TELOS VX

THE FUTURE IS CALLING.

MEET TELOS VX: THE FUTURE OF BROADCAST TALKSHOW SYSTEMS.

VX is the world's first VoIP talkshow system. It's incredibly powerful, incredibly flexible, and highly scalable -- a powerful whole-plant broadcast phone system that's also economical enough for stations with just two or three studios. VX connects to traditional POTS and ISDN telephone lines via standard Telco gateways. But it can also connect to VoIP-based PBX systems and modern SIP Trunking services to take advantage of low-cost Internet-delivered phone services. Using standard Ethernet as its data backbone, VX significantly eases the cost of phone system installation, maintenance and cabling, while making it easier than ever for talent to take control of their phone system. VX is truly the future of broadcast phones.

ABOUT TELOS

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. We 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. And 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 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 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.

ABOUT THE VX BROADCAST VoIP SYSTEM

With the VX, we've wedded the capability of modern networking to the remarkable power of today's digital processing to bring the benefits of the resulting synergy to broadcast facilities. With VX, you can move and share lines between studios at the touch of a button. VX is naturally scalable, capable of serving even the largest of facilities — while remaining surprisingly cost-effective for even single stations with more modest needs. To make the most of this networked environment, we've built VX around the VoIP (Voice over IP) standard.

WHY VOIP FOR BROADCAST?

VoIP has taken the business world by storm, increasing the flexibility of office phone systems and PBXs while simultaneously

VX: WHERE PHONES AND IP NETWORKING INTERSECT.

lowering maintenance and equipment costs. In fact, most Fortune 500 companies have replaced their older PBX systems with VoIP for just these reasons.

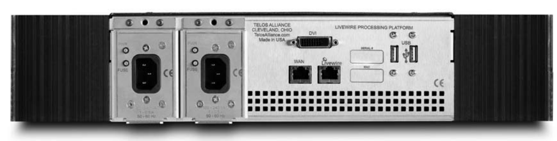
VoIP is a natural for broadcasters, interconnecting the phone system CPU with audio interfaces, phone sets, console controllers, and PCs running screening software by way of efficient, low-cost Ethernet. Using VoIP, you can finally share phone lines among multiple studios and route caller audio anywhere in your facility, easily and instantly. Got a hot talkshow 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 can even connect with your business office's VoIP PBX to allow easy call transfers.

REDUCED COST. INCREASED FLEXIBILITY.
This sophisticated networking 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 taking their attention from the board.

The VX system's standards-based VoIP architecture helps you save money, too, by widening your choices in telco providers. You can connect to traditional POTS or ISDN phone lines using standard telco gateways — or connect to pure VoIP services using modern SIP Trunking, which can deliver substantial savings to stations that need a large number of lines.

But it's not just VoIP — It's VoIP from 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 notoriously 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 should you choose to use SIP Trunking telco services, calls from mobile handsets with SIP clients will benefit from VX's native support of the G.722 "HD Voice" codec, instantly improving caller speech quality.

VX SYSTEM COMPONENTS



VX ENGINE

The VX Engine is the heart of the system. A 2RU rack-mount device with enormous processing power, the VX Engine provides all the call control and audio processing needed for the system. You can have as many as 48 lines active simultaneously — on-hold, being screened, queued for air, etc.— and up to 16 "audio connections" on-the-air at once. (Think of an audio connection as a hybrid, with each bidirectional console connection containing the audio from phone system to console and the clean feed from console back to phone system.). Its two Gigabit Ethernet ports provide a cost-effective interface to both telephone lines and studio audio via proven Livewire AoIP. VX is Web-based, so remote control and configuration are a snap — engineers can work with it from any place they can get online.

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 as well.

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

You'll notice that there are no audio I/O or telco ports on the VX Engine. 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, VX Producer PC applications, console-integrated controllers, etc.

For traditional phone services, you can choose standard telco gateways from Patton, Cisco, Grandstream and others to connect to T1/E1, ISDN, and POTS providers. And, if you have a VoIP-based PBX or SIP Trunking telco service, the VX uses standard SIP (Session Initiation Protocol) and RTP (Real-time Transport Protocol) to direct up to 48 simultaneous on-air phone calls.

VX SYSTEM COMPONENTS

VSET12

The VSet12 phone controller is an IP-based phoneset 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 system has 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. You can even map groups of lines to a single fader.

VSET6

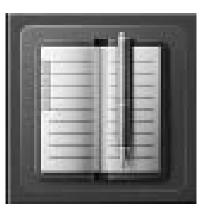
VSet6 is a six-line phone controller for VX. Like the VSet12, it has a bright, attractive LCD color display with Status Symbols that feed talent instant information about line and caller status, and controls that enable talent to step through queued calls, busy incoming lines, lock calls on-air, start an external recording device, et cetera.

VSET1

VSet1 is a single-line phoneset that's perfect for news booths or production facilities where multiple lines are not required. As with its multi-line brethren, VSet1 has a big LCD color display that helps users intuitively navigate through available options, and provides information such as caller ID time ringing and time on-hold, and even screener comments from the VX Producer software application. All VSet phones can be powered by PoE from a Telos-approved switch, a PoE port on an Axia console engine, or by using the included power injector.

VSET CONTROLS

TRANSFER



ADDRESS BOOK

BUSY ALL

DEVICE CONFIG.

DIALED CALLS

ON AIR

AUTO ANSWER

RECORD

CHAT

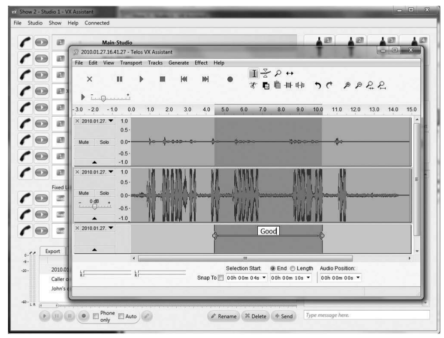
MISSED CALLS

VX SYSTEM COMPONENTS

VSET CONTROLS

Easy-to-understand VSet controls, with a friendly visual interface, let talent manage incoming lines, lock calls on-air, start an external recording device, and take a queue of calls to air sequentially, for precise management of multi-call interviews or conferences. The LCD displays deliver detailed line status, caller information, caller ID, time ringing-in or on-hold, and even comments entered in the VX Producer screening software application. Shown above are a few of the attractive, instantly-understandable Status Symbols that help talent run tight, mistake-free shows.

A built-in address book and call history log round out VSet12's features. As with the rest of the VX system, each VSet12 has its own web server for easy remote configuration and software upgrades.



VX PRODUCER

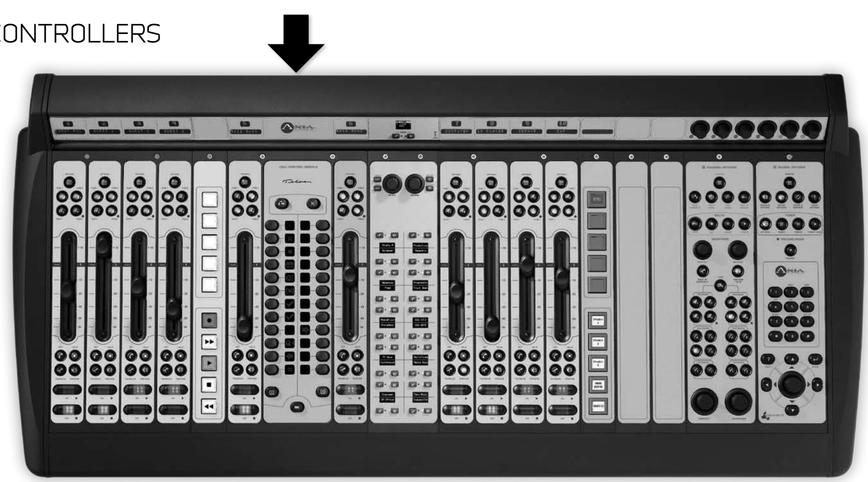
VX Producer comes with every VX Engine purchase, and has a site license for deployment throughout your entire facility. Its intuitive, user-friendly interface provides the usual call screening functions for phone-active broadcasts (caller status, screener's comments, Caller ID, time-per-call for up to 12 lines) – but with a number of enhancements enabled by the IP nature of the VX system.

For instance, VX Producer's integrated softphone uncomplicates the producer's life, since the PC interface is used for all operations, including answering and making calls, assigning priority, writing notes, etc. It also reduces cost, as no hardware phone is needed.

VX Producer also has a built-in audio recorder/editor with which a producer can record and edit a phone call without leaving the application. A producer can readily record calls for later play, which can be edited with a PC application running on the same machine, then sent to the on-air studio or talent's PC over the network for quick and convenient airing.

Like the VSet12, VX Producer includes an address book and call history log. There's also a Chat function that lets producers and talent communicate quickly via off-air instant messaging. And VX Producer is included at no extra cost with every VX Engine purchase.

CONSOLE CONTROLLERS



VX SYSTEM COMPONENTS

CONSOLE CONTROLLERS

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: IP-Audio networking technology provides the ideal way to integrate broadcast phones into the on-air console — the control center of every studio.

VX connects directly to Axia Element 2.0, 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. And soon, VX console controllers will be available for use with other console brands, too.

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. And since the console now communicates directly with the phone hybrid, mundane tasks such as mix-minus generation, starting recording devices, and playback of recorded off-air conversations can all be automated.

VX INTERFACES

VX Audio and Logic Interfaces let you connect VX to any non-networked radio console or other broadcast equipment, using standard analog or AES/EBU interfaces. A GPIO Logic interface provides control logic where needed.

VX ANALOG AUDIO INTERFACE

The VX Analog audio interface has eight balanced stereo inputs and eight balanced stereo outputs, presented on easy-to-install RJ-45 connectors. The inputs are switchable to accommodate consumer level -10dBv or professional level +4dBu. Outputs are short-circuit protected and capable of delivering up to +24dBu before clipping. Superior performance specs include 102dB of dynamic range, <0.005% THD.

VX AES/EBU AUDIO INTERFACE

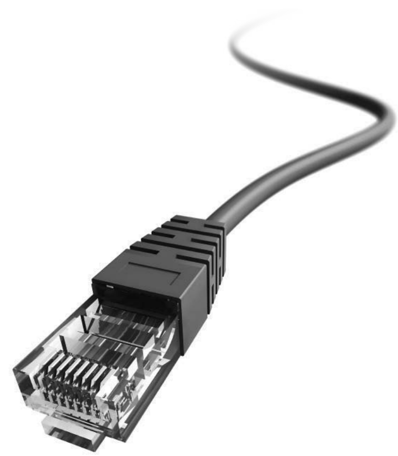
The VX AES/EBU audio interface provides eight digital AES3 inputs and outputs, each on a separate RJ-45 connector. Studio-grade performance specs, like 138dB of dynamic range and <0.0003% THD.

VX GPIO LOGIC INTERFACE

Each VX GPIO logic interface has eight assignable logic ports. Each port contains 5 opto-isolated inputs and 5 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.

THE POWER OF IP REALIZED

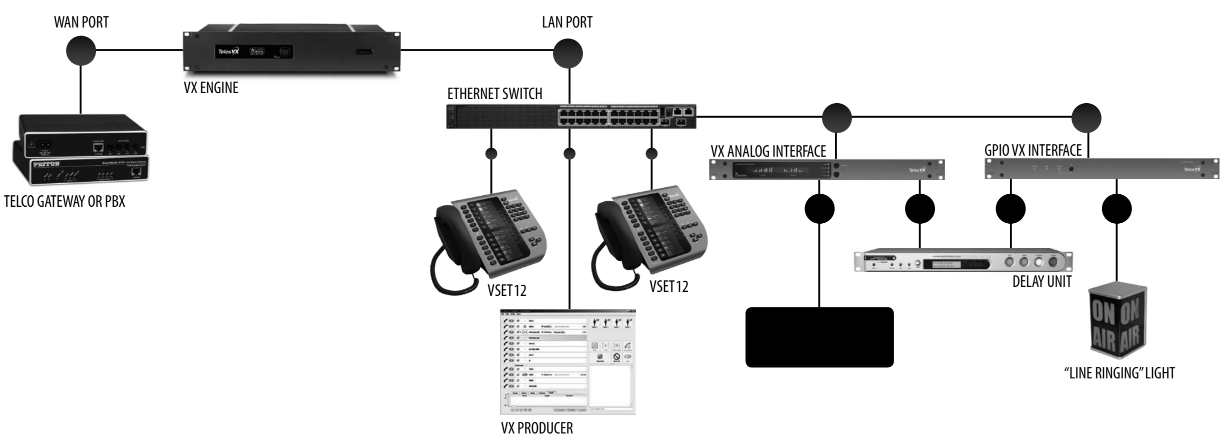
With VX, there's no need for the maze of discrete cables once required by a multi-line talkshow system. All VX components are linked with standard Ethernet, so a single CAT-5 cable provides:



- » 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



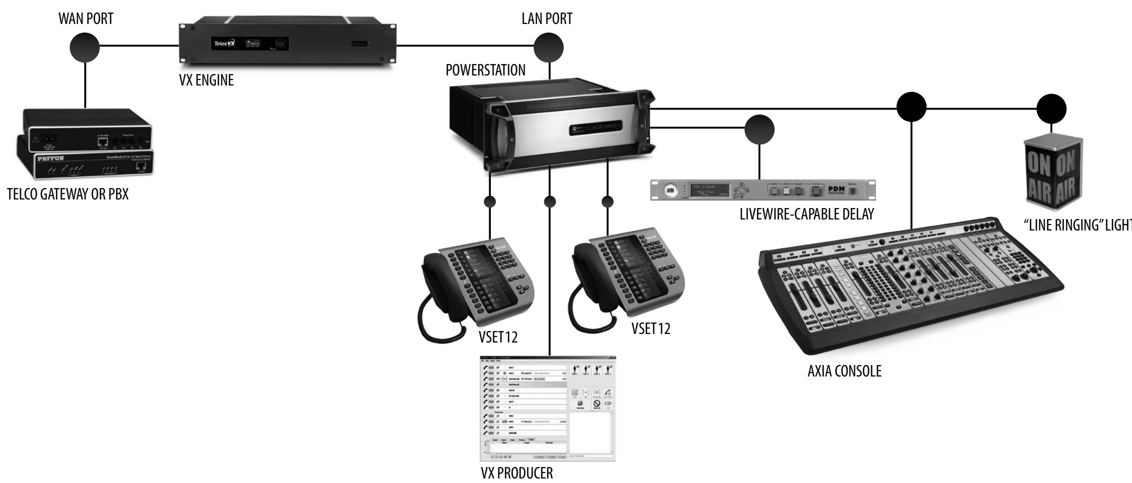
NON-AXIA INSTALLATION DIAGRAM

Got an Axia Livewire AoIP studio network? Telos VX will plug right in. It's the seamless integration of studio phones, mixing consoles and routing network you've dreamed about!

Don't have IP-Audio networking yet? Not to worry. . . VX will work with all console brands, networked or not, via VX Audio and Logic interfaces – compact 1RU breakouts that put multiple I/O channels right where you need them.

The Telos VX is a "facility wide" on-air telephone system. Facility wide means multiple studios, multiple stations, multiple shows and with minimal hardware requirements. Telco is delivered via IP through a POTS, ISDN or T1 gateway device, a SIP PBX, or a dedicated IP circuit using SIP Trunking. All line and audio connectivity is via Ethernet. This diagram shows a typical studio with an analog mixer, using VX Analog and GPIO logic interfaces to connect to the console and other broadcast equipment.

HOOK IT UP YOUR WAY - AXIA INSTALLATION DIAGRAM



AXIA INSTALLATION DIAGRAM

Installing VX in facilities with an Axia network requires even less time and hardware. The audio inputs and outputs used and produced by VX are Livewire real-time audio channels and travel over your existing Axia system just like the rest of your audio. Axia console GPIO ports can be used for "phone ringing" tallies or remote control of the 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 amazing amounts 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 .

NEO SOFT

NeoSoft offers NeoWinners, a full-featured contest-management software that integrates with Telos VX to add sophisticated screening and caller demographic features to your phone segments. NeoWinners' powerful suite of software tools can automatically schedule contests, record winner information, automatically track prize inventory, and easily generate winner lists, confirmation e-mails and even mailing labels. Contest and winner data is stored in a central, networked database for instant access by air talent, Promotions personnel and reception staff.

VISIT WWW.NEOGROUPE.COM TO LEARN MORE.

Fraunhofer
IIS

ECHO? IT'S OUTTA HERE.

The Fraunhofer AEC in the Telos VX is a remarkable new development. It's quite frankly the most stunning echo cancellation that we've ever heard. Its performance is shockingly impressive, permitting very high loudspeaker volume with no noticeable feedback or return echo, and it completely solves the longstanding problem of feedback and echo when a loudspeaker-to-microphone acoustic path is required in the studio, such as when DJs prefer to record calls without using headphones or when guests need to hear calls without headphones.

FEATURES AT A GLANCE

- The first VoIP telephone system designed and built specifically for broadcasting.
- Works with POTS, T1/E1, ISDN and SIP Trunking telco services for maximum flexibility and cost-savings.
- Standards-based SIP/IP interface integrates with most VoIP-based PBX systems to allow transfers, line sharing 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 easily manage a whole facility's studios, with support for up to 100 control interfaces (VSet phones, console controllers, screening software connections).
- System capacity of up to 48 standard phone lines; supports up to 250 SIP numbers.
- Up to 48 calls may be placed on-air concurrently – truly a "whole-plant" solution for on-air phones.
- 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 to any radio console or other broadcast equipment using available Analog, 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 modules can integrate VX phone control directly into your mixing consoles.
- Included VX Producer screening software with built-in softphone 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.
- Wideband acoustic echo cancellation from Fraunhofer IIS completely eliminates open-speaker feedback.
- Support for G.722 "HD Voice" codec enables high-fidelity phone calls from SIP clients.

FAQs

The Telos VX uses VoIP. What does that mean to me?
Let's define what "uses VoIP" means. The VX uses it in two distinct ways: One: it can connect to Telco services using standard SIP VoIP. You benefit from having options –connecting to PBXs digitally, to ISDN and analog lines via gateways, etc. With VX you can finally integrate your on-air phones with office phone systems from a variety of vendors. Getting Telco service from VoIP dial tone providers means that your audio quality and hybrid null will be much better as VoIP dial tone is delivered "4 wire" without hum, noise, and loop loss. Building on ubiquitous VoIP standards means a variety of third party hardware can offer flexibility. And you might save a lot of money getting service this way. Two: VX system components connect to each other over standard IP/Ethernet networks with all the advantages that brings. For example, in Livewire-equipped facilities, one RJ-45 jack connects dozens of audio channels and rich control to phones-like controllers, PC applications, integrated console controllers, etc.

So I can use the VX with my regular POTS lines? How? Can I use ground start lines for incoming calls only?
You can do that, and it's not difficult. You only need a POTS gateway device. However, we encourage using digital delivery for the best sound quality.

What about ISDN BRI and PRI lines?
If you have ISDN now and want to keep it, again there are gateways available. However, it is often cheaper to port these numbers to a VoIP dial tone provider. All of these options are worth considering.

But you must have old-fashioned analog and ISDN circuit switched connections covered somehow, eh?
Yes, VX works with these by means of "gateways" made by many third-party providers such as Grandstream, Patton, and others. See the "Gateway Products and Suppliers" section of the VX User Manual for some suggestions.

How do I know if my existing PBX is VoIP capable?
Since VoIP depends on SIP, and SIP is a defined standard, the only question is if your PBX can provide a trunk between itself and your VX system. If so, your PBX will work with VX – but if you have any doubts, call Telos Support and we'll help you figure it out.

My PBX is not VoIP capable. What else do I need to order, and from whom?
If your existing PBX cannot provide a SIP trunk line, it may be too old to offer such service, or might need upgrading. Contact your

FAQs

PBX provider and ask if your PBX can be upgraded for use with VoIP systems.

If you do not need to connect VX to your office PBX, however, you might opt for pure SIP Trunking service. We have experience with several SIP Trunking providers, such as EndStream Communications and Bandwidth.com, but you can obtain service from any VoIP provider that will give you a "my own device" plan. Be aware, though, that a simple Vonage account will not work – they limit the number of calls you can take based on the number of trunks you have (usually, in this case, just one). This type service is a home phone replacement — not a true SIP provider.

So I can experiment with SIP VoIP trunks? Why would I want to do that?
Actually, we recommend it! We think that you'll find that they work better than you may have expected, as many of the VoIP problems we have seen are caused by limitations of analog terminal adaptors (IP to POTs gateway devices)! Since these are not needed with a VoIP-based system such as the VX, that class of problem is eliminated. There are a number of inexpensive ways to try VoIP without risk. You can also get VoIP-delivered numbers from distant area codes and exchanges. If you're paying mileage for foreign exchange lines or have national toll free numbers, you'll definitely want to consider this option.

Let's say I decide to use SIP trunking. What sort of line provisioning do I need to order from my VoIP provider?
Unlike ISDN, SIP trunks are actually pretty simple. Your VoIP provider will give you the IP address and registration information you will need. Once you have this, enter it into your VX Engine using the SIP Configuration web page in the VX Control Center.

Will a SIP trunk ordered from, say, AT&T work directly with VX as long as it has QoS?
Sure! AT&T (or any SIP Trunking provider) will provide you with the authentication info necessary for the connection between VX and your VoIP provider.

I can't put a flasher across a VoIP line, so how can I flash a light when the hotline rings?
This same issue arises with ISDN, so beginning with our TWOx12 we included a GPIO output for this function. Not to worry – The VX has this capability. In fact, it has multiple outputs which can be assigned to any of your VoIP lines.

I have been watching VoIP with interest. But reports I have heard about services such as Vonage are that sometimes they work well, but other times not. Frankly, I am surprised to see Telos advocating it.
We get this concern often and understand why you ask. The term Voice over Internet Protocol (VoIP) does not distinguish between "VoIP over the Internet" versus "Voice over other (managed) IP networks". If you have been keeping up with the transition to IP Codecs, you probably have noticed the same terminology issue there. IP-based STLs over an IP T1 are just as reliable as traditional STLs over traditional TDM T1 circuits. You are completely right to be concerned about a VoIP trunk (or an STL) over the Internet, as that is not at all the same thing, and performance in that case could be variable. Inside the facility, on a LAN, all problems disappear, since you have plenty of bandwidth and full control over the network.

So shared bandwidth is the only problem with VoIP?
Well, there's audio quality. In the early days VoIP used a lot of compression, with bit rates being as low as 6kbps. Needless to say, the resulting audio was not impressive. These low-rate codecs have mostly fallen by the wayside. The lowest-grade codec the VX supports is g.711, the standard for digital audio in the telephone network pre-IP. And you will eventually benefit from higher-fidelity codecs as these proliferate in the VoIP world.

How does AoIP relate to VoIP?
Despite the similar names and underlying technologies, they are very different with regard to performance and application. An analog phone line and a balanced 600_ studio audio circuit are pretty much the same tech, but the applications and performance are very different. "AoIP" has come to mean professional studio-grade audio networking - full-fidelity and usually with no compression. Low-delay and synchronized channels are other distinguishing characteristics.

Another way the two differ is that AoIP uses an advertising/discovery protocol for receivers to find sources instead of the Session Initiation Protocol (SIP) that VoIP employs. AoIP uses a system-wide clock mechanism to support low-delay and tightly synchronized channels.

Finally, AoIP often takes advantage of IP's multicast capability to permit multiple receivers to listen to an audio source efficiently. AoIP is intended for managed, guaranteed-bandwidth networks, such as with an Ethernet switch as the core of a local area network.

And what about "IP codecs?"
Now you are very close to VoIP! These use SIP for call setup and various codecs for compression, so are similar to VoIP telephones. In fact, they actually are VoIP telephones and can sometimes interoperate with them. They have better codecs than VoIP phones, though. AACELD, in particular. Advanced IP codecs, such as our Z/IP, employ sophisticated technologies to overcome the Internet's deficiencies. Dynamic buffering, error concealment and more clever stuff.

I have read that some VoIP PBXs use IAX trunking. Can I use the VX with these?
IAX is a protocol invented by the Asterisk people. It provides functions similar to SIP, but with more bandwidth efficiency. The

FAQs

VX doesn't support IAX trunking at this time. But you can connect the VX to an Asterisk with SIP trunking or as multiple SIP extensions. There's plenty of bandwidth on a LAN, so this works fine, while staying with a standards-based approach. We like Asterisk as a VX adjunct. It can add voice mail, automated attendant, blocking callers from caller ID, off-premise SIP extensions, and more, to a VX installation. Asterisk is free Linux-based PBX software that runs on a PC. The VX and Asterisk PBX are an attractive combo we expect will become popular within the broadcast industry.

I would like to use an Asterisk PBX with my VX system. Does Telos offer a pre-configured Asterisk installation?
A pre-configured Asterisk box just isn't practical due to the variety of configuration options needed to tailor such a system to your station's specific equipment and Telco service. We'll be happy to provide resources that can help you get up and running, though.

I see that Asterisk can host POTS and ISDN interface cards. Should I use those to interface my old-tech lines, or should I use a stand-alone gateway?
This is a matter of taste, but a general guideline is that analog lines are best interfaced with an external gateway to optimize audio quality, while ISDN BRI and PRI can use Asterisk cards to have a lower-cost solution.

How can I get reliable VoIP trunk lines? What is involved? Can you recommend a vendor?
Yes, we can assist. We have been working closely with the VX beta sites and other early adopters of VoIP, so we have plenty of experience to share. There are several types of VoIP dialtone providers. You'll want to consider how the service will be delivered to you; via the Internet (like Vonage), or via a dedicated IP circuit from the provider that includes a Service Level Agreement and guaranteed Quality of Service (as offered by a number of vendors including most of the traditional Telcos). For discussion of this and other matters, check the Telos web site on a regular basis as we continue post material on this and related topics.

I know that SIP is supported by the new IP codecs. Will the VX be able to connect to my Zephyr/IP in the field? Other codecs?
As we hinted above, Yes! The VX supports g.722 (7khz, 'wideband audio' and g.711 (3.4khz, "phone quality").

What about SIP, SDP, RTP, ENUM and UDP?
We know that engineers are lifetime learners and encourage that. However, just as you probably don't know much about "SS7" or "IUP" in the telephone network, understanding these details is optional. We do have White Papers on our web site to educate you on these, and other, terms. Start with the one below. You could also read Steve and Skip's AoIP book, too, for a fun and comprehensive coverage of this stuff.

Steve Church's paper on the adaptive IP codec, Advanced Tech for IP Remotes, can be downloaded from Telos-Systems.com/techtalk/ .

Does Livewire technology come in to the VX picture?
Yup. The VX takes VoIP on the Telco side and Livewire AoIP on the studio side. This makes integration with Axia consoles and networks easy and efficient. If you don't already have a Livewire network, you would use Axia analog or AES audio 'nodes' to provide I/O in either format. Each node provides eight stereo inputs and eight stereo outputs to and from the system. Each Axia GPIO node provides 8 "groups" of 5 inputs and 5 outputs, covering the needs of 8 studios. Telos Support is always available to help you specify exactly what you need. If you are new to Livewire Technology you may wish to skim through our Primer at AxiaAudio.com/manuals/files/IntroToLivewire2.1.pdf .

So the two can live together side-by-side on the same LAN?
Yes they can.

So let's talk caller audio quality. What does VX offer compared to the Nx series and your legacy products?
Advanced audio processing, and the fact that you never have to overcome Telco loop losses or extra two- to four-wire conversions means that the voice quality is as good as it can be. Calls from mobile phone calls will be less than perfect at times, but VX extracts the best possible from them. Caller audio is maintained digital and four-wire end-to-end, and VX does a clean sample-rate conversion from the Telco rate to Livewire's 48khz. There is also a sophisticated dynamics processing section and automatic EQ designed with help from our colleagues at Omnia.

How are callers on the VoIP trunks going to sound? What about echo? Won't cell phones sound even worse than usual?
The VX has enough processing horsepower to deal with even extreme echo situations, and four-wire Telco delivery means that the only external echo path is from the caller's line, when it's analog, and the caller's handset. The VX uses Telos' latest hybrid technology (5th generation), enhanced with the latest state of the art acoustic echo cancellation. Even when using open speakers, and changing levels during a call, the new algorithm makes feedback nearly impossible.

I heard that amazing Acoustic Echo Canceller on your Axia intercom system. No feedback at all with the speakers blasting and the mic a few inches away. Why don't you put that in the VX so that DJs can answer calls on the cue speaker without feedback?
Done! The VX has it.

FAQs

Steve Church once told me that IP cell phones can sound better than usual 3 kHz circuit-switched phone technology - something about G.722 dot something. Is this true?
Right. Current Cisco VoIP phones, for example, support the g.722 codec. The VX supports this, as well. However, Steve was probably referring to "AMR Wide Band" also known as G.722.2., sometimes called "HD Audio". It's 7khz and doesn't sound at all like "phone audio" - in fact, it sounds better than regular G.722! AMR-WB is part of the new ITU standard for mobiles, so should grow over time. Meanwhile, some IP-based apps for mobiles are starting to use wideband codecs, such as MPEG-ELD in Apple's 'facetime' app.

OK, I am starting to see the light. Cool stuff, but where's the catch? Is VX hard to install and configure?
Setup is via web. It may be little different than what you're used to (or not) but it's not difficult, and some customers never crack open the book to set it up. Power and flexibility do come with a little complexity, but we'll always be at your side should you need us.

There is no such thing as a free lunch - it must be hard to use then. I know there's a catch... I really don't have time to explain a new system to the air staff.
We know! Rest assured it's easier for your talent, not harder! We recognize that any time you change anything in a studio, there can be some transition time. While there are a lot of new features in the VX, your staff can use the basic stuff immediately because it works just like familiar and comfortable Telos gear. The color, hi-rez LCDs and seamless console integration (to Axia Element and iQ) enhance the user experience. As you read this, systems around the world are screening calls and putting them on the air without drama. Jocks and Talk hosts alike praise the VX! Operators familiar with our longstanding two-column line selection will be right at home.

Is there any support for VIP lines?
Yes. We call these "fixed" lines, and you can have as many as you want. They are used for callers who stay on-air while other callers are coming and going.

What about conferencing?
This is one of the strengths of the VX. Since VoIP calls are four-wire, multiple lines can be conferenced with very high quality. The user interface lets operators assign selectable lines to multiple faders.

If I use an Axia console, it gets even better?
Yes – that's the ultimate. You start with the most flexible console/audio-platform and then add smoothly integrated phones with the IP network powering it all. Sweet! The network delivers any of your Telco lines to any of your studios, in any combination. Any line is available in any studio at any time.

I notice the VX engine has both a LAN and WAN connector; why is that?
It's a built-in firewall, isolating the VoIP connection from your studio network. We use this same approach in the iPort Livewire-WAN MPEG gateway.

What about call screener and database functions?
A basic Call Screening app, VX Producer, comes with the system. Other networked, PC-based apps, such as Broadcast Bionics' PhoneBOX VX or NeoGroupe's applications put information about your callers in front of your producers without the need for caller ID boxes, serial cables or other hassles.

What about SMS messages and chat - Can they be integrated into my phone system?
Using Broadcast Bionics Phone Box VX, yes! Telos has always built open systems to allow others to create their own visions around our gear.

OK, so the catch has to be the price?
The VX lets you leverage cheap networking to serve your entire facility. Since you don't need hardware boxes for each studio, cost is surprisingly reasonable. You'll use the VX in your on-air studios to replace older multi-line systems, and you'll use it to replace hybrids in newsrooms and production studios. You might also decide to eliminate walls full of "couplers" for pre-delay IFB dial-up lines, and transitioning your Telco service might save you a lot of scratch. We've seen stations saving thousands of dollars a month (no kidding) by eliminating POTS lines, with their taxes and fees.

Anything else cool about the VX?
Did we mention the color LCD user interface on the new VSet phone/control surface? Producers and talent love it!

TELOS VX SPECIFICATIONS

SYSTEM

- +Maximum number of phone lines: 48, when used with aLaw or uLaw codecs for VoIP lines. (Higher quailty codecs, such as G.722, consume more system resources and result in a decreased number of total lines available.)
- Maximum number of SIP numbers: 250
- Maximum active on-air calls: 48
- Maximum number of simultaneous audio connections (Livewire I/O channels): 16 systemwide
- Maximum on-air calls on one fader: 4

AUDIO PERFORMANCE

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)
- 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 AND 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

TELOS VX SPECIFICATIONS

VX ENGINE

IP/ETHERNET CONNECTIONS

- One 100BaseT/gigabit Ethernet via RJ-45 LAN connection
- One 100BaseT/gigabit 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
- Line Echo Canceller (hybrid)
- Acoustic Echo Canceller (wideband)

POWER SUPPLY AC INPUT

- Auto-sensing supply, 90VAC to 240VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 100 Watts

OPERATING TEMPERATURES

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

DIMENSIONS AND WEIGHT

- 3.5 inches x 17 inches x 15 inches, 10 pounds

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.
- LW-equipped studios may take the audio directly from the network. Interface Nodes are available for pro analog and AES3. (uses standard Livewire Nodes)

TELCO CONNECTIONS

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