Telos VX® Prime+

Big Performance for Small Facilities

OVERVIEW

Telos VX® talk-show systems are the world's first true VoIP-based broadcast phone systems and have been proven to deliver the power of VoIP to the broadcast studio like no other. The Telos VX Prime+, with built-in support for AES67, is the next evolution of Telos VX VoIP phone systems in a powerful new 1RU hardware unit. Additionally, support for the G.722 voice codec ensures the highest quality calls from supported mobile devices. With capacity of 8 fixed hybrids/faders, VX Prime+ is ideal for facilities with 2 to 4 studios. (For larger facilities, check out VX Enterprise with up to 120-hybrid capacity.)

AES67 support brings a new level of compatibility and flexibility to VX phone systems. Support for AES67 gives broadcasters the flexibility of integrating VX Prime+ into *any* AES67 environment, in addition to our own Axia® Livewire® network. With plug-and-play connectivity, you can network multiple channels of audio with any manufacturer's AES67-compliant hardware. Beyond AES67, Livewire users have the added convenience and power of networking control (GPIO), advertising/discovery, and program associated data throughout the network.

Using VoIP, VX Prime+ gives you remarkable-sounding on-air phone calls with no 'gotchas'. It uses standard SIP protocol that works with many VoIP PBX systems and SIP Telco to take advantage of low-cost and high-reliability service offerings. VX Prime+ can also connect to traditional telco lines via Asterisk PBX systems, which can be customized for specific facility requirements.

VX Prime+ gives you incredible operational power, flexible, adaptable workflows, and superior audio quality, while making it easier than ever for talent to have complete mastery of their callers. With VX Prime+, the world's leading broadcast phone system is now available to those with smaller budgets, offering Big Performance for Small Facilities.

Why VX Prime+?

Cost-Efficient Way to Upgrade to IP

- Lower-capacity alternative to VX Enterprise Broadcast System
- Potentially save thousands monthly on expensive ISDN/POTS lines
- Ideal for smaller studios (2-4 studios) & smaller budgets

Audio Quality

- Native support of G.722 "HD Voice" codec
- Fifth-generation Telos Adaptive Digital Hybrid on every line for cleanest, clearest caller audio
- Smart AGC ensures consistent caller audio levels
- Digital Dynamic EQ (DDEQ) by Omnia® adjust EQ automatically to ensure call-to-call consistency and the best intelligibility

Simple Setup

- Connects to your Axia Livewire—or other AES67 network—with a single Ethernet cable
- Non-Livewire studios can use Telos Alliance Mixed Signal xNode for audio and GPIO connectivity to studio consoles
- Provides phone hybrids for each of your studios without need for any additional wiring or physical audio connections

Flexible Use

- No restriction to the number of SIP lines or phone numbers that can come into the system
- 8 fixed hybrid/faders (not expandable)

Support

- Industry-leading 24/7 support
- Free Xscreen Lite call-screening software from Broadcast Bionics

FEATURES

Features At A Glance

- A true VoIP telephone system designed and built specifically for broadcasting; VX Prime+ is ideal for small to medium facilities with 2 to 4 studios.
- Includes support for AES67, giving broadcasters added flexibility of integrating VX Prime+ into any AES67 network, in addition to our own Axia Livewire network.
- SIP call-handling throughout—no internal conversion to analog call handling like some other so-called "VoIP" systems.
- Standards-based SIP interface integrates with Asterisk open-source SIP phone servers and most VoIP-based PBX systems to allow transfers and common telco services for business and studio phones.
- Standard Ethernet backbone provides a common transport path for both studio audio and telecom needs, resulting in cost savings and a simplified studio infrastructure.
- System capacity of 8 hybrids. Each call placed on the air receives a dedicated hybrid for unmatched clarity and superior conferencing.
- Native Livewire integration—one connection integrates caller audio, program-on-hold, mix-minus, and logic directly into Axia AoIP consoles and networks.
- Connect VX systems to any third-party radio console or other broadcast equipment using available Telos Alliance Mixed Signal, AES/EBU, and GPIO xNodes. xNodes feature 48 kHz sampling rate and studio-grade 24-bit A/D converters with 256x oversampling.
- Powerful dynamic line management enables instant reallocation of call-in lines to studios requiring increased capacity.
- VSet phone controllers with full-color LCD displays and Telos Status Symbols present producers and talent with a rich graphical information display. Each VSet features its own address book and call log.
- The "Drop-in" Vset Call Controller™ modules can integrate VX phone control directly into your mixing consoles.
- XScreen Lite screening software included.
- Clear, clean caller audio from 5th-generation Telos Adaptive Hybrid technology, including Digital Dynamic EQ, AGC, adjustable caller ducking, and send- and receive-audio dynamics processing by Omnia.
- Support for G.722 codec enables high-fidelity phone calls from iPhone and Android SIP softphones using an SIP server.
- Wideband acoustic echo cancellation from Fraunhofer completely eliminates open-speaker feedback.
- Works with POTS, T1/E1, ISDN and SIP Trunking telco services for maximum flexibility and cost savings, via Asterisk servers.*

* Due to the wide variation in how traditional phone service can be delivered, and the complexities that can be involved in converting those services to SIP, we really want to talk with you about your system design before you order. Telos has VX System engineers standing by to help you draw up a configuration that will ensure your VX purchase will perform to your expectations when using traditional POTS and ISDN lines.

IN DEPTH

This Is Where It All Started

Steve Church founded Telos Systems® in 1985. As both a talk-show host and radio group Technical Director, Steve was only too familiar with the frustrations of "bad phones" and even less responsive equipment manufacturers, so he set about eliminating the technical problems that plagued radio call-in segments. In 1984, he invented the Telos 10, the first DSP-based telephone-to-broadcast interface system—allowing radio stations to significantly improve the technical quality of call-in segments. The overwhelming response to Steve's economical and technically elegant solution to a nagging problem provided the spark from which Telos was born.

A lot's happened since then. Telos pioneered the use of MPEG Layer 3 coding in the revolutionary Zephyr ISDN codec. We produced the first hardware MP3 streaming encoder for broadcast. We developed the world's first "whole-plant" broadcast phone system. We invented the IP-networked radio console, and then integrated broadcast phones into that network via Ethernet.

Telos has grown steadily since our initial production run of 25 Telos 10 units in 1985! With tens of thousands of systems in the field, it's now is hard to find a broadcast facility in the world without at least one piece of our gear. Our organization, now called the Telos Alliance, includes the Omnia®, Axia®, 25-Seven®, Minnetonka Audio®, and Linear Acoustic®, and our R&D department— the largest research team in broadcasting— continues to develop innovative audio products for radio and television broadcasting, telephony, and the Internet.

VX Prime+ Broadcast VoIP Phone System

Telos VX marries the flexibility and capabilities of IP networks to the remarkable power of today's digital signal processing, and brings the benefits to broadcast facilities. With a VX system, you can move and share lines between studios at the touch of a button. VX Prime+ is an 8-hybrid system that brings VoIP flexibility to medium and small facilities without breaking the bank.

VX systems are naturally flexible, naturally powerful. Your broadcasts benefit from superb, crystal-clear caller audio while callers hear clean, intelligible audio from your console. VX systems are surprisingly cost-effective, even when deployed in single-station facilities.

Why VoIP For Broadcast?

VoIP has taken the business world by storm, increasing the flexibility of office phone systems and PBXs while simultaneously lowering maintenance and equipment costs. In fact, most Fortune 500 companies have replaced their old PBX systems with VoIP for just these reasons.

VoIP is a natural for broadcasters, interconnecting the phone system with audio interfaces, phone sets, console controllers, and PCs running screening software by way of efficient, low-cost Ethernet. Using VoIP, you can finally share phone lines among multiple studios and route caller audio anywhere in your facility, easily, and instantly. Got a hot talk-show that suddenly needs more lines in a certain studio? Just a few keystrokes at a computer and you're ready...no delays, no cables to pull. VX systems can even interconnect with your business office's VoIP PBX to allow easy call transfers.

Reduced Cost. Increased Flexibility.

The use of sophisticated, modern IP networking for Telos VX Prime+ allows rich communication between devices. For example, caller information entered by a producer is displayed on the studio phone's color LCD. Caller audio is available on studio PCs for easy recording. Operators at mixing consoles can directly control line switching without diverting their attention from the board. The result? Talk shows that run like clockwork, sound better, and flow without errors.

This standards-based VoIP architecture helps you save money, too, by widening your choices in telco providers. Most carriers now offer VoIP services using the SIP protocol, which can deliver substantial savings to stations that need any number of lines. (You can also connect to traditional T-1/PRI, POTS or ISDN phone lines using open-source Asterisk-based phone servers.)

But VX systems don't stop at providing the benefits of VoIP—they also carry the broadcast-phone technology expertise of Telos.

Every incoming line has its own 5th-generation Telos Adaptive Digital Hybrid, our most advanced ever—packed full of technology engineered to extract the cleanest, clearest caller audio from just about any phone line, even cellular calls. Multiple lines can be conferenced with superior clarity and fidelity. Smart AGC ensures consistent caller audio levels. And calls from mobile callers using SIP clients on their smartphones benefit from native support for the G.722 "HD Voice" codec, improving caller speech quality and intelligibility.

Added Flexibility to Connect to Livewire or Any Other AES67 Network

AES67 support brings a new level of compatibility and flexibility to VX phone systems. Support for AES67 gives broadcasters the flexibility of integrating VX Prime+ into *any* AES67 environment, not just our own Axia® Livewire® network. With plug-and-play connectivity, you can network multiple channels of audio with any manufacturer's AES67-compliant hardware. Beyond AES67, Livewire users have the added convenience and power of networking control (GPIO), advertising/discovery, and program associated data throughout the network.

VX System Components

VX Prime+ Engine

The heart of any VX system is the Engine. The fixed-capacity VX Prime+ system is powered by a 1RU rack-mount Engine with enormous processing power. In fact, the VX Prime+ Engine provides all the call control and audio processing needed for your entire on-air phone system.

With VX Prime+, you are equipped with 8 high-performance VoIP hybrids, to support multiple lines of concurrent on-air phones for two to four studios (depending on configuration).

Each VX Prime+ Engine features two Gigabit Ethernet ports, a high-density, cost-effective interface to both telephone lines and studio audio via proven Livewire Audio over IP (AoIP). VX systems are web-based, so remote control and configuration are easy—engineers can work from any place they can get online.

Call workflow for VX users is sophisticated and flexible. Lines may be readily shared among studios; the Web interface allows easy assignment of lines to "shows," which can then be selected by users on VSet phone controllers and console drop-in modules. Each studio may be configured with its own Program-on-Hold as well.

The processing power of the VX Engine provides sophisticated DSP hybrids for every line, allowing multiple calls to be conferenced and aired simultaneously with excellent quality. The hybrids are equipped with a rich processing toolbox to make caller audio sound its best, no matter what kind of line or phone the caller uses.

Caller audio benefits from Smart AGC coupled with famous Telos three-band adaptive Digital Dynamic EQ and a three-band adaptive spectral processor. Call ducking and host override are part of the VX audio toolkit as well.

You'll notice that there are no audio I/O or telco ports on VX Engines themselves. That's because they're meant for fast connection to Livewire AoIP systems; using Livewire, all I/O is handled via Ethernet. The Livewire network supports a wide variety of peripherals such as Axia audio consoles, VSet phones, PC-based screener applications, console-integrated controllers, and more. SIP servers and telecom providers connect through a dedicated WAN Ethernet jack for routing simplicity and easy maintenance. For traditional phone services, VX works seamlessly with open-source Asterisk SIP servers, and most SIP PBXs. Telos VX experts speak fluent Asterisk, and are ready to assist you in specifying and configuring an installation to suit your studio's requirements. VX also works with standard telco gateways to connect to T1/E1, ISDN, and POTS providers. And, if you already have a VoIP-based PBX or SIP endpoint service, VX systems can work with those as well.

VSet12

The Telos VSet12 phone is beautifully designed, with a friendly LCD color display that uses exclusive Status Symbols to let talent know what's going on instantly. VSet12 works with up to 12 phone lines; the info-rich display provides caller ID for each line, along with time ringing-in or on-hold, and even screener comments from the screening software applications.

VSet12 gives talent unprecedented flexibility. You can map groups of lines to a single fader, making it simple to take a queue of calls to air sequentially. One-touch controls let talent step through queued calls, "busy out" incoming lines, lock calls on-air to prevent unintentional disconnection of a VIP. Telos-exclusive "Next Call" key speeds workflow for producers, screeners, and talent. But because VX systems provide a hybrid per line, much more functionality is unlocked: You can now spread multiple calls over a number of faders, using one for each call so that operators can control each line's level individually. You can hard-assign individual lines to fixed faders, such as for VIP calls. A built-in address book and call history log round out VSet12's features.

VSet6

VSet6 is a 6-line phone controller for VX systems. Like the VSet12, it has a bright, attractive LCD color display with Status Symbols that feed talent instant information about line and caller status and controls that enable talent to step through queued calls, busy incoming lines, lock calls on-air, activate the dump button on a profanity delay, and more.

VSet Phone Controls

The LCD displays deliver detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in screening software applications. Shown above are a few of the attractive, instantly-understandable Status Symbols that help talent run tight, mistake-free shows.

Each VSet phone has its own web server for easy remote configuration and software upgrades, and flexible power options include PoE (Power over Ethernet) from a Telos-approved PoE switch or Axia xSwitch, or an in-line power injector.

On-Console Control

Whether calls are live or pre-recorded, interviews or audience participation, one thing's certain: phone segments are an integral part of today's fast-paced radio. But up to now, the phone system was separate from the on-air console; audio was shared, but little else. Wouldn't it be great if talent could take control of phones without ever having to divert their attention from the board? They can: the Axia Console Controller provides the ideal way to integrate broadcast phones into the on-air console—the control center of every studio.

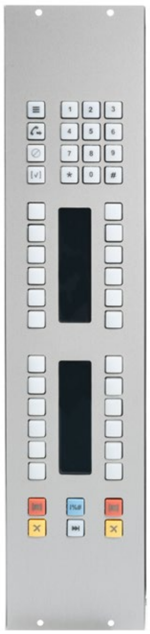
There are plenty of advantages to melding phones with consoles. Like ease of installation: IP-Audio consoles with built-in phone controllers don't need any additional wires or connections. Their control signaling, caller audio, and backfeeds ride on the network connection that's already there. Bringing caller audio into the IP-Audio domain makes it routable like any other audio source. You can even dynamically conference multiple lines using just a single fader.

VX systems connect directly to Axia Fusion, Element, iQ and Radius mixing consoles using Livewire+™ AES67 IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. And now, VX Systems have the added flexibility of AES67 support. Multiple phone lines— each with a dedicated hybrid—can automatically map to individual console faders for complete control of caller audio. And users enjoy seamless console integration, with phone controls right on the board so that talent can dial, answer, screen, and drop calls without ever diverting their attention from the console. Information about line and caller status can be displayed right on the console as well.

Axia Call Controller

For VX clients with an Axia Fusion mixing console. The Axia Call Controller module puts control of VX telephone systems right into the console. The two-fader telephone control module features an integrated Telos Call Controller with renowned Status Symbols visual call management. These include Transfer, Drop, and Block All keys plus the Telos-exclusive "Next Call" key that allows fast airing of pre-screened calls. The rotary Options Control knobs can be programmed to trim source or fader gain when turned, and alphanumeric channel displays give complete information on source assignment, channel options, and more.

VSet Call Controller

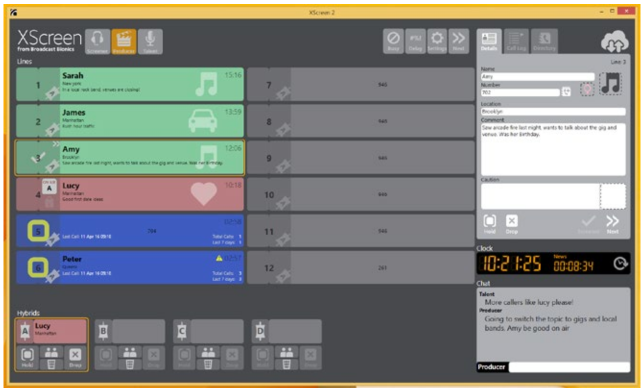


Want a VX system, but don’t have an Axia mixing console? No problem — Telos provides VSet Console Controller electronics packages, which may be fitted to your console using panels supplied by your OEM console provider or preferred third-party fabricator. Like the VSet12 phone set, the VSet Console Controller provides visual line-status indicators and fast-take keys for selection and control of up to 12 callers, along with standard controls such as Take, Drop, Hold and Busy keys, and the Telos-exclusive “Next Call” key to speed workflow for producers, screeners, and talent. There’s also a built-in keypad for on-console dialing of outgoing numbers.

VSet Desktop Controller

The Telos VSet Desktop Controller with visual line-status indicators provides selection and control of up to 12 callers. Includes standard controls to allow fast, error-free talent operation, including Take, Drop, Hold, and Busy keys. Telos exclusive “Next Call” key speeds workflow for producers, and talent; built-in keypad allows on-hybrid dialing of outgoing numbers. VSet Desktop Controller works in producer mode only. Make and answer calls using VSet6, VSet12, or on hybrid with VSet Desktop Controller.

Broadcast Bionics XScreen Lite Call-Screening Software Included



XScreen Lite software comes with every VX Prime+ purchase and provides Unlimited Lite users, dial, hold, hang up, screened hold and next, conference control, dump mode, lock call, VSet control, telephone number, location, name, point & disposition, chat, clock, and call log (6 hours only) functionality. Please download your XScreen software from www.xscreen2.com.

VX Prime+ Interfaces

Telos Alliance xNode Audio Interfaces

Telos Alliance xNodes let you connect VX Prime+ to any non-networked radio console or other broadcast equipment, using standard AES/EBU interfaces. A GPIO Logic xNode provides control logic where needed. To cover all your bases, the Telos Alliance Mixed Signal xNode provides one mic/line analog input (switchable); two analog line inputs (dedicated); three analog line outputs; one AES3 input, one AES3 output, and two GPIO ports, each with five opto-isolated ins and outs.

The Telos Alliance AES/EBU audio xNode 4 AES/EBU inputs and 4 AES/EBU outputs. Left and right input signals may be split and routed independently as mono signals. Stunning performance specs include 48 kHz sampling rate, 126dB of dynamic range, and <0.0003% THD.

Each Telos Alliance GPIO logic xNode interface provides six general-purpose logic ports each with five opto-isolated inputs and five outputs. A logic port can be associated with any audio input or output and routes control data transparently along with the audio.

Telos Alliance xSwitch Zero-Configuration Ethernet Switch

xSwitch is the world's only zero-configuration Ethernet switch optimized for Livewire IP-Audio applications. Fast setup requires only IP address assignment via front-panel OLED display or Axia iProbe software. Features 8 10/100MBit Ethernet ports — 4 with Power-over-Ethernet to power Axia xNodes, Telos VSet phones, and other networked devices compatible with the IEEE 802.1af PoE standard. Noiseless and fan-free, xSwitch can be conveniently placed adjacent to your audio devices, rack-mounted using included hardware, or wall-mounted (with an accessory kit available separately).

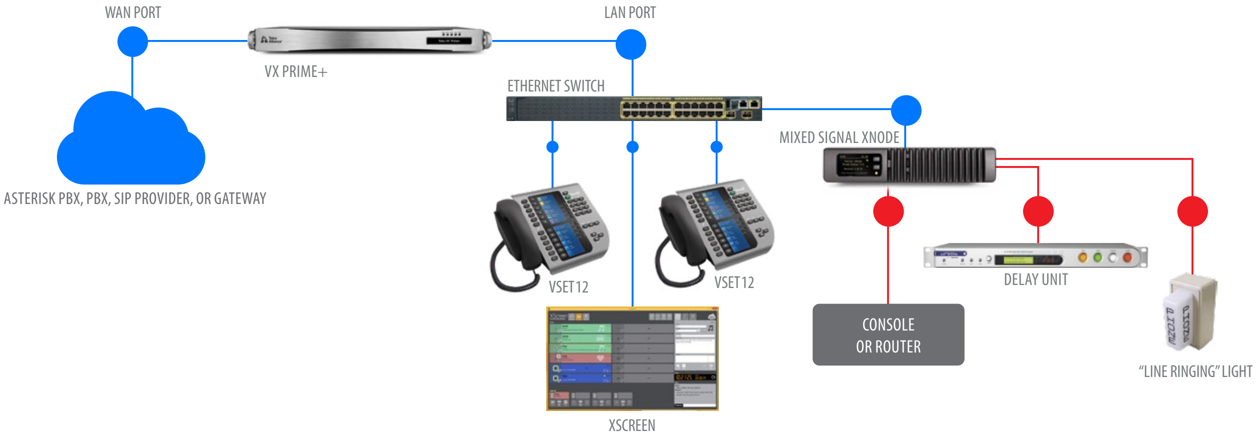
The Power of IP Realized

With VX Prime+, there's no need for the maze of discrete cables once required by multi-line talk show systems. All VX components are linked with standard Ethernet, so a single CAT-5 cable provides:

- Connection to the telco interface
- Line switching commands
- Data communication between the VX Prime+ and VSet12 phones
- Transport of caller audio to mixing consoles
- Return of mix-minus and program-on-hold audio to the caller
- Data messages (such as call notes and IM) between producer and talent
- Livewire audio for the recording of calls
- Transfer of recorded call files from the producer to the studio

Now… how many discrete cables does that save you from having to wire up?

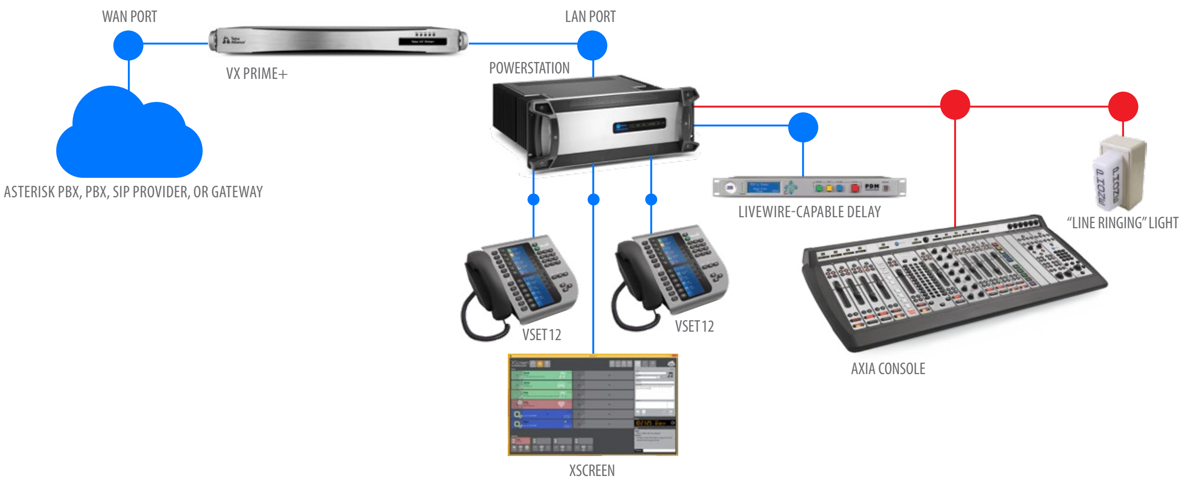
Hook It Up Your Way: Non-Axia Installation Diagram



Got an Axia Livewire AoIP studio network? Great! Your new VX Prime+ phone system will plug right into it. It's the seamless integration of studio phones, mixing consoles, and routing network you've dreamt about.

Don't have IP-Audio networking yet? Not to worry... VX will work with all console brands, networked or not, using Telos Alliance xNodes, and the VX Call Controller drop-in controller for your console. Telos VX systems are "facility wide" broadcast phone products. That means multiple studios, multiple stations, multiple shows—with minimal hardware requirements. Telco is delivered via IP from your SIP PBX, or through a dedicated IP circuit using SIP trunking. POTS, ISDN, or T1 phone service can be brought in using an open-source Asterisk server or a standalone gateway device. Once connected, all line and audio connectivity flows via Ethernet. The diagram above shows a typical studio with an analog mixer, using a Telos Alliance Mixed Signal xNode to connect to the console and other broadcast equipment.

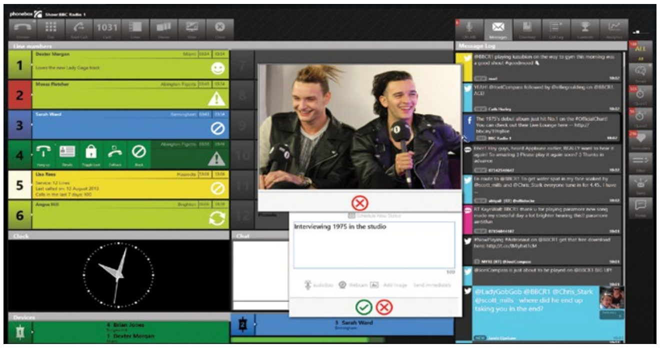
Hook It Up Your Way: Axia Installation Diagram



Installing a VX phone system in facilities already powered by Axia Livewire+™ AES67 networks requires even less time and hardware than described above. The audio inputs and outputs used and produced by VX are Livewire real-time audio channels and travel over the Axia AoIP network just like the rest of your audio. Axia console GPIO ports can be used for "phone ringing" tallies or remote control of profanity delay units.

VX Gives You Options

Broadcast Bionics



Broadcast Bionics offers PhoneBOX VX, a tailored-for-VX version of their original PhoneBOX software. PhoneBOX VX gives VX users an amazing amount of information and a high level of control over the VX system. There's prize management, call editing, and recording, sophisticated visual talkback, including a drag-and-drop database your show's calls, plus a rich phonebook and visual warnings, tied to Caller ID, for persistent or nuisance callers.

Find out more from www.phoneboxvx.com .

NeoScreener by NeoGroupe

NeoGroupe offers NeoScreener, a call management solution that interfaces Telos NX12, NX6, IQ6, VX, HX6, 2x12 and 2101 systems, allowing for line control and database lookup using caller ID. The solution can interface to NeoWinners which is NeoGroupe's contest management software. It is designed for radio and television stations that need to manage their flow of incoming phone calls.

NeoScreener also handles external inputs, like SMS, Website, iPhone. Database driven, it enhances the phone-call workflow. With NeoScreener, call screeners can easily welcome calls and present them to the Talent on a specific display. Visit www.neogroupe.com to learn more.

Arctic Palm CS Call Management

The CS Call Management package provides producers and talent with the tools to capture and control callers while staying in touch with each other in a single Caller Control window. Designed for the VX VOIP systems, both local and remote users are in constant communication.

For more information, visit www.arcticpalm.com/CSScreener.htm.

SPECIFICATIONS

System

- Maximum number of simultaneous calls on-air, VX Prime+: 8 (more with conferencing)
- Maximum number of SIP numbers, VX Prime+: 96
- One rack unit - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)

Audio Performance (Node)

Analog Line Inputs

- Input Impedance: >40 k ohms, balanced
- Nominal Level Range: Selectable, +4 dBu or -10dBv
- Input Headroom: 20 dB above nominal input

Analog Line Outputs

- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu

Digital Audio Inputs And Outputs

- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR) h Signal Format: AES-3 (AES/EBU)
- AES-3 Input Compliance: 24-bit with selectable sample rate conversion, 32 kHz to 96kHz input sample rate capable.
- AES-3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling
- Latency <3 ms, mic in to monitor out, including network and processor loop

Frequency Response

- Any input to any output: +0.5 / -0.5 dB, 20 Hz to 20 kHz

Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB “A” weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB “A” weighted
- Digital Input to Digital Output: 138 dB

Total Harmonic Distortion + Noise

- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation, Stereo Separation And CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20Hz to 20 kHz
- Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz

VX Prime+ Engine

IP/Ethernet Connections

- One 1 Gigabit Ethernet via RJ-45 LAN connection (livewire)
- One 1 Gigabit Ethernet via RJ-45 WAN Connection (SIP provider)

Processing Functions

- All processing is performed at 32-bit floating-point resolution.
- Send AGC/limiter
- Send filter
- Gated Receive AGC
- Receive filter
- Receive dynamic EQ (3 band)
- Ducker
- Sample rate converter

Power Supply AC Input

- Hot-swap capable dual-redundant internal auto-ranging power supplies. 90 – 132 / 187 – 264 VAC, 50Hz/60Hz. IEC receptacle, internal fuse.
- Power consumption: 100 Watts

Operating Temperatures

- -10 degree C to +40 degree C, <90% humidity, no condensation

Dimensions and Weight

- Rackmount, 2RU
- 3.5 inches x 17 inches x 15 inches
- 10 pounds

Studio Audio Connections

- Via Livewire Ethernet. Each selectable group and fixed line has a send and receive input/output.
- Each studio may be configured with its own Program-on-Hold input.
- Livewire-equipped studios take audio directly from the network.
- Telos Alliance xNodes are available for professional-level analog and AES3 connection breakouts for clients without Livewire AoIP networking.
- VX Prime+ supports AES67 connectivity.

Telco Connections

- Audio: standard RTP. Codecs: G.711u-Law and A-Law, and G.722.
- Control: standard SIP endpoints, ISDN PRI/T-1, ISDN BRI and POTS may be supported with the appropriate interfaces using an Asterisk Open source PBX.