecom, press (OK)

Press ³ to highlight POTS, press (OK)

Press ¹/₇ to highlight Country, press (OK)

Use $\sqrt[3]{}$ and \hat{U} to select your country, press (OK). If you do not see your country listed you should use CTR21 or USA.

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The module has six modular jacks. The type of jack will be a six position four pin RJ-11 style. The connection is to the center two pins of this jack. Note: Unlike our older products, the iQ6 does not have the loop through provision for POTS lines. The POTS lines used with the iQ6 should be dedicated. The yellow and black leads (pins 2 and 5) of the modular connectors should be disconnected and insulated.

IMPORTANT

When we say POTS lines, we mean just that – plain old analog loop start telephone lines. A rule of thumb – if a line works with an analog modem or fax machine, it will work with the iQ6.

Analog Ground Start lines can be used for incoming calls only. However you will need to set the "Loop Check" option in the Telco menu to "No" for these lines to be recognized properly.

Be careful not to connect the iQ6 to PBX ports intended for proprietary telephones. These sometimes have voltages that could damage the iQ6.

2.2.2 Integration with a PBX

The iQ6 will usually be used independently of any other phone system in your facility. But it is possible to have one or more of the lines come from another phone system.

Consider having at least one line directly from the Telco to keep your iQ6 up and running even if the PBX or its Telco connection goes down.

Using POTS through PBX

Most PBX or VoIP telephone systems can provide analog ports for analog equipment such as modems and fax machines. You will need such ports to connect a POTS iQ6 to a PBX.

A few years ago, we advised that hybrid performance was best when equipment was connected directly to the lines from the Telco. Today, this is not always true. If the PBX is connected to the Telco using digital lines (such as T1, E1, PRI, S2M, etc), performance through a digital PBX will generally be better than direct connections to copper analog circuits. This is particularly so if you are located a significant distance from the Telco central office.

The biggest variable when connecting to a PBX analog port is the quality of the analog port. We have found that many PBX manufacturers offer more than one analog port option. The better port can usually be distinguished by its higher cost, and the fact that it may offer 48 volts rather than 24 volts. The iQ6 will work with either voltage; however experience has shown that the 48-volt version usually has better quality transformers and other components, resulting in better hybrid performance.

The other thing to investigate is whether the PBX offers 'Loop Current Wink' or 'CPC' (Calling Party Control) on the analog ports. Without this supervision, the iQ6 will not detect that a caller on hold has hung up, and your talent will get a recording "please hang up and dial again..." or dial tone when attempting to air such calls.

2.3 Basic Web Interface Configuration

Some basic settings need to be entered into the web interface before you can really do anything with the iQ6. With a PC that is connected directly to the iQ6 with a cross over cable, or connected through a LAN and with the correct network settings, open a web browser application and in the address field, type the IP address which you have assigned to the iQ6.

Telos iQ6 Talk Show System

1	
<u>Status</u>	Current state of the system, network interfaces, controller connections
<u>Telco</u>	Country selection, Hookflash duration, Auto drop enable and threshold, Loop current checking, Caller ID configuration
Audio	Input/Output levels and Meter ballistics Hybrid: Caller ducking, EQ, AGC, Gate, Mix Minus Routing, Feedback Reduction
Show Setup	Configure show behavior, line assignments, line sharing, phone book
Livewire Configuration	Configure Livewire Sources and Destinations: Names, Channel numbers, Modes
Livewire GPIO	Configure Livewire general-purpose I/O: names, channel numbers, format
<u>Livewire QOS</u>	Livewire clock mastership settings 802.1p priority and VLAN tagging, DSCP Class of Service
<u>System</u>	IP network address, Software update and maintenance
Backup	Backup and restore system settings
Troubleshooting	Syslog messages, State dump
Help	Documentation for Telco, Audio, and Show Setup pages.

The Home screen is shown above. First select the Livewire Configuration link. *Username* is user and the *password* should be left blank.

Assign the desired Livewire Channel number in the Channel field so that the iQ6 will produce audio and send it to the appropriate unique channel. You can also assign the Livewire channel for the Program On Hold (POH) source, if you have one. Once the correct values are in place, press the Apply button and continue to the Telco configuration.

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From the Telco page, you have the option to configure the POTS interface.

POTS Interface

POTS Interface Settings ?	
POTS Country USA 💌	
CallerID alert signal Default 💽	
CallerID Type Default 💌	
Show ringing before CallerID: No 💌	
Hookflash duration: 520ms 💌	
CPC loop drop threshold (ms) 60	
Loop current checking: Yes 💌	
Enable auto call drop: Both loop drop and CPTD 💌	

Verify the correct country setting is selected and any of the other desired settings. For reference on the other settings, refer to the detailed configuration section.

2.4 Client Configuration

There are three different clients which can connect to the iQ6. The client limit connection is twelve.

- ♦ Axia iQ surface (2.4.1)
- ♦ VSet (2.4.2)
- ♦ VX Producer (2.4.3)

2.4.1 Axia iQ Surface client

For details on the Axia iQ console, please refer to the appropriate documentation for the product. This section is here to assist with interfacing iQ6 to your iQ console.

Connect to the iQ console's web interface and select a Show Profile. The Show Profile page has Phone Connection settings to be filled in.

- Phone Server IP: Enter in the user name, password, and IP address of the iQ6 in the following format:
- User: password@ip.address (telos@192.168.2.106)
- ◆ Studio Name: Enter "Hybrid 1&2"
- ♦ Show Name: This is used for TelosVX systems and may be left blank

Save your Show Profile and reload it using the Profile key on the console. When the show is loaded, the iQ will log into the iQ6 as a client.

2.4.2 VSet client

For details on the VSet, please refer to the appropriate documentation for the product. To set up your VSet for use with iQ6, follow these steps:

- Press the Setup softkey
- ◆ Select the VX Engine IP soft key
- ♦ Enter in User, Password, and IP address
- ♦ The VSet will log into the iQ6 as a client

2.4.3 VX Producer client

For details on the VX Producer, please refer to the appropriate documentation for the product.

If VX Producer is not connected to a server, options will be available at the top page that need to be filled in.

- ◆ *VX Engine IP*: enter the IP address of the iQ6.
- ♦ Port: defaults to 20518 and should be maintained.
- ♦ Username: can be left blank
- ◆ *Password:* can be left blank

VX Producer will log into the iQ6 as a client and show the active studio. There is only one studio associated with the iQ6. Select the "Hybrid 1&2" Studio.

VX Producer is now connected to the iQ6

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3 User Operation

Telco lines used with your iQ6 can be in many different states, simultaneously. The iQ6 has a front panel OLED display which will show the status of the lines. Also, the web interface on the Status page gives remote access to these same states. The three different clients, Axia iQ console, VSet, and VX Producer, will also show the states of these lines. The following symbols define the states.

Line is ready for incoming or outgoing calls (you may be able to observe a small pulse occur with this symbol)



Line is ringing



Call is on hold



Call is ON AIR - Hybrid 1



Call is ON AIR – Hybrid 2



Call is ON AIR & Locked - Hybrid 1



Call is ON AIR & Locked - Hybrid 2



Screened Hold - Call is on hold and has been screened by Producer



Block All Mode (busy lines)



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Call is in progress (either with VX Producer or VSet)



iQ6 can be controlled by using either the VSet phone, VX Producer software, or the phone controls built into your iQ console's Telco frame. Let's look at each of those controllers.

Axia iQ Console

The Axia iQ console Telco expansion module is designed to interface with the iQ6. Its basic controls are as follows.

- ♦ Keypad: Used for placing outbound phone calls
- Transfer: Used to create a "hook flash" in POTS analog phone systems. Depending on the way you've ordered your Telco service, this key may serve to activate features such as transfer or 3-Way Calling.
- Block All: Pressing this key once will "busy" all phone lines (except those that are on-air, on hold, or on a handset), thus blocking incoming calls. Pressing the key a second time will release call blocking.
- Line Buttons: Each line has 2 buttons, the left column for Hybrid 1 and the right column for Hybrid 2. Pressing a line button places the caller on that line on-air using the corresponding hybrid. Pressing a line button when a call is already on-air locks the call on-air, preventing accidental disconnection. To unlock, press the line button a third time.
- Drop 1 and 2 Buttons: These buttons will drop calls (in their respective column) that are currently on-air, if that call is not in the "locked" state described above.
- Next: Selects the line which has been ringing longest. If no line is ringing, it selects the line that has been on (normal) hold the longest.

VSet

Set Talent or Producer Mode

The VSet should be set to Talent mode when it is located in the studio and is used to put calls on the air. Producer mode is for a VSet that is being used by a producer to screen calls. In Producer mode, calls cannot be put on-air and on-air calls cannot be dropped. The function of the right line button column is different in the two modes to serve the needs of each type of operator, as described below.



Press the menu button to access the menu functions. The LCD will show the various items that can be changed. Select Talent or Producer and then exit the menu functions by pressing the menu button.

The Line Info Field

Each line has a corresponding portion of the LCD display to the right of the line button that shows status and info about that line.

Next indicator

The '>>' symbol. Marks the line that will be taken when the next button is pressed.

Fader number

The yellow/orange rectangle near the status icon.

Line name

The first text line. This is the line name given during configuration.

Caller ID

This is either the incoming or outgoing caller ID, depending upon whether this line is incoming or dialedout. The green arrow to the left of the text points left for incoming or right for outgoing.

Time

The length of time the call has been ringing-in, on hold, on-air and on-handset.

Action 'hint'

Shows what line will be affected by drop and hold. Choose the line you want to drop or hold after pressing either of those buttons. This shown only when more than one is possible; this step is not needed when only one line is possible, when the line would be dropped or held immediately.

Line Button Columns

Each line has 2 associated buttons to the left of the LCD.

Pressing a left column line button puts a held or ringing caller on the handset.

Press the right column button to put a held or ringing line on air, dropping any other unlocked calls. If a call is already on the air, pressing the right button locks the line, or if already locked, unlocks it. That is, it toggles between the two states. Not to worry; you cannot drop a call.

In Producer mode you use the Ready Hold button (the one with the checkmark) to put lines in ready hold mode and assign priority for the Next function. See Next Button and Function. Ready hold works like normal hold, except the line status icon has the checkmark to indicate to operators that a call has been determined to be ready for air, normally by a producer/screener.

Hold Button

Holds a call that is ringing, on the handset, or on-air. If more than one line is in this category, there will be a small hold icon near the status icon on the lines that could be held. Press the line that you want to hold.

Drop button

Drops a call that is ringing, active on the handset, or on-air. If more than one line is in this category, there will be a small drop icon near the status icon on the lines that could be held. Press the line that you want to drop.

Lock

Normally, taking a call to air causes any others on-air to be dropped. If you need to conference two or more calls, you can use the lock function.

Press an already on-air line button to lock it. The line status icon will change to display the locked symbol. Pressing the button again unlocks the line and the lock icon goes away.

Locked calls remain on air, and cannot be dropped until they are unlocked.

The Drop and Hold buttons have no effect on a locked line.

Next Button and Function

The Talent next priority is:

- 1. longest waiting ready hold
- 2. longest waiting hold
- 3. longest ringing-in

The producer can manually override these and assign priority as desired.

The producer mode next priority is:

1. longest ringing-in

In producer mode, Next does not take any held lines.

Busy All

Pressing this button will cause all inactive and ringing lines to be dropped and blocked from accepting any calls. Calls on-air, on the handset, on hold and Priority lines will not be affected. The usual application for this function is to let you prevent early callers from getting in on contests until after you've made the announcement and released the lines to accept calls.

Pressing Block All again will release the lines and allow incoming calls.

Numeric Keypad

For dialing out, either on the handset or on-air. It works during an active call to generate tones for voice mail and other services.

While a number is being composed, the digits will appear in the line info field. After a number is entered, the GO button causes the connection to be made, in similar fashion to a mobile phone.

Re-dial Function

Any number previously dialed may be re-dialed from the CALL HISTORY menu on the VSet. Enter the CALL HISTORY menu and use the round keys to the right of the screen to select the call you would like to make and press GO. Reboot the VSet to clear it's call history. The call history is shared with all VSets connected to the hybrid system.

Fader Assign

Normally, when a call is taken, it goes to fader 1, but you can assign it to the second console fader. When a call is on-air, or before it is taken, the round fader assign buttons to the right of the LCD can be used to move the call to the desired fader. The number in the yellow rectangle in the line info field shows you which fader is or will be used.

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Take it Easy

In the simplest case, when you are a talent supported by a producer, you could just use the Next and Drop buttons. You press the Next button to take calls in the sequence determined by your producer and you press the **Drop** button when you are finished with your last call and you want to stop taking more.

One step up would have you pressing the right column line buttons to select the calls you want, one at a time. Again, when you are finished taking calls, you would press **Drop**. You could use the left column line buttons to talk with callers on the handset.

VX Producer

Introduction

The VX Producer application integrates:

- ♦ Call screening and production communication
- Recording, playing, and editing calls
- ♦ A 'softphone' for talking with callers off-air

Normally, it would be used by producers or call screeners to communicate with callers and the on-air talent. Each would have a copy running on a PC near to them.

Managing Calls

The main window shows a list of the lines that are available.

Near the bottom of the screen is a tabbed list box, controls, and an audio level meter that are used for the recording and play functions.

At the lower right is a window used for text chatting between the producer and talent.

Handset

Press the handset button to the left of any line to take the call to the 'handset' that is connected to the PC running the VX Producer application. (Often this will actually be a headset/mic.)

If no call is ringing-in, the system assumes you want to dial out and opens a window to let you do that.

On-air

Pressing this button takes a call to the studio mixing console. Your operation of the console determines whether the call is actually on-air or sent to preview or to somewhere else. Usually the studio mic (or microphones) are used to talk with a caller, whether on-air or not.

The Line Info Field

- Fader number. Before a call is taken to air, this shows which fader it will be assigned to when it is put on-air. After a call is on-air, this shows which fader is assigned to the call. The number changes when you move a call using the fader assign buttons.
- Next indicator. Shows which call will be taken when you press the Next button.
- ♦ Line status icon.
- Line name. This is the name that was given to the line during show configuration.
- Caller info. Text you enter into this area will appear on any other VX Producer applications and VSets that are assigned to the same studio. It automatically replaces the line name text.
- Telephone number. Shows incoming (caller ID) or outgoing (dialed) number. The small arrow near the number points left for incoming and right for outgoing numbers.
- Time. Shows time that a call has been waiting on hold, on-air, on handset, or ringing.

Drop, Hold, and Hold Ready Buttons

Drop disconnects a call that is ringing, active on the handset, or on-air. If more than one line is in this category, they will all be highlighted. Press the line that you want to drop. Or select the line you want to drop to highlight it and then press Drop.

Hold puts a call that is ringing, active on the handset, or on-air into hold. If more than one line is in this category, they will all be highlighted. Press the line that you want to hold. Or select the line you want to hold to highlight it and then press Hold.

Hold Ready puts a call that is ringing, on-air, on the handset, or on hold to the hold ready (screened) state.

Lock/Unlock Button and Function

Normally, taking a call to air causes any others on-air to be dropped. If you need to conference two or more calls, you can use the lock function.

Press an already on-air line button to lock it. The line status icon will change to display the locked symbol. Pressing the button again unlocks the line and the lock icon goes away.

Locked calls remain on-air until unlocked and then another call is taken to air or the call is explicitly dropped or held.

The Drop and Hold buttons have no effect on a locked line.

Fader Assign Buttons

Normally, when a call is taken, it goes to "fader 1", but you can assign it to the other console fader.

Next Button and Function

The Talent next priority is:

- 1. longest waiting ready hold
- 2. longest waiting hold
- 3. longest ringing-in

The producer can manually override these and assign priority as desired.

The producer priority is:

1. longest ringing-in

Block All

Pressing this button will cause all inactive and ringing lines to be dropped and blocked from accepting any calls. Calls on-air, on the handset, on hold and the priority lines will not be affected. The usual application for this function is to let you prevent early callers from getting in on contests until after you've made the announcement.

Pressing Block All again will release the lines and allow incoming calls. The specific lines blocked are determined by configuration. Normally, VIP and hotlines are configured as priority lines, and therefore not busied.

Recording, Editing, and Playing Calls

Pressing the Record button starts a recording of the currently active phone call.

In talent mode, this will be the "on-air" call. Usually, the audio will have been routed via the studio mixing console.

In producer mode, this will be the call on the handset.

You can choose if both the local mic audio and the phone are recorded or only the phone. In the former case, a stereo file will be produced.

Enabling Auto Record causes the recorder to automatically make a new recording each time a call is taken.

File names are automatically created by the system, but can be changed by writing over the default text after a right mouse click. The file will appear in the tabbed list box determined by Settings. Normally, a VX Producer in producer mode will put recorded files under the Producer tab and in studio mode, under the Studio tab (naturally). You can move files among the tabbed lists by either dragging them to the destination tab or using the Send button and drop-down list.

The meter to left of the tabbed file list window shows the audio level as a recording is being made.

Playing

Select a file by clicking on it. Press the Play button. The audio will be sent to the PC's output configured in Settings. Depending on the Settings configuration, the playback will be available on PC-connected loud-speakers or headphones, or it may be routed to a studio mixing console fader.

Editing

Select a file by clicking on it. Press the Edit button. The audio editor configured in Settings will open and the selected file will appear in the editing window (Audacity is the default). Audio input and output will be according to the editor's configuration. Depending on the configuration, the payback will be available on PC-connected loudspeakers or headphones, or it may be routed to a studio mixing console fader.

A stereo file will be passed to the editor when both a mic and phone audio are recorded. A mono file will be passed when phone-only record mode has been selected.

Host/Producer Text Chatting

The window at the lower right lets people within a Studio communicate with each other. Type text into the lower field and it will appear on any other VX Producer applications connected to your studio.

The 'Studio' and 'Producer' identifiers are added automatically. The colors for them can be configured in settings.

4 Detailed Configuration & Reference

4.1 Front Panel

The iQ6 includes four LED bargraph meters. The ballistic (time) characteristic of these meters changes depending on the Audio menu Input Level setting. In VU mode, they have the usual 300ms time-constant. In PPM mode, they have a 10ms attack and 2.8sec release characteristic.

Two meters are labeled input for hybrid 1 and 2.

Two additional meters are labeled for output for hybrid 1 and 2.

There is a single LED labeled POH which shows the presence of Program on Hold audio.

The OLED display of the iQ6 provides status information as well as some basic configuration.



The status screen provides the state of each of the six telco lines (states are indicated by icons described in chapter 3), input state, digital sync, and lock.

Pressing any of the three buttons will enter into the main configuration screen where the options available are:

- ◆ Audio (audio settings and adjustments)
- ◆ Telecom (telco settings and adjustments)
- System (base system options)
- ♦ Status return to default status page

If no activity happens for 2 minutes while on this screen, the display times out back to the Status screen.

4.1.1 Audio Menu



4.1.2 Telco Menu







4.2 Web User Interface

A web browser can be used to remotely configure and administer the system. All front panel configuration options are duplicated on the web pages. Certain advanced options such as entering Livewire configuration require the web interface.

You can connect a PC directly to the iQ6 with an Ethernet cross-cable. Or you can connect both the PC and the iQ6 to an Ethernet switch or the switch in the Axia iQ Core.

After you have a network connection between your PC and the iQ6, open a Web browser and enter the IP number assigned to the iQ6 into the address entry field. You will then see the iQ6 main Web screen:

<u>Status</u>	Current state of the system, network interfaces, controller connections
<u>Telco</u>	ISDN: SPIDs, DNs, Protocol type POTS: Country selection, Hookflash duration, Auto drop enable and threshold, Loop current checking, Caller ID configuration
Audio	Input/Output levels and Meter ballistics Hybrid: Caller ducking, EQ, AGC, Gate, Mix Minus Routing, Feedback Reduction
<u>Show Setup</u>	Configure show behavior, line assignments, line sharing, phone book
LiveWire Configuration	Configure Livewire Sources and Destinations: Names, Channel numbers, Modes
Livewire GPIO	Configure Livewire general-purpose I/O: names, channel numbers, format
<u>LiveWire QOS</u>	Livewire clock mastership settings 802.1p priority and VLAN tagging, DSCP Class of Service
<u>System</u>	IP network address, Software update and maintenance
Backup	Backup and restore system settings
Diagnostics	Syslog messages, State dump

Telos iQ6 Talk Show System

Upon clicking any of the menu links, you will be prompted to enter a user name and password.

The default user name is "user" and the password is <blank> (no password).

The Status, Administration, Telco, Audio, Show Setup, Backup, Miscellaneous, Livewire Configuration, GPIO, and QoS menus provide access to the full set of configuration options, a superset of those available on the front panel. (It's not very convenient to enter numeric data on the LCD/ button interface, so we thought you'd appreciate having a PC's keyboard for that.)

Some pages have a "?" in the title bars. These are links that open information pages. Some of the features described in this manual are also described in these help pages. If a new feature is added to the product prior to a manual update, the help page will provide the details to the new feature.

4.2.1 Status

The status page does not provide any configuration options, but it provides useful information as to the status of your iQ6. Take note of the version of code running on the iQ6, VSet (connection counts), and the Line Interface states (handy remote access information).

Cur	tom Ctai	tue.	
Version:	stem sta	1.6.0r	
Uptime:		0-02:54:28	3
Au	dio Sourc	es	
Hybrid 1:		Livewire	
Hybrid 2:		Livewire	
Program on Hold 1		Livewire	
Line	Intorfac	o 4	
LINE	interiac	ea	
Туре:		POTS	
Line 1:		Idle	
Line 2:		Idle	
Line 3:		Not Ready	
Line 4:		Not Ready	
Line 5:		Not Ready	
Line 6:		Not Ready	
Con	trollor Ct.	atur	
Cun	Max I	Current	
VX connections:	12	2	
Audio connections:	6	0	
	Name	Studio Audio (han
	manne	stadio Addio t	mail
192.168.100.209:	VX0	1 No Audio	2

4.2.2 Telco

POTS Interface Settings

Telco	
POTS Interface 9	Settings ?
POTS Country	USA 💌
CallerID alert signal	Default
Show ringing before CallerID:	Yes 💌
Hookflash duration:	520ms 💌
CPC loop drop threshold (ms)	60
Loop current checking:	Yes 💌
Enable auto call drop:	Both loop drop and CPTD 🔽

POTS Country

This setting configures the analog Telco line interface for proper impedance matching and other characteristics for different countries. If your country is not listed, try using: CTR21 (European standard) or USA.

Caller ID alert signal

The behavior of "Default" is determined by the POTS Country setting. If "Default" does not work, ask your telco what Caller ID alert signal they send.

Show ringing before Caller ID

Normally the ringing indication is passed to the VSet and other devices after the Caller ID has been received. This delays the ring indication until after the second ring. You may want to have a faster ring indication, and choosing "Yes" here lets you have it. But you risk losing the CID if an operator answers the call quickly. Some lines don't have CID and in this case this should be set to Yes since there is no reason to wait for the second ring.

Hookflash duration

Adjusts the duration of the "hook flash" (loop current interruption) when the Transfer key is pressed. The default value is 720 milliseconds. The following table indicates typical values for various countries:

USA	720 ms
Belgium, Netherlands, Switzerland	120 ms
France, Luxemburg, Portugal, Germany	270 ms
Rest of Europe	90 ms

The hook flash is sometimes used as a signal to initiate transfer or other features, with the specific effect being dependent on the Telco or PBX capabilities and configuration.

CPC loop drop threshold (ms)

Sets the threshold time value for loop drop disconnect.

WARNING

If CPC time is set to less than 250 ms, disconnection can occur on polarity reversal/answer supervision on outgoing calls in some countries.

Loop Current Checking

The iQ6 periodically checks each analog line port for loop current. When current is detected, the single dot icon appears on status screens and the line is ready. This process is called Loop Check. *Yes* is the default.

When you are using "ground start" lines, this should be set to *No*. Note: The iQ6 does not support outbound dialing on ground start lines. However, setting Loop Check to *No* will allow incoming calls to work normally.

Note

Ground Start analog lines are not fully supported. Only Incoming calls are allowed. You can, however, take advantage of this to get a feature that might be useful: preventing outgoing calls.

Enable Auto Call Drop

Calls can be automatically disconnected when the far end hangs-up. Most USA-style Telco lines provide a momentary loop drop for this purpose, but non-USA lines and most PBXs provide a tone signal (Call Progress Tone - Disconnect) instead. The iQ6 can respond to one, both, or neither of these signals. The type of tone signal detected for CPTD is determined by the POTS Country setting.

4.2.3 Audio



Nominal Input Level and Meter type

Adjusts to the level expected from the Axia iQ console. It is applied before the send AGC.

Increasing the input level past the correct value will not make the send louder due to the AGC/limiter.

The ballistic (time) characteristic of iQ6 meters changes depending on the Input Level setting. When -20dBFS is selected, the meter has a VU-standard integration time. When +9dBu PPM is selected, the meter has a PPM time characteristic.

The usual setting is -20dBFS. The +9dBu PPM setting is intended for Euro installations.

Nominal Output Level and meter type

This adjusts the level for the output.

LED Meter Channel Selection

Selects the audio that will be indicated on the LED meters.

- Studio 1&2 (default) Livewire Input and output
- Phone 1&2 Send and receive to/from hybrids 1 & 2
- ◆ POH in The Program on Hold input

Phone Send Gain Adjustment

This setting adjusts the level of the hybrid into the telephone line, after the send limiter, and is calibrated with 0dB representing the usual send level into the line.

The default is 0dB, which should be OK for most telephone lines. If callers consistently complain about low audio level and you have verified that this is a problem, you can increase the send level. However, doing so can reduce effective hybrid performance and increase leakage. Conversely, lowering this setting can reduce leakage.

Call Ducking Level

This control adjusts the amount of ducking that occurs to the received telephone audio when send-to-caller audio is present. A value of Full Duplex means that the telephone audio is never ducked or attenuated, while Half Duplex applies maximum attenuation when send audio is present. Intermediate settings range from -1 to -15dB. Ducking is desirable for a variety of reasons:

- Allows the announcer to override the caller by causing the received caller audio to be attenuated (ducked) when the announcer speaks. This is often desired for aesthetic effect and allows the announcer to remain in control of the conversation.
- Dynamically improves effective trans-hybrid loss to reduce leakage, when necessary.

There is also a ducking of the announcer audio to the caller. This is only a few dB unless Feedback Reduction is enabled, as described below. This send ducker has a fixed value.

Receive AGC

The receive Automatic Gain Control serves to improve level consistency of caller audio. Nominal levels of the telephone network vary as much as 30dB from call to call. In the past decade our experience has been that loud calls have become just as much a problem as low level callers.

Therefore we have changed the AGC to act as a platform leveler. The iQ6 uses a dB-linear approach to AGC with a feed-forward topology, which provides a consistent sound regardless of the drive level. This "smart compressor" normalizes levels while retaining the natural dynamics of the caller's voice.

This selection controls the "aggressiveness" of the AGC process, taking values from OFF, 1/4, 1/2, 3/4 to FULL. We suggest you start with the default value, FULL.

Aggressive AGC is more noticeable, but is also more effective at maintaining consistent levels. Adjustment of this setting simultaneously changes a number of parameters within the AGC function, such as attack & release times, thresholds, and compression ratio.

An important feature of the iQ6's AGC is that it is cross-coupled to other sections of the hybrid so that it can reliably distinguish between the real caller audio and residual hybrid leakage. This allows a lot of gain on low level callers while preserving excellent hybrid performance.

Noise Gate

Enables or disables a noise gate applied to the caller audio. This is a "downward expander" that has the effect of reducing the noise on the caller audio when it falls below a threshold. On some very low-level calls, this attenuation may be inappropriate because you may be able to hear the caller fade away when the level falls below the threshold. It has three settings as follows:

- ♦ Off Disables the noise gate
- ♦ Normal A slow setting, with moderate noise attenuation
- Aggressive A faster setting with deeper attenuation, for lines where noise is noticeable during normal conversation.

Receive EQ (Caller Dynamic Digital EQ – DDEQ)

Telephone audio frequency response varies widely since many factors can affect it (we've measured the response on a number of calls and the results were revealing). Consequently, some form of receive equalization is desirable. There are three choices for the type of equalization applied to the received telephone audio:

- Off The caller audio is passed through without modification. The Rcv EQ values are ignored in this mode.
- Fixed This is a simple manual equalizer mode where a fixed EQ is applied.
- ♦ Adaptive This is a three band dynamic equalizer. We've chosen frequency breakpoints, time constants, and other characteristics to optimize the tonal quality of varied telephone callers. The Digital Dynamic EQ system customizes the callers' spectral characteristics, which are maintained from call to call.
- Adaptive + Fixed This mode separately adds a fixed amount of additional gain to the high and low frequency bands of the three band Digital Dynamic Equalizer.

(Additional) Low EQ

Amount of low frequency boost or cut applied to the caller audio. Settings are in 2dB steps from 0dB to +6dB are possible with 0dB representing flat response. The default value is +4dB, which sounds good across a variety of callers, lines, and telephone sets.

(Additional) High EQ High

Amount of high frequency boost or cut applied to the caller audio. Settings from 0dB to +6dB are possible with 0dB representing flat response. The default value is 0dB.

Diagnostic Test Mode

Should always remain at "Off" unless instructed to be changed by Telos support.

4.2.4 Show Setup

Show Setup		
Show configuration		?
Hold previous when select Line:	None	~
Enable auto-answer line to hold:	Off	~
Fix locked lines to hybrid:	Off	~
Block priority lines from next queue:	Off	~

Hold previous when select Line

If a call is in progress (line selected) and another line begins to ring, selecting the second line can either end the previous call or place the previous call on hold. The option allows you to configure the hold function for clients that are set as Producer, Talent, or both.

Enable auto-answer line to hold

Enables the auto answer feature. Non-priority ringing lines are automatically answered and put on hold.

Fix locked lines to hybrid

When enabled, you will not be able to transfer a line that is locked on air to another hybrid. If not enabled, even if a line is locked, the line can be switched to the other hybrid.

Block priority lines from next queue

When turned on, priority lines will not be selected when 'next' button is pressed. As a side effect, ringing priority lines will not trigger the 'RingOutNorm' GPIO bit

Show Line Configuration				?	
	Name	Show	F	Priority	Share with 0.0.0.0
Line 1:	Line 1	Show 1	•		
Line 2:	Line 2	Show 1	~		
Line 3:	Line 3	Show 1	~		
Line 4:	Line 4	Show 1	~		
Line 5:	Line 5	Show 1	~		
Line 6:	Line 6	Show 1	~		
Lines shared with: 0.0.0.0					

Line Sharing

Line states can be shared with another iQ6 to enable studio "hot line" sharing between studios. This is generally a special DID number that only "the engineer" or GM or other would use. If using POTS, you can plug this line into two iQ6 units and configure this option so that both units know when the line is in use by the other studio. Both iQ6 units require the opposing unit's IP address to be assigned in the IP field.

To enable this functionality, enter the IP address of the "other" iQ6 and select which line (s) are to be shared.

You may also, select which lines are Priority lines. Priority lines are used for studio hotlines and will not be blocked by the Busy All function.

Phone Book ?			
	Name	Number	
Entry 1:			
Entry 2:			
Entry 3:			
Entry 4:			
Entry 5:			
Entry 6:			
Entry 7:			
Entry 8:			
Entry 9:			
Entry 10:			

The bottom of the Show Setup has a new feature set that is introduced with the VX series of products. A phone book of ten numbers can be added and each entry can be given a name. These values are used in the VSet phones.

4.2.5 Livewire Audio

The Livewire page has configuration options for the two hybrids and the single Program on Hold (POH) input. The two Hybrids require unique channel numbers to be assigned. The number can be anything from 1 to 32767, and is unique to your Livewire network.

The Mode drop down box currently has only one choice - Livestream. No configuration is needed here.

The Livewire Destination is configured for you. If there is a need to configure the Destination portion of the iQ6, select the "Expert Mode" link.

For the POH input, a typical input would be Program 1 from the Axia iQ console.

The front display of the iQ6 will indicate if each Livewire input is active. The bottom left of the display will show the Livewire icon when a stream is valid or a circle if the requested stream is not valid. There are three indicators. The first is Hybrid 1 input, followed by Hybrid 2 input, and last the POH input.



4.2.6 Livewire GPIO

The iQ6 has two virtual GPIO ports. The logic associated with Port 1 follows the logic defined in the first table, Port 2 is defined in the second table:

PORT 1

PORT 2

PIN	DESCRIPTION	PIN	DESCRIPTION
GPI-1		GPI-1	Ring Mute trigger
GPI-2		GPI-2	
GPI-3		GPI-3	
GPI-4		GPI-4	
GPI-5		GPI-5	
GPO-1	User Out (Dump)	GPO-1	
GPO-2	Ringing Out	GPO-2	
GPO-3	Priority Ringing Out	GPO-3	
GPO-4		GPO-4	
GPO-5		GPO-5	

The GPIO "channel" fields should not be assigned the same Livewire channel numbers that are used to carry the audio streams.

4.2.7 Livewire QoS

	LiveWire QoS		
Synchronization /	Livewire Clock:		
Livewire clock master priority:	3 (default)		
Livewire clock mode:	IP low rate (default) 💟		
Live Audio / Clock	Streams:		
802.1Q VLAN ID:	0		
802.1p priority:	6 (recommended) 💌		
DSCP Class of Service:	48 CS6 (recommended) 🛛 🔽		
Apply			

The default settings in the Livewire QoS page should be acceptable for the majority of all installations. We recommend changing these settings only when specifically instructed to do so by an Axia Technical Support guru.

4.2.8 System

This menu consists of items that affect basic system operation.

	System
IP settings:	
Host name:	iQ6 (1-12 characters: letters, numbers, hyphen)
Network address:	192.168.100.106
Netmask:	255.255.0.0
Gateway:	192.168.100.1
NTP server:	192.168.100.69 (takes effect after reset)
Timezone:	UTC-8 💌 (takes effect after reset)
Syslog server (IP address):	192.168.100.120
Syslog severity level filter:	Notice: normal but significant condition

IP Settings

This screen allows you to set various options pertaining to the network settings of your iQ6.

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Host Name

The Host Name is a text string you can use to easily identify your iQ6 to browsers and other devices on your Livewire network. 10 characters is the maximum field size.

IP Address

This set of four three-digit numbers specifies the unique network address of your iQ. During setup, you used the front-panel controls to enter this address, but it may be changed here as well. This is a numeric-only field, and each 3-digit sequence ("octet", in IT terms) accepts numbers from 0 to 255. Do not enter leading zeros in any of the four fields; for example, if your iQ6's IP address is to be 192.168.000.012, simply enter 192.168.0.12. Your iQ6 should have an IP address in the same range as the rest of your network (the first three sets of the numbers should match those of the rest of your network), and the last 3-digit set should be a number not used by any other network device.

Subnet

The Subnet mask allows you to administer the size of your local network more easily, in smaller chunks. All devices you wish to communicate with each other should have the same subnet mask values. Packets addressed to a destination with a different subnet mask are sent to your gateway router, and not to local devices. Like the IP address, the The subnet mask is a series of four numbers separated by periods. This is also accessible from the front panel interface.

Gateway

This is the IP address of a gateway router connecting your LAN to another LAN or wider WAN. You will need to get this value from your network administrator. Leave it blank if you don't plan on routing audio to any other networks.

To enter IP addresses using the front panel interface:

- **1.** Press down arrow key $\sqrt[n]{}$ to highlight System, press (OK)
- 2. Press ¹/₂ to highlight Networking, press (OK)
- 3. Press ¹/₂ to highlight IP address, press (OK)
- **4.** Use \mathcal{P} to delete and move the curser back.

To delete the previous incorrect digit press \Im .

- **6.** Press $\widehat{\mathbf{U}}$ until the desired digit is displayed.
- 7. Once all characters are entered, press OK twice to save your entry

NTP server

The iQ6 will provide call history to VSet units with date and time. The time stamp will not be correct unless the iQ6 is able to synchronize to an NTP server.

Syslog Server & Syslog Filter

Permit logging to an external computer running a Syslog server for troubleshooting purposes.

User password:				
New password:	(5-8 characters: letters and numbers)			
ketype new password:	••••••••• (veniy)			
Firmware version):			
Hardware revision:	iq6.r1			
⊙Bank O	ver. 1.6.0r (build Tue Sep 20 16:18:41 EDT 2011)			
OBank 1	empty update: Browse			
Warning: System will reboot after changing current bank.				
Apply				

User Password

By default, the web interface has no password. To access the pages, the username is **user** and the password field is left blank. If you desire that there be a password to access the web interface, you will enter the password here.

If you change the password and forget that password, you will need to use the front panel to restore factory defaults: (Note this clears everything!!)

System -> Diagnostics -> Restore Defaults -> Restore Defaults -> Yes

The unit will reboot and all settings that have been applied will be lost. You will need to apply the IP address and restore all settings

Firmware version

The iQ6 has two banks for firmware. To upload a new version, you will have the desired version saved to a computer, select the *Browse* button, select the appropriate file. Once the file path is in the text field, press the *Apply* button to begin the upload process.

Once a version is loaded to Bank 1, you can use the radio buttons to select which bank you wish to run from. In order to activate the chosen version, you must press the *Apply* button. This requires a reboot of the iQ6.

4.2.9 Backup

The backup page provides the option to store the Telco, Hybrid Audio, and Show setup configuration information to a PC as well as upload a configuration file to the iQ6. Livewire settings are not stored in this file as of version 1.5.9b.

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5 Technical Data & Troubleshooting

5.1 Overview

A consequence of modern surface-mount construction is that it is frequently no longer possible for localrepairs to be made. Special and expensive equipment is required to change parts. As well, today's equipment is very complicated and requires repair technicians to have detailed experience and training, and have access to high-end test equipment.

At the same time, the advent of overnight delivery services means that equipment can be returned to the factory for quick turn-around repair. Therefore, we do not expect you to fix this unit at the component level, and we do not include schematics of the unit in the manual. Upon request, our support staff can fax or mail you the needed schematic. Please see the first pages of this manual for proper procedures on returning units for repair.

There is the possibility, however, that you could repair the unit by swapping subsystems such as plug-in modules or power supplies. The Telos customer support crew is standing-by to assist with this, if you need it.

5.2 System Software & Firmware

Viewing Version Information

To assist with troubleshooting, Telos customer support may ask you to provide some software version numbers. There are two major software components to the system:

- System software This is the software that can be updated by downloading new software.
- Firmware This code is analogous to a bootstrap loader and contains certain critical low level software. This information can only be updated or changed at the factory. It is stored in a partition of the non-volatile memory on the motherboard.

The software version and software build date can be determined from the start-up screen. The software version and software build date are shown.

5.3 General Troubleshooting

Thinking about Problem Solving

Despite best intentions, something could always go wrong. Sometimes troubleshooting a balky set-up can make even the toughest engineer a Maalox and Rogaine addict.

You can't fix any system without the right world view; a zeitgeist of suspicion tempered by trust in the laws of physics, curiosity dulled only by the determination to stay focused on a single problem, and a zealot's regard for the scientific method. Perhaps these are characteristics of all who successfully pursue the truth. In a world where we are surrounded by complexity, where we deal daily with equipment and systems only half-understood, it seems wise to follow understanding by an iterative loop of focus, hypothesis, and experiment.

The notions here apply whether you are solving problems at the system level or at the component level. At the system level, the actions you might take would be very different – checking cables, trying different menu settings – but the thinking is the same.

Too many times, we fall in love with our suppositions. We are quick to overtly or subconsciously assume the problem being chased is due to lousy design, the stupid phone company, or the manager's latest memo.

Armed with a healthy skeptical attitude, the basic philosophy of troubleshooting any system is to follow these steps:

- Observe the behavior to find the apparent problem;
- Observe collateral behavior to gain as much information as possible about the problem;
- Round up the usual suspects;
- ♦ Generate a hypothesis;
- Generate an experiment to test the hypothesis;
- ♦ Fix the problem;
- ◆ Then, repeat, if necessary, to attack additional problems.

Let us now cover each step of the troubleshooting sequence in detail.

- 1. Observe the behavior to find the apparent bug. In other words, determine the bug's symptoms. Remember always that many problems are subtle and exhibit themselves via a confusing set of symptoms.
- 2. Observe collateral behavior to gain as much information as possible about the problem.

Does the LCD's problem correlate to an LED flashing? Try to avoid studying a problem in isolation, but at the same time be wary of trying to fix too many at the same time. No one is smart enough to deal with multiple problems all at once – unless they are all manifestations of something more fundamental.

- **3.** Round up the usual suspects. At the system level, always suspect the menu set-up, the cables, the Phone Company line setup, the punch-blocks, etc. At the component level, many computer problems stem from the same sources. Never, never, never forget to check Vcc!
- **4.** Generate a hypothesis. Before changing things, formulate a hypothesis about the cause of the problem. You probably don't have the information to do this without gathering more data.

Sometimes you will have no clue what the problem might be. Sometimes, when the pangs of desperation set in, it's worthwhile to try anything practically at random. You might find a bad plug, an unconnected line, or something unexpected. Look around, but be always on the prowl for a working hypothesis.

- 5. Generate an experiment to test the hypothesis. Change the telco connection to a known good line; call known good phone or hybrid at the other end; if long-distance doesn't work, try a local call. You should plan your tests to eliminate 50% of the possible problems in one test, if possible. Just keep careful track so you know what you have eliminated.
- **6.** Fix the problem.

A Final Thought...

Constantly apply sanity checks. More than three decades ago, the Firesign Theater put out an album called "Everything You Know is Wrong". Use that as your guiding philosophy in troubleshooting any set-up. For example, just because you checked the Telco line last night and it was fine does not mean that it's OK now.

At 3:00 AM when the problems seem intractable and you are ready to give up engineering, remember that the system has worked and will work. Never panic–you are smarter than it is.

5.4 Replaceable Modules and Hardware Servicing

WARNING - DANGER

Remove all connections, including the Telephone, Audio, and Electrical Mains Power cord before opening the unit. Failure to do so could result in injury or death. Do not open the unit unless you are qualified to do so.

Removal of the top plate gains access for service. To do so, remove the 10 Philips head screws.

Replaceable modules

While we do not expect you to do much repair or troubleshooting of the individual circuit boards, the system does have several removable modules which can be replaced. The system has the following replace-able modules:

- 1. Power Supply
- 2. Analog POTS Line Interface

Fuse Replacement

A fuse is located on the power supply PCB. If fuse replacement is required, note: For continued protection against fire, replace fuse only with same type and value.

Access to the Main PCB

The main PCB, closest to the bottom of the enclosure, is mounted component-side down. You can gain access to the component side of the PCB without taking it out by removing the bottom plate of the enclosure box.

5.5 Livewire Audio Troubleshooting

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Axia Livewire Audio is very plug and play, but with that plug and play is a little blind faith – unless you know where to look to troubleshoot when the audio is not working.

The iQ6 provides two audio outputs, or "sources." (In Livewire parlance, any audio that can be loaded to a fader or routed to an output is called a Source. Any output is called a Destination.) By plugging the iQ6 into an approved network switch and following the instructions in the installation chapter, you created these two sources and made them available to the network. You can route these sources to a destination , like an Audio Node output, or define them as a source profile on your iQ console.

The iQ6 has three audio "inputs". The front panel OLED display shows whether audio is present on these inputs. As mentioned in section 4.2.8, the bottom left corner of the display shows a Livewire logo for a valid stream, or a circle for a invalid stream. What is a valid stream? A valid stream is a network source that is actually present on the network. An invalid stream is a network stream that doesn't currently exist or is not available to the iQ6. If the caller is not receiving audio or not receiving POH (Program on Hold), check the state of this indicator first.

Indicator is a circle with a diagonal line (no stream available to input):

POH input: Verify that the POH Livewire channel that has been configured in the Livewire Audio page is correct. If correct, is that channel enabled on the network? Is there a network disconnection somewhere that is preventing the stream from reaching the iQ6?

Hybrid inputs ("backfeeds"): By default, the hybrid input will be the "backfeed" to the hybrid. This is a source that's dynamically created by your iQ console's mixing engine. It will only generate this source if the Hybrid source is loaded to a fader and is configured as a PHONE type. The Axia iQ console requires that you configure which DSP sources are available to the network - have the backfeeds for the iQ6 hybrids been allocated to the network?

In the Livewire page, there is a link for Expert Mode. This allows for different assignment to the inputs than the default backfeed.

	vewire					
Expert Livewire configuration mode						
	Livewire Sources		Livewire Destinations			
#	Name:	Channel:	Mode:	Channel:		Mode:
1	Hybrid1	10601	Livestream 💌	11201 <program 1@hppr<="" th=""><th>≣◄</th><th>From source 💌</th></program>	≣◄	From source 💌
2	Hybrid2	10602	Livestream 💌	10602 <to:hybrid2@hpp< th=""><th>.</th><th>To source 💌</th></to:hybrid2@hpp<>	.	To source 💌
3	POH Show 1			5001	≣▼	From source 💌
Apply						

In Expert mode, the Livewire Destination section is the input stage to the iQ6. The Channel field is used to request a Livewire channel from the network. If you don't know the channel number, use the browse button and a pop-up screen will populate with all the active channels in the network. The Mode drop down has two options, "From source" and "To source". When "From source" is selected, the channel requested is specifically the number in the channel field. For example, channel 11201 might be Program 1 from a console, which is not a mix minus but the all the audio present on the PGM 1 bus. The "To source" mode specifies the backfeed. When requesting a backfeed, the channel field must equal the same value as the source itself – since you are requesting the source that is designed to go "to" Hybrid2 (channel 10602).

So is the backfeed assigned as the hybrid input and has the backfeed been assigned as a network source in the Axia iQ core? Is something other than the backfeed assigned as an input? Is that available on the network? Try a different network source by selecting the browse button and selecting a source from the network that you are familiar with. Does the front display icon change?

Indicator is Livewire logo (but no audio heard by caller):

Can you verify there is audio present in that source? Can you route that same source to an analog output within your system and determine there is audio there or not?

In cases where you are getting a valid backfeed stream but are not able to get any audio in the backfeed, make sure you understand the "Feed to source" feature of the Axia iQ console. In a source profile for the hybrid (source type Phone), there is an option to select the "Feed to source". By default this is "Auto (Program1 / Phone)". The Axia iQ console manual fully covers these options. For the sake of troubleshooting, you might consider changing this option, or, if you don't change the option, engage the PHONE assignment button on the fader that has the audio you expect to be delivered to the caller. Note that this changes based on the ON or OFF state of the fader that the hybrid is assigned to. (The features of the Feed to source in the iQ manual explain this).

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6 Specifications

General

- ◆ Interface for connecting six analog telephone lines to professional studio equipment. Includes two high-performance digital hybrids and a digital matrix switch.
- ♦ Audio interface is Livewire Audio-over-IP.
- Up to twelve control devices including VSet phones, VX Producer software instances, and iQ console Telco controls may be connected simultaneously.
- ♦ All control and audio is via network connection.

Processing Functions

General

◆ Telos 3rd-generation adaptive digital hybrids

Send (to caller) Processing

- ♦ High-pass filter
- ♦ Frequency shifter
- ♦ AGC/limiter
- ◆ Pgm-on-Hold AGC/limiter
- ♦ Send EQ
- ♦ Ducking

Receive (from caller) Processing

- Low-pass filter
- ◆ Smart AGC/platform leveler
- ♦ Noise gate
- ♦ Telos' DDEQ (Digital Dynamic Equalization) 3-band adaptive spectral control processor

Audio

Livewire

♦ Livewire connection for all in/out and POH audio in

Switching Matrix and Conferencing

- ♦ All-digital audio routing and switching
- 6 telephone lines, 2 hybrids, 2 studio inputs, 2 studio outputs, 1 Program-on-Hold input

Analog Telephone Line Connectivity

Universal interface for worldwide application. Programmable loop current, ring signaling, and hookflash time. Includes caller ID decoding using Bellcore 212 modem standard.

Control Ports

Ethernet 100Base-T

- ♦ Web server for configuration and software updates
- ◆ Telnet for command line control and diagnostics
- ♦ VX Producer server allows up to 12 control devices connect simultaneously



Telos Alliance Limited Warranty



For the latest Telos Alliance warranty, visit: <u>telosalliance.com/warranty</u>.

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