TELOS Hx6

Six-line Broadcast Phone System

USER'S MANUAL

Manual Version 1.0.2, July 2020

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 Hx6 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. telosalliance.com/product-registration/

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Table of Contents

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.

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Telos Alliance

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1 Introduction

1.1 The Hx6

Congratulations on your purchase of Hx6, the latest in multi-line talkshow systems from Telos. Hx6 combines a six line selector and **two** third-generation Telos hybrids with Digital Dynamic EQ. Hx6 also comes with the new AEC – Advanced Echo Cancellation – from Fraunhofer Labs, revolutionary technology that eliminates open-mic feedback. The two hybrids are designed to be used within a single studio for airing simultaneous callers and achieving optimal audio quality.

Hx6 may be controlled by an attached Telos VSet6 phoneset, or by a producer's PC with the included Broadcast Bionics Xscreen Lite call screening software. A network interface connector (RJ45 Jack) is located on the back plane for direct connection to a controller or a suitable network switch for support of multiple controllers.

The Hx6 may be ordered 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 XLR connectors on the back panel.

  ♦ Two XLR-Male connectors for audio from Hybrid-1 and Hybrid-2.

  ♦ Three XLR-Female connectors for audio input to Hybrid-1, to Hybrid-2, and for POH (Program on Hold).

The Hx6 also supports Livewire audio for connection to a Livewire audio routing system through the network interface.

The Hx6 has two female D-subminiature 9 pin connectors for GPIO (General Purpose Input Output).

1.2 Features

Back in the 1980s, Telos pioneered the application of Digital Signal Processing to broadcast telephone hybrids - a huge step forward from the hot-rodded AT&T speakerphones that stations had been using. In the decades since, we've significantly advanced the art of broadcast telephony. Your Hx6 system represents the highest state of that art. It includes:

♦ The most advanced digital hybrids yet devised for POTS phone lines.New symmetrical wide-range AGC and noise gate by Omnia, with adjustable gain settings.

♦ Studio adaptation and pitch shifter to prevent feedback in situations where open speakers are required.

♦ An adjustable caller override that improves performance and allows you to individualize the degree to which the announcer ducks the caller audio.

♦ Telos-exclusive Digital Dynamic EQ (DDEQ) to keep 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).

♦ A Livewire Ethernet AoIP interface for one-click connection to Axia studio networks.

♦ Conference linking for 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 phones and call screening software which supports the VX protocol.

Hunt Groups

The most common configuration for on-air phone system is to have the different phone numbers linked in a "Hunt Group," also called "Rollover Lines," or "Incoming Service Grouping (ISG)." A hunt group allows you to give out a single number to the audience and each call will "hunt" to an unused line. Sometimes one or two numbers will be reserved for a "hot line" or "warm line" in which case those numbers would not be part of the hunt group.

2 Installation and Configuration

2.1 Getting Started

The Hx6 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 Hx6.

Connect the Telco circuits (section 2.2)

Connect the Audio input and output (section 2.3)

Connect power, and once the Telos Hx6 flash screen appears, configure the IP address of the Hx6. Press any of the three keys to access the configuration options.

Press the down arrow key (▼) to highlight **System**, press (OK)

Press the (▼) to highlight **Networking**, press (OK)

Press the (▼) to highlight **IP address**, press (OK)

Use the (▼) to delete and move the curser back. Once you press (▲), you begin to enter a value at curser position, starting with 0. Use the (▼) and (▼) to select the desired value, then press (OK). Once the value is complete, press (OK) once more.

The majority of possible configuration will occur with a PC's web interface. Use a direct connection between a PC and the Hx6 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 Network port. The PC should have an IP address that is within the same subnet as the Hx6. (section 2.4)

Configure clients to function with the Hx6. (section 2.5)

2.2 TELCO

The Hx6 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 Hx6 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 (▼) to highlight **Telecom**, press (OK)

Press (▼) to highlight **POTS**, press (OK)

Press (▼) to highlight **Country**, press (OK)

Use the (▼) and (▲) to select your country, press (OK).

If you do not see your country listed you should use CTR21 or USA. 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 Hx6 does not have the loop through provision for POTS lines. The POTS lines used with the Hx6 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 Hx6.

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

Be careful not to connect the Hx6 to PBX ports intended for proprietary telephones. These sometimes have voltages that could damage the Hx6.

2.2.2 Integration with a PBX

The Hx6 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 off another phone system.

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 Hx6 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 Hx6 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 Hx6 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 Audio

The Hx6 has three female XLR and two male XLR connectors at the rear. The XLR ports support either analog or AES/EBU digital (available option). The Hx6 also supports Livewire audio through the network connection. When Livewire inputs are used, the Analog /AES inputs are switched off. Should the Livewire inputs fail, Analog/AES inputs are switched on.

2.3.1 Analog Audio

Output 1 is associated with Hybrid 1 audio. Output 2 is Hybrid 2 audio. Connect the output ports to your mixer or device you intend to use the Hx6 with. The XLR outs are standard active balance outs:

- ◆ Pin 1 – Ground
- ◆ Pin 2 – Positive (High)
- ◆ Pin 3 – Negative (Low)

Input 1 is the audio feeding Hybrid 1 and Input 2 is audio feeding Hybrid 2. The Hx6 inputs must be fed send-to-caller audio that is free of the caller's audio, a 'mix-minus'. A mix-minus is a mix of all audio except the caller audio. The term mix-minus may also be referred to as 'M-1' or 'clean feed'.

Modern day broadcast consoles should provide an output that fulfill this need. If not, then you will need to build a mix of sources to be fed to the Hx6 with a utility bus.

2.3.2 AES/EBU digital connections

The Hx6 can be ordered with AES audio i/o. This requires additional hardware to be added at the factory. When this option is in place, The XLRs described in the Analog section are used for the AES/EBU digital connections. A single AES/EBU digital stream is two channels of audio. Output 1 has Hybrid 1 audio on Channel-1 (Left channel) and Hybrid 2 audio on Channel-2 (Right channel). Output 2 has Hybrid 2 audio on Channel-1 and Hybrid 1 audio on Channel-2.

Output 1 (XLR1)	**Ch1 (Left) Hybrid 1**
	Ch2 (Right) Hybrid 2

Output 2 (XLR2)	**Ch1 (Left) Hybrid 2**
	Ch2 (Right) Hybrid 1

Inputs have two options, 2-channel or AES POH. In 2-channel mode, left side of the AES input stream will be used to feed the caller on the Hybrid.

Input 1 (XLR1)	**Ch1 (Left) Hybrid 1**

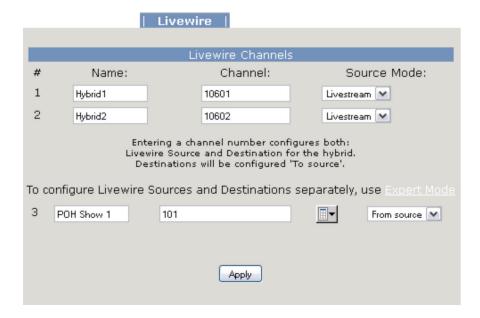|	
Input 2 (XLR2)	**Ch 1 (Left) Hybrid 2**

In AES POH mode, left side of AES input 1 is To Hybrid 1 while right side is To Hybrid 2. The second AES Input is the POH input.

Input 1 (XLR1)	**Ch1 (Left) To Hybrid 1**
	Ch2 (Right) To Hybrid 2

Input 2 (XLR2)	**POH**

Which mode used is set in the web interface.

2.3.3 Livewire audio connections

The Hx6 supports Livewire audio I/O through the network connection. To use the Livewire I/O, the Hx6 should be connected to an Axia approved network switch that supports QoS (Quality of Service) settings as well as multicast control protocols. To enabled the Livewire audio, connect to the Hx6 web interface and select the Livewire Configuration link. Login will be requested; the default *Username* is user ; leave the *password* field blank.

Provide unique Livewire channels for the two hybrids and select the Livestream mode. These will be the Livewire audio sources for Hybrid 1 and Hybrid 2. The send audio to the callers will be automatically configured for you. If you want specific control of the send audio, select the Expert Mode. This is covered in Section 4.2.5.

The POH input is configured below. You can directly type the Livewire channel of the hold audio or select the browse button to the right of the channel text box and select a source from the network in a pop window.

Don't forget to press the Apply button when you are done with your Livewire configuration.

2.4 Basic web interface Configuration

Some basic settings need to be entered into the web interface before you can operate your Hx6. You can connect a PC directly to the Hx6's Ethernet port using a crossover cable, or connect the Hx6 and your PC to your studio LAN. Once done, open your Web browser and enter the IP address you previously assigned to your Hx6.

You'll be greeted with the Home screen, shown above. Select the Telco page to configure the needed POTS interface settings. When first entering into any of the HTML pages, a login request will appear.

Username is **user** and the *password* should be left blank.

POTS Interface

For now, verify that the correct Country setting is selected in the Telco menu. For reference on the other settings, refer to the detailed configuration section. You can also select the "?" at the corner to open a pop up help screen which will provide information on the various options available.

2.5 Client Configuration

The Hx6 supports the VX control protocol allowing for control clients to connect through a TCP connection (a network connection). The Hx6 supports up to 12 simultaneous clients which may be a combination of any items listed below.

Available clients are:

- ♦ VSet6 (2.4.1)
- ♦ XScreen Lite (2.4.2)
- ♦ Axia iQ surface (2.4.3)
- ♦ Axia Element surface (2.4.4)

2.5.1 Vset6 phone

Connect the VSet6 to your network or directly to the Hx6. Note that the VSet6 is powered by Power-over-Ethernet (PoE). This requires that you have a PoE enabled switch or that you use an in-line PoE injector. Make sure you connect the data+power port on the injector into the VSet6.

After the VSet has booted, you should see the "Not connected." notification on the screen.

Press the MENU button.

Press the 6 key ("Next…")

Press and hold the 2 key ("Setup) for 5 seconds

Press the 3 key ("VSet IP Address") to edit the IP address.

Use the keypad to enter the desired IP address for the VSet. Using the # as a backspace and the * as a "dot" (.) .

You can use the softkeys to cancel the edit by moving UP or DOWN, complete the edit by pressing the ENTER soft key, or delete a character by pressing the X.

With an IP value assigned to the VSet6, press the 1 key to exit setup and follow by configuring connection to the Hx6. Press and hold for 5 second the 3 key ("Engine Setup") .

Using the keypad, enter in the IP address of the Hx6. When complete make sure to press the ENTER softkey.

Press the 6 key ("Select Studio") followed by the 2 key ("Hybrid 1&2") . Once the above steps are complete, the line status should appear on the screen.

The last step is to press the menu key to check on the mode the VSet is running in. Verify the mode is in "Talent" mode. If the mode shows "Producer", press the 3 key to toggle it to "Talent".

The VSet6 has a few setup items you need to go through to connect to the Hx6 because not only is it used with the Hx6 but can also be used with other Telos products such as the VX.

2.5.2 XScreen Lite

Before installing XScreen Lite, you must think about which machine will be your *Master* and which will be *Buddys*. You need a single *Master* machine. All other machines must be installed as *Buddys* and these will connect to the relevant *Master* in order to share information for the Hx6. The *Master* machine runs the database that stores past call information and the *Buddys* get access to that information from the *Master*. For this reason, the *Master* machine needs to always be running in order to allow the *Buddys* to connect to it. Running the main Control Room / Studio machine as the *Master* is the usual arrangement.

Once XScreen, version 2.3.00.108 or higher, is installed on the Master machine, a license will need to be generated. It is strongly recommended that the machine is connected to the internet in order to make this process as straightforward as possible.

You will need to select Hx6 from the available Telos devices and enter in the IP address of the Hx6. With the correct parameters entered and a license, XScreen will run.

Refer to: *http://wiki.bionics.co.uk/XScreen.MainPage.ashx* for the online manual of the XScreen application.

2.5.3 Axia iQ Surface client

For complete operational details of the Axia iQ console, please refer to the appropriate documentation for the product. This section is here to assist with connecting the Axia iQ surface to the Hx6.

Connected to the Axia iQ web interface, 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 Hx6 in the following format:
- ♦ user:password@ip.address (user@192.168.2.106)
- ♦ *Studio Name:* Enter "Hybrid 1&2"
- ♦ *Show Name:* This is used for TelosVX systems and may be left blank

When the show is loaded on the surface, the Axia iQ surface will log into the Hx6 as a client.

2.5.4 Axia Element Surface client

For complete operational details of the Axia Element console, please refer to the appropriate documentation for the product. This section is here to assist with connecting the Element surface to the Hx6.

In Element version 2.5.0.9 (PowerStation version 1.1.3c) the Element Show Profile page has a Phone link. Selecting this link presents the Phone configuration settings.

- ♦ *Phone URL:* Define the usage of VX protocol, enter in the user name, password, and IP address of the Hx6 in the following format:
- ♦ vx:user:password@ip.address (vx:user@192.168.2.106)
- ♦ *Studio Name:* Enter "Hybrid 1&2"
- ♦ *Show Name:* This is used for TelosVX systems and may be left blank

When the show profile is loaded on the surface, the Element will log into the Hx6 as a client.

3 User Operation

The telco lines with the Hx6 can be in various states. The Hx6 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. Front-panel Status Symbols displays are explained below.

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)

Call is in progress (with a handset or similar)

Operation of the unit can depend and change based on the needs of the facility. Basic operation can be a screener, using a VSet or other client, answering calls and placing them in Screened Hold mode or simply hold mode. The control operator or host can select a line to go to a hybrid and thus on the air.

To Screen calls

Typically calls are screened by answering without routing the caller direct to the On Air console. In the past this was done by a phone. With the Hx6, a call can be taken by a VSet or a computer running client software. The VSet allows for the call to be taken by picking up the handset and selecting the line. Likewise, a PC with a headset (earphones and microphone) and client software (XScreen) can answer calls, place them on hold, and write notes about the caller on hold. Once calls are screened and placed on hold, the screener has the option to apply the "Screened Hold" mode to the call. This will display a check mark next to the line and is an indication that the call is ready to go live on air. This is helpful in cases where many calls come in and the screener picks up each line and placed them in hold but has not had the opportunity to screen them. So the host or control operator can see many lines in hold but also see ones that are checked indicating the caller can be taken. When the caller in placed on hold, they will hear the audio that is the POH input. Typically this is the main output of the console, Program 1.

To take callers to the console

This step depends on the console being used. If the console is an Axia Element or Axia iQ with a call controller, the operator has the ability to take the call on Hybrid 1 or Hybrid 2 directly, using the on-console controls. Otherwise, the operator will need a VSet running in Talent mode. This allows the operator to select the line to Hybrid 1 or Hybrid 2 and the caller will be routed to audio outputs of the Hx6.

When a caller is routed to one of the two hybrids, the caller will hear the audio that is the input to the hybrid. This should be the Send-to-Caller audio from your console. This audio needs to be a mix-minus. When the caller is placed on hold, they will hear the audio from the POH input.

Vset operation

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 to screen calls. In Producer mode, calls cannot be put on-air and are protected from being dropped.

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 phone line has a portion of the LCD display that shows status and info about that line.

Line is ready for incoming or outgoing calls

Line is ringing

Line is assigned to local handset (with audio levels of send and receive)

Line is on hold

Line is in Screened Hold + indication of Next caller (>>)

Line is On Air – Hybrid 2

Line is On Air + Locked –Hybrid 1

Line is ready for dialing (handset or Hybrid-1)

Line name. Configured from the Show Setup page (Section 4.2.4)

TIME: *Each line will show length of time that has passed since the call has been on hold or ringing to the bottom right of the line field.*

Caller ID *information will show under the Line name*

Take Action ("Hint") *is a function that comes up when the HOLD or DROP buttons are pressed and further selection is required for the system to determine which line to apply the action to. For example if a line is applied to a hybrid and another line is applied to the handset and the HOLD button is pressed, the HOLD icon will appear on the two lines requesting further selection to be taken (as shown above). If the line 1 key is pressed, the response will be to place the handset line to Hold state while <u>not</u> changing the OnAir line state.*

If a line is assigned to Hybrid-1 and another line is assigned to Hybrid-2, but neither is locked, pressing the DROP button will request which line to drop. If one of the lines were locked, pressing the DROP button would drop the unlocked line.

This function is active only when more than one outcome is possible; this function is not needed when only one outcome 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 line to the handset.

When the VSet is in 'Talent' mode, pressing the right column button places the line to a Hybrid-1. There is an important workflow that should be given attention here:

- ♦ Pressing right button assigns line to Hybrid-1
- ♦ if another line is already assigned to Hybrid-1, it will be dropped
- ♦ if another line is already assigned to Hybrid-1 and locked, second line will be added to Hybrid-1

When a line is assigned to a Hybrid and you press the right button associated with that line, the line enters into Lock state. Pressing the button again will toggle out of Lock state.

When lines are active (on hold, ringing, or assigned to a hybrid), the softkeys to the right of the LCD will be assigned to Hybrid-1 and Hybrid-2. When a line is ringing, pressing the softkey for Hybrid-2 will assign the call to Hyrbid-2. If multiple lines are ringing, pressing the softkey for Hybrid-2 will trigger the **Take Action** function previously explained. The hybrid softkeys allow for assignment of lines to Hybrid-1 or -2.

Recall that most of the functions explained so far are for VSets that are in *Talent* mode. In *Producer* mode, the Ready Hold button (the one with the checkmark) becomes active and is intended to put lines in ready hold mode and assign priority for the Next 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 on the handset or on-air. If more than one line is in this category, the Take Action function is called and small hold icons appear 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 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 dropped. 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 until unlocked and then another call is taken to air or the call is dropped or held.

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.

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 fixed lines will not be affected. The usual application for this function is to 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.

Axia Consoles (Element or iQ)

Axia consoles have many of the same functions available on the VSet phones built into the console controllers, and they'll have a similar appearance. These controllers are thoroughly covered in those Axia products' manuals, but for conveniences' sake, we'll recap them quickly here:

♦ *Keypad:* Used for placing outbound phone calls

♦ Block All: Pressing this key once will "busy" all phone lines (except those that are on-air or on hold), 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.

Call Screening Clients

Introduction

Hx6 can be used with computer-based call screening applications. Typically, these apps are used by Producers or call screeners to communicate with both callers and the on-air talent. Producers and talent usually each have a copy running on a PC near them. Functions the software may provide include:

♦ Call screening and production communication

♦ Recording, playing, and editing calls

♦ A 'softphone' for conversing with callers off-air

Managing Calls

The application window shows a list of the lines that are available. The application provides text messaging between all active clients. As calls come in the user can answer the calls, make notes about the callers, and provide space for text dialogue between hosts and producers.

Check documentation for the PC client software you are using for detailed information on how the activate the many functions available. Most client software will be able to place calls OnAir, make line notes, drop, hold, Screener Hold, Lock, unlock, Block, etc. All the functions that are on the VSet should also be available to the PC client software like XScreen, VX Producer and others.

4 Detailed Configuration & Reference

4.1 Front Panel

The Hx6 includes four LED bar graph 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 a10ms attack and 2.8sec release characteristic.

Two meters are labeled Input for Hybrids 1 and 2.

Two additional meters are labeled for Output for Hybrids 1 and 2.

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

The OLED display of the Hx6 provides status information as well as some basic configuration.

Typical status screen for traditional I/O connections

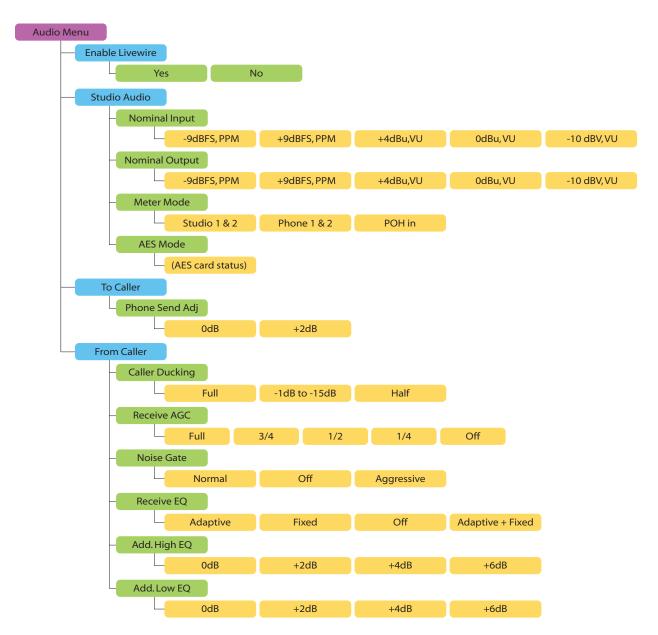*Typical status screen in Livewire environment*

The status screen provides status states for 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:

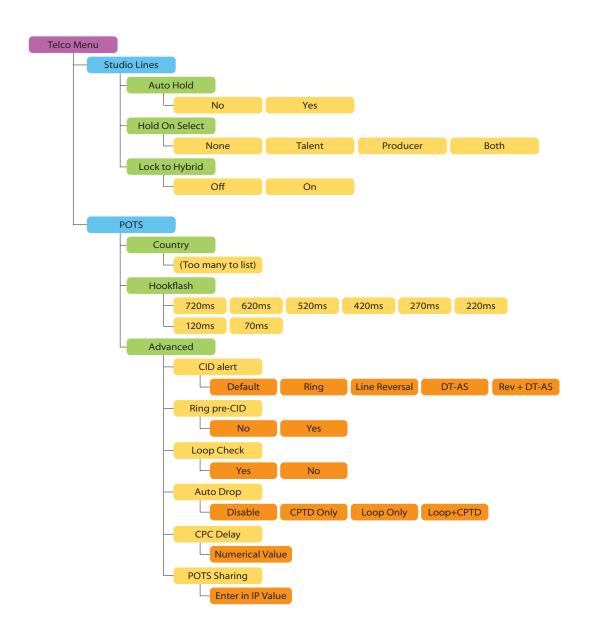♦ **Audio** (audio settings and adjustments)

♦ **Telecom** (telco settings and adjustments)

♦ **System** (base system options)

♦ **Status** return to default status page

After 2 minutes of inactivity, the display will return to the Status screen.
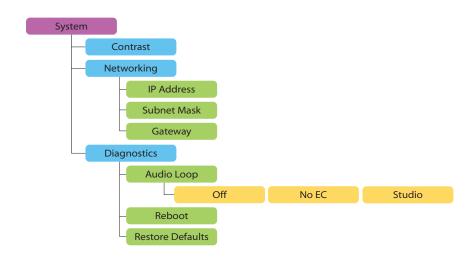
4.1.1 Audio Menu

Audio Menu

- **Enable Livewire**
  - Yes
  - No
- **Studio Audio**
  - **Nominal Input**
    - -9dBFS, PPM
    - +9dBFS, PPM
    - +4dBu, VU
    - 0dBu, VU
    - -10 dBV, VU
  - **Nominal Output**
    - -9dBFS, PPM
    - +9dBFS, PPM
    - +4dBu, VU
    - 0dBu, VU
    - -10 dBV, VU
  - **Meter Mode**
    - Studio 1 & 2
    - Phone 1 & 2
    - POH in
  - **AES Mode**
    - (AES card status)
- **To Caller**
  - **Phone Send Adj**
    - 0dB
    - +2dB
- **From Caller**
  - **Caller Ducking**
    - Full
    - -1dB to -15dB
    - Half
  - **Receive AGC**
    - Full
    - 3/4
    - 1/2
    - 1/4
    - Off
  - **Noise Gate**
    - Normal
    - Off
    - Aggressive
  - **Receive EQ**
    - Adaptive
    - Fixed
    - Off
    - Adaptive + Fixed
  - **Add. High EQ**
    - 0dB
    - +2dB
    - +4dB
    - +6dB
  - **Add. Low EQ**
    - 0dB
    - +2dB
    - +4dB
    - +6dB

4.1.2 Telco Menu

```
Telco Menu
    ├── Studio Lines
    │       ├── Auto Hold
    │       │       └── No    Yes
    │       ├── Hold On Select
    │       │       └── None    Talent    Producer    Both
    │       └── Lock to Hybrid
    │               └── Off    On
    │
    └── POTS
            ├── Country
            │       └── (Too many to list)
            ├── Hookflash
            │       └── 720ms    620ms    520ms    420ms    270ms    220ms
            │           120ms    70ms
            └── Advanced
                    ├── CID alert
                    │       └── Default    Ring    Line Reversal    DT-AS    Rev + DT-AS
                    ├── Ring pre-CID
                    │       └── No    Yes
                    ├── Loop Check
                    │       └── Yes    No
                    ├── Auto Drop
                    │       └── Disable    CPTD Only    Loop Only    Loop+CPTD
                    ├── CPC Delay
                    │       └── Numerical Value
                    └── POTS Sharing
                            └── Enter in IP Value
```

4.1.3 System Menu

4.2 Web User Interface

A web browser can be used to remotely configure and administer the Hx6. 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 Hx6 with an Ethernet cross-cable. Or you can connect both the PC and the Hx6 to an Ethernet switch.

After you have a network connection between your PC and the Hx6, open a Web browser and enter the IP address of your Hx6 into the address entry field. You will then see the Hx6 main Web screen:

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

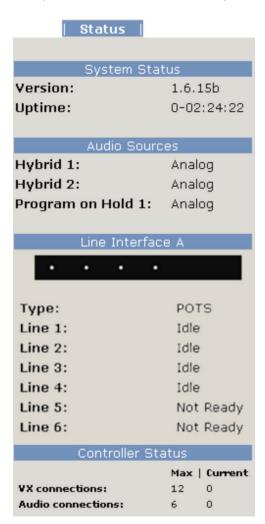The default user name is **user** and the password is <blank> (no password).

The Status, Telco, Audio, Show Setup, Livewire Configuration, GPIO, and QoS, System, Backup, Trouble-shooting menus provide access to the full set of configuration options, a subset 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.)

The Livewire menus are only available on the web interface.

Some pages have ? 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 an updated manual, the help page will provide the details to the new feature.
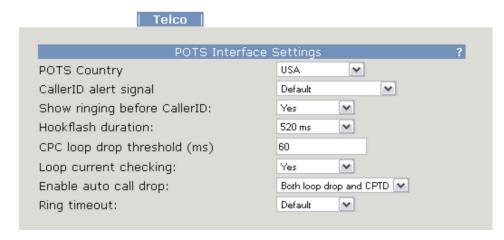
4.2.1 Status

The Status page gives you useful information about your Hx6. Take note of the version of code running on the Hx6, the VX and Audio connection counts, and the Line Interface states (handy remote access information).

| Status |

System Status	
Version:	1.6.15b
Uptime:	0-02:24:22

Audio Sources	
Hybrid 1:	Analog
Hybrid 2:	Analog
Program on Hold 1:	Analog

Line Interface A	

Type:	POTS
Line 1:	Idle
Line 2:	Idle
Line 3:	Idle
Line 4:	Idle
Line 5:	Not Ready
Line 6:	Not Ready

Controller Status		
	Max	Current
VX connections:	12	0
Audio connections:	6	0

4.2.2 Telco

POTS Interface Settings

POTS Country (Affects analog lines only.)

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 (Affects analog lines only)

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.

Loop Current Checking (Affects analog lines only)

The Hx6 periodically checks each analog line port for loop current. When current is detected, the single green dot icon appears on Desktop Directors and the line is ready for use. This process is called Loop Check. Yes is the default.

When you are using 'ground start' lines, this should be set to Off. Note: The Hx6 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 by the Hx6. Only Incoming calls are allowed. You can take advantage of this to get a feature that might be sometimes useful: preventing outgoing calls.**
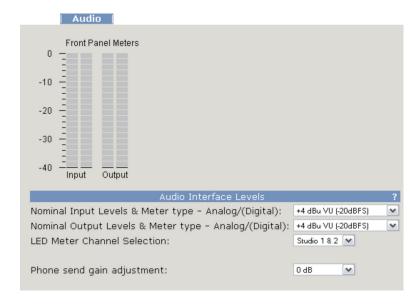
Enable Auto Call Drop

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 Hx6 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.

Ring timeout

 The time to wait after an incoming POTS ring signal before the end of the ringing state is detected and line is considered idle. We strongly recommend leaving this as 'Default' to use the default value for your country.

4.2.3 Audio

The title bars on the Audio page have ? links that open a help page with information that is contained in this section or added features that may have been added since the release of the product.

Nominal Input Level and Meter type

This option lets you set the nominal level expected from the input. 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 Hx6 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.

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) – Audio Input and output
- ♦ Phone 1&2 – Send and receive to/from hybrids 1 & 2
- ♦ POH in – The Program on Hold inputs

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 Hx6 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 Hx6'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. The high and low frequency boost or cut is

(Additional) Low EQ

Amount of low frequency boost or cut applied to the caller audio. Settings are in 2dB steps from -8dB to +8dB 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 -8dB to +8dB are possible with 0dB representing flat response. The default value is 0dB.

Enable Adapt burst

Normally when a hybrid is started, it outputs about a second of white noise to the caller. This helps improve sound quality by calibrating the echo canceller. However, some people don't like it so we have added the option to disable it.
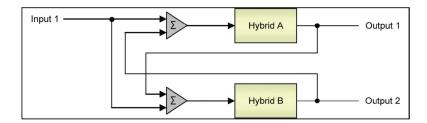
Mix Minus Routing mode

The Hx6 provides different options to connect to your mixing consoles. New consoles will provide separate mix-minus per fader. In some older mixing consoles these features are limited and the Hx6 helps to accommodate to this missing features.

Separate (default)

This option provides fully independent inputs and outputs for each hybrid. In this mode, callers cannot hear each other except through an external mix-minus. Separate requires the mixing console to provide the cross-connection for conferencing.

Single

Using this option means the mixing console only has to make a single mix-minus feed for the caller send audio, but separate the two hybrid caller audio outputs are sent to the console. This mode is useful when your console is limited in the number of mix minus feeds it can make, but you want to keep individual faders per hybrid.
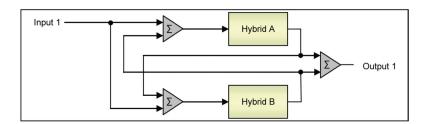

This option sends the single mix minus feed from the console to each caller. Internally within the Hx6, the cross-connected audio from the other callers is mixed in, so each caller can hear the other callers.

Single (Mono out)

This mix-minus option means the mixing console only has to make a single mix-minus feed for the caller send audio and the Hx6 makes only one mixed caller audio output of all callers, to one fader on the console.

Like the single mode, this option sends the single mix-minus feed from the console to both hybrids.

Its main purpose is to accommodate older consoles that have only one mix-minus and when only one fader is available for telephone calls. The disadvantage is that the operator has no control of the relative levels of the multiple callers. It relies upon the Hx6's AGC to automatically adjust the multiple callers to a consistent level.

AES channel mode
The Hx6 can be ordered with AES audio I/O. This requires additional hardware to be added at the factory. When this option is in place, The XLRs described in the Analog section are used for the AES/EBU digital connections. A single AES/EBU digital stream contains two channels of audio.

2-channel modeLeft side of the AES input stream will be used to feed the caller on the Hybrid.

Input 1 (XLR1)	Ch1 (Left) To Hybrid 1

Input 2 (XLR2)	Ch1 (Left) To Hybrid 2

AES POH mode

Left side of AES input 1 is To Hybrid 1 while right side is To Hybrid 2. The second AES Input is the POH input.

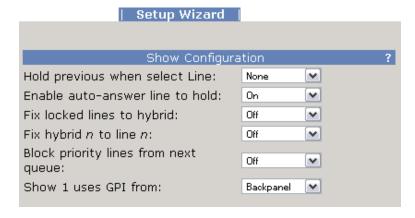| Input 1 (XLR1) | Ch1 (Left) To Hybrid 1 |
	Ch2 (Right) To Hybrid 2
Input 2 (XLR2)	POH

Diagnostic Test Mode

Should always remain at No unless instructed to be changed by Telos support.

4.2.4 Setup Wizard

Show Configuration

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 other Hybrid.
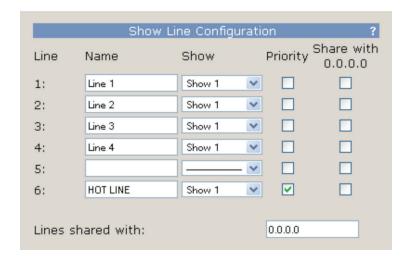
Fix hybrid *n* to line *n*

When enabled, Hybrid 1 is set to line 1, and Hybrid 2 to line 2. This means that these lines and hybrids cannot be paired with any other lines or hybrids. This function is used where a line and a hybrid are intended to be tied to each other and there is no call controller to select lines.

Block priority lines from next queue

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

Use GPI from

Selects whether to use the physical GPIO ports on the backpanel or the Livewire GPIO ports.
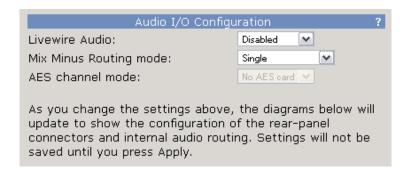

Line states can be shared with another Hx6. To do so, enter in the IP address of a second Hx6 and select which lines are shared. This is to be used when you splice an analog line between two Hx6 units. Both units require the opposing units IP address to be assigned in the IP field.

Also, select which lines are Priority lines. Priority lines are used for studio hotlines and will not be blocked by the Block-All function.

The Name field can be used to give lines names that will appear in VSet6 LCD screen as well as PC client applications.

You might be interested in looking at the ? links that are present in the title bars of the Setup Wizard page. These links open up help pages that explain current/updated feature sets. (The code team often works faster than the manual team!)

Livewire Audio

Enable or disable Livewire function from the front panel or the web interface. Default setting is to have Livewire Disabled for customers that do not have a Livewire network.

Mix Minus Routing Mode

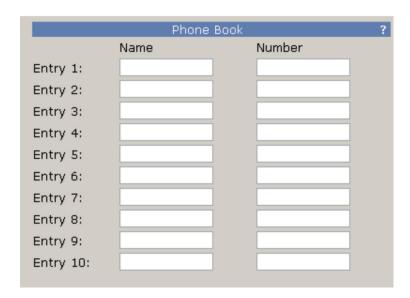This is a duplicate of the settings available on the Audio page. This topic is covered in section 4.2.3 Audio.

AES Channel Mode

This is a duplicate of the setting available on the Audio page. This topic is covered in section 4.2.3
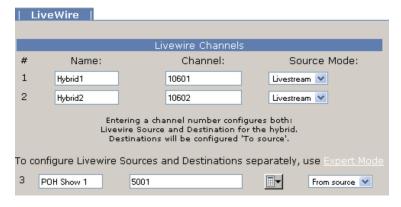
Audio Connections

The lower section of the Setup Wizard page will provide a graphic of the back panel and a flow chart of the internal audio routing to reflect the above settings that are selected.

4.2.5 Phonebook

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 phone line can be given a name. These values are used in the VSet phones.

4.2.6 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 must not duplicate channel numbers assigned to any other audio device on your Livewire network.

The Mode drop down allows for the selection of Livestream or StandardStream. Livestream provides for a low latency audio source into your network while Standard Stream is an efficient packet with 100ms delay. The Livestream will be the best selection in this application.

The Livewire Destination is configured for you. If there is a need to configure the Destination portion of the Hx6, select the "Expert Mode" link.

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

The front display of the Hx6 indicates when Livewire inputs are active. The bottom left of the display with show the Livewire icon when a stream is valid or an XLR icon if the requested stream is not valid (defaults to physical input when Livewire stream not available). There are three indicators. The first is Hybrid 1 input, followed by Hybrid 2 input, and last the POH input.

4.2.7 Livewire GPIO

The Hx6 has two virtual GPIO ports. The logic associated with the port 1 follows the logic defined in the first table; port 2 is the second table:

PORT 1

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

PORT 2

PIN	DESCRIPTION
GPI-1	
GPI-2	
GPI-3	
GPI-4	
GPI-5	
GPO-1	
GPO-2	
GPO-3	
GPO-4	
GPO-5	

The GPIO Channel fields should not be assigned the same channels as your Livewire audio - select a unique channel number for these port channels.

4.2.8 Livewire QoS

The default settings in the Livewire QoS page will be acceptable for the vast majority of installations; don't change them unless instructed to do so by Telos or Axia Support.
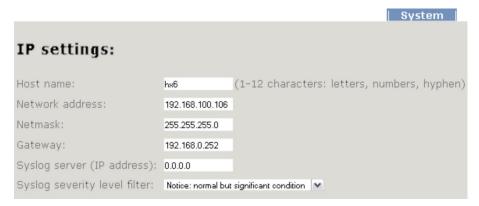
The Livewire Clock master priority option provides the Hx6's priority setting. Livewire devices must be synchronized with each other; one device on each Livewire system acts as the clock master. The Priority setting determines how a particular device will negotiate to be that clock master. The 0 option sets the device never to produce network clock, the 7 option sets the device to always produce clock; steps between the two adjust the device's hierarchy in the network. We suggest you leave this at factory default setting.

The front display of the Hx6 will display a SYNC or MASTER on the bottom of the screen based on its state of producing clock or being synchronized to the clock source.

The settings for the Live Audio and Standard Audio define how the packets are tagged and with what value of QoS tag. The settings should only be adjusted if your network does not follow the standard suggested configuration as documented by Axia.

4.2.9 System

This menu consists of items that affect basic system operation.

IP Settings

This menu section determines the Hx6's network addresses.

♦ **Host Name:** The IP network text name. Default is **Hx6**. Can be used to ID the box, example, Hx6-Stdo1. The Host name will be used to identify the sources which originate from this box.

♦ **IP Address:** This is the IP address for this Hx6. An IP address is a series of four numbers separated by periods. This is a numeric-only field. You should enter no leading zero in each of the four numbers (if you are given 192.168.000.012 enter 192.168.0.12). This is also accessible from front panel OLED interface.

♦ **Subnet:** This is your subnet mask. The subnet mask is set to the size of your local network. All packets addressed to a destination outside this local area are sent to the gateway router.

♦ The subnet mask is a series of four numbers separated by periods in the same format as the IP address. A subnet with 256 addresses would have this mask value: 255.255.255.0. This is also accessible from front panel OLED 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 from your network administrator or can be left blank if there is no routing to other networks.

♦ **To enter IP addresses using the front panel interface:** To enter an IP address, use the following procedure:

 1. Press the down arrow key (▼) to highlight **System**, press (OK)
 2. Press the (▼) to highlight **Networking**, press (OK)
 3. Press the (▼) to highlight **IP address**, press (OK)
 4. Use the (▼) to delete and move the curser back.
 5. Once you press (▲), you begin to enter a value at curser position. Use the (▼) and (▲) to select the desired value. Press OK to accept digit.
 6. To delete the previous incorrect digit press (▼).
 7. Press (▲) until the desired digit is displayed.
 8. **Once all characters are entered, press OK twice to save your entry**

NTP server

Hx6 provides a call history with date and time to VSet phones. To create an accurate time-stamp for this history log, specify the IP address of your network's NTP server. If left blank, the time stamp will be drawn from the Hx6's internal clock.

Syslog Server & Syslog Filter

Permit logging to external computers running Linux standard Syslog for troubleshooting purposes.

♦ **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 Hx6 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 Hx6.

4.2.10 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 Hx6. Livewire settings are also in the settings backup (all but network settings and call controller settings), to allow easy "cloning" of multiple units in large installations.

4.3 Back panel connections

The back panel has connections for AC-power, network connection, audio input/output, logic (GPIO – General Purpose Input Output), and telco interface.

4.3.1 Telco interface

The Hx6 can be ordered with support for POTS (Plain Old Telephone Service). The Telco interface installed at the factory determines which service is supported. An interface with 6 RJ-11 connectors at the back is for POTS.

4.3.2 GPIO ports

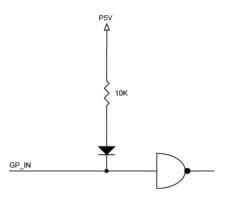Two DE-9 female connectors are present on the backpanel.

GPIO 1 (Bottom Jack)

PIN#	FUNCTION	NOTES
1	Ground	Ground Common
2	In-0	Ring Mute trigger
3	In-1	
4	In-2	
5	In-3	
6	Out-0	User Out (Dump)
7	Out-1	Ringing Out
8	Out-2	Priority Ringing Out
9	Out-3	Hybrid 1 Active

GPIO 2 (Top Jack)

PIN#	FUNCTION	NOTES
1	Ground	Ground Common
2	In-4	
3	In-5	
4	In-6	
5	In-7	
6	Out-4	Hybrid 2 Active
7	Out-5	
8	Out-6	
9	Out-7	

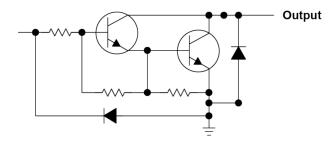### INPUTS

The inputs accept a closure to ground, which may be provided by switches, relays, or logic outputs. The inputs are active low. A built in 10k Ohm pull up resistor is provided.

OUTPUTS

Open collector to ground.

These will require a pull-up resistor to drive TTL-style logic inputs.

Sink (pull-down) current must be limited to 400mA maximum per output with total output restricted to 1 amp (200mA each output if all five will be used).

4.3.3 XLR Audio connections

The Hx6 has 3 XLR-F for inputs and 2 XLR-M for outputs. The XLR inputs are for Program-on-Hold (POH) audio, Send to Caller-1, and Send to Caller-2. The POH input is fixed as an analog input. The two Caller input XLRs can be modified to accept a digital AES/EBU signal. The audio input level is software controlled from the front panel as well as the web interface through a PC. Typical inputs used would be balanced audio with a nominal level of +4dBu.

Two XLR outputs are available and by default are set so that output-1 is audio from Hybrid-1 and output-2 is audio from Hybrid-2. The XLR outputs also have an option to be converted to an AES/EBU digital signal. As an analog output, typical level is a nominal +4dBu balance signal. The output nominal level is adjustable from the front panel as well as through the PC interface.

4.3.4 Network port

The network port on the Hx6 is a 100Mb connection suitable for configuration, client connections, and Livewire streaming if enabled.

Configuration is done using a computer within the same network domain and the user's chosen browser. Access the configuration pages by typing the IP address of the Hx6 into the browser's URL field. Authentication it typically required and the factory default username/password is *user* with no password.

The Hx6 supports up to 12 client connections (either VSets or other VX protocol clients).Livewire audio is supported when Hx6 is connected to an Axia approved network switch that supports QoS settings and multicast management.

4.3.5 AC Power

The Hx6 has an IEC C14 inlet that will accept voltage range from 90 to 264 V (AC), 47-63 Hz.

5 Technical Data & Troubleshooting

5.1 Overview

A consequence of modern surface-mount construction is that it is frequently no longer possible for local repairs 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 POTS 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.

5.5 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 replaceable modules:

1. Power Supply

2. POTS analog telephone line interface card

3. Optional AES/EBU I/O card

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.

5.6 Livewire Audio Troubleshooting

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 working.

The Hx6 provides two audio "outputs" or sources. By plugging the Hx6 into an approved network switch and following the instructions in the installation chapter, you have created these two sources to the network. From any other device in the network that is a Livewire device, you can route these sources to a destination or as a source profile into the Axia iQ console. If you route a source to an Axia node Destination, the audio of that hybrid should output from that node. This can be used to help troubleshoot if there is audio coming from the Hx6 hybrids.

The Hx6 has three audio "inputs". The front panel OLED display helps to indicate if the inputs are valid. As mentioned in section 4.2.8, the bottom left corner of the display will indicate a Livewire logo for a valid stream or a circle for a non valid stream. What is a valid stream? A valid stream is a network source that is present on the network. A non valid stream is a network stream that doesn't currently exist or is not accessible to the Hx6. If the caller is not receiving audio or not receiving POH (Program on Hold), the first question is what the state of this indicator is?

Indicator is an XLR symbol (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 to reach the Hx6?

♦ **Hybrid inputs ("backfeeds").** By default, the hybrid input will be the "backfeed" to the hybrid. This is a dynamically created source by the DSP Engine of your console. 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 Hx6 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.

In the Expert mode, the Livewire Destination section is the input stage to the Hx6. 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 is how one requests for 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 in 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. Take note this changes based on the ON or OFF state of the Hybrid fader. (The features of the Feed to source in the iQ manual explain this).

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 six Vsets, VX Producer software, or Axia iQ console may be connected.

♦ All control and audio is over a network connection.

Processing Functions

General

♦ Telos 3rd-generation adaptive digital hybrids

♦ Telos exclusive feedback reduction functions, including Acoustic Echo Cancellation

Send (to caller) Processing

♦ High-pass filter

♦ Frequency shifter

♦ AGC/limiter

♦ Pgm-on-Hold AGC/limiter

Receive (from caller) Processing

♦ High-pass "hum" filter

♦ Smart AGC/platform leveler

♦ Noise gate

♦ Telos' DDEQ (Digital Dynamic Equalization) 3-band adaptive spectral control processor

Audio Performance

♦ Frequency Response +/-0.5dB, 20Hz to 20kHz (swept sine procedure, measured from analog input to output with unit in loopback mode)

♦ THD+N: Analog to analog, studio loop mode, 1Khz 18dBu test level: <0.006%

♦ Dynamic Range: Analog in to Analog out, studio loop mode, 10Hz-20Khz a-weighted: > 92 dB

♦ SNR: Analog output, referred to -12dBm phone line signal (+4dBu studio out), 10Hz-20Khz a-weighted: > 72 dB

Inputs & Outputs

Analog Inputs

♦ 2x send analog inputs

♦ 1x Program-on-Hold analog inputs

♦ XLR female, pin 2 high. Active balanced, with RF protection.

♦ Input level Adjustable, -10dBV to +8dBu nominal level.

♦ Analog clip point +24dBu

♦ Impedance Bridging, >10K

♦ Analog-to-digital converter resolution 20 bits

Analog Outputs

♦ 2x receive analog outputs

♦ XLR male, pin 2 high.

♦ Output level adjustable from –10dBV to +8dBu nominal level.

♦ Output impedance <60 Ohms

♦ Digital-to-analog converter resolution 24 bits

♦ Headroom before clipping 20 dB from +4dBu nominal level

AES3 Digital Inputs/Outputs (optional)

The AES input/output option module substitutes 2x input and 2x output AES XLRs in place of the standard analog I/O.

♦ Conform to AES3 standard

♦ Sample rates 32kHz to 96kHz.

♦ Rate conversion Input and output, independently selectable

♦ Output Clock AES input or 48kHz internal.

♦ Input Level Adjustable -27 to –9 dBFS nominal.

♦ Output Level Adjustable –27 to –9 dBFS nominal.

Livewire

♦ Livewire connection is standard for all hybrids 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, andflash 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 instances of VX Producer, Vset, Axia iQ console to connect simultaneously

7 Telos Alliance Warranty

Telos Alliance Limited Warranty

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