TelosVX

Multi-Studio IP Phone Interface System

USER'S MANUAL

Version 2.0.2, December, 2018

THE TELOS ALLIANCE®

TelosAlliance.com

User Warnings and Cautions

The installation and service instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel

This instrument has an autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240VAC. The ~ symbol, if used, indicates an alternating current supply.

This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

CAUTION: HAZARDOUS VOLTAGES

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

WARNING:

The product's power cord is the primary disconnect device. The socket outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings on the rear and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

USA CLASS A COMPUTING DEVICE INFORMATION TO USER.

WARNING:

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC rules, part 15, subpart j, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. Note: objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of shielded cables.

CANADA WARNING:

"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the radio interference regulations of the Canadian department of communications."

"Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de Class A) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

CE CONFORMANCE INFORMATION:

This device complies with the requirements of the EEC council directives:

- ◆ 93/68/EEC (CE MARKING)
- ◆ 73/23/EEC (SAFETY – LOW VOLTAGE DIRECTIVE)
- ◆ 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to those standards: EN50081-1, EN50082-1.

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Patent information can be found at www.TelosAlliance.com/legal

Updates

Telos VX features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

We welcome feedback on any aspect of our products or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments or suggestions.

We support you...

By Phone/Fax

You may reach our Telos Support Team in emergencies by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:00 AM to 5:00 PM USA Eastern Time, Monday through Friday.

By Email.

Non-emergency technical support is available at Support@TelosAlliance.com.

By Web

The Telos Web site has a variety of information that may be useful for product selection and support. The URL is https://www.telosalliance.com/Telos .

SERVICE

You must contact Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We will issue a return authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Telos Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

Warranty

 For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Register your product

Register your product today to get the full benefits of our warranty, support, and product updates. telosalliance.com/product-registration/

The Telos Alliance

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

We're grateful that you have chosen audio tools from Telos® Systems, Omnia® Audio, Axia® Audio, Linear Acoustic®, 25-Seven Systems®, and Minnetonka Audio®. We're here to help you make your work truly shine. We hope that you enjoy your Telos Alliance product for many years to come and won't hesitate to let us know if we can help in any way.

The Telos Alliance

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1 Welcome

What's VX anyway?

The VX is a telephone system designed for broadcasters. It's different from your office phones because broadcast talent and producers have different needs than most other "typical" telephone users. We designed it to do exactly what you need in the studio. It has bright clear displays, tougher, bigger buttons and extra functionality that isn't found in a common office phone.

The VX system is typically composed of these parts: the engine, vset phones, A variety of optional call controllers, audio interfaces (for non-axia plants) and a Livewire capable switch, set up to handle the Livewire audio and control protocols.

The VX, configured appropriately, supports many studios and what we call "shows". In in VX, "Studios" are a collection of I/O configuration (audio and control) that are generally associated with a particular studio. The caller audio channels, the Mix Minus send to call channels, GPIO for flashing lights and the delay dump button. There is also the "show", that is, what lines appear on the users phones when that show is selected. A typical show would have say 6 call in lines in hunting, a couple of "warm" or VIP lines and a hot line.

The VX supports 24 "phone faders" (telephone "hybrids"). These may be divided among studios. The limitation is mainly due to CPU resources. There may be situations that allow more hybrids. It's best to arrange for a call with a Telos Field Applications Engineer if you get "close" to the maximum number of 24 or have a very large number of Vsets.

The Engine pretends to be a bunch of SIP telephones (called "endpoints" by the IT crowd) on a PBX or from a Provider. It looks and feels like the familiar "Key System" of days gone (mercifully) by, but produces the best sounding phone audio that you've ever heard. The workflow was designed for broadcast use specifically, making it easy to move studios (and take your phone numbers with you).

This is the second revision of the VX manual. The first was created back in 2012 and was co-written by Steve Church, Founder of Telos and myself, Joe Talbot. In that time, many hundreds of VX systems have been installed around the world, with great success, and we've learned a lot. Steve came up with the idea of the VX back when it sounded like it was "way off in the future". The future got here very quickly! At the time of this writing, December 2017, SIP is now a standard offering by most telephone service providers in the world. Every PBX currently manufactured supports SIP in some way, most completely. A notable exception to this is Microsoft's offering. Once called "Lync" and currently known as "Skype for Business", it's probably best to avoid it for broadcast use. Oddly SIP can be supported by it using special hardware gateways.

Reading the previous manual as this one was written, reminded me of how quickly the voice network and its operators have changed and also of the magnitude of the changes! In the previous manual, Steve talked about the archaic POTS signaling methods and the challenges of making a hybrid work on them. That all seems so distant and quaint today, but the changes that have occurred in telephony have been so ubiquitous that discussing the olden days of telecom technology is barely relevant anymore.

If you're new to VOIP and SIP, we can help. We've done so many of these systems that I like to say "We've already made the mistakes so that you don't have to." We'll try to make things clear in the pages that follow.

SIP has a lot in common with the former ISDN standard. In common are digital four wire audio, and message based call set up and signaling. These properties essentially bring the central office into your studio and improve audio and overall reliability. A SIP trunk, on an IP PBX is very similar to an ISDN PRI: It has multiple digital audio channels, supports Direct inward Dial (DID) numbers and caller ID. VOIP expands the feature set beyond what ISDN was capable of, or even imagined by its designers.

VX Provides:

♦ A single on-air phone system server that can supply all the studios in your facility with rich telephone capability.

♦ A common wiring and Ethernet switch infrastructure serves both your studio audio and telecom needs.

♦ VSet controllers, used on air or for call screening, have extensive capabilities owing to their connection over IP.

♦ Screening software running on PCs connect over the network, and may include integrated softphones

♦ Mixing console control surfaces can incorporate phone system controllers that need no additional connection; their signaling just rides on the existing network connection. Detailed status information can be displayed either on the phone control module or the console's main screen.

♦ Receiving Voice services via IP is usually much less expensive than the with legacy methods, and full office PBX integration is simple.

The diagram shows a typical installation, hinting at the possibilities inherent in the system.

WAN PORT

LAN PORT

VX ENGINE

ETHERNET SWITCH

TELOS ALLIANCE MIXED SIGNAL XNODE

ASTERISK PBX, PBX, SIP PROVIDER, OR GATEWAY

VSET 12

VSET 12

DELAY UNIT

CONSOLE OR ROUTER

"LINE RINGING" LIGHT

XSCREEN
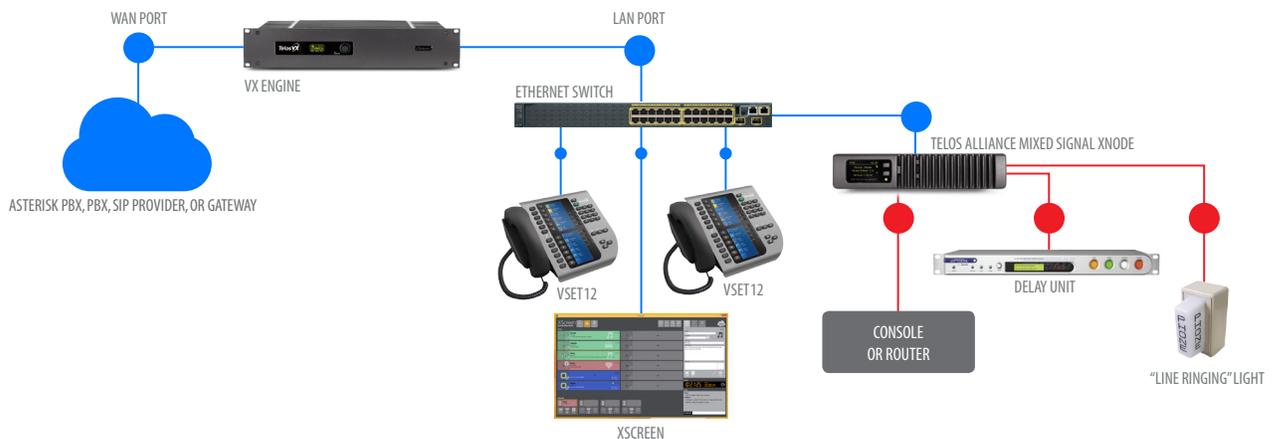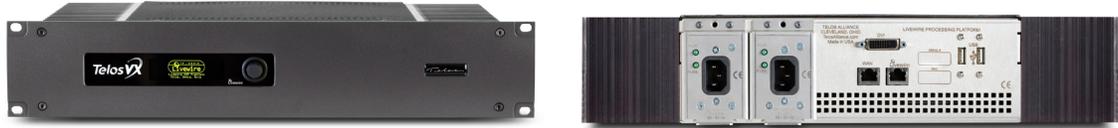
The VX Engine

A 2U rack-mount device with ample processing power, the VX Engine provides all the call control and audio processing needed for the system. It supports the SIP equivalent of dozens of telephone lines and many studios. Its two gigabit Ethernet ports provides a cost-effective interface to both telephone lines and studio audio via Livewire AoIP.

Call handling is sophisticated and flexible. Lines may be readily shared among studios. A 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 processing features also have taken a leap forward. Multiple calls can be conferenced and aired simultaneously with excellent quality. Each line is equipped with AGC, automatic EQ processing, and override ducking. Caller audio is incredibly consistent from call to call, in terms of both level and even in spectral content.

All connections to the Engine are via two Ethernet jacks that are extended with an Ethernet switch to support a wide variety of peripherals: telephone lines, Livewire studio audio, VSet phones, VX Producer PC applications, console-integrated controllers, etc.

The VX engine has hot-swappable dual power supplies and requires no cooling fans for the highest reliability.

The VSet Phone and Console Controllers

While you can control the VX with PC applications and mixing consoles. Most systems will include one or more Telos VSets. These are the phone-like controllers that have handsets for off-air conversations. These are available in 6 or 12 line models.

IP-based Telos VSet phones have large, high-contrast color LCD panels that provide line status, screening status, and caller information. Caller description text that is entered into compatible call screening applications will show up on the LCD next to the line select buttons.

For the comfort of familiarity, the VX can work like a traditional Telos controller, with calls being selected, held, and dropped in the way to which operators have grown accustomed.
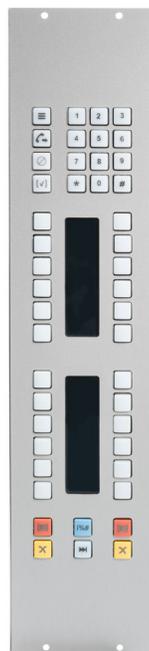
The Vset Console controller is an electronics package intended to be integrated into Audio consoles or turrets, using metalwork provided by the console manufacturer or integrator. It is an attractive and small Power over Ethernet (PoE) device with durable buttons and displays to allow Telos VX control from the board operators position. A drawing package is available for those wishing to create metalwork for the unit for their projects. Metalwork is available from most console manufacturers.

Console Controllers

Integrating phone line selectors into the studio mixing console is an operator-pleasing feature that is easy to accomplish in a networked studio. As this is written in December 2017, the following consoles have VX-compatible call control modules available.

♦ Axia Fusion

♦ Axia Element Console

♦ Axia IQ Console

♦ Other manufacturers consoles by using the vset Call Controller Electronics package described above

VX call screening software applications

PC based applications take studio phone operations to a higher plane. With many packages to choose from, most provide the usual call screening functions for phone-active broadcasts – the PC interface can be used for all operations, including answering and making calls, assigning priority, writing notes, etc. Using software may also reduce system cost, as no hardware phone need be installed.

We've partnered with User Interface and broadcast workflow expert Broadcast Bionics to offer X-Screen2, a "cloud based" call screening system. Basic operation is provided by a free version that is useful and fully capable. Though an optional paid version allows extensive database options and a ever-growing list of other helpful features.

Since every stations needs and preferences are different, third party vendors such as Neogroupe, Arctic Palm and others offer their own take on workflow, features and support options. Customizations are usually available for your own unique situation. Please have a look at them all, as they each have much to offer.

Because Livewire audio is available at the PC via standard LW Driver software, a producer can readily record calls for later play. These could be edited with a PC application running on the same machine. When a file has been produced, it can be sent to the on-air studio over the network.

Thus, depending on the software option chosen, the one Ethernet cable is used for:

- ♦ Telephone audio for the softphone, via Livewire.

- ♦ Livewire audio for the recording of calls.

- ♦ Transfer of recorded call files from the producer to the studio.

- ♦ Control messages between the PC and the VX main box for line selection, etc.

- ♦ Data messages such as call notes and IM between the producer and on-air studio.

- ♦ Database lookup of caller information, such as how many times they have called, the quality of their contribution, whether they have won any contests, etc.

- ♦ Web browsing, email, chat, etc.

The Livewire AoIP Network

The VX system components are linked via an Axia Approved standard Ethernet switch. In Livewire-enabled studios, this will already be present and no additional switch would be needed. A list of currently approved switches is available at https://www.telosalliance.com/Axia/What-Ethernet-Switches-has-Axia-Approved

It is important to understand that not all switches are created equal or are inter-changeable. A tested and approved Livewire capable switch is required to use livewire. That switch must also be set up for livewire using the local console port before it will handle multicast correctly.

The Acoustic Echo Canceller

Fraunhofer's acoustic echo canceler algorithm solves the problem of 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. The AEC in the Telos VX is a remarkable development. Its performance is shockingly impressive, permitting very high loudspeaker volume with no noticeable return echo. (If you've already experienced the Axia or Infinity intercom system, you know what we're talking about…)

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2 Installation and Configuration
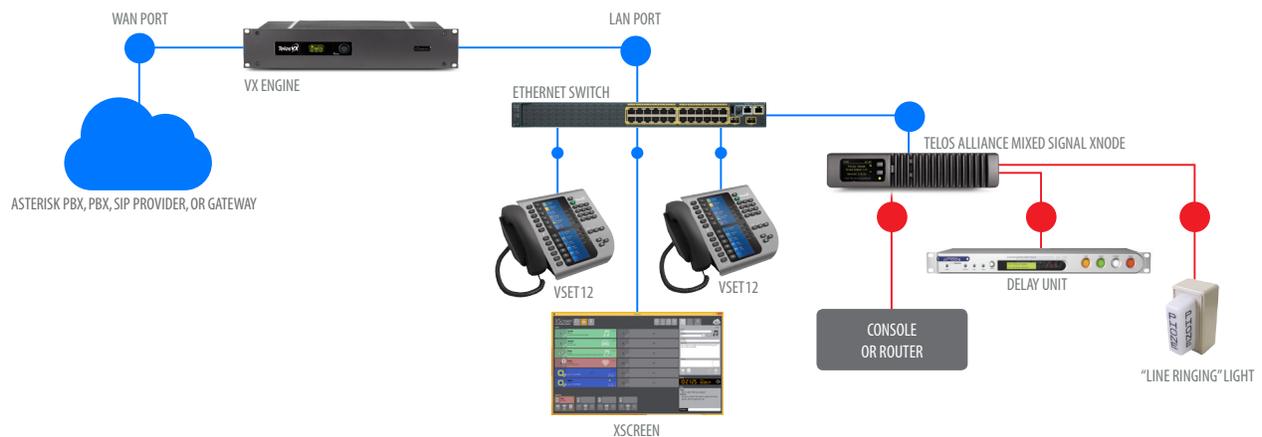
The diagram shows a generic installation that could be used in a typical studio. Two Ethernet ports are employed to keep the isolation between the WAN and LAN ports on the VX Engine. The LAN port carries the Livewire audio and everything else but the VoIP, which is on the WAN port. The livewire switch is an approved, livewire capable switch, while the one on the WAN port need be.

Telco lines enter two ways: From the PSTN via a PBX or from a SIP VoIP provider over an IP network, configured as endpoints. VX audio Interfaces or nodes provide analog, AES, and/ or GPIO connections to traditional studio equipment. Nodes are not needed when the studio equipment can accept AoIP natively. For an example of this sort of installation, see *Axia Console* as *VX Controller* in Section 5.

Connecting to the PSTN: PBXs and Providers

'PSTN' is an acronym for '*Public Switched Telephone Network*', the traditional telephone network is now made up of many types of service providers, including the Legacy Utility Telco providers often called Local Exchange Carriers (or LECs), Metro Ethernet providers (availability varies regionally), Fixed Wireless providers, Internet Telephone Service Providers (who deliver service via the public Internet) and Competitive Local Exchange Carriers (CLECs). Most providers and PBX manufacturers have adopted support for basic SIP trunks and some also provider services provisioned as "Endpoints". They also usually offer traditional interfaces like POTS, ISDN (PRI), often delivered over Ethernet or T1's/E1's as 'last-mile' connection technologies.

The VX system connects to Telco lines using industry standard SIP (Session Initiation Protocol). This means it is compatible with a wide variety of VoIP services, gateways, and PBXs. Gateways can be used to interface PSTN lines to SIP, though we discourage this, as we've found them difficult to set up and maintain, and don't provide the flexibility, audio quality and reliability of an all-digital connection.

We've had the best results with full-up IP PBXs, like Asterisk and its derivatives, Cisco, Mitel, Avaya, and many others. Some PBXs do have limitations in the number of "Third Party SIP devices" that they will support and there may be also be a variety of other concerns, so it's a good idea to arrange a free call with a Telos Applications Engineer before placing an equipment or provider service order.

Since VoIP phones are commonly used today for general office service. SIP lets you move calls between the office and studio systems with no audio degradation and a common dial plan. A PBX such as Asterisk also adds additional capability, such as automated attendant functionality, call recording, logging and time based routing. With appropriate add-on interface cards, Asterisk makes an excellent gateway to T1 and ISDN PRI Telco lines. It's especially good for high volume contesting situations.

No longer a 'niche' offering, as of 2017, SIP is now the standard for digital telephone service. The telephone service provider and PBX platform are among the most important choices that you'll make, and you'll be living with those choices for a number of years. We suggest a free call with a Telos Field Applications Engineer to help you compare and sort out those choices. Also see the app-notes in this manual for specific considerations and ideas.

What's today's equivalent of a 'Line'?

Historically, a "line" meant a pair of wires from the telephone company with a phone number. It had simple DC (on/off hook) and AC (ringing) signaling and audio all on the same pair. It could support a single call and would likely have a unique telephone number. If set up to do so by the telco, it could provide "hunting service" to allow calls to move onto another POTS line if the first one was busy. Today, everything has been "un-bundled". Signaling is separate from voice channels and the telephone number may be used my many channels at the same time. Another important distinction: VOIP audio is "4 wire", that is separate transmit and receive. Telephony has evolved, So, in the words of the Firesign Theater: "Everything that you know is wrong".

Let's look at this new reality in two very different ways: From the talent/producer point of view, and from the telecom engineers point of view.

The Talent side is the easiest to understand, so we'll start there. Since this is the engineering side of the manual, I'll let you in on our dirty little secret. We are pretending to be old POTS lines on a key system... for the user's benefit, at least as far as the workflow and feature set goes. We designed the system to look and feel familiar. Talent has different needs than the engineering side of the house does. In short: They have an interface that they've used for years and we've re-created it for them, though with many improvements, and we do all of the complex stuff "under the hood". Talent expects that each button on their Vset phone or call controller is a "line". VX works identically to a key system line from 1955, except that it has push button dialing, a color display with caller ID and an easy way to move callers to "air". It also has a skinnier ethernet type cord instead of the traditional 25 pair (or worse) amphenol plug ended cable(s).

In the original 2009 manual, Steve Church said "Users, blissfully unaware of all this, will undoubtedly continue with their conditioned habit, referring to a certain caller as being on a particular line. Thus is the word 'line' destined to join 'dial' in a peculiar departure from original meaning." As we update this manual at the end of 2017, It's clear that Steve has been proven "right " again.

From the Telecom Engineering perspective, it's very useful to keep the elements that histori-cally made up a "line" separate. Using all of these elements intelligently can save us money and give us flexibility that we never knew in the old days.

The changes that VOIP has brought us, have essentially moved the "Central office" right into our own facilities. We can now do things today that 20 years ago could only be done by the telephone company. It also means that we pick up some of the responsibilities for design and engineering and maintenance once take for granted as the domain of the "Telco".

Things that we now care about:

Direct Inward Dial (DID) numbers, sometimes called "Directory Numbers" - "Phone numbers" in plain English as spoken by most humans

Channels – Voice channels, like bearer channels in ISDN, each carries a call and uses 64-96kbs of bandwidth

Trunk Groups – Groups of voice channels, sometimes it's useful to artificially group calls to prevent calls to single number from using all available voice channels, impacting other users (like "choking").

Dial Plans – determine call routing for anything dialed. Ideally modeled after something familiar

Networks/Subnets – Because we're dealing with real-time voice, low jitter must be maintained to keep calls sounding good. This could involve separate LANS/VLANs, Quality of Service (QOS) settings, IP routing and even firewalls.

Codecs – we now have some control over what codecs are used. The PSTN as it exists in 2017 uses the g.711 codec to provide "toll grade" (3.4 khz audio, 7 or 8 bit, companded), but private networks are not subject to this limitation. Many use the G.722 "HD Voice" codec for better voice quality, and availability is expanding. Some providers also use the aggressively com-pressed G.729 codec which provides audio quality far inferior to the standard G.711 codec. Use of this codec should be avoided in broadcast use. The VX does not support the G.729 codec due to quality concerns.

Bandwidth – We need to have enough low-jitter bandwidth continuously available to accom-modate the expected number of calls, including calls on hold, and a way to deal with additional calls that exceed our expectations. It need not be "really fast", but does need to be "steady" and able to deliver constant packet streams.

Each audio channel used by a call consumes 64-96 kbps.

Network Security – We must consider everything from proper operation to "keeping the bad guys out". Please read our app-note concerning this topic which could fill many books by itself.

While this may all sound dramatic or daunting... It's not that tough, and both providers and Telos can help. Remember, the phone company used to manage all of this. Here's an example of how it's done today. Let's say that you want to emulate the existing studio phones:

5 Stations each with the following -

6 public call in numbers, request or contest lines

2 "Warm" lines for DJ business, guests or promotions

1 "Hotline" for the boss to call in on when they hear something that they don't like

Here's how we'd do it assuming ordinary levels of traffic for the typical music radio station:

Example Option 1 – Telephone company provided (typical)
From Telco or a CLEC We order:

1 or 2 T1's carrying Primary Rate Interfaces (each carries 23 bearer channels and a D or Data channel that is used for call setup). This will allow up to 46 simultaneous inbound or outbound calls through the system.

A block of 20 DID numbers, initially to test with, then used later for utility things like Hot and Warm lines, IFB lines, VOIP phones and voicemail boxes. Lets say that the telephone numbers are 311 555-2300 through 2320. DID numbers are very inexpensive (pennies a month each), so don't be afraid to order plenty.

Ported "Vanity numbers" come later, after everything is all done. We only need the published numbers so... 5 of these (1 for each of the 5 stations)

Expect this to take 15-90 days to provision and install.

From your distributor we order:

Asterisk Machines with 4 port PRI cards (1 or 2 so you have a spare)

Telos gear (after a call with a Telos Applications Engineer)

Here's how it this all works in practice once installed:

A listener dials 311 555-2300 (your studios number). That number is recognized by Telco and is connected to a (PRI) trunk to your facility. The telco connects to your PBX and sends the DID number to the PBX on the far end of the Trunk. The PBX uses those digits to route the call further, to the correct extension in most cases. The PBX rings the desired extension and causes ringback tone to be sent to the calling party until the called party answers. When the call is answered, a two way audio path is cut through and the network is signaled that answer supervision should be returned, charging the calling party of the call.

If there was more than one appearance of the called extension (more line buttons), more calls could be passed to that extension. This would continue until no more channels were available, or an artificial limit in either the PBX, Trunk or Telco would cause the "busy here" message to be sent back to the calling parties PBX or Central Office.

This means that:

1. We can now build our own hunt groups
2. We must consider the impact of getting "more calls than we can handle".
3. We can change the hunt group size on the fly in most cases

Advantages: "Safest" as the TDM PSTN network limits security threat exposure, and minimizes network design complexity. Station may already have TDM connectivity present, reducing wait for new service. Traffic constraints "built in" and understood already.

Disadvantages: Legacy technology, will ultimately become unavailable, slightly higher initial and recurring costs.

Example Option 2: Provider provisioned as SIP endpoints
From the Provider we order:

1. A circuit to carry our phone traffic. This could be anything from a T1 to a 10 mbps fiber circuit. This is a dedicated circuit that has IP addresses that are not routable from the public internet. The carrier will usually provide a router or "Integrated Access Device" (IAD) at your studio site, The termination will be an RJ-45 ended ethernet jack. The length of time needed to get this will vary wildly, depending on your location and the provider that you're dealing with.
2. A small block of DID numbers for testing and later utility use.

From your distributor we order:

Telos gear (after a call with a Telos Applications Engineer)

Here's how it this all works in practice once installed:

A listener dials 311 555-2300 (your studios number). That number is recognized by Telco and is connected to a (SIP) trunk to your facility. The telco connects to your PBX and sends the DID number to the PBX on the far end of the Trunk. The PBX uses those digits to route the call further, to the correct extension in most cases. The PBX rings the desired extension and causes ringback tone to be sent to the calling party until the called party answers. When the call is answered, a two way audio path is cut through and the network is signaled that answer supervision should be returned, charging the calling party of the call.

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This means that:

1. We can now build our own hunt groups
2. We must consider the impact of getting "more calls than we can handle".
3. We can change the hunt group size on the fly in most cases.
4. We must consider network engineering and security ramifications in design of the system because it's delivered as IP.

Advantages: Fully protected from public Internet threats. Assured performance. Most up to date technology.

Disadvantages: cost may be higher than Internet delivery

Example Option 3: Service delivered over the Public Internet

From the Internet Provider we order a circuit, to be delivered on Ethernet, to be used to telephony only. Bandwidth, jitter and a Service Level agreement should be taken into account.

From the Internet Telephone Service Provider (ITSP) we need to order a trunk, with an adequate number of channels and a block of DID numbers.

We need to order an Asterisk machine appropriately sized for the task and the Telos gear.

Here's how it this all works in practice once installed:

A listener dials 311 555-2300 (your studios number). That number is recognized by Telco and is connected to a virtual (SIP) trunk to your facility across the public Intermet. The telco connects to your PBX and sends the DID number to the PBX on the far end of the Trunk. The PBX uses those digits to route the call further, to the correct extension in most cases. The PBX rings the desired extension and causes ringback tone to be sent to the calling party until the called party answers. When the call is answered, a two way audio path is cut through and the network is signaled that answer supervision should be returned, charging the calling party of the call.

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This means that:

1. We can now build our own hunt groups
2. We can change the hunt group size on the fly in most cases.
3. It is critical to fully consider network engineering and security ramifications in design of the system because it's delivered as IP and we are fully responsible for the security risks and performance of the system.

Advantages: Probable lowest monthly cost, instant provisioning (short lead time if adequate Internet circuit is present at the site already).

Disadvantages: Riskiest type of installation for security. Riskiest type of installation for performance.

Fully considering and planning for proper Network security is absolutely critical when attaching anything to the public Internet. This cannot be overstated. If not handled correctly, huge bills from service providers, interruptions in service and very embarrassing public hacking episodes could be in your future. Your employer counts on you to minimize their exposure to fraud and embarrassment. Please take this seriously. Telos can help. Discuss your implementation with an Applications Engineer before system deployment.

App notes for your specific situation

App notes address the particulars for your situation, like the PBX that you use, or the provider that you'd like to use... You'll find some these in the back of this manual, or on-line if they're really recent. We add new ones all the time. Think of them as a cookbook for your phones and VX installation. We try to present the information in a clear and concise manner, and always "real world". If you learn best by example as many engineers do, you'll likely find this helpful.

If you have a situation not fully covered by the app notes, have a look at them anyway. They'll generally make clear the typical configuration requirements.

Livewire is used for audio I/O

The Telos VX and VX Prime accepts and generates many audio channels. Using Livewire, it's possible to make them all available through a single simple ethermet interface. A single Ethernet RJ-45 connects dozens of bi-directional audio channels and GPIO command and status indications.

The VX uses Livewire for all audio I/O. For studios man that are already Livewire-based, this approach saves money and simplifies installation.

When needed, traditional audio connections are provided via VX Interfaces, with Axia Livewire connectivity. These come in both analog and AES3 versions. Because they are networked, they can be located wherever it's convenient, either in individual studios or in a central rack room. There are different models of nodes,each with a mix of inputs and outputs, of different types. Select the right mix and quantity of inputs and outputs that is most useful for your plant.

Depending on its configuration, A VX engine can generate enough livewire channel data to fill a 100 mbps ethernet "pipe". You should always connect Gigabit Ethernet switch ports to the LAN (Livewire) port on the VX. Remember that the Livewire approved switch needs to be set up according to the app note below before proper operation should be expected.

https://www.telosalliance.com/images/Axia%20Products/Approved%20Ethernet%20Switches/Configuring_Cisco2960X_Switch_for_Livewire.pdf

Step-by-step

There is a lot of information in the pages that follow. Here's a high level and simplified view of what you need to do to get your VX system up and running.

1. If you will not be using an existing Axia Livewire network to support the VX, install the Ethernet switch, or switches. Configure these for use with the Axia Livewire standard using the console cable. Enable and configure the WAN port on your VX engine appropriately with addresses and credentials provided my your telephone carrier.
2. Assign IP addresses to all of your Livewire devices and test connectiviy.
3. Install and configure your PBX if you will be using one to interface to POTS, ISDN, or T1 PSTN lines. If you will be using VoIP trunking for your Telco connection, get it connected and gather the IP and registration credentials.
4. Get a web browser going and do the system configuration.
5. Move on to install and configure the VXset phones and call screening software software application. Then to any other controllers you have in your system, such as integrated console controller modules.
6. Test your system throughly

The Engine and Network

The Engine front panel display gives at-a-glance status information. The main screen is pictured above. It shows the IP number for the LAN (Livewire) port, its connection speed and status. The overall status and CPU temperature are also indicated. The knob lets you scroll through the menu pages. Under normal conditions, you'll see what we call "The big OK".

If you don't see "The big OK", there is an indication on the front panel and web GUI (system page) that will indicate what's wrong. It's usually a power supply not plugged powered, an ethernet not plugged in or the switch not set up for Livewire.
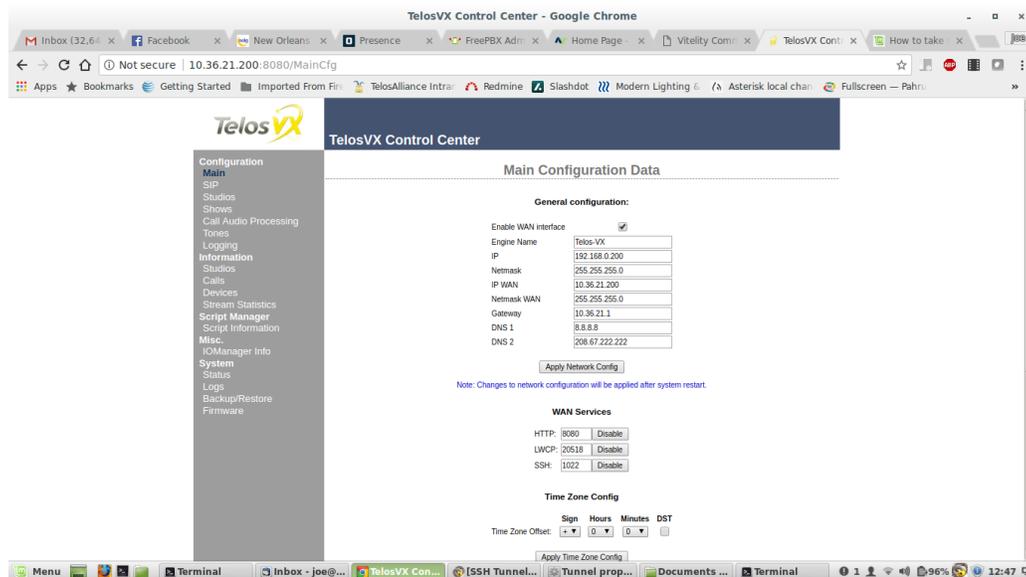
On the "main" menu page, are the options that let you enable the WAN port, change the IP addresses and another that shows the status of the Livewire audio channels

The main screen, shown at power-up, is pictured above. To see any of the other pages /menus, push, and then rotate the knob to show each in turn. Pushing the knob enters you into the menu that is displayed. Editing items inside menus, such as the IP address, is similar - navigate to the specific item you want to change by rotating the knob, then push it to choose the item, such as the IP address. In the case of the IP address, select from among the digits by rotating the knob, then pushing. Finally, rotate and push to enter the value you want. (yeah, this isn't the most convenient of user interfaces, but you shouldn't have to use it except for initial IP setup) The web interface, once available, is much more friendly. The reason for the IP address entry here on the front panel is to avoid the circularity that you need the IP address to connect a web browser, in order to enter the IP address. (should the default not work, for some reason.) We think you'll also find it convenient at times to have the IP address displayed on the front panel.

Network Connections

The VX Engine has two RJ-45 Ethernet jacks on the rear panel. Both are 100/1000 auto-speed and duplex setting. The (LAN) port is for Livewire audio and, by default, everything else, such as PCs used for web-based configuration, VSet phone controllers, producer PCs, and console controllers. This can be connected to an existing Livewire network or to a new, dedicated, network that you create to serve only the VX system, which could be as simple as a single Axia Approved Ethernet switch.

For audio to work, including Vset phone audio, there must be at least one Livewire device capable of supplying clocking on the Livewire network. Normally, this would be a Livewire audio Node, Axia Qor or Power Station.

More on the Network

An appeal of the Ethernet/IP approach to building studios is that you can make them as simple or elaborate as your needs require. The components of a VX system: the Engine, VSet phones, console controllers, and PCs are networked together using standard, off-the-shelf Ethernet switches. A small VX system might have only a single Ethernet switch as the '"network infrastructure", while a large full-facility setup could have dozens of switches and an IP router or two.

All Ethernet switches are not created equal! The Ethernet switch used by the Engine's LAN port must be Axia Approved and Livewire capable. That is, it needs to correctly support multicast. Switches that are not multicast-capable usually (most consumer unmanaged switches) flood with multicast traffic such as Livewire audio streams to all ports, potentially overwhelming devices like PCs and printers.

Never plug the VX LAN port into an office or VOIP phone network after Livewire outputs are enabled! ! Multicast-capable switches will not propagate the Livewire traffic to ports that are not subscribed to a particular audio channel. Thus blocking the high-volume traffic from places where it isn't wanted. A suitable switch would be the Cisco 2960G or another from that family. The switch that is part of the Axia Element PowerStation or Qor is also suitable. Of course; the VX and Axia consoles were designed to work smoothly together.

For the current list of Livewire approved switches please visit https://www.telosalliance.com/Axia/What-Ethernet-Switches-has-Axia-Approved

The WAN (Wide Area Network) port is isolated from the Livewire port and thus provides a kind of livewire firewall, ensuring that no livewire traffic can pass to the Wide Area network, often connected to a business network to access VoIP. Livewire traffic can easily overwhelm a network that isn't designed for it.

Out of the box, the defaults for the network connections are:

♦ The WAN port is disabled. To use it, you must enable it on the *Main* configuration page, as described below, and enable the services you want to use to it, also on the *Main* configuration page.

♦ All services – HTTP web access, SIP/VoIP, VSet audio and control, and Livewire audio I/O – are provided via the LAN port.

♦ All services – HTTP Web access, SIP/VOIP and Vset audio and the Livewire Control Protocol are available on the WAN port when enabled from the VX Web GUI.

♦ No Livewire inputs or outputs are assigned or enabled. You do this on the *Studios* configuration page.

VSet Installation

VSets are normally connected to the Livewire/LAN Engine port via the Ethernet switch linked to this port.

The VSet is powered over the Ethernet connection. This is 48 volts DC according the PoE (Power over Ethernet) standard. The Vset's power consumption is 15.4W. Power may be supplied by the standard PoE injector we include with each Vset, or it can come from an PoE equipped Ethernet switch. If a PoE switch is used, ensure that the Ethernet cabling is not so long that it causes too much voltage drop. Some Axia products, such as the PowerStation and IQ QOR, also have ports that provide PoE.

♦ Connect the Ethernet cable and confirm that that the LEDs near the jack are indicating correctly. The green LED (link light) should be on solidly, showing a good connection. The yellow LED (data) should be flashing, indicating data flow.

♦ After a few seconds, the LCD backlighting comes on, then the screens should come to life as the Vset "boots up". If they do not, the likely problem is the PoE supply or an ethernet cable issue.

♦ Press the Menu button and then navigate to the Setup Menu . (On the Vset 6, press the "more" button to display additional options including Setup.) Press and hold the button to the left of Setup Menu key for 3-5 seconds and then release it to access the setup menu for changes. This "press and hold" feature will make it less likely for unauthorized users to make changes that could disrupt normal operation. Just pressing it and releasing it (without holding it) will only show you the IP address and will not let you change it.

1. Enter the IP address that you are assigning to the phone and press the green "Go" button.
2. Do the same for the netmask value. (Most often, this may be kept at the default 255.255.255.0)
3. Do not enter anything in the "Vset Gateway" field unless directed to do so by Telos Support. The Vset will operate on the Livewire network and will not need to be routed anywhere unless you have an unusual or custom installation.

After selecting the engine, you might see 'Connected to..." for a moment at the bottom of the set screen.

If the Vset operation seems unusually sluggish or unreliable, check your network to be sure that your switch was set up for Livewire operation according to the app note for your switch.

System Configuration

Selectable and Fixed Lines

The VX supports what we call 'Selectable' and 'Fixed' lines. It is possible to have a mix of each type in any studio.

If you're in North America, you'll most likely want to use Selectable lines. Fixed lines are more commonly used in Europe. With a fixed line, A specific telephone number is always associated with a particular button on the VSet phone and a fader on the console. It is as if each line had its own hybrid. This allows VIP numbers and hotlines to have fixed and dedicated console faders.

Multiple calls that are assigned to a single fader have individual hybrids/processing and are actively conferenced within the VX Engine using an internal mix-minus matrix. Calls assigned to different faders would normally be conferenced via a mix-minus matrix of some kind within the studio mixing console.

Selectable lines use an operation style like traditional Telos systems, where there was a line selector before the hybrid or hybrids. Most operators are used to the operating paradigm that the old "1A2" key phones made so prevalent, where you have a column of buttons and pressing one takes a line and drops the one that was active before. For that reason, we have kept this operation style as the default for the VX. We have also kept Telo's "lock" function that lets an operator conference multiple lines and keep VIP callers on-air while coming and going with others. This also allows selective holding and dropping of conferenced lines.

Do keep in mind that all faders (fixed or selectable) count against the maximum total number of 24. 'Selectable' is generally a more efficient use of VX system resources.

Studios and Shows

The VX system can support multiple studios and can share lines among them.

A "**studio"** is a collection of controllers and audio or GPIO IO that are used together, usually to serve one studio: Vsets, Call screening applications on PCs, Audio input and output Nodes (if any), GPIO interfaces (if any), and console controllers (if any).

A **show** is a pre-configured profile that can be selected by users. It assigns the lines to buttons. This is the mechanism that lets you easily move groups of lines from studio to studio.

Studios and shows are configured via the web interface to the VX Engine. Interfaces such as the VSet and the VX Producer let operators select the studio and show they want to use.

Configuration Web Pages

To access the configuration Web pages, find or connect a PC to the network that the VX Engine is connected to. Start a web browser and open a connection to the VX Engine by entering http://192.168.2.200 into the browser's address field (assuming the default IP number has not been changed and that your PC is on the same subnet).

Main Page

This is the page you will see upon initial connection. Select the pages you want by clicking on the menu at the left side of the page. If this is a new installation, start with the *Main* page, then move to the *SIP* page, the *Studios* page, and then on to the Shows page. The purpose of each of the pages and how to use them is detailed below.

Main page - General Configuration

Engine Name. Assigns the text name for this VX Engine. It becomes the Linux network name for the Engine.

If you will not use the defaults for **IP**, **Netmask**, and **Gateway**, enter them here. IP addresses can be entered for both the LAN and WAN ports.

WARNING: Do not assign the "gateway" address to the Livewire network, as Livewire is not routed. What this means is if your LAN address is 192.168.2.200, make sure that the gateway does not start with 192.168.2.XXX

Web Username and **Password.** Used for web access for configuration and monitoring. User-name is always 'user' and the default password is <blank>.

LWCP Password. LWCP stands for *Livewire Control Protocol*, the way controllers communicate with the VX Engine. A password can be used to secure the Engine from being controlled by unauthorized devices. The Vset, for example, uses this password. By default, this is set to <blank>. We recommend not setting a password.

WAN Services chooses which services are available on the WAN port. HTTP is for web access, LWCP is for controllers, and SSH is shell access to the Linux OS in the Engine. SIP/VoIP are always enabled on the WAN port when the port is enabled.

NTP (Network Time Protocol) is the Internet standard for keeping devices synchronized to the correct time and date. The VX uses this to have accurate timestamps for logged events. Enter the IP number for the NTP server here An NTP server must be accessible on either the LAN or WAN port. The VX automatically finds the correct port. If the NTP server is outside of both the LAN and WAN local network segments, the **gateway** address must be set, and the NTP server must be reachable from the WAN. You can use a domain name if a DNS server is configured and accessible. Under *NTP Config* are the settings for your location.

A green check mark will appear after you enter and accept your NTP server setting, and the server connection is successful.

Default Username, Password, and IP settings

We put this info here under its own header, just to make it easy for you to find when surfing the table of contents…

VX Web GUI username/Password

Username: "**user**"

Password: <blank>, that is, no password

Network Interface default settings

-'Network' (Livewire) interface IP: 192.168.2.200

- Netmask: 255.255.255.0

- State for the WAN connection is OFF by default, with no IP set.

- Default "Gateway" address should be gateway for WAN address only

Network TCP and UDP ports used by the system

- Active ports on WAN interface by default are:

- 5004 - DSP Engine

- 40000-41000 handset audio for Vsets and call screening software

- 62000 through 62513 - RTP

- 5060 – SIP

- 10000-20000 - SIP RTP

- 8080 – WAN port HTTP Server (disabled by default, change with GUI)

- 1022 – WAN port SSH Server (disabled by default, change with GUI)

- 20518 – Livewire Control Protocol (LWCP disabled by default, change with GUI)

A computer on the same subnet as the 'Network' (Livewire) interface, (example 192.168.2.100) could log into the VX with this URL: http://192.168.2.200

while a computer on the same subnet as the WAN interface (example 192.168.5.2) could log on using http://192.168.5.200:8080 .

Default user name is: user

Default password is: <blank>

Passwords are case sensitive

If you change any of these, you might want to write the new values into this manual, here or elsewhere.

SIP Configuration

The SIP configuration page shows the global SIP settings, along with a list of all SIP servers that your VX is configured to use. You can configure one or many SIP servers. The form at the bottom of the page lets you to add another server; VX will automatically create server configuration for any line in the Show configuration. As a consequence it is not possible to delete server entry if it is referenced from at least one show. The *Lines / Refs* column shows the number of lines which are explicitly configured, and the total number of line positions in shows using this server. Clicking on the link with the server name will bring you to the server configuration page.

 The *Default Server* changes the server address which will be used for new lines in Show configuration. Note that this will affect only new lines, existing ones will keep their old settings (use the *SIP Server* option to change the server address for existing lines).

♦ Enabling *SRV Lookups* makes the VX to strictly follow the SIP standard when resolving domain names. This feature is not widely used and can be left unchecked unless required by the SIP provider.

Regarding Codecs and SIP

Some details regarding the VX's SIP implementation:

♦ As of the time of writing, the VX supports the g.711 (alaw or ulaw companding) and g.722 codecs. g.711 is the usual PSTN codec, so ISDN or T1 connections will not suffer any transcoding loss. SIP automatically negotiates and selects a codec that is supported by both ends. Implements RFC 3261 standard (but see below)

The g.722 codec is the same codec known to broadcasters from the pre-MPEG days of ISDN remotes. It has 7kHz audio bandwidth, opening the door to much better than usual speech quality, often from mobile handsets using softphone apps and connecting via internet. When there is reliable and sufficient IP bandwidth, audio quality from mobile phones can be much improved compared to the usual fuzztone. The Acrobits or Bria apps work well with G.722.

On Windows and Linux, the free Ekiga VoIP softphone works fine.

♦ Be aware that there are several other "G.722 codecs" that are proprietary for use with certain brands of SIP business telephone sets or mobile devices. These are not currently supported by the VX. The most common G.722 codec is supported. We may add other codecs to the VX over time with sufficient interest from customers.

For both of the above, you need to force the use of the g.722 codec by enabling it and disabling the others (in the PBX or providers dashboard). On the VX side there is no setting for this, everything is handled automatically during normal SIP codec negotiation. Following the usual procedure, the codec choice is made by the caller and conveyed in the SIP Invite message and procedure.

SIP-connected g.722 audio is not via the PSTN, so the VX Engine would need to connect to an Asterisk system or G.722 capable service provider, properly firewalled, with a protected public facing IP address or VPN tunnel that is accessible by the calling side. There are safe ways to do this. Please see the app note "VX with G.722" for full details. Also note that G.722 does not pass via the PSTN or gateways (which means that ISDN g.722 codecs will not work with the VX), while many IP codecs will.

We do not ever recommend exposing your VX directly to the public internet! Please read the App Note "VX with G.722" in this manual for the current "best practices".

RFC 1890, the original standard governing audio codecs in SIP, erroneously listed the clock rate of G.722 as 8kHz (the actual sampling is 16kHz). When the error was discovered, it was too late to fix it, and it was decided to keep it that way. Nevertheless, makers of some SIP devices decided "fix" this mistake on their own, breaking compatibility with most other devices that stick to the standard, including VX. As the result, VX will use G.711 to communicate with such devices.

Note that VX's call audio processing is bypassed when using g.722.

♦ RFC 2671 MD5 digest authentication is supported.

♦ RFC 2833 DTMFs is supported.

♦ Supports SRV address lookups.

♦ Has basic NAT support

♦ Unlike some other SIP devices, the VX supports non-numeric "extensions". This is sometimes used by hosted PBX services.

SIP Server Configuration

The *SIP Server* field allows you to change the server address, while keeping the other settings intact. You can also assign a more descriptive *Name* to the server, which will be then shown in place of the server address. As it is used for display purposes only, it can be anything you want, like 'Asterisk'. The *External IP* setting is discussed in the next section. Please note that to change the server settings you have to unload any shows referencing the server; it is not possible to change them while the server is in active use.

The *Extension* and server together make up the SIP address of a line. Each must be unique, as in the context of SIP the "line" *is* the address (and associated configuration).

We recommend configuring SIP extensions as Endpoints that use registration. This makes network troubleshooting easier and tests the entire IP connection to the SIP server. Be sure to tick the "*Register"* checkbox. This will make the VX register with the server (meaning essentially, log in) whenever a show referencing this line is active, The "*Expires"* field lets you to change the interval (in seconds) in which the VX will refresh its registration. When left empty, the server and VX will negotiate for a reasonable default, usually one hour. As a general rule, don't populate the expires field unless Telos support recommends it to solve a specific problem.

The other two fields allow you to set the password and change the "Auth" user name used for authentication. *Auth User*, is generally the same as the extension on most PBX's.

For quick access to the configuration, you can use the *Config* link in the Show configuration page. Each position with correct configuration has a link marked '»' next to it. Clicking on it will bring you to the corresponding server configuration page, with the matching line configuration highlighted. In case the line is not configured, the extension field for adding it will be filled in from the show.

SIP and Network Address Translation (NAT)

As SIP messages list IP addresses and ports used to transmit audio (RTP), it doesn't work well if the client is in a private LAN but needs to communicate with a SIP provider outside of the LAN. As messages pass through the router, it translates addresses in IP headers, but not the SIP message itself, giving the provider wrong connection info.

Many SIP providers use clever hacks to work around this limitation without any additional support from the client. If you are connecting VX to a SIP provider that doesn't have such service, don't worry - all is not lost, as VX has basic NAT support built-in.

For full details of using VX behind a NAT router, please see the App note on that topic.

Studio Configuration Page

The *Studios* page lists all of the studios that you have configured for your system. It lets you add new ones. And it lists the Shows that studios are using.

Individual Studio Configuration Pages
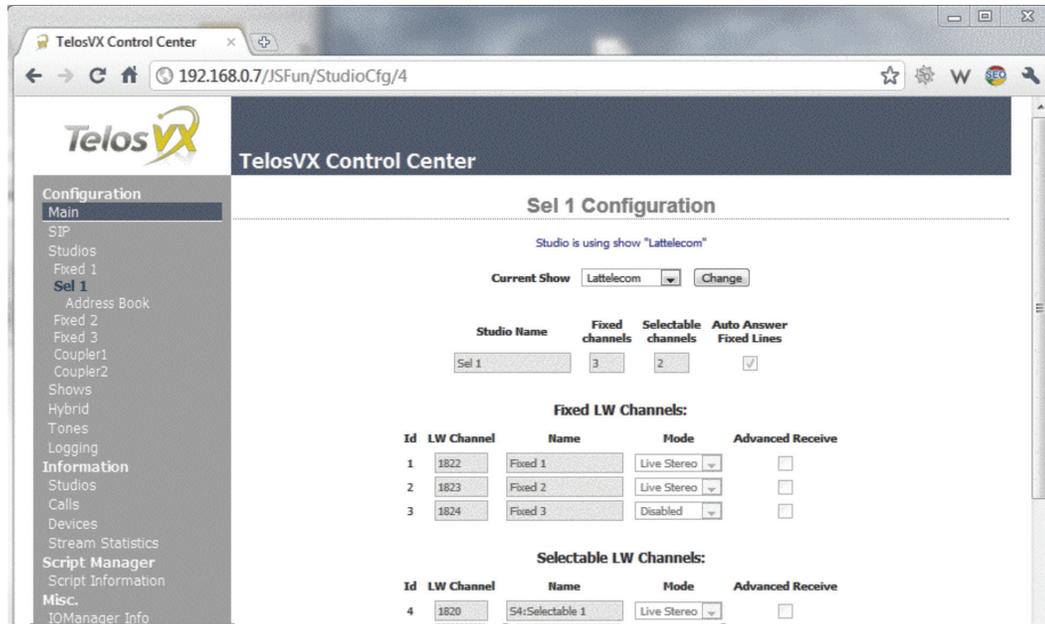
The individual *Studio* pages are to configure each studio. For each, you can choose how many console faders you want to devote to selectable and fixed lines and you assign the Livewire channel numbers to each. Note that when you assign Livewire channels to outputs, Livewire streams will start to flow on the LAN network port. Livewire streams system and network consume resources as soon as they are defined with a type and channel number.

It is suggested that you assign Livewire channel numbers according to App-note "Livewire Channel numbering. Livewire channels must be unique and should not start with "1, 2, 3...". Channel numbering is completely fliexible, but a good system of assigning these numbers is covered fully in the app-note.

Remember that in an Axia environment, Livewire channels carry audio in two directions, so both send and receive are covered at once. 'Live Stereo' should usually be chosen as the Mode to keep the delay as low as possible. Standard stereo or mono can be used for Program on Hold or IFB feeds where a few extra ms of delay is inconsequential.

One LW channel is used to feed **Program on Hold** to each studio. Since only one of the two Livewire audio directions is needed, you can choose which is appropriate for your studio set-up. *From Source* is the usual.

You must configure a Program on Hold feed for each defined studio. It could be a silent livewire channel, or a generic program or air feed but if no working channel is assigned calls on hold may drop after 30-60 seconds, typically. It's also a "best practice" to feed Program on Hold feeds "Pre-Delay" to minimize confusion for those on hold.
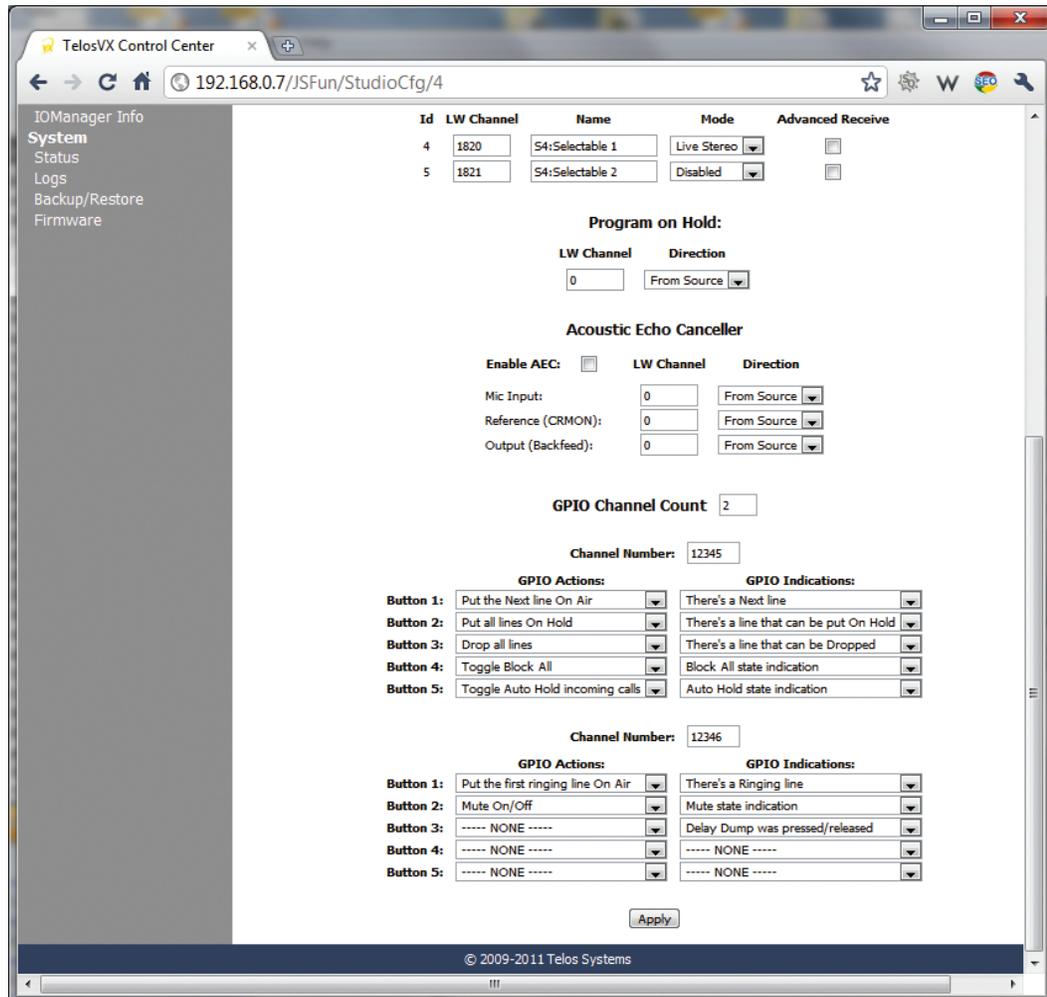
Advanced receive opens a new field where you can enter a LW channel number to which this Fader/LW channel will listen. In an Axia environment, it will ordinarily listen to the same LW number as it is sending to. (With native Livewire consoles such as the Axia Element, this is what you need). But if you want to connect the VX to another console via an interface Node, "From source" should be selected. For this case, choose Advanced receive and enter the LW channel number of one of Node's sources. This is audio that goes to the caller, it should be a Mix-Minus or "clean feed" (in other words, the phone does not feed the phone).

Each studio has an acoustic echo canceller available. The two inputs and the output are assigned to Livewire channels on this page. The output of the AEC is labeled Backfeed because the output of the AEC is what you usually would feed back to phones. For configuration purposes, you can think of the AEC as a separate functional block outside of other VX functions. There is no internal connection from/to the AEC and other VX signal paths.

The Acoustic Echo Canceler helps with the problems that occur when you need to have loud-speaker monitoring of calls in the same room as the microphone feeding the phone. Without a canceler, the received caller audio would be returned to the caller as an annoying echo. (In the old days, this acoustic coupling would more probably cause feedback howl than echo, but today's mobile phone and VoIP delays have made echo the trouble *du jour*. The canceler needs two inputs and produces one output. The Mic input is fed from the studio microphone. The reference input is the audio that needs to be can celled - the received phone audio that is going to the monitor or preview loudspeaker. The output of the canceler goes to the VX phone feed input(s). (Why didn't we just internally connect the canceler? Because the canceller works best when the reference input is after anything that is in the phone-to-speaker path, such as the volume control and mute. In a Livewire-equipped studio, it should not be too difficult to tap the needed signals. The canceler is low-distortion and full-fidelity, so it may be used with wideband codecs.

Complete information on the Acoustic Echo Canceler is in AEC App-Note towards the back of this manual.

GPIO

VX supports GPIO (General Purpose Input/Output) control. It's a useful way to control functions in the VX or to get helpful indications of VX conditions or modes. The Livewire channel assign-ment for this is done on this page.

Electrical connections are made via Livewire GPIO Nodes. They are equipped with D connec-tors using the Axia standard 5 inputs and 5 outputs per channel. You tell the system how many Livewire channels you will use with **GPIO Channel Count**. After you press apply, you will see entry fields for each of the channels. Specify the Livewire channel number you want to use for each and choose the functions you want to assign.

GPIO Actions are inputs to the VX from some external source, which could be nothing more than a pushbutton. **GPIO Indications** are outputs from the VX. For example, *there's a Ringing Line* could be used for a wall-mounted lamp and *Delay dump was pressed* to trigger a profan-ity delay unit. You can find the details regarding the corresponding pins of the GPIO connectors in the various GPIO Node manuals or in the "using-GPIO" app-note towards the back of this book.

Show Configuration

Individual Show Configuration Pages

With the individual *Show x* configuration pages, you tell the system how to assign Telco lines to VSets and other controllers. You give each line a text name that appears on controllers. You decide if the line is to be Selectable (a number of lines switched into a few faders) or Fixed (one-to-one correspondence from lines to faders). You specify if a line is to be affected by the **Block All** function.

The SIP **Extension** is like an extension number in a PBX system, with the difference that it is generally not required to be a number, though most often it is. It maps incoming calls to the line positions on the controllers. Together with **Server** it makes the SIP URI (address) identifying the line. The server is also used to form SIP addresses from numbers for outgoing calls. The checkbox in *Server* column allows you to override the default server set in SIP configuration.

You can assign the same extension to multiple buttons. Calls will ring in on the first available button, acting like a "hunt group".

Near the top of the page, you see what studios are using this show.

Using the VX to replace 'Couplers'

If your station uses "Pre-delay IFB (Interruptible FoldBack) Couplers or "listen lines", the VX can replace an entire wall of hardware, and by using DID (Direct Inward Dial) numbers instead of POTS lines, you can also save a lot of money each month by replacing those expensive analog lines.

Create dummy studios for each coupler bank and shows with the phone number or extension you want to use and tick the auto-answer box next to it. Choose the audio feed from among the Livewire sources.

Full details are in the "Using VX for couplers" App-note toward the back of this manual.

Call Audio Processing page - Audio Processing and metering

The VX has dynamics processing on both the send (from studio, to caller) and receive (caller) audio directions. Adjustments and meters are on the *hybrid* page.

We recommend using +8 on the high EQ, +4 on the low EQ, 16 on the AGC (full range), the caller ducking to 6dB, and the noise gate off.

The send (to the caller) audio processing is fixed and consists of a protection limiter and some EQ. The purpose of the limiter is to protect the Telco line from the distortion that would result from clipping due to an audio overload. Livewire studio audio has a much larger dynamic range than telephone lines. We use a limiter rather than an AGC because we don't want the system to increase low-level signals, which could cause feedback and sound unnatural to the caller.

Send EQ is a simple EQ that rolls-off the low-end and has a peak at high-frequencies to improve intelligibility. This is designed to make the audio similar to what a caller would receive from a normal telephone. When wideband audio is fed directly into a telephone connection, there are often complaints that "it is hard to understand" because the low-frequency energy is masking the higher frequencies that are needed for intelligibility.

The receive processing includes ducking, an AGC, and dynamic EQ.

The term "hybrid" is used only for convenience here, since VOIP is inherently four-wire (separate transmit and receive). The only echo possible is each from the far end which is either acoustic or reflected at a four-wire to two-wire (POTS) hybrid, found in Telco line equipment connected to bad subscriber loops.

Ducking. The purpose of this is to provide an effect that many DJs and talk show hosts prefer – that the caller level is reduced when they speak. The ducker reduces the gain in either the send or receive path depending on who is talking at a given moment. There is a smoothing filter with time-constants tuned to make the effect as natural as possible. The amount of ducking is a user-adjusted variable value and may be defeated entirely if you choose. 12 db of ducking is the most popular setting.

Caller AGC/EQ. Normalizes the level coming from telephone calls. A wideband AGC is used for consistency, followed a multi-band frequency-selective processor that provides spectral control for the best caller audio, balancing intelligibility and warmth. The Duck Control module communicates with the AGC module so that gain is not increased when the studio-side is sending audio. Leakage from the hybrid in this case could look like low-level telephone audio that needs to be boosted. If this happens, the effect of the hybrid and ducker would be undone.

The **Call Audio Processing** button allows you to enable or disable all of the audio processing features of VX, such as AGC, EQ, ducking, etc., all at once. Enabling this option will let you adjust the individual options below.

Assigning Sounds & Tones – The Tones page

SIP signaling is via message, not by tones in the audio. This means the VX has to make its own sounds in response to various line status conditions for human consumption. We provide default sounds, which are the traditional tones generated by the PSTN, but you can upload your own to create a unique on-air 'signature'.

You can also use any sound you like to signal that a new call has been taken. This is actually a cool new feature made possible by modern technology. Rather than a new call appearing silently, the event can be accompanied by a signature sound. A story… Years ago, we installed a fancy new Telos system at Z-100, New York, replacing an old key phone setup. The new system had a mute function to remove the loud and annoying 'clunk' that banged out when each call was taken to air. On the first day, star morning DJ Scott Shannon called Frank Foti, the Chief Engineer, to complain he was missing his "*kerchunk kerchunk*". He liked the clunk, saying it created a kind of dramatic "yer-outta-here" punctuation when he dismissed a call and took another. He would have loved this feature of the VX. We've included a key phone button punch sound in his memory.

For line status (sometimes 'called 'call progress' tones, The *Tones* configuration page lets you choose either USA or Euro standard for the default tones You can upload your own sounds. Files to be uploaded must be mono, in the signed .au file format, linear PCM (8, 16, 24 or 32bits) or 32-bit normalized floating point. Ringtones MUST be 8kHz, 16-bit, mono, other tones 48kHz. You can use the free Audacity audio editor to convert whatever you have to this format. Use the Export command and choose the required values.

The files you wish to upload must be available on the PC that hosts the Web browser you are using to access the VX. Your tones can be any length, but should normally be kept short to conserve memory (and to avoid annoying listeners…).

Dial tone. This is only heard when a line button is pressed before a call is dialed.

Ringback tone. Heard when dialing is completed and the called phone is ringing.

Busy Tone. Heard when the called phone is busy.

Reorder tone. (fast busy) Signals that there are no call paths available in the PSTN, usually caused by congestion in links between central offices. In the USA, this is is the 'fast busy' - a dual-frequency tone of 480Hz and 620 Hz at a cadence of 0.25s on/0.25s off.

Error Tone. Usually caused by an incorrectly entered number, but can be from other problems in call setup.

Call answered (Sound effect). Played whenever a call is put on air (whether directly, from handset or from hold).

Off Hold (Sound effect). The sound that is played on-air when a call is taken to air from Hold by pressing a line-select button.

Line switch (Sound effect). The sound that is heard when there is a call on-air, and a new call is taken to air by pressing a line-select button. This is the one Scott Shannon wanted.

Caller hang up (Sound effect). A brief sound that is heard when an on-air caller disconnects. We recommend that you leave this enabled because call disconnect is completely silent when using SIP and the host might not realize that they've been disconnected. While the sound isn't loud or obnoxious, hosts pick up on it while listeners don't seem to notice it.

Caller Alert tone. Sent to caller when call is answered, or when caller is taken to air from hold, to inform the caller that they are now on-air. The Caller Alert sound is sent to the caller, not played on the air. Its purpose is to let a waiting caller know that it is his turn to talk. Since he has usually been listening on hold, he will have heard the host say "hello, you're on the air" a few times before it really is his turn. Since this sound is played only to callers being switched to air, it clears the confusion. By default, it is a short 'blast' of white noise, similar to what all

Telos hybrids have sent down the line for years. Ideally, The Producer/call screener should explain this to callers.

DTMF (on air). Choose the DTMF menu to set these tones. Dialling keypresses send call setup messages digitally to VoIP lines or gateways. Though they are not sent to the Telco line, standard DTMF (Dual Tone Multifrequency) tones are played on-air - but they are scrambled so that listeners are not able to easily detect the number being dialled. The correct DTMF tones are played to the VSet handset and loudspeaker.

Ringtones. Choosing the Ringtones menu lets you choose different ringtones for each incoming line position on the VSet. Ringtones with higher numbers take precedence. For example, If one line has ringtone #1 assigned to it, and another line has ringtone #5, and both lines are ringing, the Vset will play ringtone #5. Ringtones play through the Vset speaker if not muted via GPIO or Livewire settings.

SIP and DTMF

The VX uses standard SIP techniques for dialing. A SIP call setup message containing the number is sent to the IP network. The gateway to the PSTN generates the DTMF (Dual Tone Multi-Frequency) audio signal that the PSTN uses to direct the call. The VX generates DTMF audio just as sound effects really. Listeners expect phones to sound like phones. DTMF tones "played" for the audience are not the actual digits dialed and some are not even "real" DTMF digits, but sound like them. This is so that if a number is dialed "live, on the air" nobody can get the telephone number being dialed by recording the program audio or "decoding" the tones heard on the air.

There is sometimes the need to send DTMF to the PSTN after a call is connected, such as for automated attendant systems or "dial up" remote control systems. In this case the VX sends a special SIP message that tells the PSTN gateway to generate the corresponding audio DTMF tones, according to the standard specified in the IETF (Internet Engineering task force) RFC2833.

Logging
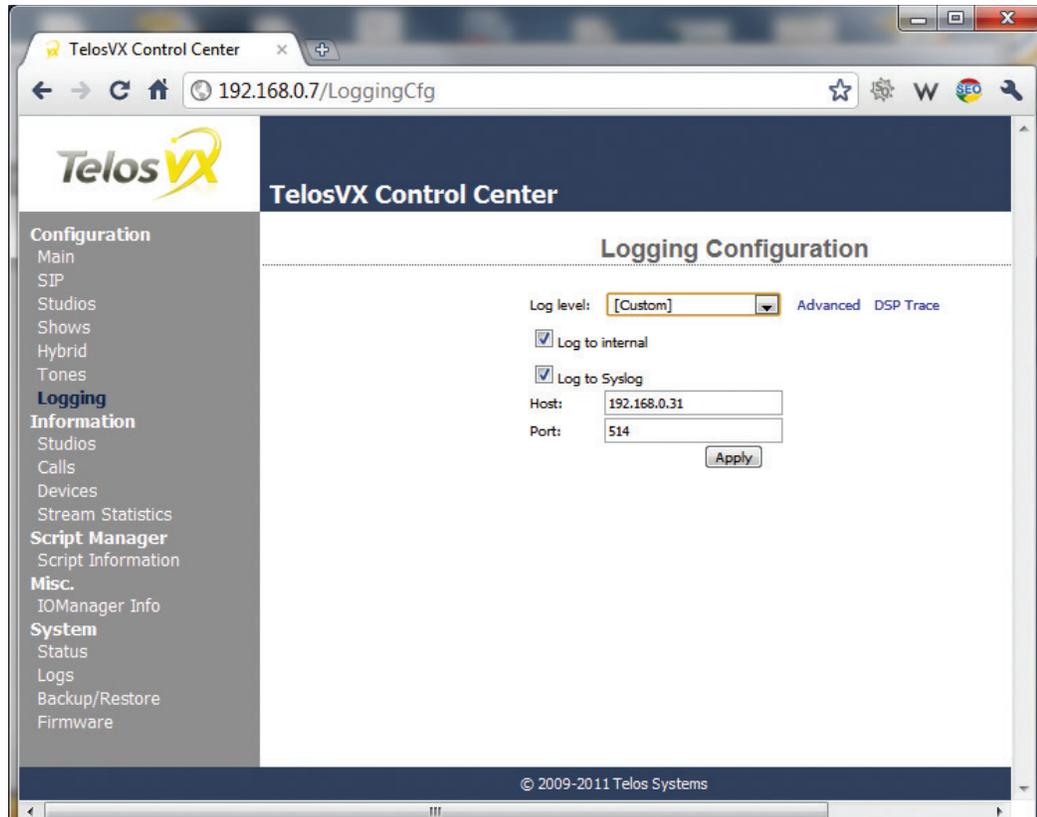
The VX's logging capability is a powerful tool to track down problems –even those due to operator error – we do recognize that operator error is rare and nearly non-existent.

In the *Logging* config page you determine where logs are stored and how much detail is recorded.

You can choose to store logs inside the VX or to an external PC on the network that is set up to handle standard syslog messages received via UDP on port 514.

If you choose the internal option, log messages will be kept in the VX's RAM, so are limited in size. The advantage is convenience: You can view internally-stored logs with the *System* page log section, by default at http://192.168.0.7/log, A typical log is shown in the screen capture below. A disadvantage of keeping the logs in RAM is that if the VX were to lose power or crash, the logs that might show you the cause of your problem would be lost. We recommend using syslog and a syslog server.

With **Log to Syslog**, log messages are sent over your network to the Syslog server. Enter the IP address for your syslog server. Syslog servers need not be Linux machines, though Linux's Rsyslogd works really well. You can install a syslog server on a windows PC. One simple Windows app is at http://www.kiwisyslog.com/ or syslog remote log reception can be enabled on your Asterisk PBX if you have one. The system runs rsyslogd as a service for it's own use anyway, it's simple to change the config to accept logs from VX or other systems. See the App Note "VX with Syslog".

For basic troubleshooting, choose **Log to Internal**, then go to the *Log* page, clear all existing messages, then place a call or make the operation that's causing trouble. Then return to the Log page to see the entries

While you might be able to understand the log messages after some effort, we don't expect you to do so. These are mostly intended for our support people, who would advise you as what settings to use for the capture and then interpret the result. These are simple text files and if you look through them you may find something interesting or useful for troubleshooting.

A typical setting for diagnosing SIP problems would be to set the log level to **Debug (extended)**. Fewer log messages will be generated if you set level to **Debug (basic)** Easier to read, but maybe missing detail that you need.

If you want to adjust logging in detail, go into the *Advanced* page and select which modules to log, then select the DEBUG level for the module you want to trace. For example, if you want to trace what VX Producer or VSet is sending to the Engine, you might chose to select LWCPBE module log level to DEBUG.

Once the problem is solved, you can reduce the chattiness or "verbosity" to a lower level to save space, as the debug level generates a lot of lines of text!

The three most interesting modules to log are:

 ◆ AIF - VX front end communications with VX DSP part

 ◆ CPSIP + SIP - SIP messages

 ◆ LWCP - LWCP protocol for VX control

Internet standard Network Time Protocol (NTP) is used to provide an accurate timestamp to the events captured in logs. For accurate timestamps, you need to have an NTP server accessible on your network and its IP address properly configured on the *System* page.

Logs can get huge, especially if you're logging LWCP or SIP at a debug (most verbose) level! We suggest that you do use a syslog server and closely monitor your disk usage. Also end the logging session or change to a less verbose debug level (like error or Info) as soon as you can. Linus has a program called logrotate that can be st up to automatically do all of this if your logging needs are long term.

Firmware updates

New firmware for the VX Engine is first obtained from Telos via email, Internet download, etc. It is then uploaded to the VX Engine using this page. You can have as many as five stored in the Engine at one time. On this page, you choose which of them will be active. Reboot the Engine after activating new firmware.

Backing up and restoring Engine configuration
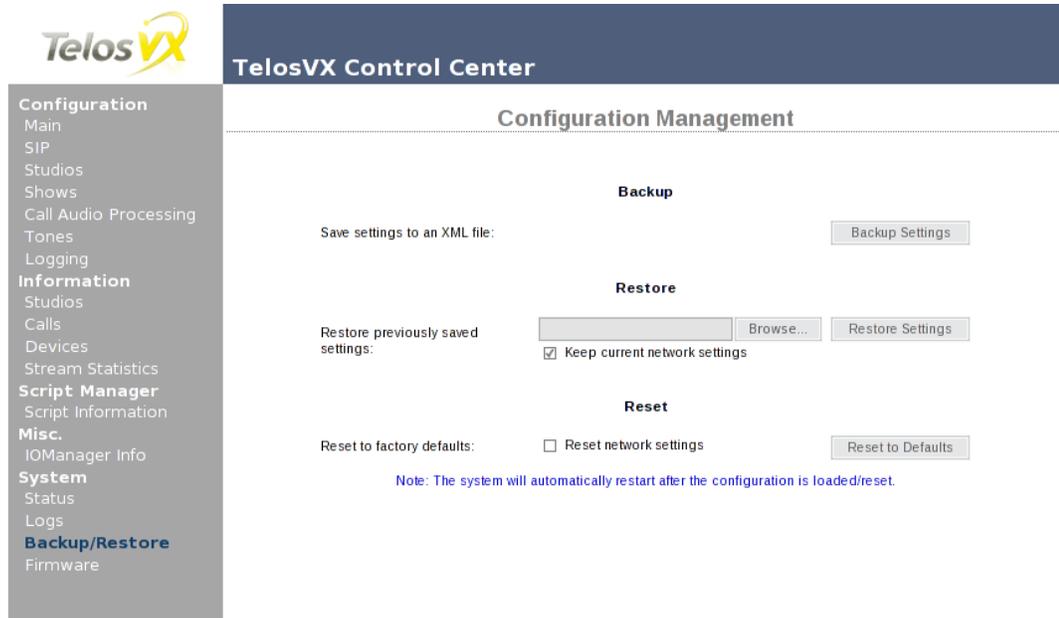
Backup/restore configuration settings

The backup file for configuration settings is stored on your local machine or on a removable drive attached to it.

When restoring, either from backup or to factory defaults, it is possible to keep the current network settings intact. Note that VX will automatically restart to apply the new configuration.

Many people have multiple VX engines, and keep a single one as a spare, with the configs of all of their live systems in a folder on their computer. This makes it easy to drop a config on a system and make it the "live" unit while you get the other repaired.

Setup instructions for Vsets, Vset Call Controllers, Console Controllers and Nodes

Instruction manuals or 'quick start guides' come with these devices or are available on the Web for download.

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3 Operation

VSet Operation

Select Studio and Show

If these have not been already selected, this should be done as the first step before using the VSet. Press the Menu button to access the menu functions. The LCD will show the various items that can be changed. Select the studio and show you want and then exit the menu by pressing the Menu button.

Set Talent or Producer Mode

The VSet should be set to Talent mode when it is located in the studio and is used to put calls on the air. Producer mode is for a VSet that is being used by a producer to screen calls. In Producer mode, calls cannot be put on-air and are protected from being dropped. The function of the right line button column is different in the two modes to serve the needs of each type of operator, as described below.

Press the menu button to access the menu functions. The LCD will show the various items that can be changed. Select Talent or Producer and then exit the menu functions by pressing the menu button.

The Line Info Display

Each line has a corresponding portion of the LCD display to the right of the line button that shows status and info about that line.

Next indicator

The '>>' symbol. Marks the line that will be taken when the **next** button is pressed.

Fader number

The yellow/orange rectangle near the status icon.

Line name

The first text line. This is defined in configuration using the VX Engine's control center web GUI.

Caller ID

Caller ID will be displayed on incoming calls if it is available. Outgoing calls will show the number dialed. The green arrow to the left of the text points left for incoming or right for outgoing. Calls that are blocked or that come from lines without caller ID may display 'anonymous'.

Time

The length of time the call has been ringing-in or on hold.

Action 'hint'

(None active in the graphic.) Shows what line will be affected by drop and hold. Choose the line you want to drop or hold after pressing either of those buttons. This is shown only when more than one is possible; this step is not needed when only one line is possible, when the line would be dropped or held immediately.

Line Status Icons

Dialing

Shows when you are using the keypad to dial an outgoing call.

Outgoing call in-progress

The icon will be animated with the white highlight moving around when a call is in the process of being connected.

Ringing-in

On-air

On-air and locked

On handset

Hold

Ready Hold (Screened Hold)

Used Elsewhere (in another studio)

Line is blocked

Line Button Columns

Each line has 2 associated buttons to the left of the LCD.

Pressing a left column line button puts a held or ringing caller on the handset.

Press the right column button to put a held or ringing line on air, dropping any other unlocked calls. If a call is already on the air, pressing the right button locks the line, or if already locked, unlocks it. That is, it toggles between the two states. Not to worry, we've made it difficult to accidentally drop a call. Call screeners or producers should use the Ready Hold button (the one with the checkmark) to put lines in ready hold mode and assign priority for the Next function. See "Next" Button and function.

Ready hold works like normal hold, except the line status icon has the checkmark to indicate to operators that a call has been determined to be ready for air, normally by a producer/screener. The Vset must be in Producer mode in order to use this key.

> **Tip: You can re-order the next priority queue at any time. Put lines in Ready Hold - even those already on Ready Hold - in reverse order from the order you wish to air them. First, Screen hold the call to be aired last, then the call to be aired second to last and so on. You would screen hold the call that you want to be aired next, last. At first blush, this may seem confusing, but it makes sense when you remember that the call held the longest goes first to air.**

Hold Button

Holds a call that is ringing, on the handset, or on-air. If more than one line is in this category, there will be a small hold icon near the status icon on the lines that could be held. Press the line that you want to hold.

Drop button

Drops a call that is active on the handset, or on-air. If more than one line is in this category, there will be a small drop icon near the status icon on the lines that could be held.

Press the line that you want to drop.

Lock

Normally, taking a call to air causes any others on-air to be dropped. If you need to conference two or more calls, you can use the lock function.

Press an already on-air line button to lock it. The line status icon will change to display the locked symbol. Pressing the button again unlocks the line and the lock icon goes away.

Locked calls remain on-air until unlocked and then another call is taken to air or the call is dropped or held.

The Drop and Hold buttons have no effect on a locked line.

> **Tip: If configured in your system, using fixed lines is another option for handling "guest" or "VIP" calls that need to say on-air while you switch between other callers. With fixed lines you'd have a dedicated console fader for each of these calls, so you can control volume, and switch them independently in and out of the conversation, etc.**

Next Button and Function

The Talent next priority is:

1. longest waiting ready hold
2. longest waiting hold
3. longest ringing-in

The producer can manually override these and assign priority as desired.

The producer mode next priority is:

1. longest ringing-in

In producer mode, **Next** does not take any held lines.

Block All

Pressing this button will cause all inactive and ringing lines to be dropped and blocked from accepting any calls. Calls on-air, on the handset, on hold and the fixed lines will not be affected. The usual application for this function is to let you prevent early callers from getting in on contests until after you've made the announcement and released the lines to accept calls.

When block all mode is selected, as lines become available (they stop ringing or drop off) they will automatically "go busy".

Pressing Block All again will release the lines and allow incoming calls.

Numeric Keypad and GO Button

For dialing out, either on the handset or on-air. The keypad works during an active call to generate tones for voice mail and other services.

While a number is being composed, the digits will appear in the line info field. After a number is entered, the **GO** button initiates the connection, in similar fashion to a mobile phone.

Re-dial Function

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

Fader Assign

Normally, when a call is taken, it goes to fader 1, but you can assign it to any of six console faders. (The number of possible faders depends upon your specific installation and configuration.) When a call is on-air, or before it is taken, the round fader assign buttons to the right of the LCD can be used to move the call to the desired fader. The number in the yellow rectangle in the line info field shows you which fader is or will be used.

Fader assign is configurable to be enabled or disabled to keep the LCD uncluttered for people who will not use the feature.

In our older on-air phone systems, we were limited by the cost of the day's technology to one or two hybrids, so we only used one or two console faders. With the VX system, this limitation is removed. That means that you can assign a fader to each call if you want to do so, allowing independent control of each call's volume.

Firmware Upgrades

You can upgrade firmware for your VSet by using a web browser and downloading the firmware from Telos. Support may give you a link in email, or you can find the latest versions at the Telos website. After downloading the firmware image, point your web browser to the IP address of the VSet that you wish to upgrade (the default password is <blank>, and the username is "user"). Then, browse to the firmware image location on your PC (desktop or downloads folder is common) and select Upload. To activate the new firmware, select it in the browser and the VSet will reboot. When it restarts, it will be running the firmware version that you selected.

Take it Easy

We've built a lot of flexibility into the VX system to support all the sophisticated things you might want to do. But you don't have to use all the fancy-pants stuff when you don't need it and a simpler subset will do.

In the simplest case, when you are a talent supported by a producer, you could just use the **Next** and **Drop** buttons. You press the Next button to take calls in the sequence determined by your producer and you press the **Drop** button when you are finished with your last call and you want to stop taking more.

One step up would have you pressing the right column line buttons to select the calls you want, one at a time. Again, when you are finished taking calls, you would press **Drop**. You could use the left column line buttons to talk with callers on the handset.

At the next level, you would use either the lock function or the fixed line capability to conference calls.

Some Common Questions

Question: "I can't place a call. After I dial, nothing happens! What's up?"

Answer: Don't forget to press the green "Go" button after dialing.

Question: "Why don't I hear any audio after pressing a line button in the handset column (left column)?"

Answer: Double-check that the handset cord is plugged into the handset jack (not the headset jack) on the VSet.

Question: "Why can't I put a call on the air? When I press the right column line key, nothing happens!"

Answer: Put the VSet into "Talent" mode. When the VSet is in "Producer" mode, calls cannot be placed on the air from that VSet (so as to prevent accidental call drops, or placing the wrong caller on the air).

Your opinions and ideas are always very welcome at Telos. Please share them with us at support@telosalliance.com .

Operation with the Element Console

The Axia Element console can be used to control both fixed and selectable lines from the VX. Entering the VX Engine IP address connects all the needed logic functions - line selection is the main one, but muting for the VX sets is also handled. Any VX set assigned to a studio and show shared with an Element will mute its ringer when the Element tells it to. Nice, eh? Selectable lines use the Element's Call Control Module as the line selector. It can be used in either 12 or 24 line mode, depending upon configuration. In the former, two columns/faders are used; in the latter, one column/fader. The same module is used for either 12 or 24 line operation. When 12 line mode, the left column of Status Symbols displays the line number and the right displays status. When configured in 24 line mode, each active line is numbered with the number being shown when the line is idle (replacing the idle "dot"). In the 12 line mode, the left row of buttons assigns lines to the left fader and the right row to the right fader. In the 24 line mode, both rows assign lines to the left fader. To assign to the right fader, push **on**, **off** or preview on the right fader which activates flashing and displays **PRESS LINE** on the alpha display. Selecting a line assigns it to the right fader. To cancel the mode before selecting a line, press the right **drop** button or the right channel **off** button.

When you need to be able to assign selectable lines to more than one or two faders, the Call Control Module can be expanded using one or more Phone Fader modules. The additional Phone fader channels will work like the right fader on the Call Control Module. You can quickly build multiple-line conferences on the fly, while keeping the usual one-call-after-another auto-drop style operation on the left fader.

In all cases, the keypad for dialing is on the Master Module. Pressing an **options** key on a fader channel with a phone line assigned activates the keypad in the master section and opens up some other phone-related options, such as the auto dial list. The keypad is also automatically activated to the fader channel that last seized a phone or codec line. However, when options is active on a phone or codec module, the keypad remains locked to that channel until options is deselected.

Usually, a VSet phone will be used in the studio along with the console to allow handset conversations. Taking a line on the VSet will disconnect it from the Element, and vice-versa. The call will not be dropped.

Fixed lines can be configured directly to Element faders. Phone Fader modules are preferred because the **set**, **hold** and **drop** buttons are useful. Note that using only fixed lines and Phone Fader modules would give you "Euro-style" operation, where each line has its own hybrid/fader and there is no line selector. There is no limit to fixed lines vs. selectable in the VX, so you can have as many as you have available console faders.

Mix-minus is generated automatically in all Axia consoles. In Element mix-minus AUTO mode, the feed-to-source will be generated by the PGM-1 bus "minus" the caller audio when the channel is ON. When the channel is OFF, Talkback will be the source. Alternatively, Feed-to-Source can be fixed to P1, P2, P3, P4, SA, SB, SC or SD.

You can assign a line to a specific fader by pushing the Set, OFF, ON, Hold, or PVW button on the fader channel before selecting the line. The fader channel alpha display will toggle between source type and LINE N after a line is assigned.

4 The VX Producer Application

Introduction

The VX Producer application integrates:

♦ Call screening and production communication

♦ Recording, playing, and editing calls

♦ A 'softphone' for talking with callers off-air

Normally, it would be used by Producer/call screeners to communicate with callers and the on-air talent. Each would have a copy running on a PC near to them.

Installation

The standard Windows installer is used to install the VX Producer application and the .Net framework, if needed. It also can install the Audacity audio editor.

Remember that since the VX uses Livewire audio to carry the "handset audio" to your VX producer PC, that the PC needs to be on a LAN subnet that carries Livewire traffic (the same subnet as the VX engine Network LAN port).

You'll need at least Windows XP Service Pack 2.
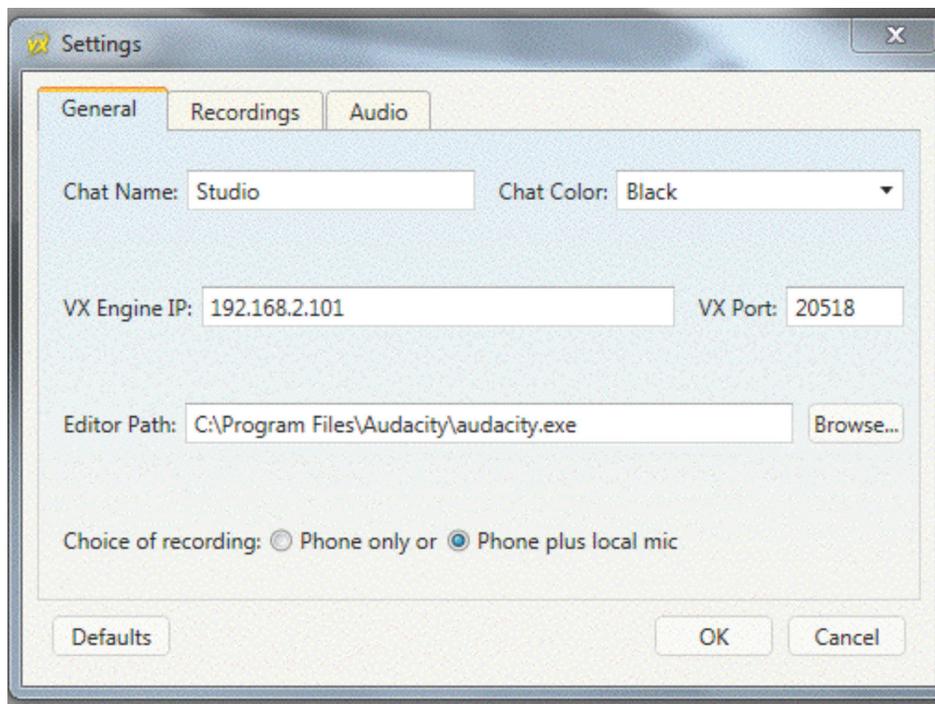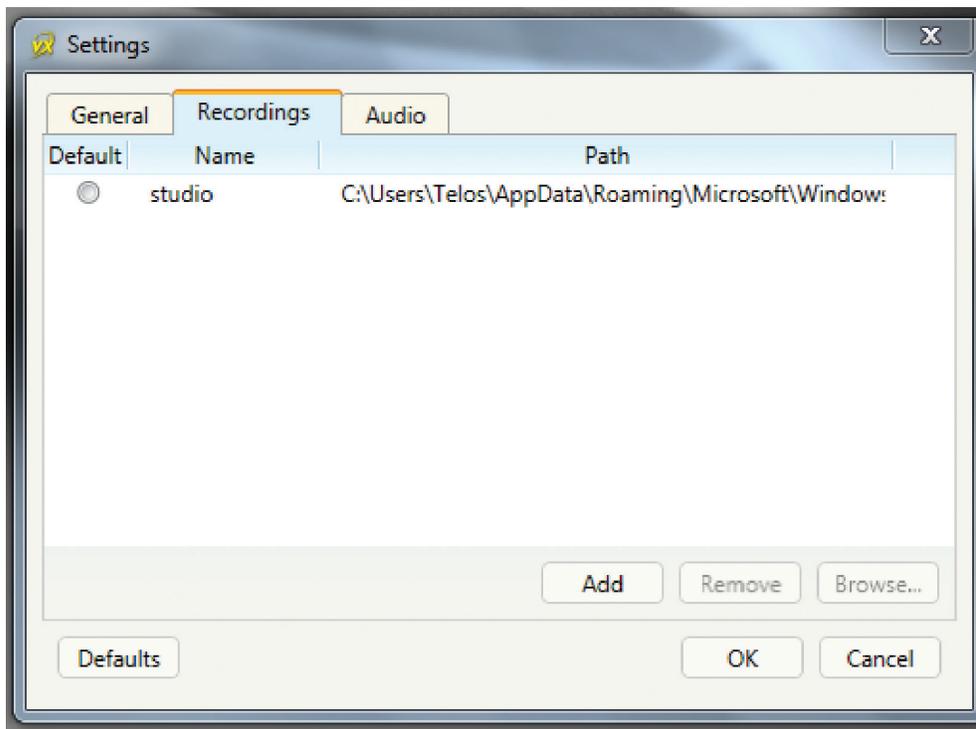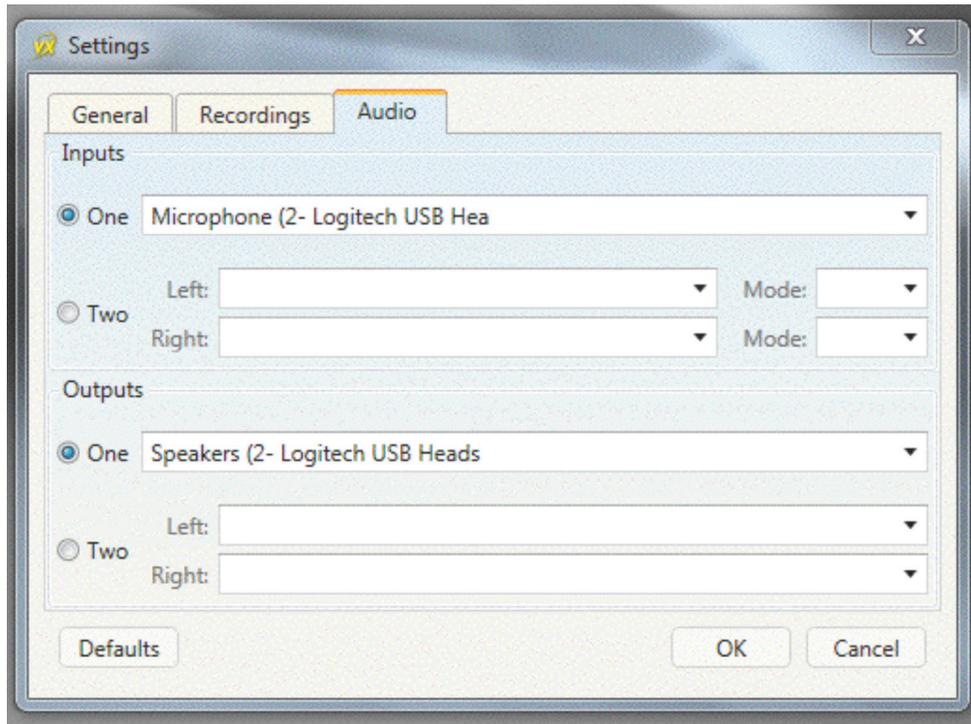
Set-up: Studio and Show
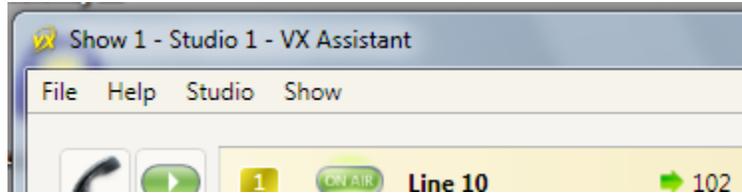
Managing Calls

The main window shows a list of the lines that are available for the selected show. It is divided into *selectable* and *fixed* line groups. (if both are configured for the selected show).

Near the bottom of the screen is a tabbed list box, controls, and an audio level meter that are used for the recording and play functions.

At the lower right is a window used for text chatting between the producer and talent.

Handset

Press the handset button to the left of any line to take the call to the 'handset' that is connected to the PC running the *VX Producer* application. (Often this will actually be a headset/mic.)

If no call is ringing-in, the system assumes you want to dial out and opens a window to let you do that.
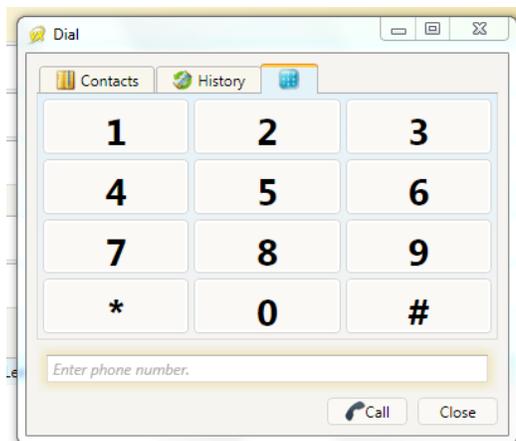
Contacts

History

Dialing

On-air

Pressing this button takes a call to the studio mixing console. Your operation of the console determines whether the call is actually on-air or sent to preview or to somewhere else. Usually the studio mic (or microphones) are used to talk with a caller, whether on-air or not.

If no call is ringing-in, the system assumes you want to dial out and opens a window to let you do that.

The Line Info Field

Fader number. Before a call is taken to air, this shows which fader it will be assigned to when it put on-air. After a call is on-air, this shows which fader is assigned to the call. The number changes when you move a call using the fader assign buttons.

Next indicator. Shows which call will be taken when you press the **Next** button.

Line status icon.

Line name. This is the name that was given to the line during show configuration.

Caller info. Text you enter into this area will appear on any other *VX Producer* applications and VSets that are assigned to the same studio. It automatically replaces the line name text.

Telephone number. Shows incoming (caller ID) or outgoing (dialed) number. The small arrow near the number points left for incoming and right for outgoing numbers. It automatically replaces the line name text.

Time. Shows time that a call has been waiting on hold, or if on-air, how long it has been on-air.

Drop, Hold, and Hold Ready Buttons

Drop disconnects a call that is ringing, active on the handset, or on-air. If more than one line is in this category, they will all be highlighted. Press the line that you want to drop. Or select the line you want to drop to highlight it and then press **Drop**.

Hold puts a call that is ringing, active on the handset, or on-air into hold. If more than one line is in this category, they will all be highlighted. Press the line that you want to hold. Or select the line you want to hold to highlight it and then press **Hold**.

Hold Ready puts a call that is ringing, on-air, on the handset, or on hold to the hold ready (screened) state.

Selectable Lines

Lock/Unlock Button and Function

Normally, taking a call to air causes any others on-air to be dropped. If you need to conference two or more calls, you can use the lock function.

Press an already on-air line button to lock it. The line status icon will change to display the locked symbol. Pressing the button again unlocks the line and the lock icon goes away.

Locked calls remain on-air until unlocked and then another call is taken to air or the call is explicitly dropped or held.

The **Drop** and **Hold** buttons have no effect on a locked line.

> **Tip: If configured in your system, fixed lines will probably be the better option for VIP calls that need to say on-air while you come and go with others. You will have a dedicated console fader for each of these calls, so you can control volume; switch them independently in and out of the conversation, etc.**

Fader Assign Buttons

In older on-air phone systems, we were limited to one or two hybrids (audio paths), so we only used one or two console faders. With the VX system, this limitation is removed. That means that you can assign a fader to each call if you want to do so, allowing independent control of each call's volume.

Normally, when a call is taken, it goes to "fader 1", but you can assign it to any of six console faders. (The number of possible faders depends upon your specific installation and configuration.) When a call is on-air, or before it is taken, the round fader assign buttons to the right of the LCD can be used to move the call to the desired fader. The number in the yellow rectangle in the line info field shows you which fader is or will be used.

Fader assign is configurable to be enabled or disabled to keep the LCD uncluttered for people who will not use the feature.

Next Button and Function
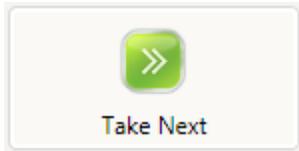
The Talent next priority is:

1. longest waiting ready hold
2. longest waiting hold
3. longest ringing-in

The producer priority is:

1. longest ringing-in

The producer can manually override these and assign priority as desired.

Block All

Pressing this button will cause all inactive and ringing lines to be dropped and blocked from accepting any calls. Calls on-air, on the handset, on hold and the fixed lines will not be affected. The usual application for this function is to let you prevent early callers from getting in on contests until after you've made the announcement.

Pressing Block All again will release the lines and allow incoming calls. The specific lines blocked are determined by install-time configuration. Normally, VIP and hotlines are excluded.

Fixed Lines

These are usually used for VIP and hotlines. They have a different operating style from selectable lines and are independent from them. Here, each line has its own - fixed - fader. These will have been assigned when the system was installed. Taking a line will not drop others that are active. Drop and Hold need to be explicitly applied.

Lock has no effect on fixed lines, since they are effectively always locked. Block All is usually configured to have no effect on fixed lines. Fader assign has no effect, since faders are permanently assigned.

Note that a system could be configured to have only fixed lines. This would result in the operating style that many large European broadcasters favor. Each line would have its own 'hybrid' and associated fader.

Fixed lines also offer a selectable "auto answer" feature. This can be used for automatic feeds, news recording or "filing lines", or "listen line coupler replacement.

Recording, Editing, and Playing Calls

Pressing the **Record** button starts a recording of the currently active phone call.

In talent mode, this will be the "on-air" call. Usually, the audio will have been routed via the studio mixing console.

In producer mode, this will be the call on the handset.

You can choose if both the local mic audio and the phone are recorded or only the phone. In the former case, a stereo file will be produced.

Enabling **Auto Record** causes the recorder to automatically make a new recording each time a call is taken.

File names are automatically created by the system, but can be changed by writing over the default text after a right mouse click. The file will appear in the tabbed list box determined by Settings. Normally, a VX Producer in producer mode will put recorded files under the Producer tab and in studio mode, under the Studio tab (naturally). You can move files among the tabbed lists by either dragging them to the destination tab or using the **Send** button and drop-down list.

The meter to left of the tabbed file list window shows the audio level as a recording is being made.

Playing

Select a file by clicking on it. Press the **Play** button. The audio will be sent to the PC's output configured in Settings. Depending on the Settings configuration, the playback will be available on PC-connected loudspeakers or headphones, or it may be routed to a studio mixing console fader.

Editing

Select a file by clicking on it. Press the **Edit** button. The audio editor configured in Settings will open and the selected file will appear in the editing window (Audacity is the default). Audio input and output will be according to the editor's configuration. Depending on the configuration, the payback will be available on PC-connected loudspeakers or headphones, or it may be routed to a studio mixing console fader.

A stereo file will be passed to the editor when both a mic and phone audio are recorded. A mono file will be passed when phone-only record mode has been selected.

Files

The recorded audio files are physically stored on the local PC or on another PC or server accessible over the network. The location depends upon the file path configuration in the *VX Producer* settings menu.
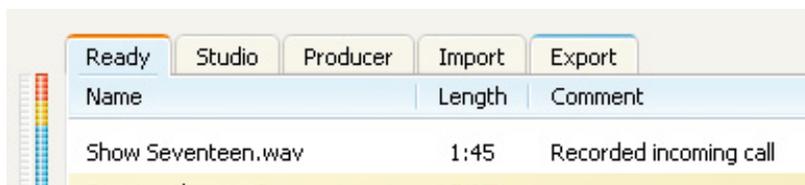
By default, files created in the studio will go the list under the Studio tab and files created by the producer will be saved under the Producer tab. The Ready tab holds files that are ready for air. The Import and Export tabbed lists are used to accept and send files to other people on the network. Again, the tab names and file locations depend upon your settings.

The files are in standard .wav format that may be accessed by a variety of audio applications. Standard Windows file sharing allows access to the files across the network at the various machines.

Host/Producer Text Chatting

The window at the lower right lets people within a Studio communicate with each other. Type text into the lower field and it will appear on any other *VX Producer* applications selected to your studio.
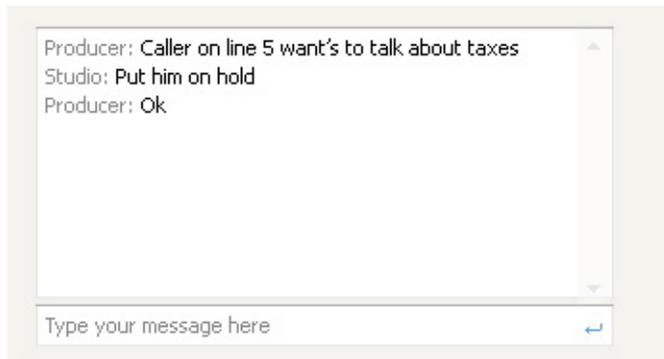
The 'Studio' and 'Producer' identifiers are added automatically. The colors for them can be configured in settings.

Tech note: The VX Engine works as the "chat server".

5 Notes, Resources, Additional Information

The Acoustic Echo Canceller

You can learn how to connect the Acoustic Echo Canceller to your studio equipment in the *Studio Configuration* section. Here, we provide more detail on how it works and how to use it.

A common annoyance in radio studio and TV studio operations is the feedback that often results from using a loudspeaker to listen to telephone calls. This comes from the acoustic coupling of the audio from the loudspeaker to the studio microphone that feeds to the caller. As well, echo might be heard by the caller. When a caller talks, the phone audio bounces around the studio and gets sent back. Due to the time-dispersion caused by the room and the round-trip transmission delay, this acoustic echo is distracting to the caller.

In the past, when analog Telco lines were the norm and mobile phones were not yet attached to everyone's ears , feedback was the usual problem, not echo per se. In earlier Telos hybrids, we addressed that with a combination of ducking, frequency shifting, and a basic canceller. That worked pretty well in its day, but Mobile phones and VoIP connections impose much more delay – and the problem has become more challenging. With the multiple-hundreds of milliseconds latency that can occur these days, an effective canceller must be part of the system when loudspeaker monitoring is expected.
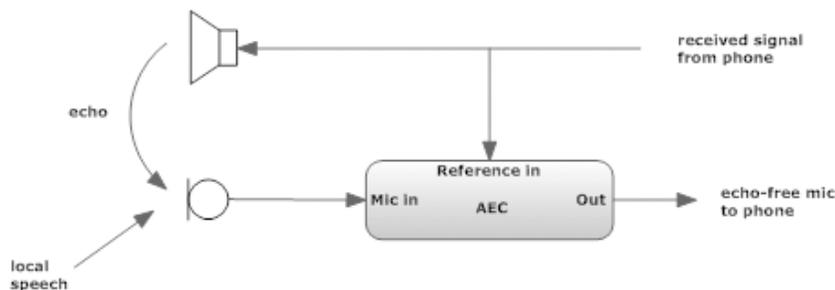
Fortunately, technology has come to the rescue. Modern Acoustic Echo Canceling is the answer. The audio at the studio microphone consists of the host's voice combined with the unwanted telephone audio that is delivered to the room via the loudspeaker. An AEC removes the loudspeaker audio, leaving only the host. It does so by synthesizing the transfer function of the acoustic path. The reference (loudspeaker) signal is passed through this function and subtracted from the microphone signal, thus canceling the echoed telephone part of what the microphone picks up. AECs have been used in high-end audio and video conferencing systems for many years. High-end broadcast hybrids and on-air systems such as the Telos Delta, 2x12, and Nx12 have included a limited form of AEC. But only recently has AEC technology advanced to the stage where it is truly effective. Thankfully, it comes just when the added delay of mobile and VoIP connections make it near essential. The good news has been a result of both break-throughs in the design of AEC algorithms and the ever-increasing power and lower-cost of the processor chips that are needed to implement them.

These latest-generation cancellers are a miracle. You can have ear-splitting volume with very little of the caller audio making it back to the other end. They work with up to 20kHz audio bandwidth, so are ready for the wideband VoIP codecs now coming online. And they solve a longstanding problem: Older 'time-domain' AECs depended upon the acoustic path remaining invariant and could quickly degenerate into feedback when a microphone was slightly moved or the acoustic path changed from some other cause. The 'frequency-domain' technology used by the new AECs works just fine with moving microphones and other echo path changes.

This new AEC technology is particularly useful for TV studio applications where it can be impractical to have talk show guests using earplugs. Today's high-performance AECs let talent and guests listen to phone calls on foldback loudspeakers.

AECs are provided within the VX Engine, but they are not connected into its internal signal paths. This is because they need to be placed in the signal paths between the studio loud-speakers and microphones. With Axia mixing consoles, the insert and return points are provided with Livewire channels and are installed by configuring the appropriate channel numbers. The AEC can be used with Non-LW consoles via LW Nodes connected to the appro-priate audio signal paths.

(??) AECs are possible from a single VX Engine. Additional Engines could be used to provide more. Please contact us to discuss your options, should you want to do this.

There are three connections for each AEC:

   ♦ Microphone input (from the studio mic)

   ♦ Microphone output (feeds the console mic input that goes to the phone system)

   ♦ Reference input (parallels the signal that feeds phone audio to the studio loudspeaker)

If multiple mics are used, there must be an AEC dedicated to each of them.

The AEC cancels audio that is put on the reference input so that it does not appear on the output. Without canceling, the output would be a mix of the studio mic and the delayed/echoed phone audio. After cancellation, only the mic audio will be present. That is; it creates a replica of the acoustic path, then passes the reference audio through it and subtracts the resulting filtered signal from the microphone feed.
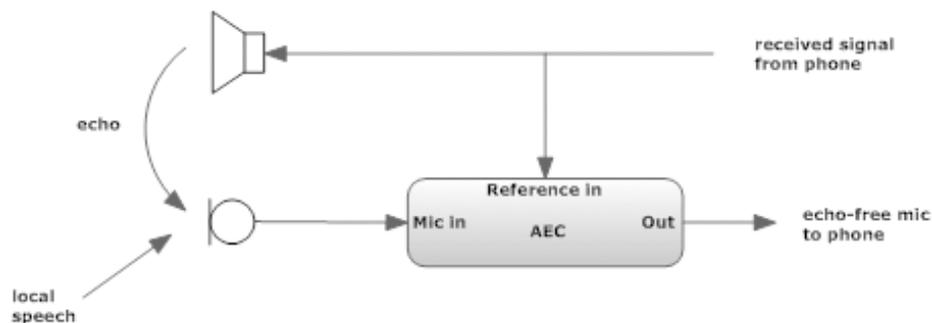
The AECs are full-range and low-distortion, so they can be put in an air signal path if need be. Nevertheless, we recommend that they be inserted only into the mic-to-phone path if possible, avoiding causing any trouble to mic-to-air fidelity. (There could be subtle frequency response alteration depending upon the echo path and other factors. Noise might also be added.)

In a studio equipped with an Axia console, the needed signal connections are available via Livewire. Configure both the console and the VX Engine with LW channel assignments so that the correct connections are made. In a non-Axia studio, route the AEC connections to a LW Node by assigning the appropriate LW channels at both ends, and then connect the audio to/from the mixing console via analog or AES3 to the Node.

To check if an AEC is working, call into the system and switch on the mic and loudspeaker. Adjust the mic level to normal and the loudspeaker to reasonable volume. Talk into the phone; listen for echo and assure that none is audible.

Telco Services and Interfaces

VoIP: SIP Trunking

While it remains a niche in early 2011, SIP trunking is growing rapidly in support from both PBX vendors and carriers. Over time, this will almost certainly appreciably reduce the use of the older POTS and T1 trunking. Eventually, it may replace it completely.

Whether the gateway to the PSTN is at your physical location or at another site should make no difference – as long as the IP path between you and the gateway has guaranteed QoS with sufficient bandwidth to support the maximum number of active connections you expect to have. In the case that the IP link is to be used for both telephony and data, the system must be designed so that phone calls have priority. In order to ensure this, there must be only one IP vendor between you and the PSTN, and this vendor must guarantee QoS in a properly written Service Level Agreement. Any time that IP service crosses from one vendor to another, all bets are off as to both the probability of achieving consistent good quality and having any chance of getting problems resolved.

The other thing to look out for is what codec will be used. For calls that ultimately are carried by the PSTN, only the native g.711 codec is acceptable for broadcast applications. Anything else would involve transcoding and an unacceptable reduction in fidelity, especially audible when mobile phone calls are involved. These already have poor quality due to their low-rate 14.4kbps codec. Passing this through g.711 within the PSTN and then yet another codec on the way to your studio over an IP link is asking for aural trouble.

The VX natively supports the g.711 A-Law, g.711 μ-Law, and g.722 codecs. Almost all PBXs and gateways support these formats.

Finally, you need to be sure that your equipment and the carrier's gear can properly communicate. While SIP is a standard, vendors often enhance it with extensions that are not universally supported.

One development that could help is a project called SIPconnect, undertaken by SIP Forum, a consortium of SIP vendors. The SIPconnect Interface Specification was launched by Cbeyond Communications in 2004 with support from Avaya, BroadSoft, Centrepoint Technologies, Cisco, and Mitel. It attempts to detail the interconnection specifications between IP PBXs and VoIP service provider networks. It specifies a reference architecture, required protocols and features, and implementation rules. It calls for the g.711 codec to be provided on all equipment and services.

IP Centrex and Hosted PBX Services

Just as it shouldn't matter whether the PSTN gateway is on your premises or not, it also shouldn't matter where your IP PBX is located. This is the principle that allows IP Centrex services or hosted PBX services to happen. These locate the hardware at the service provider's site and remove the need for phone system equipment at your location. In the pure case, you would have only IP phones at your site, which would be plugged into an Ethernet switch, which would connect to the Internet via a router. The main advantage is that someone else is responsible for installation and maintenance of the back-end equipment. It might also be that a vendor of these services has invented a suite of applications that would be difficult to replicate at individual business sites.

Number portability in the US and Canada.

Number portability regulations allow you to move your existing numbers to the service provider of your choice. Unfortunately, our experience is that "choke" exchange numbers cannot be ported. We've tried with various carriers - all with the same disappointing result: a call on the day of the port explaining that it can't be done.

Circuit-Switched Interfaces

When you order or configure a gateway, you need to know what kinds of interfaces you will be using to connect to the Telco network.

FXS/FXO

These are designations for the two ends of a standard analog POTS line. Most often these lines are used for basic home telephones. But they can also be used to link a PBX with a Telco central office.

♦ An FXS (Foreign Exchange Station) interface emulates a circuit supplied by a Telco Central Office. An FXS supplies talk battery and detects an off-hook condition. It generates 100Vac for a ringing indication. It provides dialtone and other call progress signals such as ringback and busy. It responds to DTMF (Dual Tone MultiFrequency) tones for dialing and may send caller ID information in modem-encoded audio.

♦ A telephone, and anything that looks like a telephone, is an FXO (Foreign Exchange Office) device. An FXO device signals an off-hook condition by drawing loop current. It responds to ringing voltage. It provides dialing, either by old-fashioned pulsed loop-interruption or by DTMF. It may detect caller ID.

An interesting limitation of FXS/FXO interfaces is that signaling from the FXS that a call has ended is sometimes not signaled, and the type of signaling varies around the world. In the USA, most Telco central offices interrupt the loop current when the call has ended, but some do not. And many PBXs do not. Eventually, dialtone will return, though, and this can be used as a disconnect signal, but there will be a many-second delay. This could cause *glare*, the condition where there is confusion between the CO and the PBX as to whether the line is free. A call could ring-in just when the PBX attempts to access the line for an outgoing call. In many other countries, an audio tone is sent on the line to indicate the end of the call. This lack of *disconnect supervision* results from the idea that there was no reason for a central office to hang up a phone by remote. This was a physical, human action performed by someone who knew that the conversation had ended and put reacted by replacing the receiver back in its cradle. The design was never intended to work for machine-to-machine connections.

Actually, there are two types of off-hook signaling. On a *loop-start* line, when a phone goes off-hook, the circuit is closed, and the central office detects the change in current. This is the common residential format. *Ground-start* signaling is a small modification to the scheme to permit disconnect supervision and remove the possibility of glare, where a PBX mistakenly takes a ringing-in line for an outgoing call. . In an idle circuit, the central office provides -48v on the Ring wire and an open on the Tip wire. From the PBX side, Ring is grounded first, then the central office circuit must respond by grounding Tip. The PBX senses this, releases its ground, and maintains the connection by drawing loop current.

E&M Trunks

E&M trunks use two extra wires for signaling (the so-called *Ear* and *Mouth* connections). These solve the problems with glare and disconnect supervision. This scheme is nearly obsolete, but occasionally E&M interface cards are used to connect music-on-hold to VoIP PBXs. They are convenient for this purpose because the audio path is transformer isolated and there is no need to supply talk battery from the MOH source.

T1/E1

These are basic digital interfaces to the switched voice network, and are widely used. This is especially so in the USA, where T1 is nearly standard for large PBXs. In Europe, ISDN PRI is more widely employed. T1 transports up to 24 voice channels, while E1 supports as many as 32. T1s are common in the USA and Japan, while E1s are provided by Telcos in most of the rest of the world. In addition to the audio, these digital circuits also carry basic signaling in CAS (Channel Associated Signaling) bits. This signaling emulates loop-start, ground-start, or E&M, depending upon configuration. Over a T1, 23 speech channels are offered, while an E1 provides 30. They actually have 24 and 32 channels, respectively, but one or two of the channels are reserved for signaling communications.

T1s can also be used for IP connections. In this case, all or part of a T1's 1.544kbps capacity is used as a transparent pipe from the local IP router to the ISPs equipment. The phrase *channelized T1* is sometimes used to distinguish a T1 that is intended for circuit-switched voice application. A *fractional T1* is a service that uses a portion of the line's full capacity. It is sometimes possible to order a T1 that is divided into a channelized portion and a data-transparent part for IP connectivity.

ISDN PRI

ISDN PRI (Integrated Services Digital Network Primary Rate Interface) uses the same underlying circuits as T1s and E1s. However, it employs sophisticated out-of-band signaling, which allows transfer of information such as calling number, codec type, clearing causes, and such. (It's rather strange that non-ISDN T1 sends caller ID modem-encoded in the audio channel, no?) The speech paths are called B (*bearer*) channels, while the signaling is carried in D (*data*) channels. Almost all large VoIP gateways and PBXs support ISDN PRI lines. The signaling in the USA is a slightly different protocol than that used in Europe and other parts of the world. Your gateway will need to be set to match the protocol on your line. Normally in the USA, this would be NI-1 (National ISDN - 1), while Europe would use the Euro ISDN standard.

You might hear the term *QSIG* in the context of ISDN PRI gateways. This is a signaling protocol that is yet more sophisticated than ISDN's usual Q.931 protocol, and is layered on top of it. With the ascendancy of SIP, QSIG looks to be yet another valiant attempt falling by the wayside.

ISDN BRI

ISDN BRI (Basic Rate Interface) lines offer two B channels, supported by one D channel. These were intended as a residential replacement for POTS lines or for small businesses. One application envisaged by its inventors was to allow a simultaneous voice call and data connection. With DSL providing much higher data rates, ISDN BRIs are moving ever closer to obsolescence. Nevertheless, when they are available, they could be useful for small installations that need only a few lines. Most VoIP gateways have cards to interface with these lines. As with PRI lines, take care to set the gateway's configuration to match the type of signaling that your line uses.

SIP Compatibility

VX implements the basic set of SIP functionality described in RFC 3261, together with the related standards:

- ♦ RFC 4566 and RFC 3264 for describing and negotiating supported audio codecs
- ♦ RFC 3550 and RFC 3551 for audio transmission
- ♦ RFC 2833 codec for sending DTMF tones separately from audio
- ♦ RFC 2617 digest authentication

We have tested VX with different SIP providers, gateways, PBX's, soft- and hardphones, and other gadgets, to make sure it works out of the box. You will find a list of some of these in this very manual, and on our web site. We strongly recommend using VX together with a tested device, but sometimes it might not be an option.

Telco-provided SIP trunks are one such case. Chances are that these will work, "just like that." However, it is not possible to test every SIP-talking-thing out there, and different vendors tend to interpret some parts of the standard differently. SIP started as a simple text protocol to set up phone calls, but over it has grown quite complex. While we strive to do the right thing, no one is perfect, and it's possible that we have missed a line or two. If you are experiencing problems, let us know and we'll figure it out.

That said, there are a couple of things that VX isn't likely to support in near future. One of them using multicast for both SIP signalling and RTP. This is an optional feature in the SIP standard, not widely used, and disabled in VX because it can cause conflicts with Livewire, which is multicast, too. The second is telephone URI handling, which might sound surprising for a phone system. Don't worry, it's fine. Simply put, it means that VX identifies each line with a single SIP address (extension@server entered in the configuration), and expects that any mangling of numbers (like adding or removing prefixes, or changing between internal and external numbers) is done by a gateway or PBX. For VX to do that would require it to know a lot more about the phone network, and believe us, you don't want to configure that.

Axia Element Console as VX Controller

The Element console makes a perfect partner to the VX. (With apologies to our colleagues in the Axia division, who would prefer to put that the other way 'round – "the VX makes a nice partner to the Element…" They would also point out that other Axia consoles, such as the IQ, are capable and compatible VX partners. Control and audio are tightly integrated for a talent-leasing all-in-one place control of telephone operations. Both selectable and fixed lines are supported. For the former. The Element's Call Control Module includes line select buttons and status icons. For the latter, Phone Fader modules have a couple of extra buttons to manage the Telco lines.

Livewire makes it easy to connect the multiple inputs and outputs that the VX's multi-hybrid capability affords.
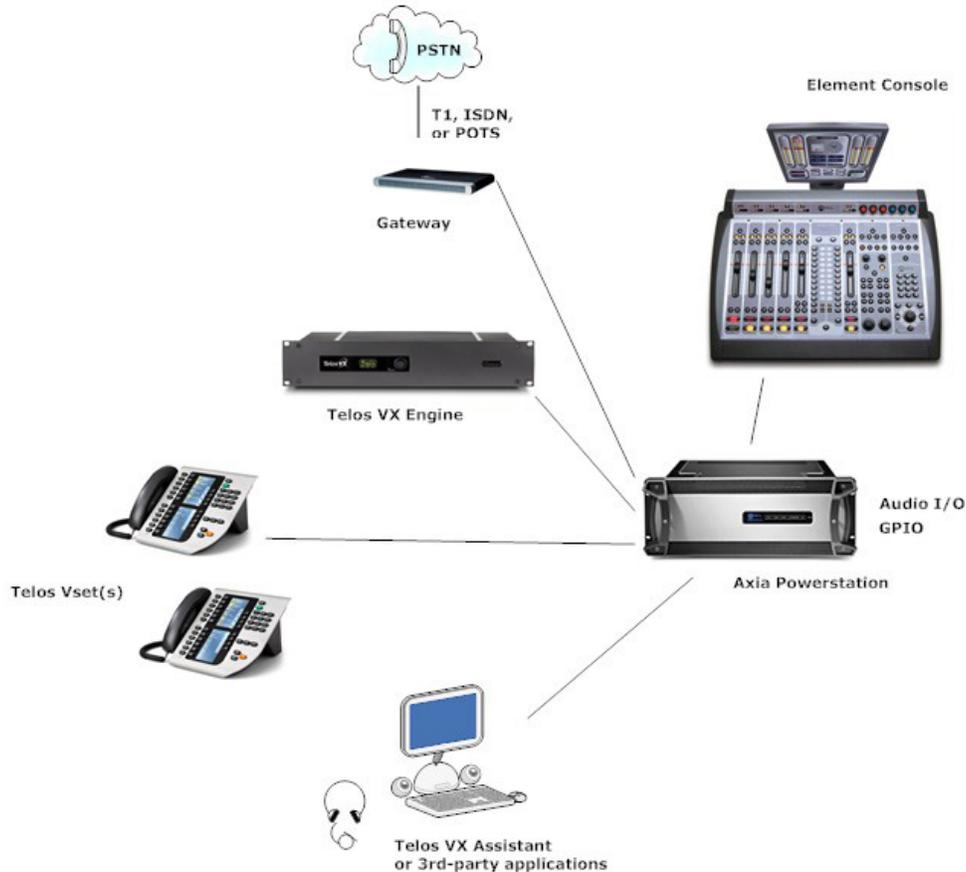
Installation

The setup in the block diagram shows the Axia Powerstation as the core for the studio, providing the Ethernet switch function. The VX Engine LAN port attaches to one of the gigabit ports and the VSets to PoE (powered) ports. The Engine WAN port could have its own switch to connect the VoIP lines and gateways. Or the VX can be configured so that the LAN port could serve everything with no additional switch. The Powerstation could also link up with a central switch to pass audio to/from other locations in the facility.

In the Element, Install or confirm the presence of the call control module and the faders you will be using for telephone operations. Some possible Element configurations are:

♦ 1 Call Control module for 12 or 24 lines, 2 hybrids/faders

♦ 1 Call Control module + 1 Phone fader module for 12 or 24 lines, 6 hybrids/faders

♦ 2 Call Control module for 12 or 24 lines, 4 hybrids/faders.

Note that a Call Control module is required even when no selectable lines will be configured. The fixed lines rely upon software linked to the Call Control Module.

Configuring the Element to Control the VX

Element configuration is via web.

Do the VX configuration first, and then move on to Element configuration.

Things you will need to know about your VX in order to configure an Element as a controller are:

♦ The VX Engine IP address

♦ User name and password to access the VX Engine.

♦ The VX *Studio* name you will connect to.

♦ VX hybrids/Livewire output names and numbers you want to use in this studio

♦ VX *Show* name if you want to load VX Shows from Element console profiles.

In order to correctly control your VX, your Element and/or PowerStation must be be running the following software versions:

♦ PowerStation: v1.1.3a or higher

♦ Element: v2.5.0.3 or higher

To obtain and install this software, visit www.AxiaAudio.com/downloads/ and follow the update instructions contained in the accompanying Release Notes.

There are three steps to configuring the Element for the VX, as follows.

1. Set up the VX-to-Element connection. Open the Element's *Module Manager/Phone Channels* web page. If you have the Caller Controller Module installed, you will see the following at the top of the screen:

If you don't see this, or if the ID or Module information is incorrect, you will need to "Capture" your current module configuration. This is normally done at the factory, but if you have added the phone module yourself, you will need to execute the Capture so that the Element can "see" the newly added module.

To Capture, press and hold the *Help* key on Element's Navigation Module for 5 seconds, until CAPTURE appears in the source name displays. On the phone module, you will see the assigned ID number in the upper left line icon position. You can change this by pushing the adjacent line button. End Capture Mode by pressing the *Enter* key on the Navigation Module.

Enter your VX's IP address. Use your Web browser to navigate to Element's *Sources and Profiles -> Configuration -> Show Profiles* web page. Select the Element Show Profile you wish to configure, and click on *Phone*. In the *Phone* URL field, enter the IP address of your VX. In the screenshot above, a VX IP address is shown as vx:192.168.0.9 where 192.168.0.9 is the IP of a VX Engine. You can also specify a username and password if needed. (If you use the default settings, where the username is "user" and the password is empty, you don't need to enter anything.)

2. Configure your VX Studio and Shows. In the same screen where you entered VX IP number, you can enter a VX Studio name. In the *Host /Studio Name*: field, simply enter the studio name you used in your VX's configuration. In the example above, the VX Studio name is "Studio 1."If you wish to change a VX Show using an Element Show Profile, simply specify the name of the VX Show you wish to load in the *Show Name* field. If you leave it empty, no VX Show will be changed, and whichever show is currently loaded will continue to be used. Leave the *Show Password* field empty. There are no passwords for Shows in the Telos VX, so none are needed from the Element.

    Configure other options. Element's Call Controller module uses the left row of telephone line selection buttons to control Hybrid 1, the right row to control Hybrid 2. If you wish to use the right row of Call Controller buttons to control Hybrid 1 and the left row to control Hybrid 2, click the *Reversed Hybrid* box.VX can be used in either 12 or 24 line mode. You can select either *Auto, 12 Lines or 24 Lines* with the Mode Selection radio buttons. In 12 line mode, 12 lines are displayed on both columns of the Call Controller module; the left key bank answers lines on Hybrid 1, the right keys answer lines on Hybrid2. In 24 line mode, both left and right key banks will answer lines using Hybrid 1. To answer lines on any other hybrid, operators will use the module's *SET* key to choose which hybrid to use. Selecting "Auto" uses the configuration of the VX Show to dynamically set the number of lines used.

Click "Save" when you're done. Note that after editing and saving a Show Profile, you must reload it from your Element console in order for the saved changes to take effect.

3. Create the needed *Source Profiles*. Navigate to Element's Source Profiles page. Click *New Source Profile* and, using the *Primary Source* drop-down box, select the audio channel that you configured in VX Engine under Studio configuration. In the list, you should see a Livewire channel number and short description of the Hybrid; give it an Element source name, such as "VX-Hybrid1," and select *Phone* for Source Type.

Primary source: 2410 <S1-Sel1@VX24>

Select Source - Google Chro...

http://192.168.0.19/srclist

VX-Hyb1

- Show sourcename
- Allow application to override
- Show both

```
1444 SRC 6@OlegAnalog
1445 SRC 7@OlegAnalog
1446 SRC 8@OlegAnalog
2401 VIP1@VX24
2410 S1-Sel1@VX24
2411 S1-Sel2@VX24
2422 S2:Selectable 3@VX24
2430 S3:Selectable 1@VX24
2440 S4:Selectable 1@VX24
2441 S4:Selectable 2@VX24
2450 S5:Selectable 1@VX24
2460 S6:Selectable 1@VX24
2470 S7:Selectable 1@VX24
```

- ☑ Channel 1
- ☑ Channel 2
- ☑ Channel 3
- ☑ Channel 4
- ☑ Channel 5
- ☑ Channel 6
- ☑ Channel 7
- ☑ Channel 8
- ☑ CR Monitor inputs
- ☑ Studio Monitor input
- ☑ V-mixer input

Source type:

- Operator
- CR producer
- CR guest
- Studio guest
- External microphone
- Line
- ◉ Phone
- Codec
- Computer player

In the source profile options, leave the *Phone hybrid IP* field empty, leave the Phone line and Phone hybrid (for NX12) fields at their default value of "0." Use 2nd show (split mode for NX12) should be left unchecked.

Phone hybrid ip:	
Phone line:	0
Phone hybrid (for Nx12):	0
Use 2nd show (split mode for Nx12):	☐
Phone Module ID:	1
Hybrid Nr.(0 - None):	1
Conferencing allowed:	No
Fixed line (1 to 24, 0 - No):	0

The Phone Module ID field lets you select from the multiple phone modules that you may have installed in your Element. Normally, you will have just one, so select 1 from the drop-down box.

For *Hybrid Nr.,* select the hybrid that corresponds to the Primary Source audio you selected above. The VX supports an unlimited number of hybrids per studio. To find out what Hybrid number corresponds to the audio source you selected, go to *VX Engine/ Studio configuration page* and check the sequence number of the *Selectable LW* channel that you use as the audio source here. The first Selectable LW channel counts as Hybrid 1.

Conferencing Allowed lets you permit "button mash" conferencing. Unless you specially wish to prohibit this type of conferencing, select *Yes* from the drop-down box.

Fixed Line lets you assign a line permanently to a specific fader channel, emulating a dedicated hybrid. Normally this is used with a 4-Phone Fader Module. If you want to use a Hybrid in this mode, specify one of your VX system's "fixed" hybrids in the *Primary Source* field. In the VX Engine studio configuration, you configure both fixed and selectable hybrids; enter a Fixed Hybrid number in this field if you wish to associate this source with a Fixed Hybrid.

The other fields in the Source Profile page are non-phone specific and can he configured as you choose. (See Chapter Two, *Configuring Inputs*, in the Element manual for info on these options.

Repeat this procedure to create a source profile for each hybrid in your studio.

You'll now need to load the hybrid sources onto the fader channels. This can be done either manually or automatically using *Show Profiles*. We recommend that you make this assignment in each Show Profile which will use the phone system, since leaving manual assignment to air talent is asking for trouble!

For normal operation, you would assign Hybrid 1 to the left fader channel of the Call Controller, and Hybrid 2 to the right fader channel. Additional hybrids, if you have them, may be assigned to any channel on a 4-Fader Phone Module.

Note: Within Element, each 'phone' type audio source has both an incoming (from caller) and associated backfeed (to caller) for automatic mix-minus generation. There are several configuration options that affect which audio sources are fed to the hybrid under various conditions; Refer to the *Source Profile Options* and *Feed to Source Mode* sections in Chapter 2 of the Element manual.

4. When you are done entering configurations, save your Show and Source profiles, and load one of your newly-edited Show Profiles using your Element console's Show Profiles command. You can use the Element Control Center Phone Channels page to confirm your VX connection, as shown below.

Push some buttons on the Element, marvel at how cool it is, and impress your staff. Use it to phone-up your Significant Other to congratulate her/him on her/his perspicacity and fine taste in mate selection.
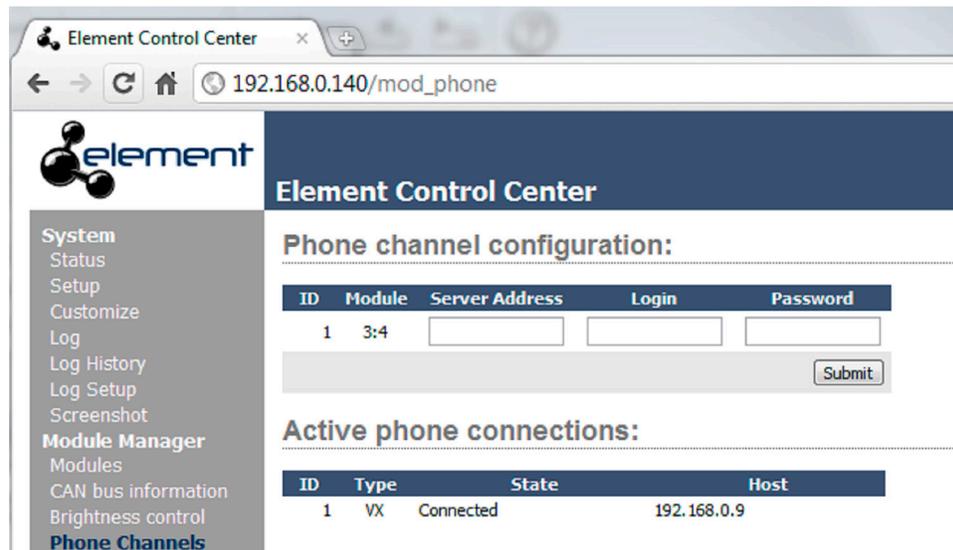
What's next?

The first stage of the application of any new technology is to replicate the function of what came before, but once the new platform is in place, creative people invent new and unexpected ways to use it. IP is a potent enabler that has already showed us plenty of surprising things; it's inevitable that more is on the way.

The Telos VX represents an exciting change in direction for studio telephone operations. The open IP nature of the system, along with its rich user interfaces and a powerful platform, offers ongoing opportunities for enhancement of its capabilities.

When the connection from a station's listeners to its studios eventually evolves to become via IP as it is near certain to do, things could change appreciably. We've already touched on the possibility for higher fidelity. But that's only the immediate and obvious next step. With an unconstrained pathway for data along with voice, a talk show's producer or host could text chat with a caller prior to his being on the air, for example. There could be an automatic updating of the time a caller is planned to be on-air. There could be instant voting. A window could be kept open on listeners' PCs that would deliver supplementary text or visual information. Peering yet further into the future, should video streaming catch-on; we might want both see and hear callers.

Here are some other things we have in mind.

♦ A gateway to Skype, allowing callers to connect via computer with higher quality than usual phones. It could be possible to exchange chat messages to update callers on how long they need to wait, etc.

♦ Build in a texting (SMS) gateway to allow this form of communication with mobile phones.

♦ Make a simpler VX phone for applications that don't need the fancy LCD displays.

We invite your suggestions and creative ideas.

Beyond Edison's Legacy

The PSTN uses the g.711 codec. Its audio frequency response is limited to 3.4kHz. A modern reader might ask, "Why was such a low fidelity considered to be satisfactory for digital telephony?" Microphones, loudspeakers, and earphones have all had much better fidelity for many decades. Indeed, when the g.711 codec was standardized in the 1960s, FM radio was just getting going with its much superior 15kHz bandwidth. Mostly, the coding choice resulted from the legacy of the carbon button microphones that were ubiquitous in telephones throughout most of the early history of telephony. (Edison invented the carbon microphone and licensed it to the Bell System, which had only an impractical liquid-based microphone in its own portfolio.) This microphone works on the principle that acoustic pressure squeezes the carbon granules more closely together, thus lowering electrical resistance in proportion to the pressure. As you can imagine, it has a non-flat response curve with a 5db peak at 2kHz and hefty roll-offs below 300Hz and above 3kHz. They were mostly abandoned for all but telephone use in the late 1920s, but were standard in phones up to the 1980s. Old analog long-distance lines also had a lot of high-frequency attenuation owing to capacitive effects. When microwave radios were introduced to long-distance telephony, a decision had to be made as to what frequency range to accommodate in their FDM (Frequency Division Multiplex) scheme. There was, as always, a trade-off: more frequency response meant fewer channels. Since the microphones weren't producing much in the way of high frequencies, why bother carrying them over the radio links? Thus, the radios were designed with narrow 4kHz carrier spacing. When the first digital T-Carrier systems where invented, it must have seemed perfectly natural to stay with the 3kHz or so audio bandwidth enshrined in the microwave link technology. A sampling rate of 8kHz with 8-bits (compressed) depth had a nice symmetry and delivered a satisfactory 4kHz Nyquist response limit , so on with the show.

The Electret capacitor microphones ubiquitous in today's phones have frequency response well beyond 3kHz. IP networking combined with modern codecs allow the better fidelity to be transported. Some mobile phone applications, such as Apple's 'facetime' are beginning to take advantage of this. The VX is ready for high-fidelity phone calls, and includes support for g.722, a codec rapidly gaining favor with VoIP providers and users. g.722 uses about the same amount of bandwidth as g.711, but samples audio at 16kHz - double that of g.711! It delivers very clean, clear, natural caller audio. Future VX software releases are likely to includes support for more high fidelity codecs.

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: 30

Number of hybrids: one per each active line

Maximum number of studios: 20

Maxiumum number of Livewire input/output channels - 20 systemwide

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

Controllers

VSet12 telephone

VSet6 telephone

VSet1 telephone

VX Producer Windows application

Axia Element studio console

Axia IQ studio console

NeoGroupe NeoWinners

Broadcast Bionics PhoneBox VX

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

Resources

Internet Speed tests

www.speakeasy.net

www.speedtest.net/

Internet VOIP Jitter Test

myspeed.visualware.com/indexvoip.php (free)

Packet "sniffer"

www.wireshark.org/download.html (free)

VoIP Soft phone" SIP PC clients

X-Lite "Soft phone" SIP PC client (free)

www.counterpath.com/x-lite.html

Ekiga, free open source SoftPhone - ekiga.org/

Linphone open source VOIP phone - www.linphone.org/

Open Source PBX Distributions

www.digium.com

pbxinaflash.net/

www.trixbox.org

www.freepbx.org/

www.asterisk.org/

Commercial PBX Products

www.microsoft.com/voip

www.lg-nortel.com

www.trixbox.org

www.siemens.com/hipath

www.mitel.com

www.digium.com

Gateway products and suppliers

www.patton.com/voip/
We have tested and support Patton gateways, such as the model 4940 for T1/E1 and ISDN PRI.
The Patton gateways for POTS (FXO) and ISDN BRI are also satisfactory.

www.grandstream.com

www.audiocodes.com

www.cisco.com

www.digium.com

www.quintum.com

VoIP/SIP equipment suppliers

www.telephonydepot.com

www.rockbottomvoip.com/

Telephony Discussion and VoIP news sites

www.sipforum.org

www.telos-systems.com

www.voipforums.com/

www.voipuser.org/forum_index.html

www.broadbandreports.com/forum/voip

forum.voxilla.com/

SIP Information and suggested reading

A Request for Comments (RFC) is a memorandum published by the Internet Engineering Task Force (IETF) describing methods, behaviors, research, or innovations applicable to the working of the Internet and Internet-connected systems. The IETF adopts some of the proposals published as RFCs as Internet standards. These are available at: www.ietf.org/rfc. Among those relevant to VoIP and SIP are:

RFC 1889: Real Time Protocol (RTP) - original version

RFC 3550 - Real Time Protocol (RTP) - latest version

RFC 1890: Real Time Protocol Audio Video Profiles (RTP/AVP) - original version

RFC 3551: Real Time Protocol Audio Video Profiles (RTP/AVP) - latest version

RFC 2327: Session Description Protocol (SDP) - original version

RFC 4566 - Session Description Protocol (SDP) - latest version

RFC 2543: Session Initiation Protocol (SIP) – original version

RFC 3261: Session Initiation Protocol (SIP) – latest version

RFC 3264: Codec negotiation

A. H. Inglis. "Transmission Features of the New Telephone Sets," *Bell System Technical Journal* 17 (1938): 358-380. (Just for fun. Waaay back, Mr. Inglis proposed "high-fidelity" phones, noting that cost was the only barrier to better audio quality in the telephone network and predicting it would become common in the future. Took awhile…)

Books

Henry Sinnreich and Alan B. Johnston. *Internet Communications Using SIP*, John Wiley & Sons, Inc., New York, 2001.

Gonzalo Camarillo. *SIP Demystified*, McGraw-Hill, New York, 2002.

Steve Church and Skip Pizzi. *Audio over IP: Building Pro AoIP systems with Livewire*, Focal Press, Burlington,MA and Oxford UK. 2010 (Mighty fine fellas, these guys. Good book, too!)

Jonathan Davidson and James Peters. *Voice over IP Fundamentals,* Cisco Press, Indianapolis, 2000.

J. Alexander, C. Pearce, A. Smith, and D. Whetten, *Cisco Call Manager Fundamentals 2nd Edition*, Cisco Press, Indianapolis, 2006.

D. Au, B. Choi, R. Haridas, C. Hattingh, R. Koulagi, M. Tasker, L. Xia. *Cisco IP Communications Express: Call Manager Express with Cisco Unity Express*, Cisco Press, Indianapolis, 2005.

Oliver Hersent, David Gurle, and Jean-Pierre Petit, *IP Telephony: Packet-Based Multimedia Communications Systems*, Addison-Wesley, Reading, MA, 1999.

Switching to VOIP – O'Reilly

VOIP Hacking - O-Reilly

Asterisk - O-Reilly

SIP Providers

Something that takes some time to get one's head around is that SIP providers usually do not provide the IP network connection. The main service they perform is giving you a gateway to the PSTN. That is, they give you telephone numbers. The gateway need not be local to you. Indeed, you may be in California and find that your provider's gateway is in New York. Providers also give you access to a "SIP server" that helps with obscure things like traversing 'NATs' and taking care of "relocation services". Because the SIP provider and network vendor are independent, it's sometimes hard to know who to blame for troubles – or praise for good service. It's a good idea to discuss network vendors with the SIP provider, since they have built up a lot of experience with clients around the country and often have insight into who best can deliver reliable Internet connectivity in your area.

Visit the Telos Website for the latest information. The short list below was created in December 2010 and the marketplace is changing rapidly. These are companies we or our clients have had direct experience with and report positively on their experiences. Most of these providers will allow you a free period to test their service in your environment (your ISP, with your switch, etc). Consider that some services work better in some areas than others. Usually it's wise to choose a provider close to you physically.

Having said that, some Telcos and other companies offer bundled IP network and SIP service. Generally, these are more expensive than the unbundled services, with the pitch that having a single vendor delivers higher reliability and one-stop troubleshooting. We have seen this approach more often outside the USA. It remains to be seen if the promise is fulfilled. We'll keep our eye on it.

BYOD (Bring Your Own device) Providers are recommended for broadcast application. Companies such as Vonage that supply an ATA (Analog Terminal Adapter) for each "line" do not provide "SIP credentials" for standards-based equipment such as the VX and are not oriented to serve customers with sophisticated requirements such as broadcast stations. They tend to be more expensive, as well. Broadcast users are better served by wholesale providers who deal with Telephony and IT professionals.

VoicePulse is a smaller company that excels at providing tech support. They have proven themselves excellent at solving obscure problems that many providers will just ignore.

They know Asterisk and can be counted on to give good advice. They don't have phone numbers everywhere, but they probably won't disappoint you. East and West Coast gateways are available. Local Number portability (LNP) is available.

Bandwidth.com is used by at least one station running a VX, with solid performance being reported.

Vitelity offers reliable and very cost effective DID numbers any many local exchange areas, and toll free and foreign numbers as well. Local number portability and SMS messaging are also offered.

SipStation - with connectivity provided by Bandwidth.com they've proven to be very reliable and low cost.

8x8 - Based in Silicon Valley, they deal mostly in hosted PBX service. Reliability is very good.

Flowroute - Based in las Vegas offers very attractively priced DID numbers and special deals on blocks of 20, 50 and even hundreds of numbers.

Local Number Portability in the USA and Canada

Telephone number porting is fairly common these days, mainly due to the large number of wireless customers changing carriers and plans frequently and wanting to keep their phone numbers.

Each carrier has its own rules and policies for number porting with the average port taking from 10 days to 3 weeks. It is important to note that high volume or "choke" numbers cannot be ported. You can try, but we haven't seen it work yet. Usually, the order will be taken, a due date given, and then on the day of the port, you'll get a call from the utility or provider informing you of the bad news.

Porting is usually done in the morning, but special arrangements can be made with the carriers to port at a specific time. Once ported, your old POTS lines are automatically canceled by the old company. We have seen situations where CLECs and Wireless companies have some problems routing calls for the first few hours after a port, though this seems to be getting better.

Don't run crazy "win a house or car" contests on these providers, though they're fine for request lines and even talk show call in numbers. If you plan to do heavy contesting with big prizes, it's not nice and potentially dangerous to bring your carriers switching office down.

Introduction to SIP's insides for the Curious

The following is a peek inside of SIP. There is no reason you need to know any of this to use the VX. It's an excerpt from Steve and Skip's book *Audio over IP*.

SIP is fast rising to be the big-daddy buzz-acronym among telecom technology acolytes. SIP is how calls are set up over IP connections, so it is actually pretty important. Together with helpers like Proxy Servers and User Agents, SIP permits all the familiar telephone-like operations: dialing a number, causing a phone to ring, hearing ringback tones or a busy signal. It also enables next-generation capabilities such as finding people and directing calls to them wherever their location, Instant Messaging, and relaying so-called 'presence' (near the phone or not, do-not-disturb, etc.) information. SIP began, rather humbly, as a simple message protocol for setting up connections. But the term has grown to be an umbrella for the family of protocols and tools that have been developed by the IETF to enable VoIP telephony and related services.

By the mid-1990s, audio and video were becoming routine on the Internet. Going beyond email, academic researchers were imagining on-line audio/video/whiteboard conferences where ideas could be shared live. It became clear that the *Mr. Watson come here, I want to see you!* function had to be done more efficiently than by shouting across the college quad or sending invitation mails. Thus was the IETF's working group MMUSIC (Multiparty Multimedia Session Control) born. There had already been work within the telephone world that had resulted in an ITU standard, but Internet types didn't much like it. "Too complicated," they averred. "Too limited," they sniffed. "Too *Phone Company*," they huffed. So off they went to do it the *Internet* way. The document describing SIP was eventually published as proposed standard RFC 2543 in 1999. Work has been ongoing, with the latest version of the specification, as this is written, being RFC 3261.

The SIP message protocol is similar to the Web's HTTP (Hypertext Transfer Protocol), and shares some of its design principles: It is human readable and request-response structured. SIP even shares many HTTP status codes, including the familiar "404 not found."

Here is a typical SIP message:

```
INVITE sip:skip@there.com SIP/2.0

Via: SIP/2.0/UDP 4.3.2.1:5060

To: Skip Pizzi <sip:skip@there.com>

From: Steve Church <sip:stevec@here.com>

Call-ID: 4678995554545@4.3.2.1

CSeq: 1 INVITE

Contact: <sip:stevec@4.3.2.1>

Content-Length: 126
```

This is how Steve's SIP client would signal to Skip's that he wants to connect and speak with him.

SIP works together with several other protocols and is only involved in the signaling portion of a communication session. SIP is a carrier for the SDP (Session Description Protocol), which describes the media content of the session, e.g. the codec being used, the bitrate, etc.

SIP provides the following capabilities:

♦ Determines the location of the endpoint: SIP supports address resolution, name mapping, and call redirection.

♦ Determines the media capabilities of the endpoint, i.e., which codecs are available and supported: During a negotiation, SIP determines the best codec that can be used by the parties on the call.

♦ Determines the availability of the called endpoint: If a call cannot be completed because the target endpoint is unavailable, SIP returns a message indicating this and why.

♦ Establishes a session between the originating and called endpoints (if the call can be completed).

♦ Handles the transfer and termination of calls: SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP establishes a session between the transferee and a new endpoint (specified by the transferring party), and terminates the session between the transferee and the transferring party.

The Parts of a SIP System

Like most things based on IP, SIP was designed to be modular. Implementers can pick-and-choose among the following elements to build the system they need.

SIP Clients: Sometimes called User Agents or endpoints. These can be hardware phones or "softphones" (phone applications running on PCs).

Gateways: When needed, gateways translate between the IP network side and the switched-circuit Telco side, providing physical, electrical, signaling and audio interface.

Proxy server: Receives SIP requests from a client and forwards them on the client's behalf. Basically, proxy servers receive SIP messages and forward them to the next SIP

server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission and security.

Redirect server: Provides the client with information about the next hop or hops that a message should take, and then the client contacts the next-hop server or client directly.

Registrar server: Processes requests from clients for registration of their current location. Registrar servers are often co-located with a redirect or proxy server.

While SIP-enabled endpoints are able to connect directly to each other, SIP servers provide a number of valuable services, including the following:

♦ Register SIP client devices

♦ Register individual human users for access to their services

♦ Perform authentication, authorization, and accounting (when needed)

♦ Look up the address of the far-endpoint

♦ Support user mobility across networks and devices

♦ Support for multipoint conferencing

♦ Support presence information

♦ Inform users as to call progress

♦ Communicate requests for QoS to various network elements, such as IP routers

While the various servers above could run on individual machines and could even be physically separated by thousands of kilometers, in usual practice they are often just software elements of an application running on a single machine. As we'll see, many small IP PBXs include the Gateway as well, making a one-box solution that includes everything needed for a small office installation.

An example a SIP Server being used in the broadcast world comes from the Telos Z/IP codec family, which uses a Telos-developed, enhanced SIP Server, called (naturally) Z/IP Server. The server is provided as a service on the Internet, but may also be installed by users who prefer to maintain their own. In addition to basic SIP functions (registration, address look-up), the Z/IP server offers additional services:

♦ Allows display and dialing by simple text name. Keeps a database of names and performs DNS/IP look-up upon a dialing request from an endpoint codec.

♦ Maintains group lists created by users. Upon entering a group name and password, the list is displayed on endpoint codecs so that users aren't burdened with having to enter or upload lists manually.

♦ Provides geolocation services by associating IP numbers with physical location. Allows display of a routing map on the codec LCD display.

♦ Upon request, keeps a record of network performance in order to assist in troubleshooting problems caused by QoS impairments.

Many products that support SIP for its standards-based interconnection capability do not have an internal architecture corresponding to the SIP specifications, so you would not see these SIP

server components listed therein – or they might be included, but not labeled according to the standard names bulleted above. Instead these functions would just be provided as part of the system "black box." Cisco and Microsoft VoIP products fall into this category, for example.

Addressing

SIP addresses, also called SIP URIs (Uniform Resource Identifiers), are in the form sip:user@ host. The user portion of the address can be a text name or a telephone number, and the host portion can be a domain name or an IP address. The address resolution process normally begins with a URI and ends with a username at an IP address. Just as with email, the sender needs no information about the physical location or IP address of the receiver. This is one of the powerful features of SIP – it automatically implements portability and mobility.

Examples of valid SIP addresses:

The usual form is an email address prefixed by 'sip:':

sip:joesmith@company.com

You can call a PBX telephone at a business this way:

> `sip:123@telos-systems.com` (extension phone 123 at Telos Systems)

If you don't have a name or extension, you might want to contact the receptionist:

> `sip:receptionist@telos-systems.com`

Here's an internal machine-to-machine message, such as from an on-air phone system to a PBX or gateway to initiate a PSTN call:

> `sip:12162417225@168.123.23.1`

Note that in this case, an IP number is provided to identify the concrete machine that is to receive the message. You usually don't want to use DNS for this because it takes time for the look-up step, and because there may well not be a DNS name associated with a machine being used as a telephone server.

To assist readability, SIP lets you use '+', '-', and '.' separators. It removes them prior to processing:

> `sip:+1-216-241-7225@telos-systems.com`

As you can see, SIP bridges the telephone and Internet worlds. Both Web-type and PSTN telephone number addresses are possible, and users on either network can reach those on the other.

Often, address resolution involves multiple steps and SIP-message hops. A DNS (Domain Name Service) server, a SIP proxy server and a SIP redirect server might all be involved in a single name resolution, for example.

A few other points of interest regarding SIP: Some servers associated with SIP systems can accept unformatted text names, but this is not part of the standard.

URIs are not URLs. URIs are independent of the location of the named object. Email addresses are an example of URIs. In SIP, a Request-URI is defined to indicate the name of the destination for the SIP Request (INVITE, REGISTER, etc.). URLs (Universal Resource Locator) describes the location of a resource available on the Internet. For example, http://www.telos-systems.com is the URL for a Web home page. It is resolved by DNS to a concrete IP address.

PSTN telephone numbers are sometimes called 'E.164' numbers, a designation applied in an ITU-T standard that describes the format of telephone numbers to be used worldwide. ENUM

(**E**.164 **NU**mber **M**apping) is the Internet service used to look up the URI associated with a particular E.164 telephone number. It's part of the DNS system. SIP can use ENUM to locate the VoIP system associated with a telephone number that accepts incoming calls.

How SIP Works

As we've seen, SIP is a simple, text-based protocol. It uses requests and responses to arrange for communication among the various components in the network, and ultimately to establish a connection between two or more endpoints.
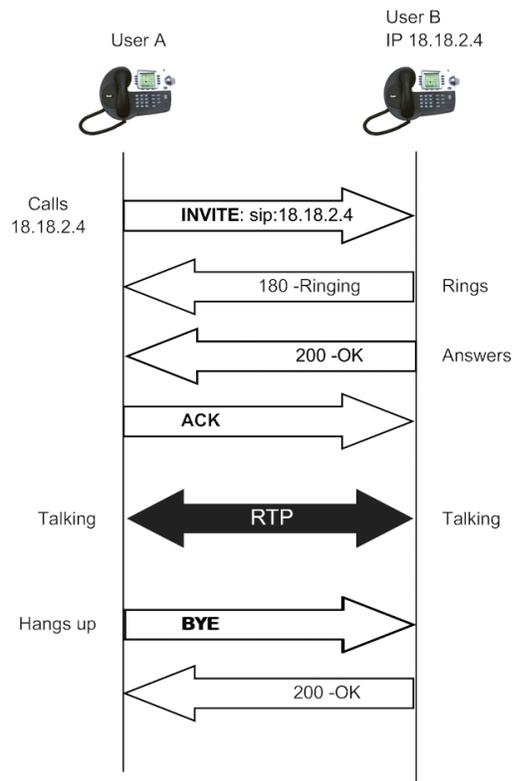
SIP at its simplest. An IP phone calls another directly.

But this is not the real world. Almost always, there are SIP servers of various kinds in the picture. When a user initiates a call, a SIP request is sent to a SIP server (either a proxy or a redirect server). The request includes the address of the caller and the address of the intended called party. In more sophisticated scenarios, users register with a registrar server using their assigned SIP addresses. The registrar server provides this information to the location server upon request.

From time to time, a SIP user might move between end systems. The location of the user can be dynamically registered with the SIP server. Because the end user can be logged in at more than one station, and because the location server can sometimes have inaccurate information, it might return more than one address for the end user. If the request is coming through a SIP proxy server, the proxy server tries each of the returned addresses until it locates the end user. If the request is coming through a SIP redirect server, the redirect server forwards all the addresses to the caller in the Contact header field of the invitation response.

When communicating through a proxy server, the caller sends an INVITE request to the proxy server and then the proxy server determines the path and forwards the request to the called party.

Since we usually need to reach phones that are connected to the PSTN, gateways will be involved in real world systems. These translate SIP signaling to the PSTN's requirements: loop current, DTMF, and ring detect for POTS lines, set-up messages for ISDN, etc.
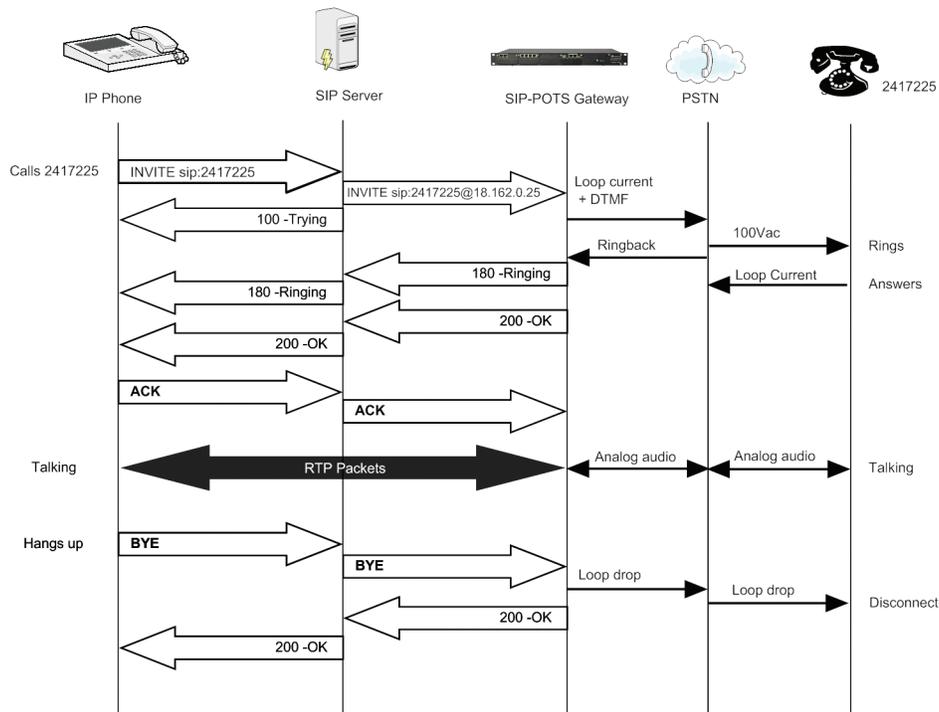
A SIP call set-up to the PSTN via a SIP server and a gateway to POTS lines.

SIP messages may be carried by UDP or TCP. SIP has its own built-in reliability mechanisms, so it doesn't need TCP's reliability services. Most SIP devices such as phones and PC clients use UDP for transmission of SIP messages. PBXs on LANs almost always use UDP because LANs don't drop packets and there is no need to incur the overhead of TCP. Transport Layer Security (TLS) protocol is sometimes used to encrypt SIP messages. TLS runs on top of TCP. (This is the protocol used with HTTP to make the secure HTTPS used for secure Web transactions.)

Not shown in our transaction examples is the media negotiation that is part of the INVITE/200 OK/ACK sequence. Through this process, endpoints decide which codec to use. The *Session Description Protocol* (SDP) defined by RFC 2327 is the way codecs are offered and (hopefully) accepted by the other end. Usually, the caller sends an SDP message along with its INVITE, listing the codecs it is prepared to use. The far end chooses one of them and tells the caller which it prefers in the 200 OK response. The caller can let the far end propose a codec by not sending an SDP message in its INVITE. It is possible that the two endpoints have no codec in common and the connection is unable to proceed, but systems are designed so that this does not happen. For example, almost all phones, gateways and SIP Telco services have G.711 as a supported codec, so this is an insurance policy that two endpoints will find common ground. Within a PBX system, designers usually choose one codec as a standard for the system and stick with it for all connections. For example, the Telos VX studio system uses 8kHz sampling-rate, 16-bit uncompressed PCM internally for all calls that connect to the PSTN.

The State of SIP, and its Future

SIP is not often used as it was intended by its developers. Most PBXs that have SIP interfaces don't use SIP servers at their core. All use their own rough equivalents, designed independently. So what went awry? The SIP schemers were certainly far-ahead thinkers, who wanted their protocol to support rich media, mobility, portability, sophisticated endpoints, etc. The problem seems to have been a lack of a certain "practicality." For example, consumer PC-to-PC VoIP products needed to solve the problem of firewall and NAT traversal, which has been addressed quite slowly within the SIP working groups. Meanwhile, Skype's developers solved it quickly and effectively. Then there is the problem of supporting all the features a vendor wants to employ to differentiate its product. It's unsurprisingly faster to just implement it your own way rather than waiting for the idea to make its way through a committee, who might well not see things your way.

At Telos, we faced this problem in the design of the VX system. We needed a lot of things specific to the studio environment that are not supported in SIP's structures. So we designed our own protocols for use within the boundaries of our system. But we use SIP at the border of the system to connect with other vendors' products – and eventually to the Telco network. This is just the strategy Cisco, Microsoft, and almost all PBXs vendors have followed.

And now *this* process is emerging as SIP's great value. It's the glue that ties systems together. Studio systems can talk with PBXs for the first time. PBXs can talk with each other. And eventually they will all be able to talk to Telco networks, smoothly and fluently. SIP's inventors got something right.

IAX as a SIP Alternative

SIP is not the only game in town. The **I**nter-**A**sterisk e**X**change (IAX) protocol is an alternative to SIP for interconnections between both VoIP servers and for client-server communication.

IAX2 (as the current version is named) uses a single UDP data stream (usually on port 4569) to communicate between endpoints, both for signaling and data. The voice traffic is transmitted in-band, in contrast to SIP, which uses an out-of-band RTP stream for audio. IAX2 supports multiplexing channels over a single link. When trunking, data from multiple calls are merged into a single set of packets, meaning that one IP datagram can deliver control and audio for more than one call, reducing the effective IP overhead without creating additional latency.

As IAX's name indicates, it was invented by the Asterisk people as a way to trunk calls between one Asterisk server and another. It has escaped from Asterisk and is now supported in a variety of softswitches and by a few VoIP carriers. Its main advantage is its bandwidth efficiency and simpler firewall configuration, since all traffic flows through a single port.

Using VX with the Asterisk PBX

Asterisk is a great way to provide extra flexibility and power to your VX installation. The software is inexpensive or free and runs on a standard PC. It's an open source project, having been created by teams and individuals from around the world, often volunteers. Despite this, we've found it to be reliable. At Telos we count on it daily to network our sites throughout the world. A few VX installations are already using it. You'll find that it's a useful tool if you want to do Interactive Voice Response systems, support on or-off-site VoIP telephones, call detail recording, failover systems, ISDN PRI channel splitters, and such. With appropriate hardware, Asterisk can also be used as a gateway. For example, you may need to peel a few channels off of a PRI to send them over to the VX, while the others feed a PBX. An ISDN PRI card in an Asterisk box and some configuration would get this done.

Because it connects to the VX using Ethernet, audio remains clean. Many channels of audio and SIP signaling are interfaced over a single Ethernet link.

Telos has people on-staff and access to consultants who are experienced with Asterisk and who can help you with regard to integrated Asterisk/VX installations. Feel free to call us with your questions and suggestions.

Here are some ways our clients are using Asterisk:

♦ To automatically play a disclaimer or informational message to callers before your producer or talent answer.

♦ As a programmable ISDN PRI gateway. Use with the PRI/T1 that feeds your office PBX, and redirect some of the DID numbers to your VX or to a voice mail/Interactive Voice Response system running on Asterisk.

♦ To block unwanted callers by implementing a 'blacklist' based on caller ID.

♦ To create your own off-site 'extensions' over the Internet. It's great for news bureaus, sales offices, or even reporter's homes.

♦ To create a private 'tie trunk network' between co-owned stations to reduce costs

♦ To have a backup Telco service by ordering a few inexpensive SIP lines or trunks to use in case of loss of a T1.

♦ To get rid of expensive toll free numbers or foreign exchange lines. Order SIP lines from a distant city that your station covers. SIP lines can cost as little as $1.50 a month. Also, SIP lines don't have all the extra charges for city taxes and other "junk fees".

♦ Use the Asterisk call detail reports as a research tool. The reports tell you who called, when they called, where they called from, and how long the call was. The data can be exported to a csv formatted file that is imported into a spreadsheet or database application for further processing.

Asterisk software: Distributions

Asterisk runs on Linux. You can download Linux from many websites and then add Asterisk to the operating system. Or you can load The OS and Asterisk from an all-in-one distribution disk. You can also buy an Asterisk "appliance" pre-loaded from companies like Fonality (Trixbox) and others. If you have one of these, Skip on to the next section titled "Log in to the web GUI for the first time".

A 'distribution' is an organized and maintained collection of open-source programs, these are available on an ISO image file that is burned to a CD and booted, automatically formatting the hard drive and creating the entire system from "bare metal."

Distributions can include the Linux OS, the Asterisk components, GUI tools, and specialty modules that the creators of the distribution deem useful. Choose one that has what you value. If this is your first experience with Asterisk, we suggest keeping it simple. If you know a "Linux guru," his assistance may be helpful, but not essential. Linux has come a long way over the past few years, and is now perhaps even easier and faster to install than Windows. You'll also find that you can go years without a reboot due to its excellent stability.

We've had success using the "PBX in a Flash" distribution. It is feature-rich, well maintained and documented, and tends to be kept more current than other distributions. It's more maintainable than some other distributions and tough to "break". Basic configuration is done through a built-in web server and a powerful "happy face" web front end called FreePBX.

Another choice that looks good is the Digium product called Phonevox. This is a ready-to-go software and hardware VoIP PBX from the main developers of Asterisk. Because it is a commercial product, it has support from the vendor.

Hardware

Unless you are experienced and confident, we advise you to buy a pre-built, configured, s/w-loaded, and supported Asterisk machine such as the one Digium sells. Nevertheless, we offer the following gratuitous advice for those of you who decide to go it alone.

You are building a machine that will need to be more reliable than most of your regular file servers. You don't want to be rebooting your phone system! We like to use a 4u Rack Mount Chassis with generous cooling and a higher end power supply. 4U cases are inexpensive, roomy, usually have good cooling with low noise fans, have good filtering options and can accommodate just about any mother board. We like to use SATA drives in removable caddies for easy full system backups. Choose a mother board with PCI slots if you intend to use expansion cards such as for T1 line interfaces.

Choose a server-grade motherboard. We usually go with on-board video to reduce costs. The CPU need not be the biggest, baddest, fastest thing out there, but it should be in the "sweet-spot" with regard to price/performance.

We often put an extra Ethernet card in the machine or buy a motherboard with two Ethernets on the board. One is configured one for the "inside" (LAN)" and one for the "public" (WAN), similar to the VX.

2-4G of RAM is fine, but more is better. Hard disks should be server-grade and kept well-cooled for longest life. The goal is to assemble a machine that can run for 5 years or more. We are aware of Asterisk servers that have run continuously for three years or more, and hardware that is still in service after 15 years.

Server grade doesn't mean exotic RAIDs or multiple processors; a 'low-end' basic server will be fine. You'll find that the more common the hardware, the better supported it will be by Linux and the fewer problems you will have with compatibility. Stay away from 'Bleeding Edge' hardware, but spend a few extra bucks on a better CPU heatsink/fan or power supply. We've learned over the years that the most common failures are fans, power supplies and optical disks. If you treat hard disks well (mainly, keep them cool) they'll be more likely to last. Also many newer power supplies have an '80+' logo. This means that they are better than 80% efficient, which means less power consumed and less waste heat. As a general rule, buy a larger power supply than you think you'll need.

If you are the extra-careful sort, consider building two identical boxes - or at least keep a spare identical motherboard and set of peripherals, such as Telco interface or Ethernet cards, on-hand. The convenient availability of a given motherboard and some kinds of PC hardware will usually be less than a year at the current pace of development.

Keep any filters clear and inspect the machine periodically to see if 'dust bunnies' are collecting anywhere. Do a full backup using the Linux 'DD' (full disk image bit copy, or 'clone') command once in a while (quarterly/annually), or after you are satisfied that any major change you've made should be retained (after a trial period). Consider that the hard drives might hold users' voicemail, asterisk configuration, and call records. There is a good backup utility that will automatically make regular backups of the unique data (and whatever you specify). Think of this as an 'incremental backup'.

There will probably be spare capacity, so you could use your Asterisk server to provide other services you need, such as centralized logging using Syslog, or a Network Time Protocol (NTP) server.

Expansion Cards

Because Asterisk has become so popular, there are a variety of expansion cards available that can be used with it.

The one we use most often is the ISDN PRI / T1/ E1 "Tormenta 2 or 3" type card, available from a number of vendors. ISDN PRI and T1 or E1 use the same card, since ISDN employs T1 or E1 as the underlying transport. A four-port PCI card is available from Phoniceq, inc (www.phoniceq.com) for about $350. Quality and service from the company have been excellent. They also offer a number of other cards, including for ISDN BRI. The company offers examples of use and driver packages that simplify installation.

Asterisk creators Digium also offer a wide range of card types and support options. www.digium.com/en/products/digital/

We don't recommend installing analog interface cards within an Asterisk server. While there's nothing wrong with the cards, we think it's better to use external gateways to interface POTS lines and Telephones such as the small 'appliance' gateways from Cisco/Sipura, Grandstream, AudioCodes, Patton, and others that go straight to SIP/Ethernet. Another way to go for more than 8 lines is to repurpose a 'Channel Bank' with a T1 interface and connect it to A T1 card in the Asterisk. Channel banks are available at reasonable cost on ebay and from used equipment suppliers. It should be equipped with an FXO (Foreign Exchange Office) interface if you plan to terminate analog lines from a Telco. Channel banks use different interfaces, so be sure that you get the right hardware configuration.

Better is to avoid analog lines altogether and use a digital connection, either IP/SIP, ISDN PRI, or T1/E1. All are 'four wire", maintaining send/receive isolation.. ISDN PRI is the usual choice in both the USA and Europe.

Following is a step-by-step guide to setting up Asterisk. With the hardware in place, you are ready to move on to getting the software installed and configured.

1 Get your 'distro'

The most popular are at:

- www.digium.com
- pbxinaflash.net
- www.trixbox.org
- www.freepbx.org

Download your distribution. We'll proceed assuming that you're using PBX in a Flash, hence-forth 'PIAF'. It's a few hundred megabytes. Once you have it saved on your hard drive, you'll need to burn it to a CD. It'll have a file name that ends with .iso and you'll have to use your CD burning software to "burn an ISO image". This procedure varies depending on what burning software you have installed on your machine (Nero, etc), so if you haven't done this, we suggest 'googling' the search term "burn an iso image with" and add your burning software name at the end. Many people have had difficulty with this process in the past and you'll find clear instructions there. It's important that you burn the 'image' and not just copy files. The CD that you create will boot up by itself and will wipe out anything on the installed hard drive. Before booting, check the bios in your machine to be sure that CD-ROMs are bootable. If the machine is recent, no action should be necessary.

2 Get loaded: Install Asterisk

Put your freshly-burned PBX-In-A-Flash disc into the CD-ROM drive and start the machine. You may need to press the space bar when the machine asks if you would like to boot from the CD-ROM. If you have trouble, check the boot menu in your BIOS. Sometimes you have to set a BIOS menu option to allow CD-ROM booting. Most newer boards allow CD booting because Windows and Linux are distributed on CD-ROM. Have an Ethernet cable plugged into your new machine, as the system will attempt to download the latest updates at the end of the installation. You should have the machine connected to a router that has a DHCP server so that the machine can configure itself with the parameters that your router gives. If you are using two or more Ethernet cards, the new system will only attempt to use one of them for the automatic updates, usually the onboard one or the card with the lowest MAC address. It may ask you to choose one. It's no big deal If you guess wrong because you can change it later (and get the updates) or configure it all manually.

The procedure will start by copying all of the files needed for the basic operating system into a RAM disk that is used for the install. You'll be asked a few questions along the way:

From the text menu, select an install type. Take the default by pressing the ENTER key. This selects the 'Default' installation. It will remind you that the machine will be wiped completely.

Select the US keyboard (or the keyboard of your choice) when prompted, then press the <TAB> key to advance to the "OK" prompt. After this selection, the GUI starts up. Move your mouse to verify that it works.
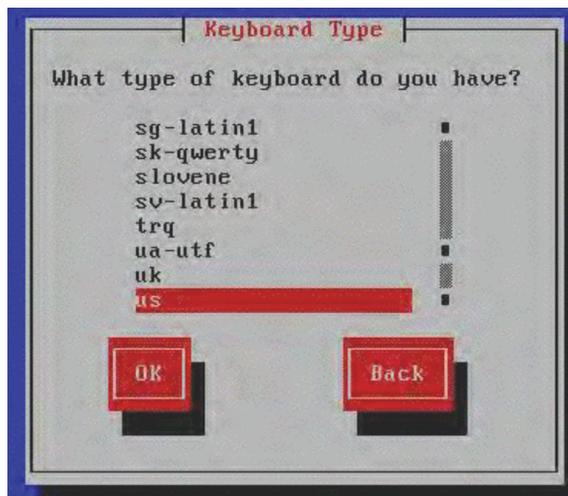
The system asks how you want to use the disk. It points out that *You have chosen to remove all partitions (ALL DATA) on the following drives* <lists drives>. If you're sure that this is OK (and it should be), select "Yes".

You'll be asked your time zone. Select the correct one and click "Next"

Next, you'll be asked to enter your root password. It is case sensitive! Enter and confirm it as asked. Then click Next

> **TIP**
>
> Be sure to remember your root Password. You'll need it later to continue the system setup.

Centos installation commences (this is the operating system). The drive will be formatted and the installation and program files are copied. The operating system will be installed and

then Asterisk and the PBX in a Flash files. Enjoy a sip of your favorite beverage as the process continues over the next few minutes,

Eventually, you'll see a message congratulating you and that "Installation is complete". Not so fast... The operating system is installed, but there is the second part of the installation: the Asterisk and PIAF! Don't worry, the system is running from a "kickstart" script that that will do all of the ugly stuff for you. Keep that beverage handy.

Remove the CD-ROM and click "reboot"

The system will shut down and restart. It will then offer you *Please choose one of the following options:* Choose 'A' (recommended option.This will begin a download of the latest version of PIAF. At this point, the system expects that you have it plugged into a network with a router that has DHCP and it will attempt to grab an IP address, netmask, gateway address, and Domain Name Server (DNS) address. If your Internet connection isn't working, the install will appear to bomb at this point. This is not a disaster! But more than likely it will just work, then you'll see it download the updates and continue to compile the program over the next 20 minutes or so (depending on your machine and connection speed).

Don't interrupt the process. It may look like not much is going on, but watch the activity of your hard drive LED. It should blink periodically. Messages on the screen will update every once in a while. The system will firmly suggest that you do not press any keys to interrupt this process.

Upon completion, the system will reboot by itself and you'll then see the colorful PIAF "splash screen" followed by a normal boot up that should leave you with a login prompt.

TIP

You can always choose the defaults unless you've "gone off the reservation" and are trying to do something unusual or complex (in which case you're probably smarter that your writer. In other words, "you're on your own!" Google and your "Linux geek" are your best friends. The good news is that many people have done what you're doing and have had the same problems that you have. Fortunately, they often share solutions on the web.

Log in to the Console for the first time

No, not the mixing console! *Console* is the Linux term for the command line interface. You'll be greeted with a friendly (if spartan) login screen that looks like this:

CentOS release 5.5 (Final)

Kernel 2.6.18-194.8.1.e15 on an x86_64

pbx login:_

Enter 'root' and hit <enter>

Password:_

Enter the password that you set for root earlier in the setup and hit <enter>.

You should get a full page status screen that shows version numbers, services running and an **OK** or other summary.

Then, at the bottom of the screen you'll see:

root@pbx:~ $_

Type 'help-pbx' and <enter>, you'll get a list of commands. First, Let's set up your ethernet IP address and get this box on your network so that you can get it going fully.

At the '$' prompt, type 'netconfig' and <enter>. You'll get a blue screen (not like in windows, don't worry!) asking if you'd like to set up networking. Press the space bar if the "YES" button is highlighted, (the <TAB> is used to toggle between yes and no). The next screen has a place to enter an IP address, netmask, default gateway and name server, or a check box that you can select with a spacebar tap that enables DHCP so that your router or DHCP server can assign an address for you. We suggest that you assign a static address, as this machine is a server that will always be online and running. Below is our configuration; change this for your network

On our lab network, the router is on 192.168.0.1. We use a class C netmask of 255.255.255.0 and the DNS server at 8.8.8.8 works almost everywhere and is fast, so it's used often.

Use the <TAB> key to advance to the OK button and <enter> to select it.

At the prompt root@pbx:~$_

Type 'ifup eth0' and hit return. There will be a short pause and then you'll get the prompt back. Your ethernet interface (eth0) should now be up and working.

If you're connected to the Internet, type 'ping yahoo.com <enter>' and if you're successful, you'll get your ping returned from yahoo.com. To stop the pinging, hold down control and hit 'c' (Ctrl-c). You should get the prompt back again. This little test proved two things: that you have internet connectivity and that your DNS name resolution worked. (it looked up the IP address for yahoo.com and gave your box the ip address for them). If you reboot the machine, the eth0 interface will come up automatically from now on.

Please be aware that this tool only configures a single Ethernet card. If you have more, you can configure them later.

TIP

Is your system really slow at certain points or does it seem to stop? DNS is used for many things in Linux. If it isn't available, many services will be VERY slow. Minutes-long pauses waiting for name resolution timeout are not uncommon and some things will appear to stop working entirely. So if you have long pauses while working with your system, lack of DNS is a probable cause. If you plan to run your system without DNS, you'll want to fill in the 'hosts' table with any addresses that might be logged or 'talked to' by the system. It is found at '/etc/ hosts'. You can edit the file using 'mc' (Midnight Commander) or other editors like pico and vi. Follow the examples in the file.

Set the passwords for various features

At the shell (command) prompt, type passwd-master and then <enter>. You'll be prompted to enter the password twice. If they match, the password is set. Save these or choose passwords that you won't forget. Do the same for the following commands:

passwd-maint

passwd-wwwadmin (you'll use this one most often)

passwd-meetme

passwd-webmin

Turn some services off for now

At the shell prompt, type 'disable-iptables' and then <enter>, then type 'disable-fail2ban' and hit <enter>. These are security tools that could get in your way until you're fully set up. If you run your machine behind a router or firewall, you won't need these anyway.

3 Configure Asterisk

Asterisk needs to be configured for your application. There is plenty of information on this topic on the Asterisk site. In brief, the steps are:

1. Set up trunks.

2. Set up outbound routes.

   We'll work through a simple example in the following pages.

   We prefer to use SIP extensions rather than SIP trunks to connect Asterisk lines to the VX because they are more flexible. For example, station SIP signaling conveys caller ID, which the VX can display and use. SIP trunks, on the other hand, do not pass CID. The VX supports the required SIP registration when you enter the authentication password on the show configuration pages.

By default, Asterisk uses SIP Reinvites. This causes the VX or VoIP phone to make a direct connection to the Telco, bypassing Asterisk's processing when compatible codecs are present at both ends. This is generally a good idea, but it can be troublesome because it forces the VX to change the IP address and other parameters for the line source. While Reinvites are supported in VX since April 2011, we have discovered an issue with Asterisk which can cause audio problems in some cases. If Asterisk is used as a PSTN gateway or connected to a telco, it should be fine, but if it is accessible from the public internet (ie. anyone with a softphone can call it), it is better to disable Reinvites. If one is not sure, better to be on the safe side. Until re-invites are supported, the Asterisk 'canreinvite' parameter should be set to 'no' (in later versions of Asterisk, those are two options: 'directmedia' and 'directrtpsetup')

(In case anybody is wondering - Asterisk doesn't re-negotiate the codecs. For example: VX supports g.711 and g.722. Let's say someone is calling from a softphone supporting g.711 and Speex. Asterisk supports all four, and will advertise it this way, transcoding if necessary. However, when it makes a direct connection, it doesn't change the codecs to those actually supported by each party. Thus, VX will end up thinking that the other end supports g.722, and the softphone will think that VX supports Speex - resulting in silence in both ways.)

When Asterisk and VX are on the same subnet, Asterisk's NAT (Network Address Translation) support should be set to 'no'.

It remains to be seen if, and how well, Asterisk will support wideband codecs. Until this is clarified, and when the VX supports wideband codecs, it might be required to keep Asterisk out of a wideband call path.

Let's get started…

Log in to the web GUI

Point your browser to your Asterisk server address, being sure that your computer is on the same subnet. For example, if your Asterisk box is on 192.168.0.248 with a netmask of 255.255.255.0 and your browser computer is on 192.168.0.11 or any IP with the same first three bytes 192.168.0 and the last byte between 2 and 254, accompanied by the same netmask as the Asterisk box's, you are good. Supply the login credentials and you should see the screen below:

Click on the
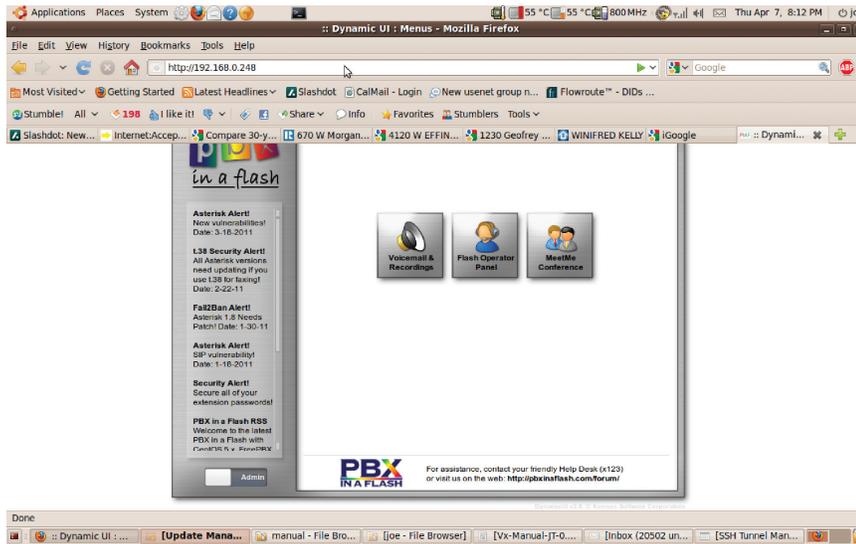
admin virtual slide switch at the bottom left of the page and the menu will change to the one below. You may be asked for your username and password to access this. Use 'wwwadmin' and the password that you set earlier, then click OK. You should see the screen below.
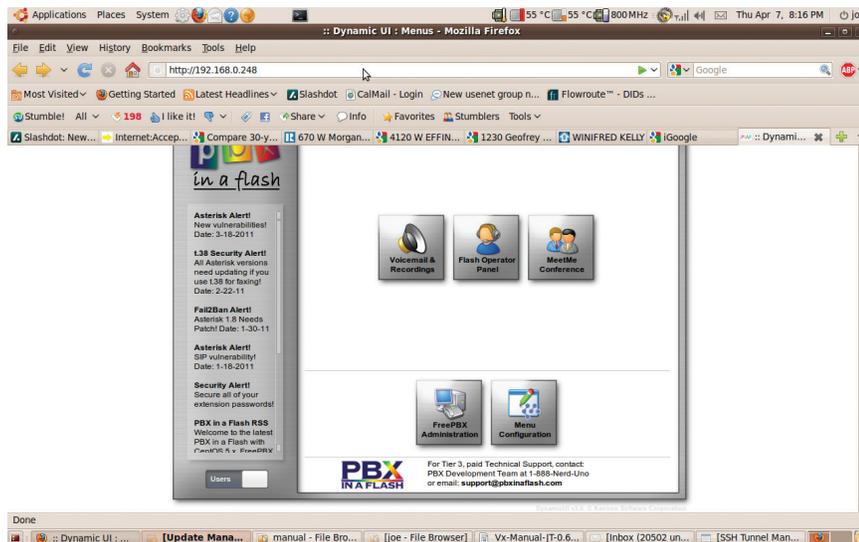
Use 'FreePBX' to do some administration chores

Click on FreePBX Administration and you'll see a new menu that supports almost all of the features you'll regularly use in Asterisk. You'll do most of your setup from this menu.

Here's what we're going to do next:

♦ Create an extension that we'll later use with the VX.

♦ Set up a trunk to accept incoming calls and to be used to make outgoing calls.

♦ Set up an 'inbound route' to tell incoming calls how to get to our extension on the VX.

♦ Set up an outbound route so that you can make calls out

Then we'll use the VX web interface to use the line that we created and then make a test call.

Finally, we'll create a trunk from the PSTN and route calls from that trunk to our extension, all by using the web GUI. We'll go a step at a time from this point.

Create an extension or two

To get comfortable with Asterisk, create an extension or two to use for testing. You can use the extension to test your connection with the VX and just to check for signs of life. You deserve to experience some action after all that work loading and configuring.

Click on **Extensions** and then click on **Add Extension**. The "Device field will default to **Generic SIP Device**. Accept this by clicking on the **Submit** button. Then, Fill in the extension number you'd like to use, then the display name and outbound caller ID. This is what callers will see when you call them from this extension. Interestingly, it can be just about anything. Some phone companies screen to be sure that you're sending a properly formatted area code and phone number, while others will take anything - a 4 digit extension, a single digit, your lucky lotto numbers, whatever.
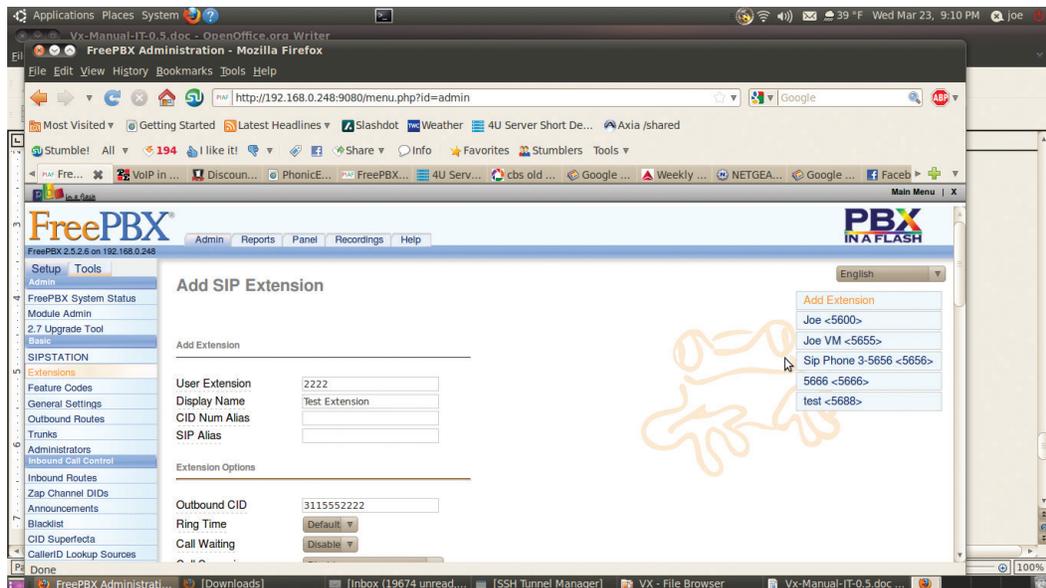
Scrolling down the page some more we see some more options. Most of these are optional, except for "secret". By default the 'nat' field is set to "Yes", be sure to change it for use with the VX or any extension to be used "inside" your firewall. This is another name for password. Scrolling further...
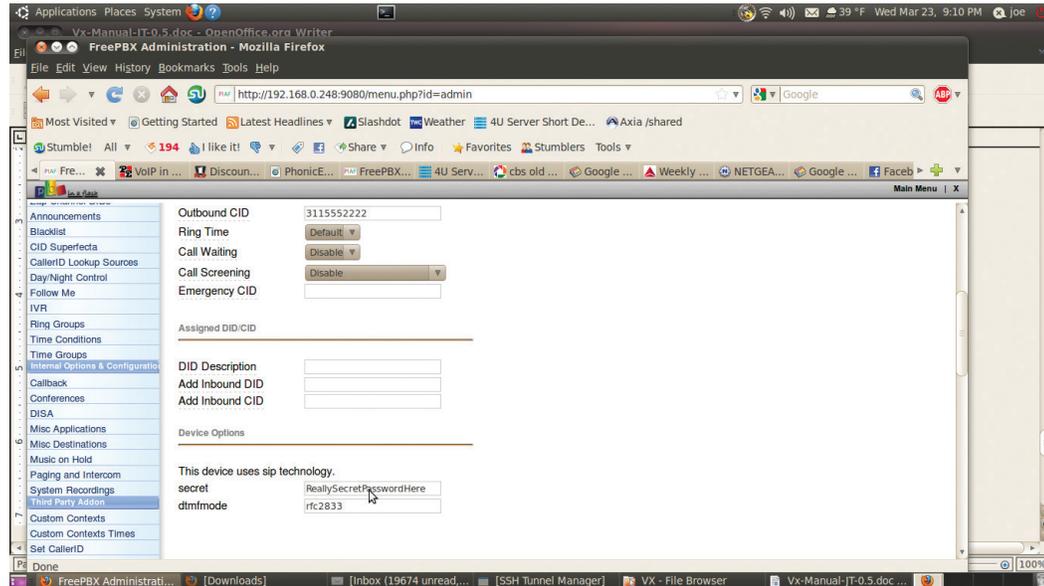
We see **Recording Options** and **Voice Mail settings** (You can record all calls or the extension user can dial a code while on the line to initiate recording of that call. Recording is optional, of course. Below the field marked **Secret** is another called **DTMFmode**, which should default to RFC2833. Leave this as is. The default is the preferred method of handling DTMF tone transmission. RFC2833 uses SIP messages to pass detected DTMF tones along to the other end of a call, rather than generating audio tones locally.

The Voicemail system in Asterisk offers all the usual capability. You could enable it on our about-to-be-created extension 2222 here so that you could see some of what it can do. Set the password. We set up this mailbox to automatically send us a wave file of any messages left to the email address in the field below. The audio quality of the recordings is impressive and usable for on-air playback. This is also optional. scrolling further down...
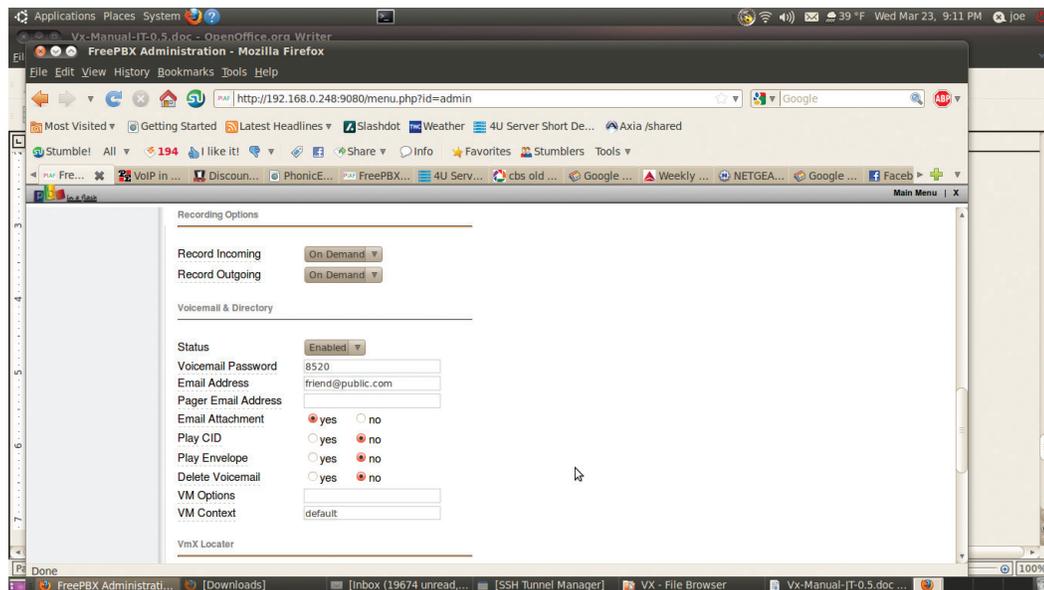
We are finally at the bottom of the page for extension 2222.
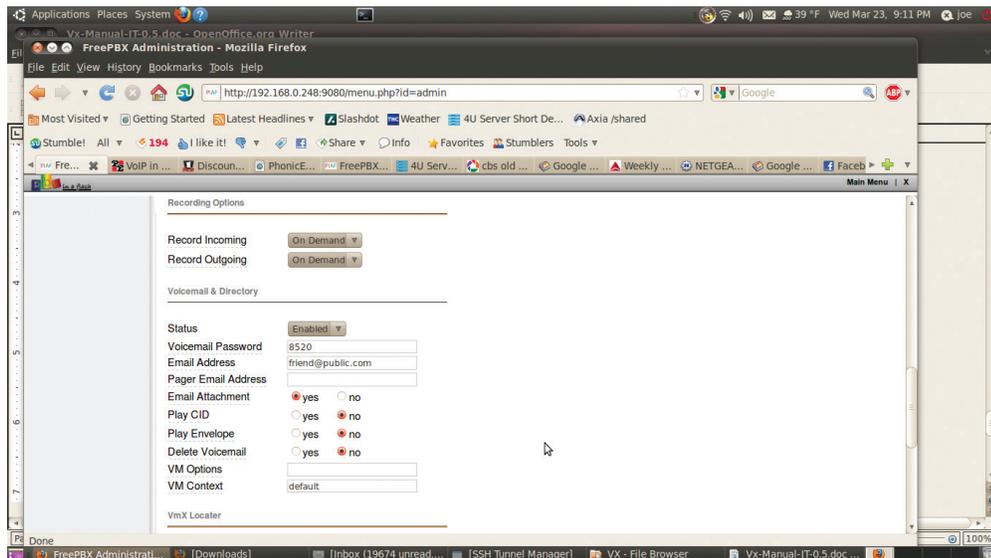
We enable the Vmx Locater. This is useful as it allows callers to be transferred to number inside or even outside of the system if 0, 1 or 2 is pressed. Here '0' is programmed to send callers to the station operator, '1' send the caller to our imaginary cell phone and '2' transfers the caller to our imaginary "personal assistant" at extension 5667. If you do use these options, be sure to mention pressing '0' for the operator in your outgoing voicemail message. You probably don't want to use these optional features for request lines or public call in numbers, but... they're there and work well if you have the need.

If you're happy with all of the above, click on the Submit button. The entry will be created and stored but not activated until you click on the orange bar at the top of the screen that appears when ever you have 'unapplied' changes. Don't worry, applying changes won't drop any calls or interrupt any conversations in progress.

Trunks

Since we now have an extension, let's make it so you can call somebody or get called. To do this, we'll need to way to and from the Public Switched Telephone Network or PSTN. The provider of SIP here was Vitelity, who provides "cut and paste" trunk set up data on their website for subscribers. Most providers do this. The next screenshot shows how Vitelity suggests that their trunk be set up.
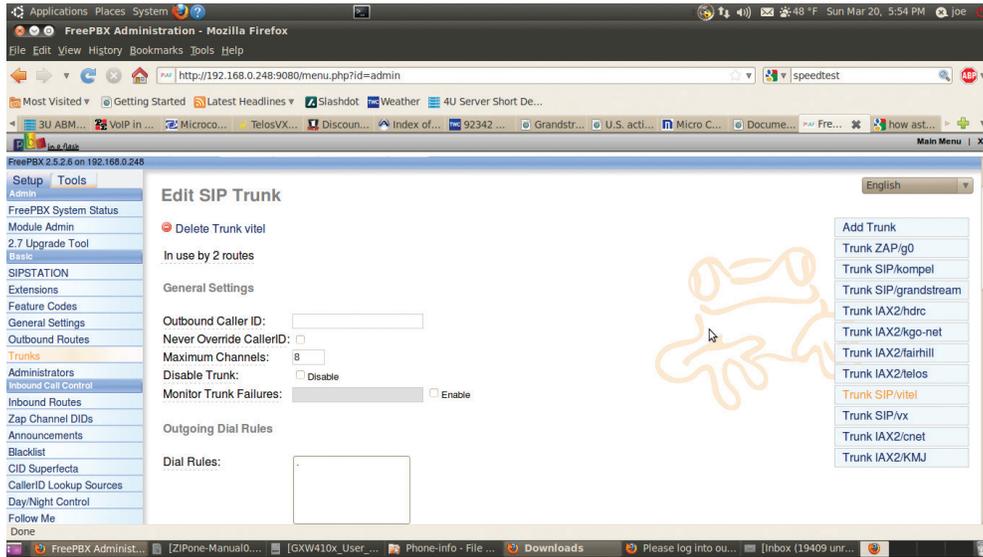
In our example here, we only use Vitelity for incoming calls, so the 'Outgoing Dial Rules' are irrelevant. We could insert a period in the Dial Rules box that would simply pass whatever digits were dialed down the trunk to Vitelity. Keep in mind that this is just the trunk setup, and that there still has to be inbound and outbound routes to get calls into and out of the switch.
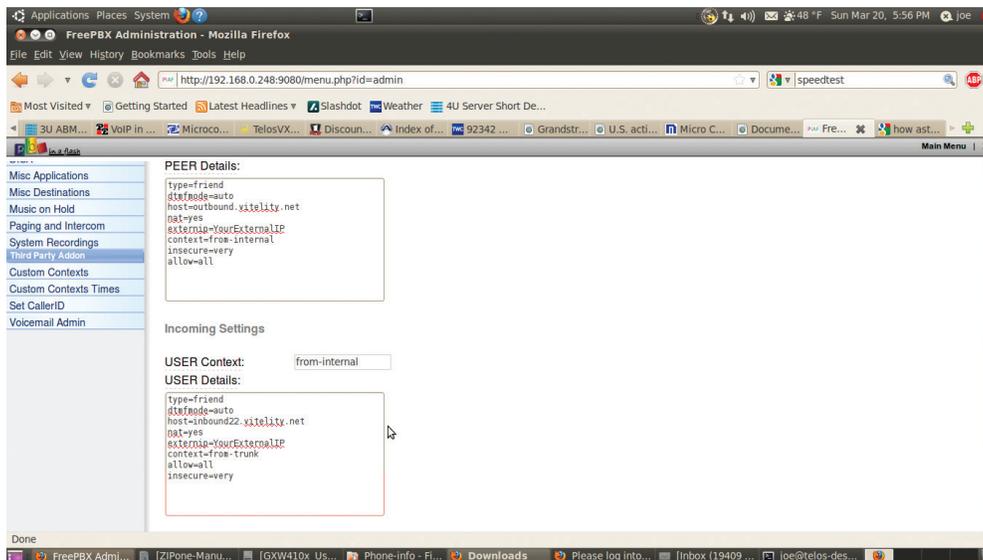
For incoming calls, this is done with an inbound route that matches digits sent be the provider, the Direct Inward Dial number itself and telling it "where to go", in our case, to an extension.
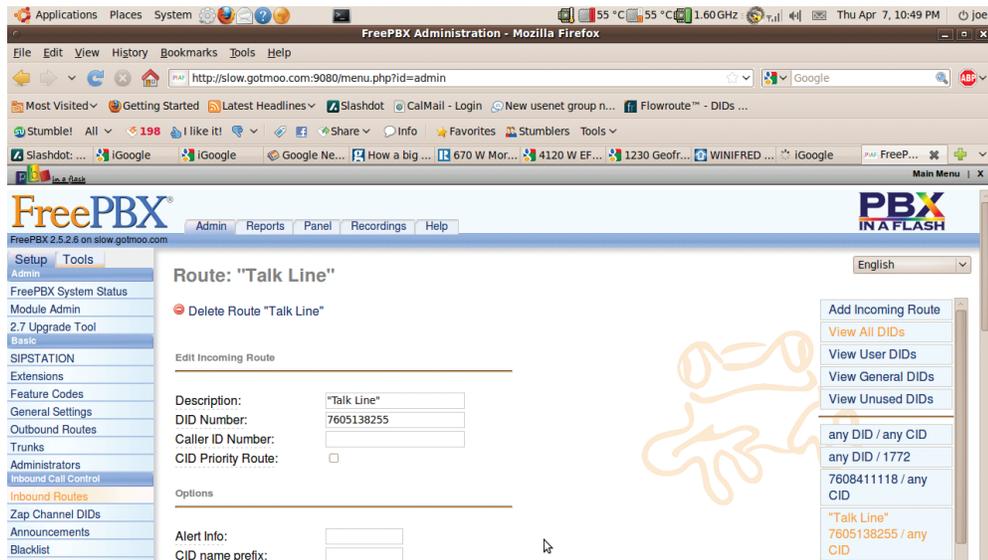
The DID number for our example "Talk Line" is 7605138255. We leave the Caller ID Number blank because we want it to work from ANY number, and entering a number would cause it to ONLY work from the number in that field. Rumor has it that a FreePBX programmer added this feature so that his "X Girlfriend" could get "special treatment", like being sent to a special 'ring forever' extension. We don't need this right now, but it could come in handy later (possibly depending on your personal situation). Leave the rest alone and scroll down further...
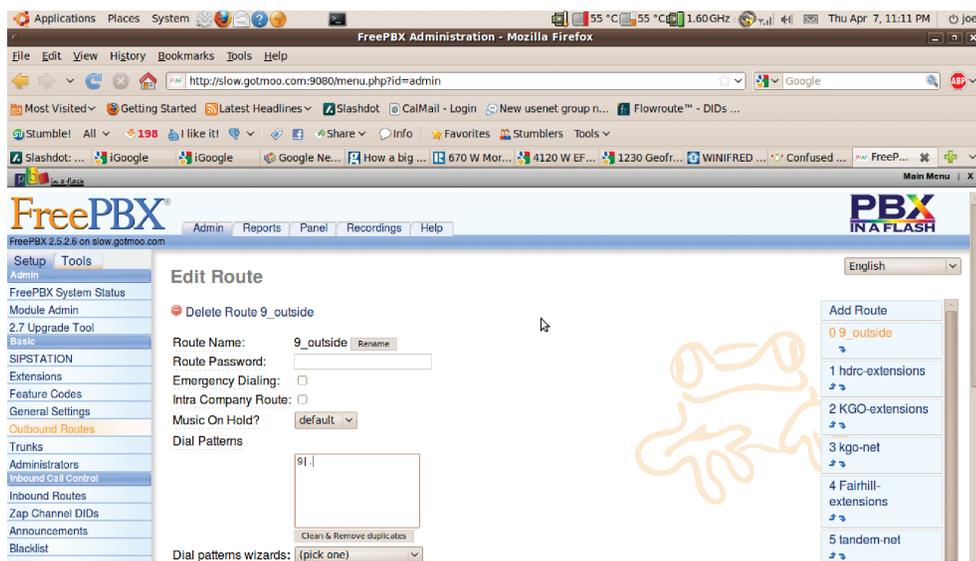
We are routing callers who dialed 7605130999 directly to Extension 2222 on the VX. Click **submit** and then apply the changes. We could also send those calls to any of the other options, such as "Misc Destinations: Mental Health Hotline" or "Joe's Voicemail" These destinations can be created and added by you in the FreePBX GUI.

Remember that if the show that uses extension 2222 is not selected on the VX (through the VSet or VX Producer software) calls to 2222 or the DID number routed to it will receive an "All

Circuits Busy" recording from Asterisk, because all circuits are busy as there is nowhere for the call to go.

Outbound Calls

Below, we decide to use '9' to dial outside the PBX, so we cleverly name the route '9_outside' to make it clear later what the thing does. In the "Dial Patterns' box we enter '9|.' What this does is match the leading 9 from the number dialed, let's call it 913115552368, deletes the 9 (everything ahead of the pipe symbol '|' is deleted) then the '.' represents the rest of the number dialed (13115552368) and passes that number to the trunk selected at the bottom of the outbound route page, in this case, our Vitelity trunk.

Connect your VX to Asterisk

We're going to assume that you've done the basic setup of your VX. That means you've got audio going and a VSet phone connected and talking to your VX engine, and it's on the same network as your Asterisk box.

Log into your VX engine with the web browser, and select SIP. Enter the IP address of the Asterisk in the *Server* field at the bottom of the page, and click *Add*. The configuration page for the server will open, where we can change the server settings and add new lines.

Earlier, we created extension 2222 using FreePBX. Now we need to add a corresponding line configuration for VX to register it with the server. *Expires* allows you to change how often VX will refresh it's registration. Leave it empty and the VX will register as often as Asterisk says it must. *Auth User* is typically your extension number and thus can be left empty as well, *Auth* Password should be the same as your 'secret' from the PBX in a Flash Extensions setup page.

Though the VX supports SIP trunks as well as SIP extensions (lines), extensions are prob- ably better for you to use than trunks. First, SIP lines each have their own outgoing caller ID, programmable per line in Asterisk. Trunks will only send the extension number for caller ID, or the same the caller ID for all of the numbers on the trunk, which can be confusing. People have become used to getting accurate caller ID on their mobile phones, etc. You may wish to hide or substitute caller ID on 'Private lines' or hotlines. Another feature of SIP lines is the registration feature. With it, the system knows that an extension is ready and available for calls.

After adding the line, you can assign the extension to line buttons in show configuration. Once the show is loaded, the information page on VX will show the current status of your lines and trunks. Notice below, that extension 2222 (and all of its multiple appearances) is OK and registered. The others say 'OK' and trunk. This is because the trunk is 'assumed' to be present. Extension 2222 has been verified as present.

Now we'll go to the VX *Shows* page and look at the show that we built earlier. It has a 6 line 'hunt group' for extension 2222 and a couple of "back lines" or hot lines. It's a 'hunt group' because there are 6 line keys set to extension 2222. Optionally, we could have the 'busy all' box checked only for extension 2222, our 'public caller' lines. Only these lines will be busied out when we press the "block all" key on the VSet or activate that feature from VX producer. In this example, that's what we did.

Make a test call on your VSet

Pick up a line using your VSet, and dial '1234' and press the GO button. If you are using PBX in a Flash, you should hear a congratulatory recorded message from the Asterisk. Then try dialing '2222' One of your 2222 lines should ring and you'll hear a ringback tone in the handset or on the studio output. Congratulations!

VX Tech Cheat sheet – Default IP numbers, passwords, etc.

Default Passwords:

VX Engine **username** = **user** , **password** = **<blank>**

VSet Telephone **username** = **user** , **password** = **<blank>**

Default IP Addresses:

VX Engine 'Network' interface = 192.168.2.200 (front panel settable)

VX Engine 'WAN' interface IP address: OFF by default, no IP set.

SIP Parameters

Codecs supported as of early 2011: g.711 alaw and ulaw. g.722.

SIP Trunks

SIP Stations

TCP/IP Ports and protocols used by the VX System

- ♦ 80 HTTP (TCP/Network Port only)
- ♦ 22 SSH (TCP)
- ♦ 5004 used internally by DSP engine (UDP)
- ♦ 5060 SIP (UDP)
- ♦ 8080 HTTP (TCP/WAN Port only)
- ♦ 20518 Livewire ports
- ♦ 62000 through 62513 RTP (UDP)
- ♦ Port numbers may be changed on the WAN interface for security and flexibility.

VSet telephone

VSet Telephone default IP address: 192.168.0.201 (front panel settable)

Default username and password:

username = **user**

password = **<blank>**

telnet requires no password

TCP/IP Ports and protocols used by the VSet Telephone:

80 HTTP (TCP/Network Port only)

23 Telnet (TCP)

20518 Livewire port

Power Over Ethernet (PoE)

The VSet uses standard 802. Power Over Ethernet (POE) and consumes 15.4 watts Please consider this relatively high power consumption rate when choosing your POE source.

VX FAQ

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., 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 1MB 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.

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.

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. The Not to worry – The VX has this capability. In fact, it has multiple oututs 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 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 600Ω 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. AAC-ELD, 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 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.

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 here: You could read Steve and Skip's AoIP book, too, for a fun and comprehensive coverage of this stuff.

Here's a paper on the adaptive IP codec: http://telos-systems.com/techtalk/Advanced%20 Tech%20for%20IP%20Remotes_Church_Final.pdf

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, here: http://www. 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.

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.

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.

Note that VX's call audio processing is bypassed when using g.722.

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!

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 news-rooms 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 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!

Fancy gear like this has to be trouble, no?

VX is simpler than a multiple-box approach. With fewer components, it's more reliable. At the time of this writing, no VX engine at a radio station has crashed, ever. We'll probably have a contest to see who has the longest 'uptime'. Right now, it's 6 months - but that system was installed 6 months ago…

Application Example - WKSU

WKSU is a non-commercial FM radio station affiliated with Kent State University. It features NPR, APM and PRI programming, classical music, regional news and, on weekends, folk music. The station serves the Akron and Cleveland radio markets with its main FM service and six repeaters The main FM and four full power repeaters each host four full-time HD Radio channels: a simulcast of the main service, a folk music channel, a classical service, and The WKSU News Channel. All of the programs are also streamed for Internet listeners.

The office and studio facility is housed in a stand-alone building at the edge of the Kent State University campus.

The two largest studios are equipped with Axia Element consoles. One of these is the main on-air control room, while the other is used for producing a variety of long-form programs, including an evening interview and call-in program.

The VX currently serves these two studios.

The main station call-in numbers are delivered via a SIP service offered by the VoIP provider bandwidth.com. IP network service is from the local Telco and is delivered over a T1 circuit, with the University's Internet connection being a backup. A router located on the WKSU premises is configured for automatic backup. 20 telephone numbers are delivered over the SIP service, which may be used for both incoming and outgoing calls. Despite the leading-edge nature of SIP Telco service, CE Chuck Poulton and network administrator Dan Kuzinsky ?? report that it has proven reliable and has delivered excellent audio quality, perhaps because call audio is maintained 4-wire (independent send and receive audio paths) and digital from the PSTN to the studio. The PSTN g.711 coding is maintained until it is converted to the high-fidelity Livewire format in the VX system.

Two analog lines from the local Telco connect to the studio IP system via a Grandstream 4-line-FXO gateway. Another two lines are brought into the gateway from the University's NEC PBX. An FXO interface card could have been installed into the Asterisk, but a dedicated external device was chosen as a measure to optimize audio quality. Fortunately, the NEC PBX provides "disconnect supervision" via a momentary break in loop current. The gateway was configured to respond to this. (if that hadn't been possible, the gateway would have to be configured to respond to the return of dialtone.)

In the other direction, two dialtones are provided from the VoIP side to the Nortel PBX via a Linksys two-line ATA (Analog terminal Adapter) – an FXS gateway that converts VoIP to analog, including talk battery, generating ringing, etc. WKSU has a number of phones on the PBX and this puts a couple of the VoIP numbers on phone buttons.

An Asterisk PC-based PBX is installed functionally "in front" of the VX to provide a basic automatic answer and IVR (Interactive Voice Response) function. Callers are greeted with a welcome message and a choice of leaving a message, going on the air, or speaking to the receptionist. When a caller chooses the option "go on the air," the call is passed to the VX, where it may either be directly answered on-air or be fielded by a producer. This IVR function is created through a web GUI for Asterisk called 'FreePBX'

Calls are routed from Asterisk to the VX via SIP/VoIP over the LAN. The Asterisk is configured to provide a SIP extension for each line that connects to the VX. (SIP trunks could have been used, but extensions are more capable - for example, they can pass Caller ID.

The Asterisk PC is equipped with three Ethernet ports. One is connected to the LAN and normally used for all input and output. Another is directly connected to the networks that deliver VoIP calls as a backup. pfsense, a software router, is installed on the machine to perform the routing to select the active VoIP path and provide the automatic backup.

The VX Producer application is used for call screening and communication between the producer and air talent.

Two Cisco 2960G Ethernet switches provide the networking. One is on the Livewire (LAN) side of the VX and the other is on the VoIP (WAN) side.

Incoming Call flow to VX via Asterisk Open Source PBX

DID numbers from SIP Provider
(311) 555-2368
(311) 555-9467
(311) 555-1067

DID Numbers from ISDN PRI
(311) 555-7200-7299
(311) 555-1300-1399

Numbers from POTS Gateway
(311) 950-1022
(311) 976-1234
(311) 853-1212
(311) 958-1114

Asterisk Switch

Asterisk SIP Trunk
Digits sent include
Caller ID and 10
Digit DID number

Asterisk Zaptel or
DAHDI type trunk
Digits sent include
Caller ID and 4-10
Digit DID number

Asterisk SIP or IAX2
type trunk. Digits sent
include Caller ID and
extension number that
calls to POTS line are
"auto forwarded" to.

Asterisk Extensions
May be SIP or ZAPTEL
or DAHDI hardware such
as expansion cards or
a Channel bank. Most often
used to create analog
extensions.

Asterisk Trunk or trunks:
May be SIP, IAX2, ZAPTEL
or Dahdi Hardware such as
expansion cards.

Vx Engine or
Telephone sets

Vx Engine or
trunks to other
PBX's

Example Call:
(311) 555-2368 called from a
cell phone in Barstow, California
(760) 256-8463

(311) 555-2368 with
Caller ID of (760) 256-
8463 is received.

(311) 555-2368 is routed
to extension 2368 on the
Asterisk switch. All info sent
is passed onto the Vx.

Ext 2368 rings
and shows
(760) 256-8463
as calling number
on Vx.

Cell carrier routes the call to the SIP
provider over the Public Switched
Telephone Network as the number
translates to the SIP provider.

The SIP provider sends
the call to the Asterisk SIP
trunk because 555-2368
is registered to the trunk
belonging to the subscriber.
Caller ID and the 10 digit
DID number are sent to the
Asterisk SIP trunk belonging
to the Subscriber.

The Asterisk switch receives
the digits and attempts to
"match" the DID number to
an "Incoming route". If matched
the call is sent further to either
and extension, trunk or special
treatment in the system, such as
voice mail or a busy signal if there
are no free trunks or buttons at the
far end.

The Vx receives 2368
and ringing commences
while the Caller ID data
is displayed along with
the ringing indication.
A "ringing line" lamp
flashes, triggered by a
GPIO interface as set in
the Vx engine "Studio"
definition.

6   Telos Alliance Warranty

Telos Alliance Limited Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty.

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