TELOS iQ6

Multi-Studio IP Phone Interface System

USER'S MANUAL

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iQ6 Manual
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Warranty
This product is covered by a two year limited warranty, the full text of which is included in this manual.

Updates
The operation of the iQ6 is determined largely by software. We routinely release new versions to add features and fix bugs. Check the Telos web site for the latest. We encourage you to sign-up for the email notification service offered on the site.

Feedback
We welcome feedback on any aspect of the iQ6, 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.

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You must contact Telos before returning any equipment for factory service. We will need the serial number, located on the back of the unit. Telos Systems 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 at Telos. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers please contact Telos technical support at +1-216-622-0247. All other customers should contact your local representative to make arrangements for service.

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Notices and Cautions

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:

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.

WARNING:

TO REDUCE THE RISK OF ELECTRICAL SHOCK, DO NOT EXPOSE THIS PRODUCT TO RAIN OR MOISTURE.

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 present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de Class A) prescrites dans le reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

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A Letter from our CEO...

"What if?" A singularly great question, because it opens the door to so many other questions and ideas. Those who ask it are sometimes called dreamers, as if that were somehow bad, but history has often proven dreamers to be the avatars of new and exciting things.

"What if?" was the query asked by Steve Church that led to the start of Telos Systems. In 1984, Steve – then a young broadcast engineer and part-time talk show host – wondered whether emerging Digital Signal Processing (DSP) technology might be useful to clean up the terrible caller audio then the norm at radio stations. The result of asking it was the Telos 10, the world's first digital telephone hybrid, a product that changed the face of radio forever.

Looking back at Telos history, "What if?" is found at the root of many more technologies broadcasters have since made ubiquitous. "MP3 bit rates are perfectly matched to the bandwidth of ISDN; what if we combined them?" "What if we built a box that let radio stations stream audio on the Internet?" "What if we could use Ethernet to treat audio as data and move it around the radio station in real time?" (By the way, if you recognized the answers to those questions as the Zephyr ISDN codec, AudioActive MPEG encoder and Livewire IP-Audio standard, give yourself a cookie.)

Today, as technology roars ahead, we're constantly seeking new ways to apply it to the problems of everyday broadcasting. Voice over IP (VoIP) is particularly compelling for its ability to packetize high-quality phone calls and direct them anywhere in the broadcast plant using ubiquitous, standard Ethernet as a backbone. A new digital tech called AEC (Advanced Echo Cancellation) is helping to literally eliminate the age-old problem of feedback in open-speaker environments. And the prize of using the public Internet for remote connections as reliable as those of switched circuits is within broadcasters' grasp, thanks to ever more sophisticated coding algorithms.

Pioneering ideas aren't limited just to technology, though. We believe that investing in people pays off big, so we've assembled the largest R&D team in the industry, a talented staff of engineers, scientists and broadcast professionals. We're also the first and only broadcast equipment maker with a 24/7 support team ready to offer assistance any time of the day or night. As they say, radio never sleeps – so neither do we.

As Telos navigates its third decade in broadcasting's choppy waters, the culture of innovation and creativity that propelled Steve in the early days still guides us today. We're still in love with radio, still turned on by new technology, still driven to ask "What if?" If you're reading this, you are too — for which we thank you, from the bottom of our hearts.

Michael "Catfish" Dosch

CEO

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 or ISDN（Integrated Service Digital Network）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 both ISDN and 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.

1.3 Integrated Service Digital Networks (ISDN)

The iQ6 works with either traditional analog Telco lines or with ISDN（Integrated Services Digital Network）. We recommend ISDN service for telephone connections whenever possible. There is no comparable technology for getting audio to and from the public telephone network.

We provide plenty of advice right here in this manual and live support by people with years of ISDN experience.（Thousands of Telos ISDN interfaces and Zephyrs are deployed around the world!）Currently, the dial-up telephone network is nearly entirely digital. Only the "last mile" copper connections from the telephone Central Office exchange to the customer's site remain with the archaic analog scheme. ISDN offers us a way to link the studio directly into the Telco digital infrastructure without the usual impairments of analog connections. iQ6 continues the digital path with the use of the Livewire audio connection to the iQ console.

While the application of digital signal processing to the problem of separating announcer and caller audio – pioneered by Telos and used in all of our telephone interfaces – has made a dramatic improvement over analog-only hybrids, using digital phone lines for on-air calls further improves performance for several reasons:

1. **Exceptional send-to-receive isolation.** Traditional analog phone lines multiplex both speech directions in order to use a single pair of wires for a conversation. This causes what's referred to as 'leakage' – when the announcer's audio is present on the hybrid output, creating an annoying 'hollow' or 'tinny' sound. Telos digital adaptive hybrids reduce this problem. But ISDN offers independent and separated signal paths, so our hybrids only have to operate on the far-end analog line, if any. The result is much lower leakage.
2. **Higher send levels.** We don't have to be concerned about regulations designed to prevent crosstalk on analog lines, so we can increase send-to-caller levels.
3. **Lower distortion.** The analog-digital conversion chips used in telephone central offices are poor compared to the converters used in professional audio equipment. Fidelity is not an important consideration when telephone equipment designers choose parts for this function. In a professional interface for studio applications, we can afford to design-in much better converters. Noise-shaping functions permit a larger word-length converter to provide significantly better distortion and signal-to-noise performance.
4. **Lower noise.** Because they are digital, ISDN lines are not susceptible to induced noise. Analog lines are exposed to a variety of noise and impulse trouble-causers as they snake across town on poles and through your building. Hum is the main problem, given most lines. Digital lines convey the bits precisely and accurately from the network to your studio equipment without any perturbation – so the audio remains clean. Even when the caller is using an analog phone line, the digital connection on the studio side makes for noticeably lower noise and better overall quality.
5. **Higher gain and reduced feedback during multi-line conferencing.** When conferencing is required on analog circuits, hybrids are needed to separate the two audio paths in order to add gain in each direction. When the gain around the loop exceeds unity, the unpleasant result is feedback. With digital telephone lines, the hybrid function is more effective – and more reliably so across a variety of calls. That means more gain can be

inserted between calls before feedback becomes a problem.

6. **Digital call setup and supervision.** Analog lines use a strange mix of signaling to convey call status. Loop-current drop signals that a caller has disconnected, and blasts of 90 Volts at 20 Hz mean someone wants you to answer. ISDN uses a modern digital approach to controlling calls and conveying status information about them. ISDN call set-up times are often only a few tens of milliseconds, enhancing production of fast-paced shows. Perhaps more importantly, when a caller disconnects while waiting on hold, the ISDN channel communicates this status change instantly. This contrasts with the usual 11-second delay on most analog lines. One of the most common complaints of talk hosts is that they go to a line where they expect a caller to be waiting, only to be met with a blaring, annoying dial tone. The chance of this happening with an ISDN line is nearly zero.

ISDN lines come in two varieties: Basic Rate Interface (called BRI, SØ, or ISDN 2 in various parts of the world) and Primary Rate Interface (PRI, S2 or ISDN 30). BRI lines are the kind we normally see in broadcast stations, as these are what are used with MPEG codecs such as the Telos Zephyr and Zephyr Xstream. BRIs have a capability of one or two active 64 kbps channels.

Since the Telos iQ6 is used with BRI lines, we will only consider that type here.

An ISDN line from the central office is a single copper pair identical to a normal (unloaded) analog line. When it arrives at the subscriber, it is called the U-interface. It is a two-wire connection, usually via an RJ-11 style modular jack.

The S-interface is at the user side of the Network Termination Type 1 (NT1) device. The NT1 is sometimes generically called a NCTE (Network Channel Terminating Equipment) or, in some countries an NTBA. It is a four-wire connection, via an 8-pin RJ-45 style modular jack. (Sometimes the S-interface is called S/T. There is a subtle distinction between the two, but it is not relevant for our purposes here, and the two may be considered to be the same.) In the USA & Canada NT1 functionality is usually included in the terminal equipment, and indeed the iQ6 in these countries supports the U-interface. In other parts of the world, the telephone company provides the NT1. Only one NT1 may be connected to a U-interface. However, as many as eight terminals may be paralleled onto the S bus.

In the USA & Canada a direct connection to the "bare copper" 2-wire U interface is required. Therefore, the USA & Canada interface module includes an integral NT1 and has RJ-11 style jacks. In the rest of the world, the telephone company provides the NT1, and the 4-wire ISDN S-interface will be used with an 8 pin RJ-45 style jack. Normally we ship to North America with the U-interface module and to other areas with the S-interface module.

In either case each ISDN interface has three connectors. Since each ISDN connection has two channels, this means each interface module handles 6 'dialtones' or 'lines'.

Data and Voice

ISDN lines may be used for voice signals encoded in standard fashion to allow inter-working with analog telephones, or may be used to transmit digital data streams. The latter mode is used for such applications as high-speed Internet access. It is also the mode used with MPEG codecs, in which case the ISDN line may be carrying voice signals, but is doing so in a format that is not compatible with the analog part of the telephone network. The distinction is made in the automatically conveyed Setup message that begins each call.

Normally, the Telos iQ6 uses only the voice mode, so data capability is not necessary. Often voice costs extra, whereas this is rarely true for data. Of course, you may use a line with both capabilities. Just be sure the BRI circuit supports the Circuit Switched Voice (CSV) capability as well as data.

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.

In some cases ISDN lines configured to hunt may deliver all calls to a single DN（phone number）. In this case, you must enter this same DN for each of the DNs on each of the hunting channels.

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 ⇩ to delete and move the curser back. Once you press ⇧, you begin to enter a value at curser position, starting with 0. Use ⇩ and ⇧ 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 either a POTS or ISDN 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 Integrated Service Digital Networks – ISDN
- 2.2.3 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 ⇩ to highlight **Telecom**, press（OK）

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

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

Use ⇩ 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 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 Integrated Service Digital Networks - ISDN

The iQ6 uses a plug-in module to attach to ISDN. The ISDN module has three modular jacks. In the US and Canada, the jack type will be a six-position RJ-11 style jack; in the rest of the world the jack type is an eight-position RJ-45 style.

The correspondence between the interface slot to the system line number for is from left to right when viewing the iQ6 from the rear. Therefore, the leftmost jack would be lines one and two. All ISDN BRI circuits must use the same ISDN protocol setting.

ISDN in USA & Canada: Using the U Interface

Connect the ISDN line from the telephone central office directly to the RJ-11 style U-interface modular jack on the rear panel. Each module has three jacks and each jack has an associated LED.

IMPORTANT

Remember that the U interface connects to a long telephone line and can convey lightning surges into the iQ6. Surge protectors intended for analog lines also work to protect U-interfaces. We recommend that you install one for each line.

U-interface Status LEDs

The small green LED on the ISDN interface card near each U jack indicate the status of the corresponding ISDN circuit. Rapid flashing（about five times per second）indicates a loss of the ISDN at the lowest（physical）level. If the iQ6 can contact the central office or an active U repeater, the LED will blink slowly – about once per second. The LED will come on solid when handshaking is completed and all is OK. If these LEDs do not light continuously once you have connected your ISDN circuits, you may wish to skip ahead to Section 5.4 for help with troubleshooting the problem.

iQ6 U-interface (RJ-11)

PIN	DESCRIPTION
1	N/C
2	N/C
3	LINE R
4	LINE T
5	N/C
6	N/C

Note that the polarity of the line connections doesn't matter.

ISDN Worldwide (outside the USA & Canada): The S/T Interface

The S-interface version of the iQ6 connects to the NT1 at one of its terminal jacks. The NT1 is sometimes generically called a NCTE（Network Channel Terminating Equipment）or, in some countries an NTBA.

iQ6 S-Interface (RJ-45)

PIN	DESCRIPTION
1	N/C
2	N/C
3	TRANSMIT TO NETWORK +
4	RECEIVE FROM NETWORK +
5	RECEIVE FROM NETWORK -
6	TRANSMIT TO NETWORK -
7	N/C
8	N/C

ISDN S-interface cable
Use only an 8-conductor RJ-45 style cable. Unshielded twisted pair Category 3 or better cable should be used.

ISDN S-Interface Cable (RJ-45)

PIN	COLOR	DESCRIPTION
1	WHITE/GREEN	PS3 POWER +/GROUND (OPTIONAL)
2	GREEN	PS3 POWER – (OPTIONAL)
3	WHITE/ORANGE	TRANSMIT TO NETWORK +
4	BLUE	RECEIVE FROM NETWORK +
5	WHITE/BLUE	RECEIVE FROM NETWORK -
6	ORANGE	TRANSMIT TO NETWORK -
7	WHITE/BROWN	PS2 POWER - (OPTIONAL) **
8	BROWN	PS2 POWER +/GROUND (OPTIONAL) **

TIA 568A Category 5 cable colors shown. Both ends are wired identically - not crossover.
*** The iQ6 does not use these power connections*

When fully wired, this cable has four twisted pairs "straight through" just as with cables used for Ethernet. Both ends are wired identically. The outside pairs are not required and may be omitted.

2.2.3 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. Depending upon the PBX and your preferences, this can be either ISDN BRI or POTS.

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 ISDN through PBX

Telos equipment has been used satisfactorily using BRI off the following PBX Switches:

USA and Canada

Nortel Meridian Option 11, 2 wire（U interface）. Protocol ID = 6. [user report]

Lucent/Avaya Definity, 2 wire（U interface）[user report]

Avaya IP Office（must use Q.931 mu setting on Telos）[user report]

Outside of the USA

Siemens HiCom series, 4 wire（S interface）[user report]

ETS300 Euro-ISDN is available from many PBX systems outside the USA and compatibility is generally good.

Unless your PBX is listed above, its support of US/Canada National ISDN-1 is not certain. We recommend you arrange a test for compatibility before committing to the purchase of the PBX ISDN module.

Contact Telos support for the latest information, or to report your experiences with other switches.

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.

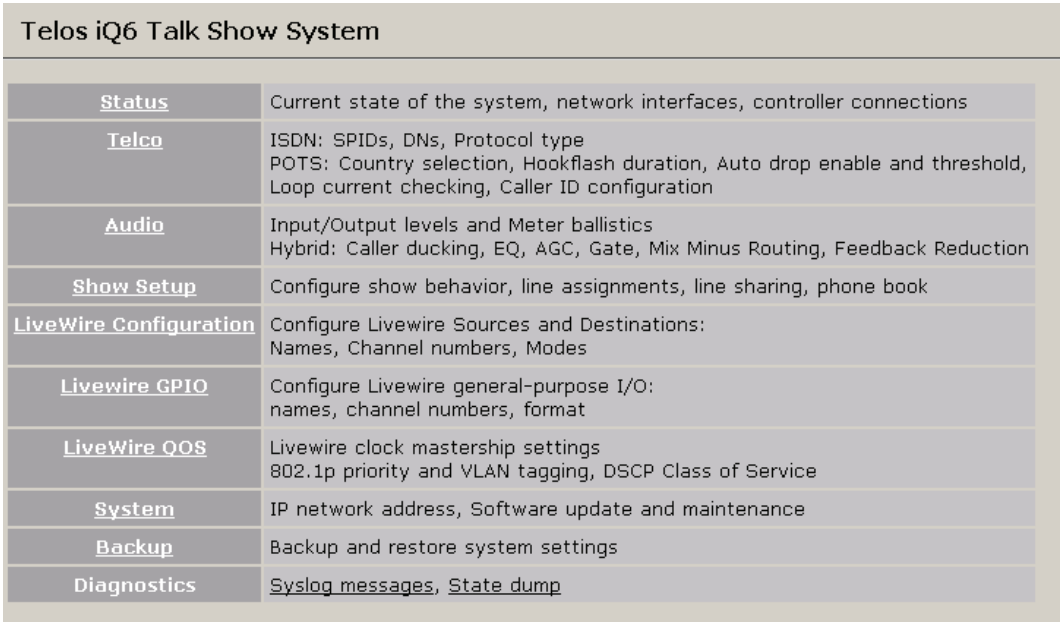
A few years ago, we advised that hybrid performance was best when equipment was connected directly to the lines from the Telco. Today, this is not always true. If the PBX is connected to the Telco using digital lines（such as T1, E1, PRI, S2M, etc）, performance through a digital PBX will generally be 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.

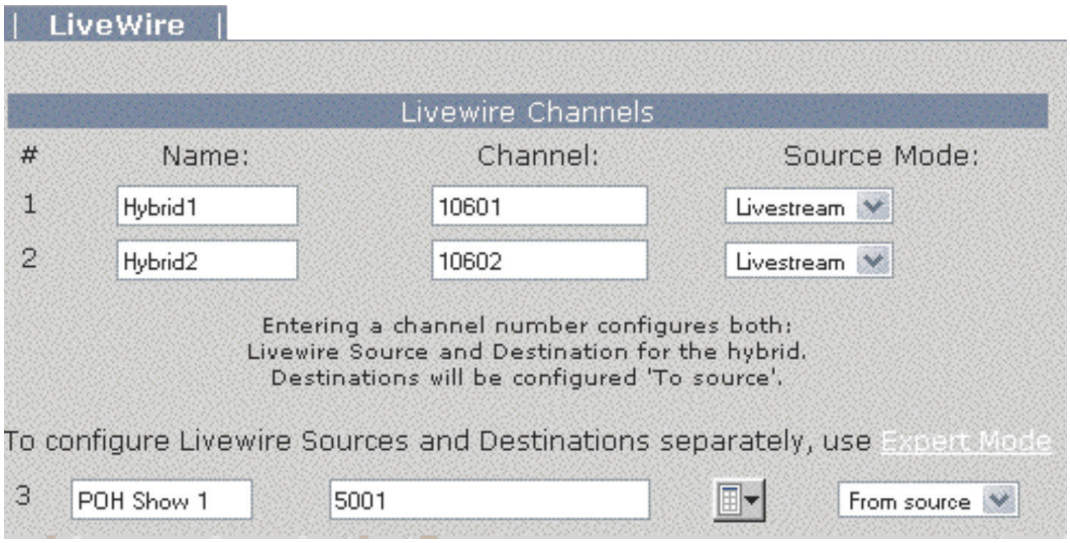
The other thing to investigate is whether the PBX offers 'Loop Current Wink' or 'CPC'（Calling Party Control）on the analog ports. Without this supervision, the iQ6 will not detect that a caller on hold has hung up, and your talent will get a recording "please hang up and dial again…" or dial tone when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



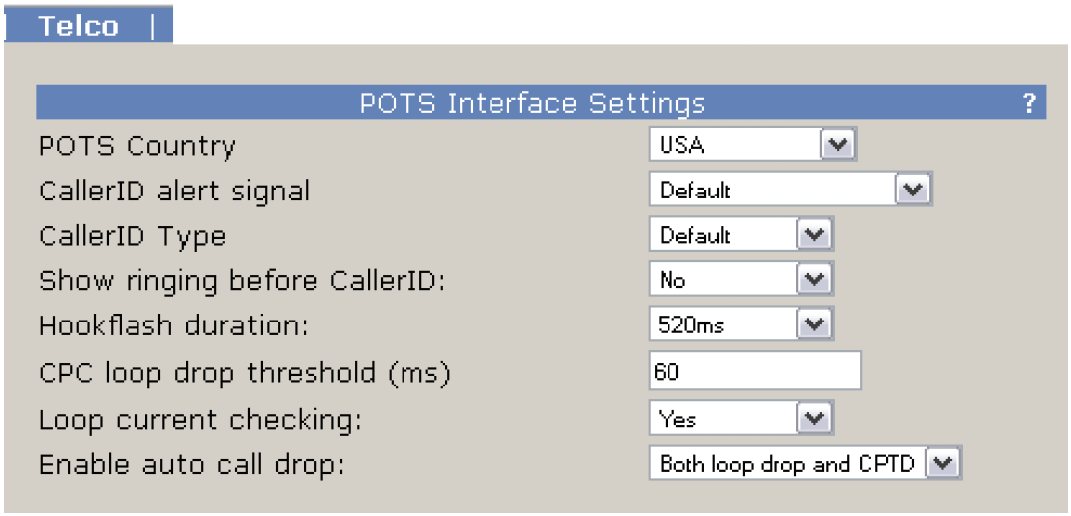
The Home screen is shown above. First select the **Livewire Configuration** link. *Username* is **user** and the *password* should be left blank.



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.

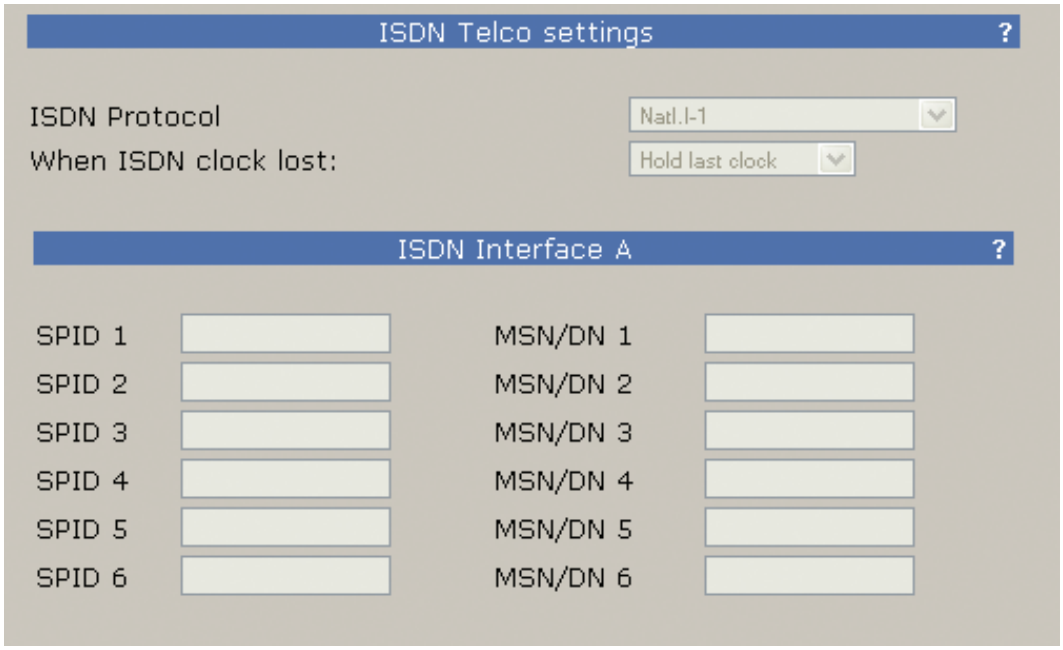
From the Telco page, you have the option to configure the POTS or ISDN interface.

POTS Interface



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.

ISDN Interface



Select the required ISDN protocol. For the USA and Canada, this will usually be Natl.I-1. For Europe and much of the rest of the world, it will be ETS300.

For Natl.I-1, enter the SPIDs for each voice channel given to you by the telephone company. Leave this blank for other protocols.

Enter the DN（Directory Number）or MSN for each voice channel. The DNs, sometimes called Listed Directory Numbers or LDNs, are your（usually 7-digit）phone number as listed in the telephone directory.

They normally do not include the area code.

In some cases, you may need to enter 10 digits for your DNs.

Click "**Apply**", then "**Reboot Now**" to activate the ISDN circuits.

Successful initialization of a line will be indicated by the idle line state on the front display of the iQ6. This state is shown with a single dot for each active line in the center row of the front display.

The same indicator is also available on the Status page of the web interface.

4 active lines shown initialized

Tips

Incorrectly entered SPIDs are the most common problem users in the USA & Canada have when placing ISDN calls. Compare the SPIDs you have entered with the SPIDs provided to you by the phone company to be certain that you have entered them correctly. Do not add anything to your SPIDs. Note that there are no hyphens, dashes (-), or spaces in SPIDs so the system will not allow you to enter any non-numeric characters. If your installer has included them in the SPID, ignore them.

If you have PTP (AT&T Point-to-Point) or ETS300 (Euro-ISDN) ISDN protocol, you should not enter anything in the SPID fields.

If you experience problems only with incoming calls, check your DNs.

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

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)

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

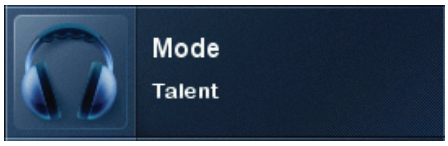
The Axia iQ console Telco expansion module is designed to interface with the iQ6.
Its basic controls are as follows.

- **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. This button does not function with ISDN phone lines.
- **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 dialed-out. 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

After selecting a line, press the **GO** button before any digits are dialed. The last dialed number appears in the line info field. Pressing **GO** again causes dialing to start.

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.

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.

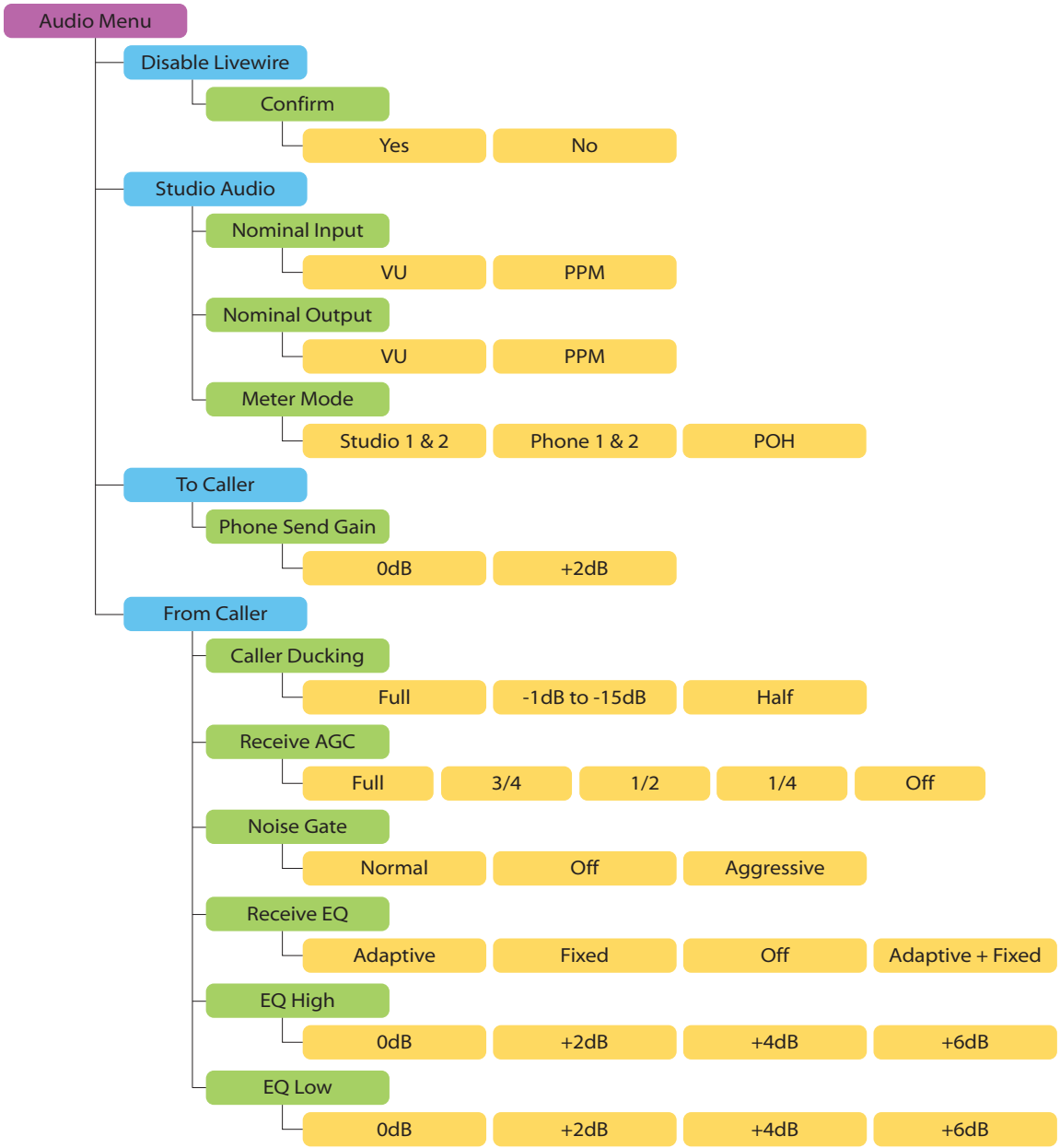
The status screen provides the state of each of the six telco lines (states are indicated by icons described in chapter 3), input state, digital sync, and lock.

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

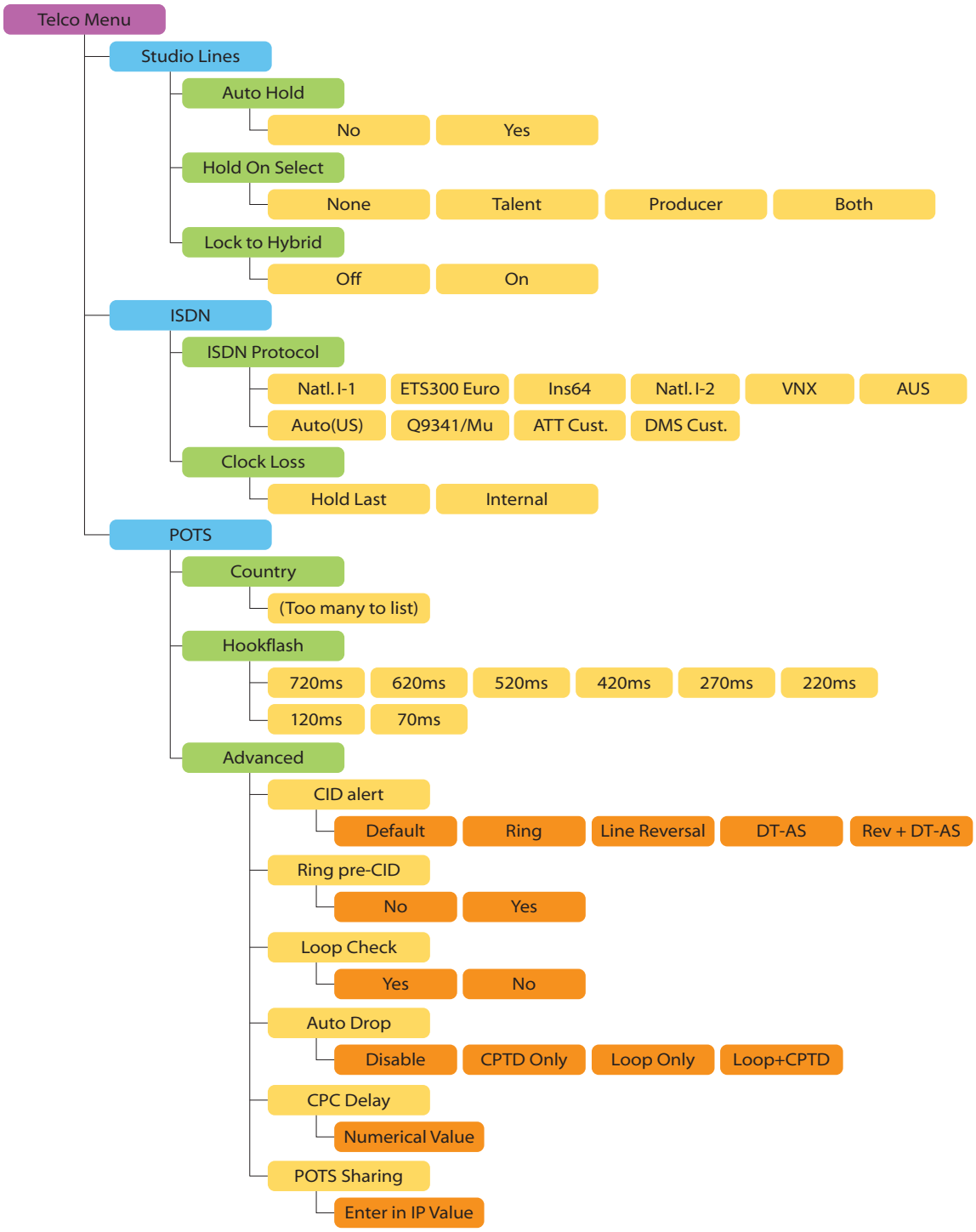
- **Audio** (audio settings and adjustments)
- **Telecom** (telco settings and adjustments)
- **System** (base system options)
- **Status** return to default status page

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

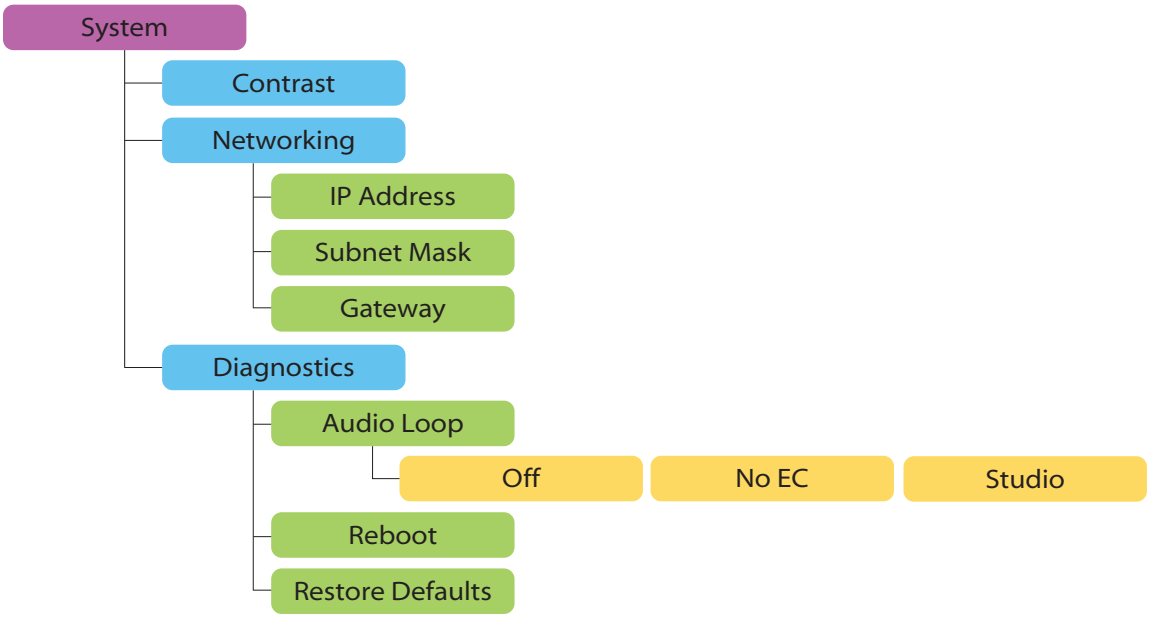
4.1.1 Audio Menu



4.1.2 Telco Menu



4.1.3 System Menu

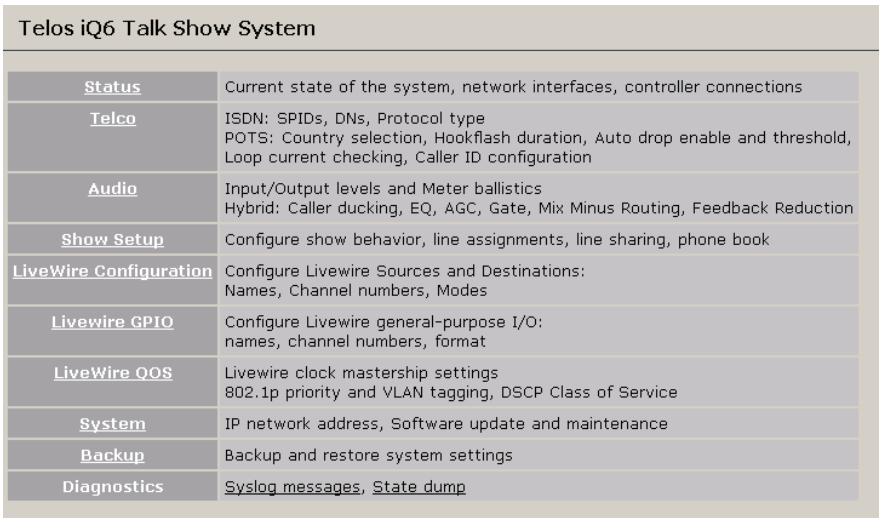


4.2 Web User Interface

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 ISDN DNs/SPIDs and 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:



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

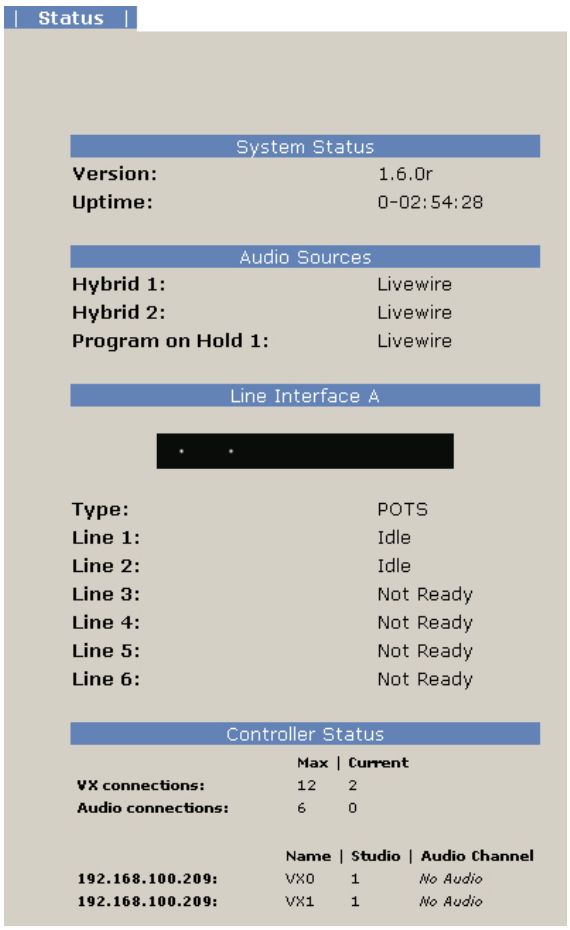
The default user name is "**user**" and the password is <blank> (no password).

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. The ISDN SPID and DN entries are only available on the web interface. (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.

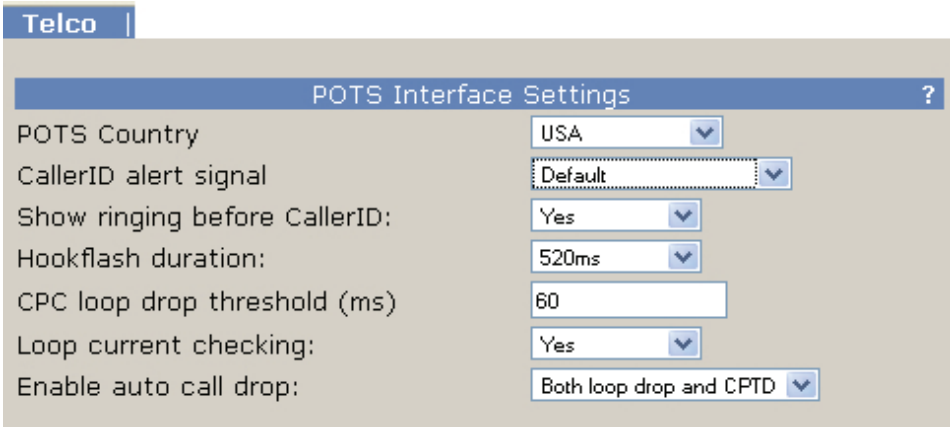
4.2.1 Status

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).



4.2.2 Telco

POTS Interface Settings



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.

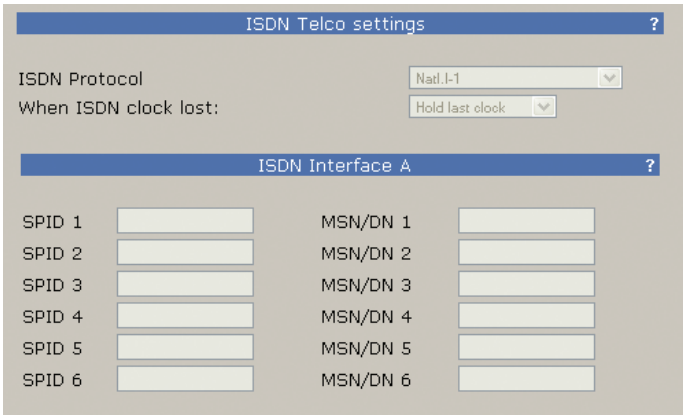
Note

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.

ISDN Interface Settings



ISDN Protocol

Selects the ISDN protocol, which must correspond to the ISDN circuits you are using. It will apply to all ISDN lines.

There are a number of choices, but the most common are described below.

For Europe and many other parts of the world

- ETS 300. This is the very common Euro-ISDN, according to the ETS 300 standard.

For USA & Canada and a few other parts of the world

- Natl. I-1. National ISDN-1 is the most common protocol for the USA & Canada and some other parts of the world. May also be used with lines conforming to National ISDN-2. See the notes about SPIDs, below.

- DMS Cust. For 'custom functional' lines on the Nortel DMS-100 switch. See the notes about SPIDs, below.

- AT&T Cust. This is the AT&T PTP protocol, sometimes called 'AT&T custom Point-to-Point'. This protocol generally does not allow two simultaneous voice calls. For this reason we recommend that the National ISDN-1 protocol be used if you are on the AT&T/Lucent 5ESS Telco switch.

If you are in Europe or most of South America, you should choose ETS 300.

If you have Net64 ISDN in Japan you should select INS64.

If you are in Hong Kong you should choose AT&T Cust.

If you are in the USA or Canada, use whichever protocol your phone company has provided. If you do not know, you may be able to guess...

Guessing the ISDN Protocol setting in Canada & the USA

- If you were given only a single phone number and no SPIDs try AT&T Cust

- If you were given 2 SPIDs with the format: 01+7 digit phone #+0 try AT&T Cust

- If you were given 2 SPIDs with the format: 01+7 digit phone #+000 or 01+7 digits +011 try Natl.I-1

- If you were given 2 SPIDs with the format: area code+7 digit phone #+4 digits（for example 0101, 0100, 0111, or 1111）try Natl.I-1. If that fails Try DMS Cust

- If you were given 2 SPIDs with the format: area code+7 digit phone #+1 digit（for example 1, or 0）try DMS Cust. If this fails try Natl.I-1

- If you were given 2 SPIDs with the format: area code+7digit phone #+2 digits（for example 01, 11, 02, or 00）try DMS Cust. If that fails try Natl.I-1

- If you were given 2 SPIDs with the format: area code+7digit phone #+3 digits（for example 000,001）try Natl.I-1. If that fails try DMS-Cust

- If you were given a single phone number and a single SPID with the format: of 01+7 digit phone#+0 try AT&T Cust（Do not enter this SPID）

- If you were given a single phone number and a single SPID with the format: of 01+7 digit phone#+00 try Natl.I-1. If that fails try AT&T Cust

- If you were given a single phone number and a single SPID with the format: of area code+7digit phone#+4digits（for example 0101, 0100, 0111, or 1111）Try Natl.I-1. If that fails try DMS

Note

After you enter (or change) your SPIDs or DNs, you must reboot. Therefore, once you have entered SPID1, SPID2, DN1, and DN2 you should turn the power off and then back on. Then go to the Status screen and verify Ready status for both channels.

Clock Loss

This setting is relevant only when there are ISDN lines that "go to sleep" or there is some other unusual ISDN Telco condition. Most users should leave this set to the default setting of **Hold Last**.

SPIDs and MSN/DNs

There will be entry locations for SPIDs and MSN/DNs for all line positions, whether needed or not. After entering the numbers, you must click **"Apply"** and then **"Reboot Now"** at the bottom of the page to activate the ISDN lines.

When you are using Natl. I-1 or DMS-100 for your ISDN Protocol mode, you must enter the correct Service Profile ID (SPID) numbers for each ISDN channel, normally two per physical line. The telephone company should tell you this number when they confirm your order. Often it is found written on the line jack or punch block.

Usually the SPID is the area code+phone number+0101. However, we have seen some that include a two-digit prefix and a two-digit suffix, and many other variations are possible. See Appendix 1 for a list of known working SPIDs. If this works, consider yourself lucky; if it doesn't work, there is no substitute for getting the correct SPID from your Telco!

ETS 300 users should disregard all references to SPIDs. Euro ISDN does not have SPIDs.

Notes on SPIDs

1. Incorrectly entered SPIDs are the most common problem USA users have placing ISDN calls. Compare the SPIDs you have entered with the SPIDs provided to you by the phone company to be certain that you have entered them correctly. Do not add anything to your SPIDs.
2. Note that there are no hyphens, dashes (-), or spaces in SPIDs, so the system will not allow you to enter any non-numeric characters. If your installer has included them in the SPID, ignore them.
3. If you have any questions about your SPIDs, call your phone company.
4. If you have AT&T Point-to-Point or ETS300 Euro-ISDN Protocol mode, you should not enter anything in the SPID fields.

MSN – Multiple Subscriber Number

This item, if required, must be configured from the web interface. Used only when you are using the ETS 300 Euro-ISDN Telco mode and another ISDN device is sharing the line with the iQ6. MSN1 corresponds to the first B channel. This is simply one of your assigned telephone numbers. Normally, for ETS 300 these should be left blank.

DN – Directory Number

This item, if required, must be configured from the Web interface. When you are using an ISDN protocol other than ETS 300 or AT&T Cust. PTP, you must enter the 'Directory Number' (DN) here. This is simply the telephone number associated with the channel.

Tip

Each directory number will correspond to one SPID. You must enter them to reflect this relationship. The DN corresponding to SPID1 must be entered as DN1 while the DN corresponding to SPID2 must be entered as DN2.

If you experience problems only with incoming calls check your DNs. Normally 7 digits should be entered, but you might need to enter 10 digits.

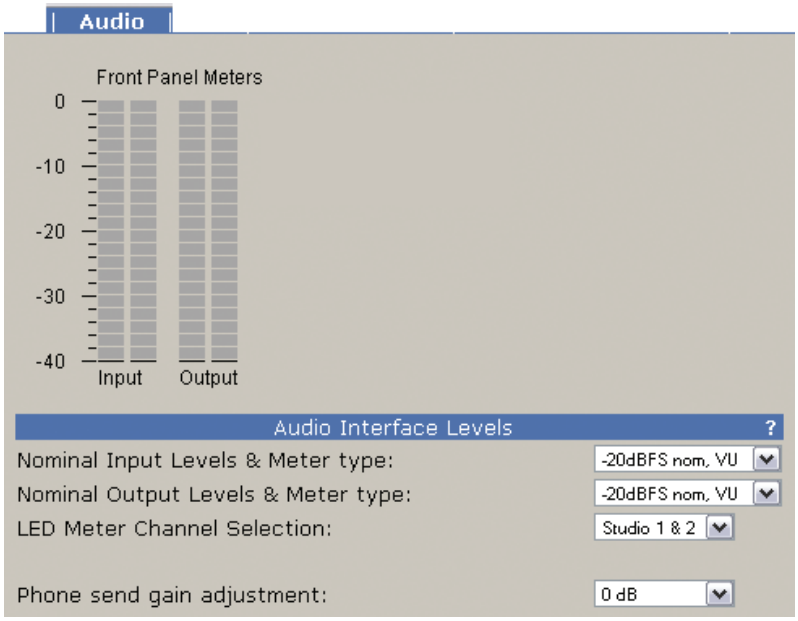
In some cases, lines in a hunt group will require that the same (primary Listed Directory Number) be entered for all lines that are part of the hunt group.

Some versions of the Telos Zephyr and Zephyr Express do not require that Directory Numbers be entered. This is not true for the Telos iQ6 and other newer products. You must correctly enter these numbers for the system to work.

IMPORTANT

After you enter (or change) your SPIDs or DN's you must restart the system to use the new information. Therefore, once you have entered SPIDs, DNs, or MSNs cycle power off and then back on. Then go to the ISDN status screen and verify "Ready" status for all channels.

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.

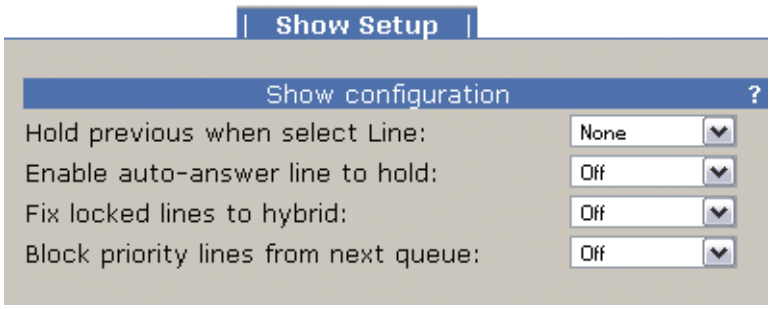
(Additional) High EQ High

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



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

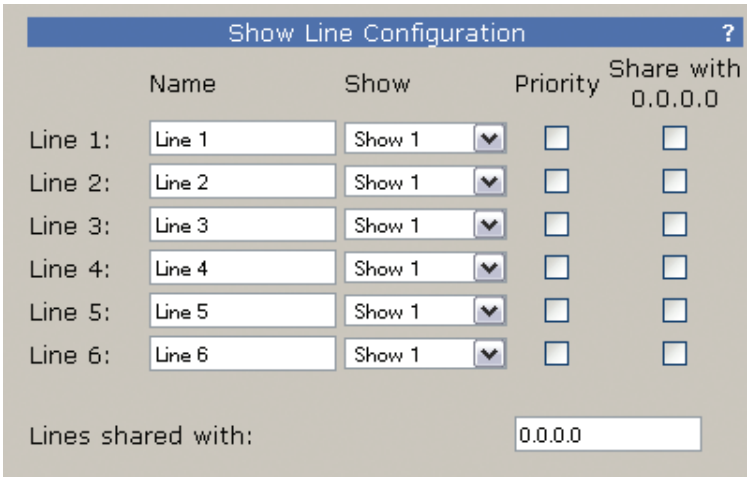
Enables the auto answer feature. Non-priority ringing lines are automatically answered and put on hold.

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

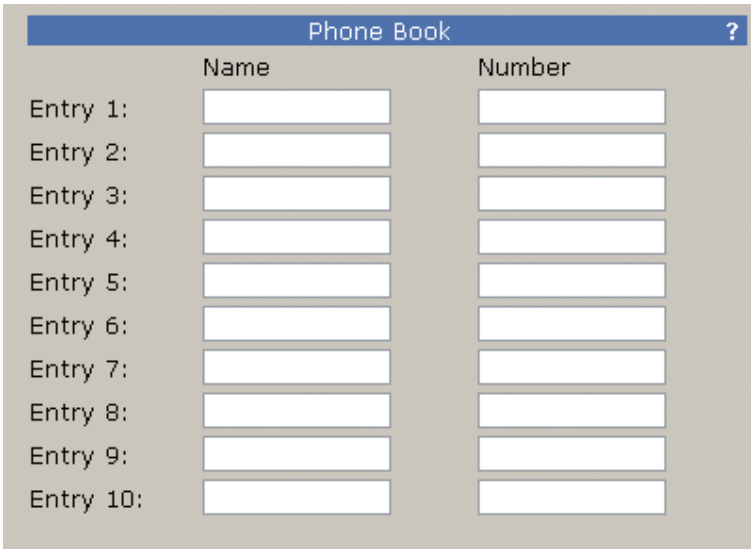


Line Sharing

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.



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

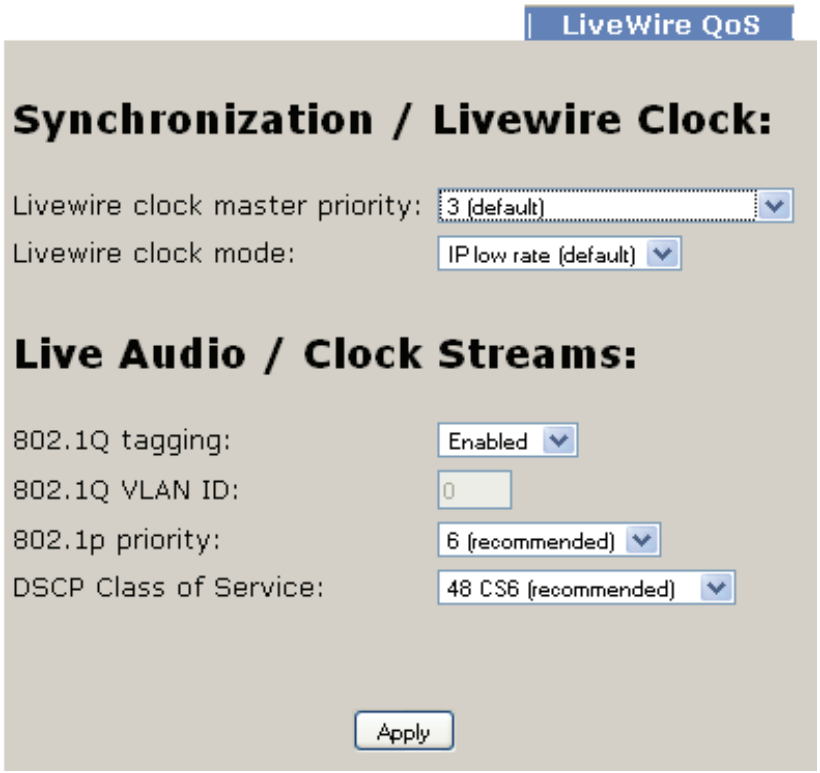
PIN	DESCRIPTION
GPI-1	
GPI-2	
GPI-3	
GPI-4	
GPI-5	
GPO-1	User Out (Dump)
GPO-2	Ringing Out
GPO-3	Priority Ringing Out
GPO-4	Record Start
GPO-5	Record Stop 1

PORT 2

PIN	DESCRIPTION
GPI-1	Ring Mute trigger
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 Livewire channel numbers that are used to carry the audio streams.

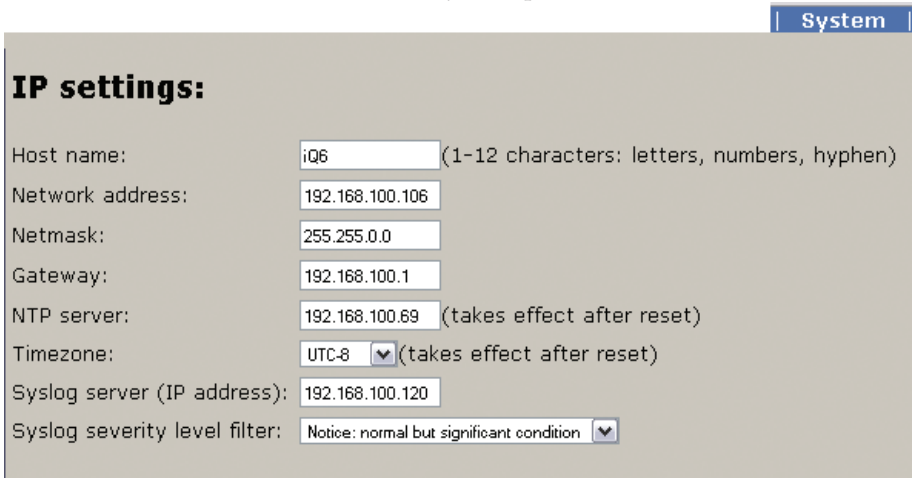
4.2.7 Livewire QoS



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.



IP Settings

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

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 ⇩ to highlight **System**, press（OK）
2. Press ⇩ to highlight **Networking**, press（OK）
3. Press ⇩ to highlight **IP address**, press（OK）
4. Use ⇩ to delete and move the curser back.
5. Once you press ⇧, you begin to enter a value at curser position. Use ⇩ and ⇧ to select the desired value. Press OK to accept digit.

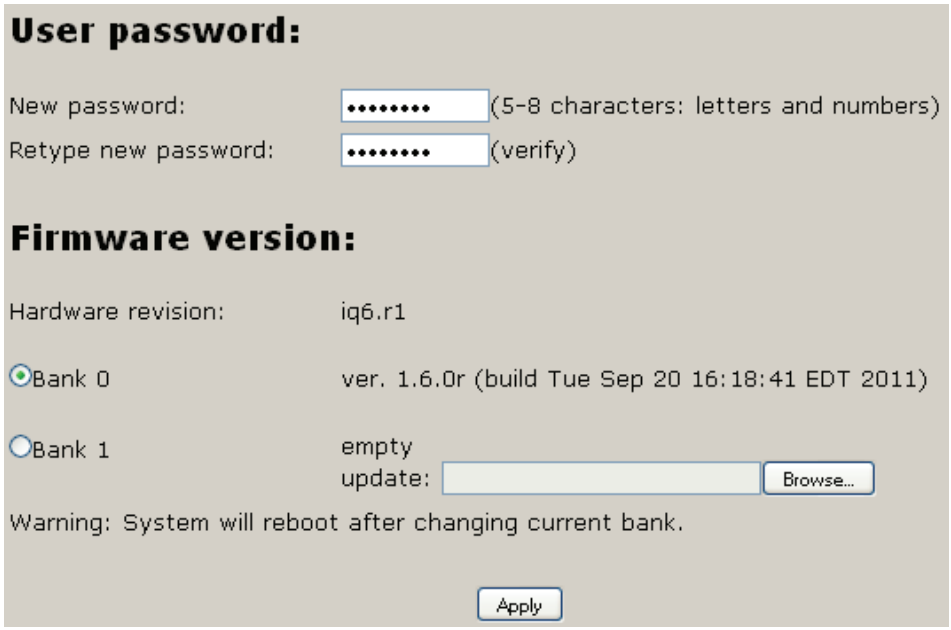
   To delete the previous incorrect digit press ⇩.
6. Press ⇧ until the desired digit is displayed.
7. **Once all characters are entered, press OK twice to save your entry**

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

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.

5 Technical Data & Troubleshooting

5.1 Overview

A consequence of modern surface-mount construction is that it is frequently no longer possible for localre-pairs 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 ISDN 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 Troubleshooting ISDN Problems

Status Screen for ISDN troubleshooting

This screen shows the status of the ISDN connection. You can determine if the line is connected and whether the line and the Telos device are successfully configured. In addition, information about calls is displayed here. One of the following status words will be shown for each active ISDN channel:

Deactivated
No line connected, or connected to wrong jack on the Telos. OR, ETS 300 permits the Telco to put a line to sleep. Deactivated will be displayed when this occurs.

SPID Pending
The SPID has been sent to network（DMS Cust. And Natl I-1 only）. Network has not approved SPID.

SPID Error
The SPID is incorrect or line is grossly misconfigured

Ready
The unit sees a live ISDN S interface（ie external NT1 is present or internal NT1 has completed startup）. If the SPIDS are present, they have been sent and approved.

Pending Seize
Outgoing call in process; Request for call has been sent to Telco switch.

Seized
Outgoing call in process; Telco switch has acknowledged the call request.

Call Proceeding
Outgoing call in process; Telco switch waiting for routing info from network.

Outgoing Ring
Outgoing call in process; far end has generated a call alert signal

Disconnecting
Request to disconnect has been made. Awaiting acknowledgment.

Incoming Ring
An incoming call request has been detected and is being processed.

Accepting Call
iQ6 has accepted an incoming call and is waiting for the Telco switch to connect it.

Connected
All signaling is complete - the voice channel is available for use.

A caveat regarding the Ready indication: If you have this indication with the iQ6 Telco mode set to AT&T Cust and you cannot dial, it is likely that your line is Natl. I-1 or DMS-100. The AT&T Custom PTP protocol does not require SPIDs, so the iQ6 does not send or check them; it just assumes the line is OK in this regard, so it gives the ready indication, though the line is actually waiting for the SPIDs.

Note

It is normal for Euro-ISDN ETS300 lines to go to the Deactivated state when not used. If the system can dial normally under these circumstances, this indication is not a cause for concern.

Is the problem only with one BRI circuit or does it move when you switch lines around?

If the problem remains with the line, then call your Telco. On the other hand, if a line that works in one interface fails to work in another interface of the same iQ6 then you have a problem with your ISDN interface card (or possibly elsewhere in the iQ6).

The LED indicator of the status of the ISDN connection is on the rear panel near each of the the U jacks. If the circuit is inactive, the LED will remain in the rapidly flashing state. This state indicates a loss of the ISDN BRI circuit at the lowest (physical) level. Try another connecting cable or another line.

If the NT1 can contact the central office, the LED will blink slowly (about 1 Hz). If the LED remains in this state there is an internal ISDN problem or you could be connected to the wrong jack on the back (Users in USA & Canada should be using the lower, RJ11 style jack). The LED will come on solid when all handshaking is completed and the basic (low level) line connection is good.

If the LED stays on, with the ISDN disconnected, or does not come on at all after cycling power, there is a hardware problem.

Condition: Can't Get ISDN Ready Indication

First try disconnecting the ISDN circuit(s) in question and waiting a full 60 seconds (leave the iQ6 on) and then reconnect the circuit(s).

What kind of ISDN do you have, and is the iQ6 set to match it?

The telephone company should have given you the ISDN protocol when the line was ordered.

If you find yourself in the field without this information, it is usually possible to piece together what you need.

- If you are given SPIDs, you probably have Natl ISDN 1 or DMS 100 for your ISDN protocol. One SPID means one channel was activated, rather than the usual two.
- No SPIDs and only one phone number generally means AT&T PTP protocol. You must not have SPIDs entered if using this protocol.
- In Europe, you have either ETS300 (EuroISDN) or the local protocol, such as ITR6 in Germany or VN2 in France. The iQ6 only works with ETS300. In the case of Natl.I-1 or 2, and DMS-100 carefully verify that you have the correct SPID from the Telco, and that you have properly entered it into the system. It must be correct in order for anything to work! We've seen a few cases where it was given incorrectly by Telco personnel, so it might be a good idea to re-check with them if things are pointing in this direction.

Hot Tip

If you suspect the SPIDs given you are wrong, or the basic line provisioning (configuration) is incorrect, call the Telco and ask to speak to someone who can "log into the switch". They will probably need to have someone call you back. Only once you have this person on the line, and they are logged into the switch, ask them to look at the SPIDs programmed there.

Also get this person's fax number and send them the ISDN ordering information from the Appendix of this manual.

Sometimes the information in the business office computer is wrong. By verifying what is programmed into the actual switch, you can save time. The AT&T PTP protocol generally does not allow two simultaneous voice calls. For this reason we recommend that the National ISDN-1 protocol be used if you are on the AT&T/Lucent 5ESS Telco switch.

We've included some useful diagnostic aids in the system to help you determine where a problem is originating. A good starting point is the ISDN status, accessible on the LCD by pressing any menu button twice.

Condition: System says Ready but attempted calls do not complete (goes back to Ready or stays at Seize pending)

Are your SPIDs and Telco setting correct?

Can you dial from Line 1 to Line 2?

If you can successfully call from one line to the other, you know that your SPIDs and ISDN protocol settings for that line are correct and that the ISDN circuit itself is ok. Try a call to another local number.

If you cannot complete this call try dialing with a 9, 8, 7, or 6 before the number. Sometimes ISDN lines are set up as Centrex service even if you did not request this.

Is your long-distance carrier OK?

If you can successfully call locally, but long-distance calls don't go through, the iQ6 usually displays Far end disconnect, or No route available as the ISDN "Cause" phrase.

You may try another by dialing the appropriate 10XXXXX prefix（in the USA & Canada only）.

In the USA you can determine who your long distance carrier is by dialing 1+700+555+4141.

If Intra-LATA long distance competition has reached your area you can determine your Intra-LATA long distance carrier by dialing 1+area code+700+4141.

What does the ISDN "Cause" phrase say?

This phrase, which will appears on the ISDN protocol log after dialing, comes directly from the telephone company equipment, and can be valuable for troubleshooting. Generally, these phrases are self-explanatory. Getting them means you are at least talking to the Central Office equipment.

There is a full description of the phrases in Section 6.7.

Condition: Intermittent ISDN Problems

These are among the most difficult kinds of problems to solve. Rest assured, however, that we have yet to lose one!

First try disconnecting the ISDN line and waiting a full 60 seconds (leave the iQ6 on) and then reconnect the line.

Does the problem occur on all of the ISDN BRI circuits or just a single BRI circuit (2 "lines")?

If the problem is limited to a single BRI circuit (ie both "lines" on that BRI are showing the problem) then the problem may be with that BRI line or with that ISDN interface card. Move the BRI line in question to a different interface card (don't forget to reprogram the SPIDs).

Are the problems confined to incoming calls only?

In this case, check the ISDN (Telco) protocol setting in the TEL. If you believe your line is DMS Custom try the National ISDN setting, or vice versa.

Are the DNs entered correctly? If so, try using 10 digits rather than 7 digits, or vice versa. What is consistent about the problem (look for patterns)?

Keep watching for a pattern. Enlist the help of the users. We've seen apparently random problems that we eventually discovered only occurred when it rained, or on a certain day each week, or at a certain time of day, or only when another line was in use. In other cases, it was related to temperature or dirty AC power. Assume nothing and suspect everything. Leave no stone unturned while searching for the answer.

Any other ISDN troubleshooting tips?

An important tool for troubleshooting incompatibility/programming problems is the iQ6's integrated ISDN analyzer. This enables you to look at the transactions between the Telco line and the iQ6, so that a judgment can be made as to whether the problem lies within the iQ6 or with the programming of the Telco switch.

The usual cause for an inability to dial is a Telco problem, such as a configuration error.

If you are convinced that the problem lies with the ISDN line configuration, you may be able to convince the Telco to have an engineer look at the line with an analyzer. This works much like the iQ6 internal analyzer, but with more detail, and in a format that is more readily understandable by telephone people. This procedure may sometimes be the only way to get attention to a line problem. It is hard to deny evidence provided by their own gear.

In rare cases, there could be a software bug in the iQ6 that reacts badly to something coming from the ISDN line. Telos customer support will communicate the problem to our engineering for resolution.

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. ISDN Line interface card
3. 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.

5.6 ISDN Cause Phrases/Values

When there is an ISDN problem in the network, a phrase appears in the ISDN protocol log that communicates information about the cause of the trouble.

CLASS 1.1 (normal)

Cause No. 1 - Check number, redial (unallocated number)

This cause indicates that the called party cannot be reached because, although the called party number is in a valid format, it is not currently allocated（assigned）.

Cause No. 2 - No route to network /Prefix 0 dialed in error

This cause indicates that the equipment sending this cause has received a request to route the call through a particular transit network which it does not recognize. The equipment sending this cause does not recognize the transit network either because the transit network does not exist or because that particular transit network, while it does exist, does not serve the equipment which is sending this cause.

or

This cause indicates the prefix 0 is invalid for the entered number

Cause No. 3 - No route to dest./Prefix 1 dialed in error

This cause indicates that the called party cannot be reached because the network through which the call has been routed does not serve the destination desired. This cause is supported on a network-dependent basis.

or

A 1 was dialed when not required. Redial without the 1.

Cause No. 4 – No prefix 1/Special Information Tone

The prefix 1 is not required for this number

or

This cause indicates that the called party cannot be reached for reasons that are of a long term nature and that the special information tone should be returned to the calling party.

Cause No. 5 – Misdialed trunk prefix (National use)

This cause indicates the erroneous inclusion of a trunk prefix in the called party number.

Cause No. 6 – Channel unacceptable

This cause indicates a called user cannot negotiate for a B-channel other than that specified in the SETUP message.

Cause No. 7 – Call awarded and being delivered on an established channel

This call indicates the user has been awarded the incoming calls, and that call is being connected to a channel already established to that user for similar calls（e.g. packet-mode, X.25 virtual calls, etc）.

Cause No. 8 – Call is proceeding/Preemption

Call in process, please standby.

or

This cause indicates a call has been preempted.

Cause No. 9 – Preemption, reserved

This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 10 – Prefix 1 not dialed [DMS Custom only]

This cause indicates that the prefix 1 was not included in the called party address and is required.

Cause No. 11 – Excessive Digits Received, call is proceeding

More digits were dialed than expected. Called number has been truncated to the expected number.

Cause No. 14 – Excess digits received, call is proceeding

More digits were dialed than expected. Called number has been truncated to the expected number.

Cause No. 16 - Far end disconnect (Normal call clearing)

This cause indicates that the call is being cleared because one of the users involved in the call has requested that the call be cleared. Under normal situations, the source of this cause is not the network.

Cause No. 17 - Busy, try again later (User busy)

This cause is used to indicate that the called party is unable to accept another call because the user busy condition has been encountered. This cause value may be generated by the called user or by the network. In the case of user determined user busy it is noted that the user equipment is compatible with the call.

Cause No. 18 - No far end response (No user responding)

This cause is used when a called party does not respond to a call establishment message with either an alerting or connect indication within the prescribed period of time allocated（before timer T303 or T310 has expired）.

Cause No. 19 - No answer (No answer from user, user alerted)

This cause is used when the called user has provided an alerting indication, but not a connect indication within a prescribed period of time（before timer T301 has expired）.

Cause No. 20 – Subscriber absent

This cause is used when a mobile station has logged off, radio contact is not obtained with a mobile station or a personal telecommunications user is temporarily not addressable at any user-network interface.

Cause No. 21 - Call rejected

This cause indicates that the equipment sending this cause does not wish to accept this call, although it could have accepted the call because the equipment sending this cause is neither busy nor incompatible.

This cause may also be generated by the network, indicating that the call was cleared due to a supplementary service constraint.

Cause No. 22 - Number changed

This cause is returned to a calling party when the called party number indicated by the calling party is no longer assigned, The new called party number may optionally be included in the diagnostic field. If a network does not support this cause, cause no: 1, unallocated (unassigned) number shall be used.

Cause No. 26 – Non-selected user clearing

This cause indicates the user has not been awarded the incoming call.

Cause No. 27 – Destination out of order

This cause indicates that the destination indicated by the user cannot be reached because the interface to the destination is not functioning correctly. The term "not functioning correctly" indicates that a signal message was unable to be delivered to the remote party; e.g. a physical layer or data link layer failure at the remote party, or user equipment off-line.

Cause No. 28 - Incorrect number (invalid number format, address incomplete)/Special intercept announcement

This cause indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.

or

This cause indicates the user should be returned a Special Intercept Announcement.

Cause No. 29 – Facility rejected/Special Intercept announcement: undefined code This cause indicates a facility requested cannot be provided by the network.

or

This cause indicates that a user in a special business group (i.e. Centrex) dialed an undefined code.

Cause No. 30 – Result of a STATus ENQuiry/Special intercept announcement: number unassigned

This cause is included in the Status Message when the reason for sending the Status Message was previous receipt of a Status Enquiry message.

or

This value indicates that a user from outside a basic business group (i.e. Centrex) has violated an access restriction feature.

Cause No. 31 - Network disconnect (Normal, unspecified)/Special intercept announcement:

Call blocked because of group restrictions

This cause is used to report a normal event only when no other cause in the normal class applies.

or

This value indicates that a user from outside a basic business group (i.e. Centrex) violated an access restriction feature

CLASS 1.2 (resource unavailable)

Cause No. 34 - No circuit available (circuit/channel congestion)

This cause indicates that there is no appropriate circuit/channel presently available to handle the call.

Cause No. 35 – Call Queued [5ESS Custom only]

This indicates the call has been queued.

Cause No. 38 - Net out of order

This cause indicates that the network is not functioning correctly and that the condition is likely to last a relatively long period of time e.g. immediately re-attempting the call is not likely to be successful.

Cause No. 39 – Permanent frame mode connection is out-of-service [Q.850 only]

This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 40 – Permanent frame mode connection is operational [Q.850 only]

This cause indicates a calls has been preempted due to the circuit being reserved for reuse.

Cause No. 41 – Net problem, redial (Temporary Failure)

This cause indicates that the network is not functioning correctly and that the condition is not likely to last a long period of time; e.g. the user may wish to try another call attempt almost immediately. May also indicate a data link layer malfunction locally or at the remote network interface or that a call was cleared due to protocol error(s) at the remote network interface.

Cause No. 42 – Net busy, redial (Switching Equipment Congestion)

This cause indicates that the switching equipment generating this cause is experiencing a period of high traffic.

Cause No. 43 – Access/user information discarded

This cause indicates that the network unable to deliver user information (i.e user-to-user information, low level compatibility or sub-address, etc) to the remote users as requested.

Cause No. 44 – No channel available (Requested circuit/channel not available)

This cause is returned when the circuit or channel indicated by the requesting entity cannot be provided by the other side of the interface.

Cause No. 46 – Precedence call blocked [Q.850 only]

This cause is used to indicate that there are no pre-emptable circuits or that the called user is busy with a call of equal or higher pre-emptable level.

Cause No. 47 – Resource unavailable/New Destination

This cause is used to report a resource unavailable event only when no other cause in the resource unavailable class applies.

or

This cause is used to indicate that the original destination is unavailable and to invoke redirection to a new destination.

Cause No. 49 – QOS unacceptable

This cause is returned when the Quality of Service is unacceptable.

CLASS 1.3 (service or option not available)

Cause No. 50 – Requested facility not subscribed

The cause is used to report that the user cannot use this feature because s/he has not subscribed to it.

Cause No. 51 – Bearer capability incompatible with service request

This cause indicates a user request for action was rejected because the action was incompatible with the capability of the call.

Cause No. 52 – Outgoing calls barred

This cause indicates that because of call screening provided by the network, the calling user is not permitted to make a call.

Cause No. 53 – Service operation violated

This cause indicates that the user has violated the service operation.

Cause No. 54 – Incoming calls barred

The user will not accept the call delivered in the SETUP message.

Cause No. 55 – Incoming calls barred within CUG at destination

This cause indicates that incoming calls are not permitted to members if this Closed User Group

Cause No. 57 – Bearer capability (Data/voice) not authorized.

This cause indicates that the user has requested a bearer capability which is implemented by the equipment which generated this cause but the user is not authorized to use it. This is a common problem caused by wrong Telco provisioning of the line at the time of installation.

Cause No. 58 – Bearer capability not presently available

This caused indicates the users has requested a bearer capability which has been implemented by the equipment, but is not available at this time.

CLASS 1.4 (service or option not implemented)

Cause No. 62 – Inconsistency in designated outgoing address information/Service Not Authorized [VN4]

This cause indicates an inconsistency in designated outgoing address information and subscriber class.

or

This cause indicates that a Terminal Diversion in a VN4 network has been rejected.

Cause No. 63 – Service or option not available, unspecified

This cause is used to report a service or option not available, only when no other cause in this class applies.

Cause No. 65 - Bearer Capability not implemented (Incompatible bearcap)

This cause indicates that the equipment sending this cause does not support the bearer capability requested.

Cause No. 66 – Channel type not implemented

This cause is returned when the called party has reached a channel type not supported.

Cause No. 69 – Requested facility not implemented

This cause indicates that the network（or node）does not support the requested bearer capability

and therefore cannot be accessed at this time.

Cause No. 70 - Restricted only

This cause indicates that the calling party has requested an unrestricted bearer service but the equipment sending this cause only supports the restricted version of the requested bearer capability.

Cause No. 79 – Service or option not implemented, unspecified

This cause is used to report a service or option not implemented event only when no other cause in this class applies.

CLASS 1.5 (invalid message; e.g. parameter out of range)

Cause No. 81 – Invalid call reference value

This cause indicates that the equipment sending this cause has received a message with a call reference which is not currently in use on the user-network interface. This value only applies if the call reference values 1 or 2 octets long and is not the global call reference.

Cause No. 82 – Identified channel does not exist

This cause is sent when the equipment sending this cause has received a request to use a channel not active on the interface for a call

Cause No. 83

A suspended call exists, but this call's identity does not

Cause No. 84 – Call identity in use

This cause indicates that a call identity is in use.

Cause No. 85 – No call suspended

This cause indicates that no call is suspended.

Cause No. 86 – Requested call identity cleared

This cause indicates that the call having the requested called entity has cleared.

Cause No. 87 – Destination not a member of CUG

This cause indicates that the address of an outgoing call attempt is not a member of the Closed User Group.

Cause No. 88 - Incompatible destination

This cause indicates that the equipment sending this cause has received a request to establish a call which has low layer compatibility, high layer compatibility or other compatibility attributes (e.g. data rate, DN subaddress) which cannot be accommodated. This call can also be returned by a switch to a CPE when trying to route a call to an incompatible facility, or one without data rate.

Cause No. 90 – Destination address incomplete/Non-existent CUG

This cause indicates that the destination address is missing or incomplete

or

Indicates a non-existent Closed User Group

Cause No. 91 - Invalid transit network selection

This cause indicates that an Invalid transit network selection has been requested.

Cause No. 92 – Invalid supplementary service parameter [VN4 only]

Used to indicate an invalid parameter for supplementary service

Cause No. 95 – Invalid message, unspecified

This cause indicates that the entity sending this cause has received and Invalid message, only when no other cause in this class applies.

CLASS 1.6 (protocol error; e.g. unknown message)

Cause No. 96 – Mandatory Info missing

This cause indicates that the equipment sending this cause has received a message which is missing an information element which must be present in the message before that message can be processed.

Cause No. 97 - Message type non-existent or not implemented

This cause indicates that the equipment sending this cause has received a message with a message type it does not recognize either because this is a message not defined, or defined and not implemented by the equipment sending this cause.

Cause No. 98 – Message not compatible

This cause indicates that the message received is not compatible with the call state or the message type is non-existent or not implemented.

Cause No. 99 – Info non-existent or not implemented, call processed

This cause is sent when the equipment sending this cause has received a message which includes the information elements not recognized because the information element identifier is not define or it is defined but not implemented by the equipment sending the cause. However, the information element is not required for the equipment sending the cause to process the message.

Cause No. 100 – Invalid information element contents

This cause is returned when the equipment sending this cause has received an information element that

it has implemented; however one or more field of the information element are coded in such as way (e.g., truncated, invalid extension bit, invalid field values, etc) that has not been implemented by the equipment sending this cause.

Cause No. 101 – Message not compatible with call state/Protocol error threshold

This cause is returned when the equipment sending this cause has received a message that procedures indicate is not a permissible message to receive at this time.

or

This cause indicates that the switch sending this cause is clearing the call because a threshold has been exceeded for multiple protocol errors during an active call.

Cause No. 102 - Timeout disconnect (Recovery on timer expiration)

This cause indicates that a procedure has been initiated by the expiry of a timer in association with error handling procedures.

Cause No. 103 – Parameter non-existent or not implemented – passed on (National use)/ Mandatory Information Element of incorrect length [VN 4 only]

This cause indicates that the equipment sending this cause has received a message which includes parameter(s) not recognized because the parameter(s) are not defined of are defined but not implemented by the equipment sending this cause. This cause indicates that the parameter(s) were ignored. In addition, if the equipment sending this cause is an intermediate point, then this cause indicates that the parameter(s) were passed unchanged.

or

This cause indicates that the equipment sending this cause has received a message with a mandatory information element of incorrect length.

Cause No. 110 – Message with unrecognized parameter discarded [Q.850 only]

This cause indicates that the equipment sending this cause has discarded a received message that includes a parameter that is not recognized.

Cause No. 111 – Protocol Error, unspecified

This cause is used to report a protocol error event only when no other cause in this class applies. Among other things, this cause may be displayed if you failed to dial a "9" or "8" for an outside line, if required. Also may be returned if you have some types of restrictions as to the number of calls, etc.

Cause No. 113 – Bearer service unavailable [VN 4 only]

This cause indicates that the bearer service requested is not available or is unimplemented.

Cause No. 114 – End-to-End Information transfer impossible [VN 4 only]

This cause indicates that the requested network does not support the user end-to-end transfer capability.

CLASS 1.7 (interworking class)

Cause No. 126 – Switch to conversation mode [VN 4 only]

This cause is used to indicate that conversation mode has begun.

Cause No. 127 - SW56 disconnect (Internetworking, unspecified)

This cause indicates that an interworking call (usually a call to SW56 service) has ended. May also be seen in the case of a non specific rejection by your long distance carrier (try again at a different rate)

5.7 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 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.

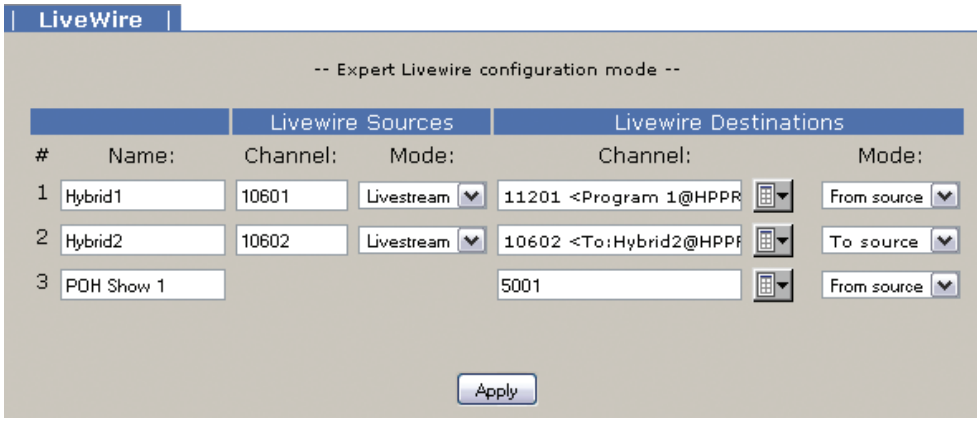
The iQ6 has three audio "inputs". The front panel OLED display shows whether audio is present on these inputs. As mentioned in section 4.2.8, the bottom left corner of the display shows a Livewire logo for a valid stream, or a circle for a invalid stream. What is a valid stream? A valid stream is a network 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.



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).

6 Specifications

General

- Interface for connecting three ISDN BRI (S0) or 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

ISDN Telephone Line Connectivity

Protocol Compatibility:

- National ISDN-1 and 2
- DMS-100 Custom Functional
- AT&T 5ESS Custom Point-to-Point
- Euro-ISDN conforming to the NET 3/ETS300 protocol
- Q. 931 µ-law

Interface (one of the following):

- **USA & Canada** Integrated NT1 for direct connection to ISDN line via the 2-wire U-interface (6-position/2-pin RJ-11 connector). 2B1Q line coding.
- **Worldwide** 4-wire S interface (8-position/8-pin RJ-45 connector)

Telephone Coding Modes

- µ-Law (ISDN Proto set to Natl I-1, AT&T Custom, Q.931µ-law or DMS Custom)
- A-Law (ISDN Proto set to ETS-300)

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

7 Warranty and Application Caution

iQ6 LIMITED WARRANTY

This Warranty covers "the Products," which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp.,d/b/a Telos Systems (hereinafter "Telos Systems").

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of two years from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

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

In no event will Telos Systems, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Telos Systems within the above-stated warranty period and warranty coverage must be authorized by Telos Systems. If Telos Systems authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Telos Systems, 1241 Superior Ave., Cleveland, Ohio 44114.

Telos Systems at its option will either repair or replace the Product and such action shall be the full extent of Telos Systems' obligation under this Warranty. After the Product is repaired or replaced, Telos Systems will return it to the party that sent the Product and Telos Systems will pay for the cost of shipping.

Telos Systems' authorized dealers are not authorized to assume for Telos Systems any additional obligations or liabilities in connection with the dealers' sale of the Products.