

Telos VX® Enterprise

The Whole-Plant Broadcast Talkshow System



OVERVIEW

Telos VX® is the world's first VoIP (Voice over IP) talkshow system — a broadcast phone system that's so powerful, it can run all the on-air phones for your entire plant. Telos VX Enterprise™, with built-in support for AES67, is the next evolution of Telos VX VoIP phone system 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 expandable to up to 120 hybrids/faders, VX Enterprise is ideal for medium to large facilities and can grow with your station over time. (For smaller facilities, check out VX Prime+ with 8-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 Enterprise 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.

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Using VoIP, VX Enterprise gives you remarkable-sounding on-air phone calls with no 'gotchas'. It weds modern networking to the remarkable power of digital signal processing. VX Enterprise uses Ethernet as its connection backbone, significantly cutting the cost of phone system installation, maintenance, and cabling. 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 Enterprise can also connect to traditional telco lines via Asterisk PBX systems, which can be customized for specific facility requirements.

Don't have an IP-Audio network yet? Optional Telos Alliance xNodes, like the Telos Alliance Mixed Signal Node, break out audio into analog and digital formats, along with GPIO logic commands. And with informative VSet phones, talent finds it easier than ever to take control of their callers, moving and sharing lines between studios at the touch of a button.

FEATURES

- VX is the world's first VoIP telephone system designed and built specifically for broadcasting.
- Includes support for AES67, giving broadcasters added flexibility of integrating VX Enterprise into any AES67 network, in addition to our own Axia Livewire network.
- Works directly with SIP endpoint telco or PBX services, and in conjunction with a PBX may support POTS, T1/E1, and ISDN BRI for maximum flexibility and cost savings.*
- Standards-based SIP/IP interface integrates with most VoIP-based PBX systems to allow transfers, line-sharing, Caller ID 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. Connection of up to 100 control devices (software or hardware) is possible.
- Modular, scalable system can be easily expanded and configured to manage a network of 60 or even more studios, each with a dedicated Program-On-Hold input—truly a "whole-plant" solution for on-air phones.
- Base system is licensed for up to 24 hybrids, may be expanded in license increments of 8 up to a total of 120 hybrids. A Telos system engineering consultation is required for any system configuration over 72 hybrids. Please contact us at vx-presales@telosalliance.com for assistance.
- Each call 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 Enterprise to any radio console or other broadcast equipment using available Telos Alliance AES/EBU, Mixed Signal, and GPIO xNodes. Audio interfaces 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.

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- VSet Call 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.
- Drop-in modules can integrate VX Enterprise phone control directly into your Axia 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®.
- Wideband acoustic echo cancellation from Fraunhofer completely eliminates open-speaker feedback.
- Support for G.722 codec enables high-fidelity phone calls from iPhone and Android SIP softphones using an SIP server.

* 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

VoIP for Broadcast. From Telos, Naturally.

VX is the world's first VoIP (Voice over IP) talkshow system, and VX Enterprise is the next evolution of this legendary system, now with built-in support for AES67 in a 1RU chassis. This whole-plant broadcast phone system is incredibly powerful, very flexible, and highly scalable.

VoIP has already 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 older PBX systems with VoIP for just these reasons. There's no reason broadcasters shouldn't take advantage of this cost-saving technology as well. In addition to cost savings from digital phone service provisioning, VX Enterprise significantly eases the cost of installation, maintenance, and cabling by using standard Ethernet as its data backbone.

As a result VX Enterprise is naturally scalable, capable of serving even the largest of facilities. There are major operational benefits as well. VX Enterprise combines the flexibility and economy of modern SIP networking with powerful digital signal and audio processing—making it easier than ever for talent to take control of their phone system. You can move and share lines between studios at the touch of a button. VX Enterprise is truly the future of broadcast phones.

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Why VoIP for Broadcast?

VoIP is a natural for broadcasters. Using VoIP, you can interconnect the phone system CPU with audio interfaces, phone sets, console controllers, and PCs running screening software using efficient, low-cost Ethernet. 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, and no cables to pull. VX Enterprise can even connect with your business office's VoIP PBX to facilitate easy call transfers.

Of course, it's got to sound good. And it does, thanks to more than two decades of DSP hybrid technology developed by Telos. Every incoming line has its own 5th-generation digital hybrid, our most advanced ever, packed full of technology engineered to extract the cleanest, clearest caller audio from any phone line—even noisy cellular calls. Multiple lines can be conferenced with superior clarity and fidelity. Smart AGC ensures consistent caller audio levels. New Acoustic Echo Cancellation from FhG removes feedback and echo in open-speaker studio situations. And if you choose to use SIP Trunking telco services, calls from mobile handsets with SIP clients, HD capable telephone sets and PC apps will benefit from VX Enterprise's native support of the G.722 codec, instantly improving caller speech quality.

Since VX Enterprise uses Ethernet as its network backbone, it naturally plugs right into Axia IP-Audio networks, connecting multiple channels of audio and control using a single Ethernet cable. If you don't have an IP-Audio network yet, that's OK; Telos Alliance xNodes provide AES audio and GPIO connections that work with your existing studio equipment.

VX Enterprise Components

VX Enterprise



VX Enterprise 1RU rack-mount device is the heart of the system. It provides all the call control and audio processing needed for the system, and supports up to 120 active calls on-air simultaneously. Its two Gigabit Ethernet ports provide a cost-effective interface to both telephone lines and studio audio via proven Livewire AoIP. VX Enterprise is Web-based, so remote control and configuration are a snap — engineers can work with it from any place they can get online.

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Call-processing 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 the studio controllers. Each studio can provide its own Program-on-Hold audio to callers.

Audio processing features also have taken a leap forward. The processing power of VX Enterprise allows 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. Send audio gets its own sweetening with an AGC/limiter and FhG's Acoustic Echo Cancellation technology that literally eliminates open-mic feedback. Call ducking and host override are part of the VX Enterprise toolkit as well, and talent can manage and customize their telephone settings and workflow using VX Enterprise Show Profiles to store and recall commonly used show configurations.

You'll notice that there are no audio I/O or phone jacks on VX Enterprise. All connections to the Engine are via the two Ethernet jacks that connect to your system's Ethernet switch to support a wide variety of peripherals: telephone lines, Livewire studio audio, VSet phones, console-integrated controllers, etc. If you have a VoIP-based PBX or SIP endpoint telco service, VX Enterprise uses standard SIP (Session Initiation Protocol).

The Coolest Broadcast Phone Controllers Ever

With decades of experience designing broadcast phone systems, it's no wonder broadcasters agree that Telos makes the industry's most powerful, most flexible system controllers. All VSet phones can be powered by PoE from a Telos-approved switch, a PoE port on an Axia console engine, or by using a power injector.

VSet12



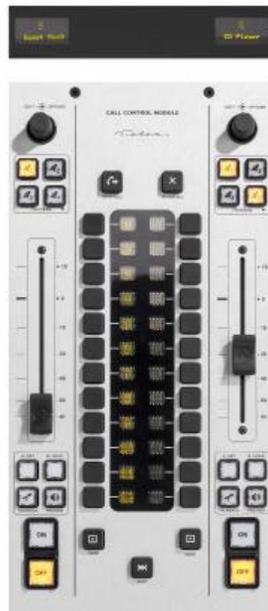
The VSet12 phone controller is an IP-based phone set with two large, high-contrast color LCD panels that provide line status and caller information. VSet phones can work like a traditional Telos controller, with calls being selected, held, and dropped in the way to which operators have grown accustomed. But because the VX Enterprise system is so powerful, 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.

VSet6



VSet6 is a 6-line phone controller for VX Enterprise. 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, start an external recording device, et cetera. Next Call functionality speeds workflow for producers, screeners, and talent. With all the control functions of the VSet12, it's great for smaller or secondary studios.

On-Console Control



Live calls 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. VX systems connect directly to Axia 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. 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.

There are plenty of other 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. With the Virtual Mixers built into Axia consoles, you could even choose to dynamically conference multiple lines and control their gain with a single fader.

Telos Alliance xNode Audio Interfaces

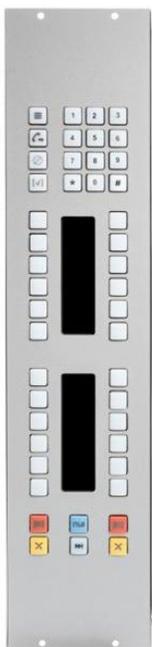


Telos Alliance xNodes let you connect VX Enterprise to any non-networked radio console or other broadcast equipment, using standard AES/EBU interfaces. A GPIO Logic xNode provides control logic where needed. To cover all your bases, the Telos Alliance Mixed Signal xNode provides one mic/line analog input (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.

VSet Call Controller



Want a VX Enterprise 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 phoneset, 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.

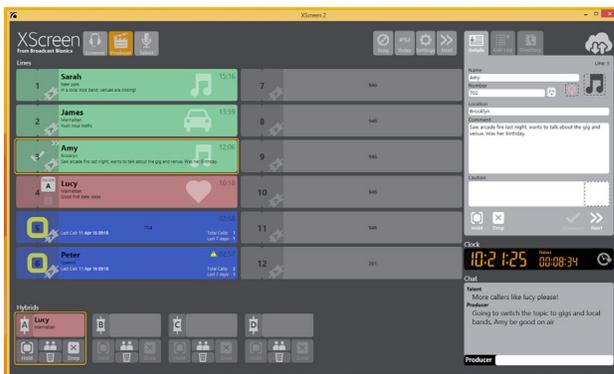
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VSet Desktop Controller



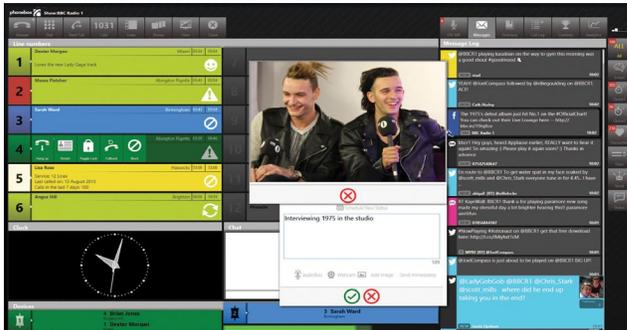
The Telos VSet Desktop Controller with visual line-status indicators provides selection and control of up to 12 callers. Includes standard controls to allow fast, error-free talent operation, including Take, Drop, Hold, and Busy keys. Telos exclusive "Next Call" key speeds workflow for producers, and 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 Enterprise 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.

Broadcast Bionics PhoneBOX VX



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 NeoGroup



NeoGroup 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 NeoGroup'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

General

- Telos 5th-generation Adaptive Digital Hybrids
- Maximum number of hybrids: 120, when used with a-Law or u-Law codecs for VoIP lines. (Higher-quality codecs, such as G.722, consume more system resources and result in a decreased number of total lines available.) A Telos system engineering consultation is required for any system configuration over 72 hybrids. Please contact us at vx-presales@telosalliance.com for assistance.
- Maximum number of SIP numbers: unlimited
- Maximum active on-air calls: 120
- Maximum on-air calls on one fader: 12
- One rack unit - 1.75"H x 19"W x 15.5"D (44 x 483 x 394 mm)

Analog Inputs (with Telos Alliance xNode)

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

Analog Outputs (with Telos Alliance xNode)

- 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)
- 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

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- 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, 20 Hz to 20 kHz
- Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz

VX Enterprise

IP/Ethernet Connections

- One 100/1000BASE-T Ethernet via RJ-45 LAN connection
- One 100/1000BASE-T Ethernet via RJ-45 WAN connection

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
- Ducker
- Sample rate converter

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- Line Echo Canceller (hybrid)
- Acoustic Echo Canceller

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: 150 Watts

Operating Temperatures

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

Studio Audio Connections

- Via Livewire IP/Ethernet. Each selectable group and fixed line has a send and receive input/output
- Each studio has a Program-on-Hold input
- Each Acoustic Echo Canceller has two inputs (signal and reference) and one output
- Livewire+™ AES67 equipped studios may take and supply audio directly to/from the network. Telos Alliance xNodes are available for pro analog and AES3 breakout.
- VX Enterprise supports AES67 connectivity.

Telco Connections

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

Regulatory

North America: FCC and CE tested and compliant, redundant power supplies are UL approved.

Europe: Complies with the European Union Directive 2002/95/EC on the restriction of the use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/ EC, 2005/747/EC (RoHS Directive), and WEEE.