Telos VX Prime

Broadcast VoIP Phone System

OVERVIEW

Telos VX talk-show systems are the world's only true VoIP-based broadcast phone systems. The VX Prime gives you incredible operational power, flexible, adaptable workflows, and superior audio quality—a powerful broadcast phone solution that's economical enough for stations with two to four studios. VX Prime connects to VoIP-based PBX systems and SIP carriers to take advantage of low-cost and high reliability service offerings. VX Prime also makes it easier than ever for talent to have complete mastery of their callers. No other broadcast phone system delivers the power of VoIP to the broadcast studio like Telos VX. With VX Prime, the world's leading broadcast phone system is now available to those with smaller budgets, giving you big studio sound at a small studio prices. Simply put, you're paying for the capability you need, and nothing extra.

Why VX Prime?

Cost-Efficient Way to Upgrade to IP

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

Audio Quality

- Native support of G.722 "HD Voice" codec
- Smart AGC ensures consistent caller audio levels
- Digital Dynamic EQ (DDEQ) by Omnia adjusts EQ automatically to ensure call-to-call consistency and the best intelligibility

Simple Setup

- Connects to your existing Livewire network with a single Ethernet cable
- Non-Livewire studios can use Telos Multipurpose Node audio interface 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
- Eight fixed hybrid/faders (not expandable)

Support

- Industry-leading 24/7 support
- Telos standard 5-year warranty
- Free XScreen 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 studios with two to four stations.
- SIP call-handling throughout—no internal conversion to analog call handling like some other so-called "VoIP" systems.
- Works with T1/E1, ISDN, SIP telco services and even POTS for maximum flexibility and cost-savings.
- 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 eight 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 Multipurpose, AES/EBU, and GPIO interfaces. 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.
- 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.
- Drop-in Call Controller modules can integrate VX phone control directly into your mixing consoles.
- Included XScreen screening software with built-in soft-phone allows a "phone" connection on any networked PC. Integrated recorder/editor simplifies recording of off-air conversations.
- Clear, clean caller audio from fifth-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 "HD Voice" codec enables high-fidelity (7 khz) phone calls from SIP telephone sets and softphones.

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 Audio, Axia Audio, 25-Seven Systems, Minnetonka Audio, and Linear Acoustic brands, 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.

Introducing 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 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 Prime offers all of this in an eight-hybrid system that brings VoIP flexibility to medium and small facilities without breaking the bank.

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, and 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.

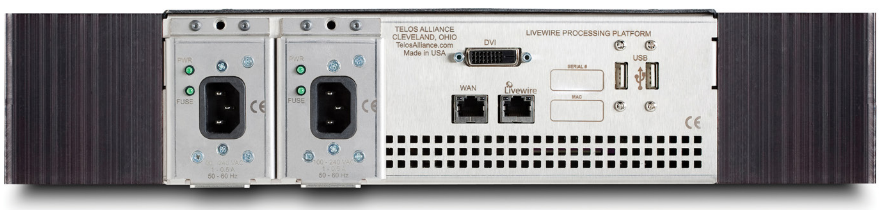
This standards-based VoIP architecture helps you save money, too, by widening your choices in telco providers. Most carriers now offer VoIP services using the SIP protocol, which can deliver substantial savings to stations that need any number of lines. (You can also connect to traditional T-1/PRI, POTS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

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 2RU rack-mount Engine with enormous processing power. It provides all the call control and audio processing needed for your entire on-air phone system.

With VX Prime, you are equipped with eight high-performance VoIP hybrids, to support multiple lines of concurrent on-air phones for from two to four studios (dependent upon 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 anyplace they can get online.

Call workflow for VX users is sophisticated and flexible. No matter which system you choose, 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 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 PBX's. 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 six-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

TRANSFER

ADDRESS BOOK

BUSY ALL

MISSED CALLS

ON AIR

AUTO ANSWER

DEVICE CONFIG.

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 switch or Axia xSwitch, or the included in-line power injector.

Console Controllers

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.

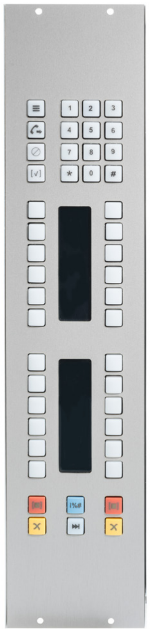
VX systems connect directly to Axia Fusion, Element, iQ and Radius mixing consoles using Livewire IP-Audio to eliminate the cost and complexity of old-style inputs, outputs, and mix-minuses. Multiple phone lines— each with a dedicated hybrid—can automatically map to individual console faders for complete 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.

Drop-in VX control modules are available for use with other console brands, too.

Axia Call Controller

For VX clients with Axia Fusion or Element AoIP mixing consoles, 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, 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 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.

Broadcast Bionics Xscreen Call-Screening Software Included

XScreen software comes with every VX Engine purchase and provides call control, call screening, data capture, and chat functionality enabling you to quickly answer, screen, and route calls using multiple PC clients. The cloud-based database keeps a log of calls and provides further alert and directory functionality.

XScreen can record and manage caller audio (Livewire systems only) and can additionally act as a softphone for talking to and screening callers directly through a USB headset or soundcard on your XScreen client PC.

XScreen is available in free (Lite) and full (subscription) versions. When you install XScreen for the first time you will receive a 90-day free trial license for the full version. After 90 days you can continue to use the full version with an annual subscription, or use the reduced, Lite functionality free of charge. Please download your XScreen software from www.xscreen2.com.

VX Prime Interfaces

VX Audio and Logic Interfaces let you connect VX to any non-networked radio console or other broadcast equipment, using multipurpose or AES/EBU interfaces. A GPIO Logic interface provides control logic where it's needed. Our new, space-efficient Telos Multipurpose Node offers a mix of mix of analog, AES, and GPIO connections—a perfect "utility node" for any VX system.

VX AES/EBU Audio Interface

The VX AES/EBU audio interface provides eight digital AES3 inputs and outputs, each on a separate RJ-45 connector. Studio-grade performance specs, like 138dB of dynamic range and <0.0003% THD.

VX GPIO Logic Interface

Each VX GPIO logic interface has eight assignable logic ports. Each port contains five opto-isolated inputs and five opto-isolated outputs, which can be associated with audio input peripherals and/or output destination devices to provide machine start/stop pulses, lamp drives, and transport controls. Once a port is configured to be associated with a particular device, it automatically activates with that device.

Telos Multipurpose Node

Compact, AES67-compliant 9.5" x 11" third-generation AoIP interface with a mix of analog, AES and GPIO connections. There's one mic/line analog input (switchable), two analog line inputs (dedicated), three analog line outputs, one digital AES3 input, and one AES3 output and two GPIO ports, each with five opto-isolated inputs and outputs. Dual Ethernet ports allow connection to fully redundant networks. It can even be powered by AC mains or PoE from Ethernet switches compatible with the IEEE 802.1af PoE standard. 1RU height and half-rack width allows side-by-side mounting of two units in one rack space.

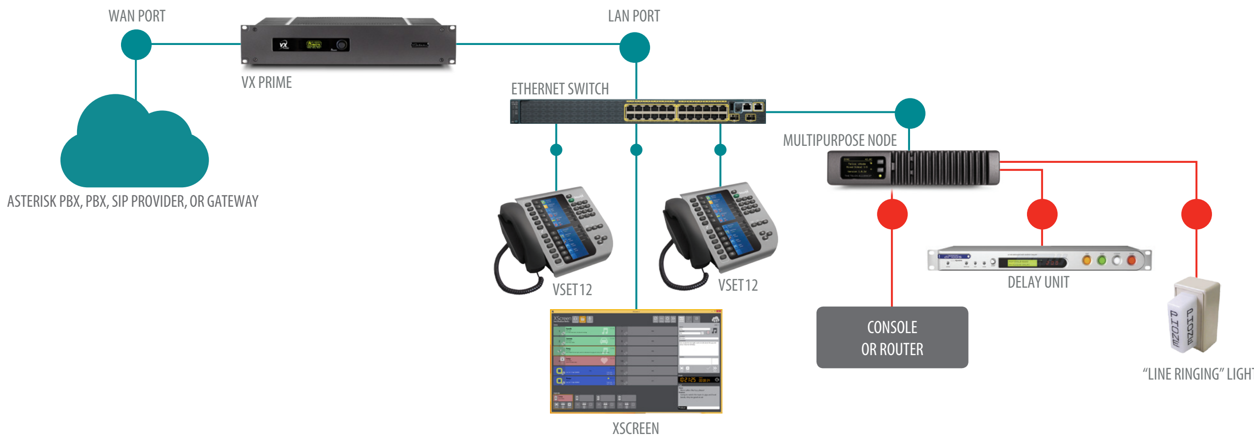
The Power of IP Realized

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 Engine 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

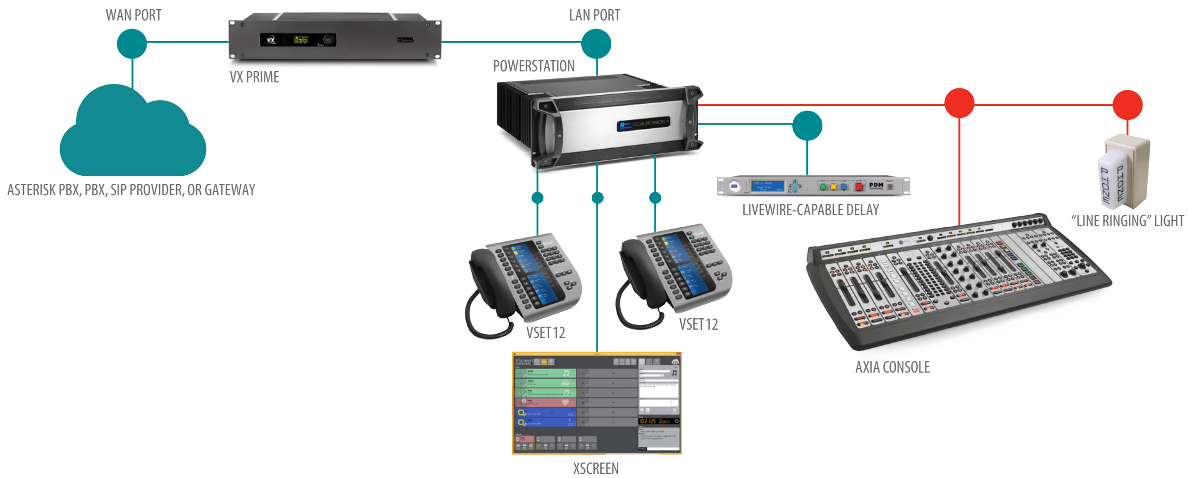


Got an Axia Livewire AoIP studio network? Great! Your new VX Prime phone system will plug right into it. It's the seamless integration of studio phones, mixing consoles, and routing network you've dreamt about.

Don't have IP-Audio networking yet? Not to worry… VX will work with all console brands, networked or not, using VX Audio and Logic interfaces, 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 Telos Multipurpose Node to connect to the console and other broadcast equipment.

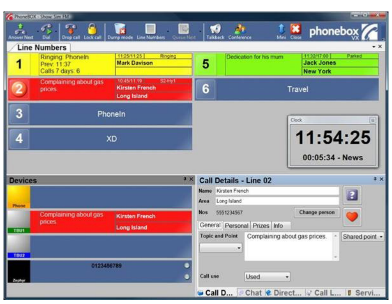
Hook It Up Your Way: Axia Installation Diagram



Installing a VX phone system in facilities already powered by Axia Livewire+ networks requires even less time and hardware than described in the previous section. 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 of 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 .

NeoSoft

NeoSoft 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

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
- Fanless, convection-cooled

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 Audio Interface Nodes are available for professional-level analog and AES3 connection breakouts for clients without Livewire AoIP networking.

Telco Connections

- Audio: standard RTP. Codecs: G.711u-Law and A-Law, and G.722.
- Control: standard SIP endpoint

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