



Universal Service Node Product Information Folder Release 9.1

Improving Productivity
Through Effective Communications



We develop and market next generation value added service platforms for small, medium and large enterprise and service providers—worldwide. Our Universal Service Node product supports Audio Conferencing, Web Collaboration, Desktop Video Conferencing, Enhanced Firebar/Dial-out Ring down conferencing, Command & Control Conferencing, Hoot-n-Holler Conferencing, Mass Notification, Enhanced Voicemail and Remote Control applications. All of these applications are accessible over legacy TDM and next generation VoIP networks. Our products serve a variety of customers including Service Providers, Independent Telcos, Military establishments, City/County governments and small, medium large Enterprises. Our applications support the needs of three major verticals, these are Digital Collaboration, Public Safety and Financial Services.

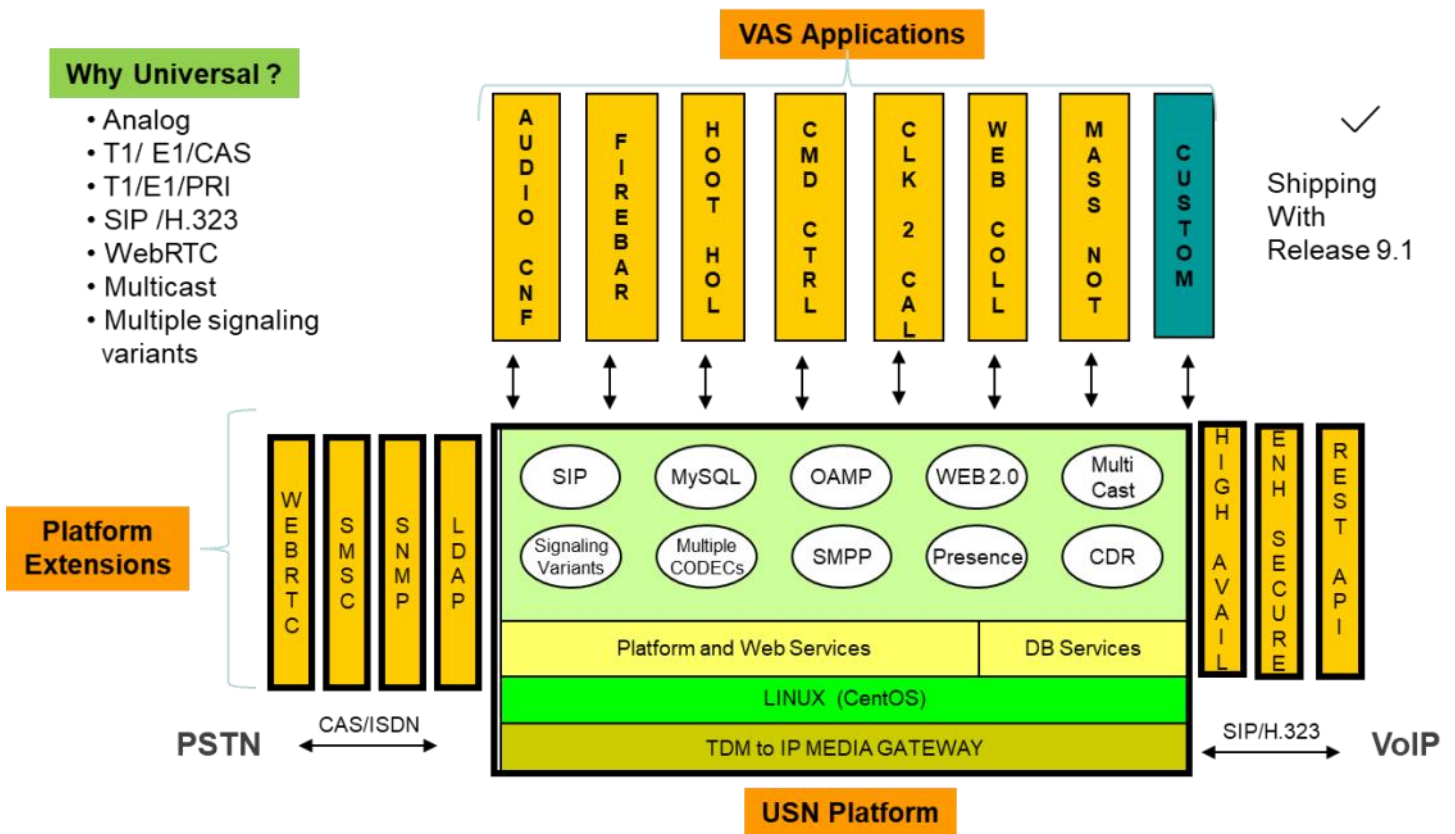
Major Customers include:





Universal Service Node - Architecture

The Universal Service Node (USN) enables improved productivity by providing multiple value added service applications on a single platform. The primary benefit of this multi-application platform approach being savings of 75% on CapEx and 50% on OpEx for our customers. The USNs are based on the Linux operating system and hence are extremely robust and resilient. The USN software can be installed on physical servers and on virtual machines. Based on customer's need, the USN can be shipped with TDM only, VoIP only or TDM and VoIP mixed mode configurations. This architectural flexibility facilitates an organization's migration from the legacy circuit switched TDM environment to next generation packet switched environment.

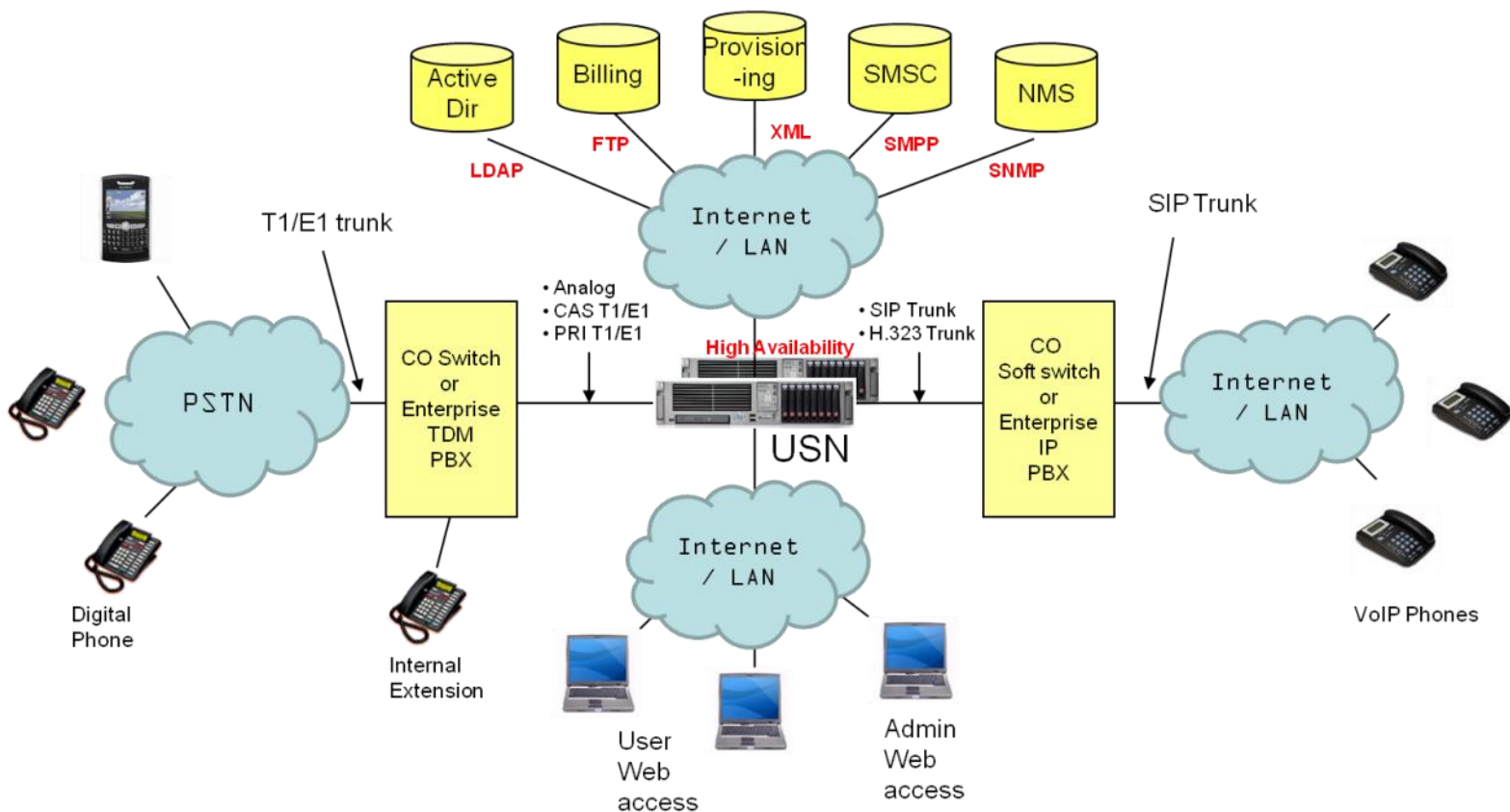


The USN Platform, Platform Extensions, and VAS Applications are licensed separately. TDM and VoIP ports can be shared dynamically across multiple applications or segmented per application.

A USN can be shipped with any one or more applications. Each application is individually licensed. All applications share the available TDM and IP ports dynamically thereby maximizing the use of network resources. Customers can also purchase Platform Extensions as needed.

Universal Service Node - Deployment

The Universal Service Node (USN) supports practically all relevant voice signaling interfaces on the TDM and well as the VoIP side. The product can be deployed in North America as well as in other countries that follow CCITT/European signaling standards. In addition, the USN supports LDAP for authentication via Microsoft Active Directory, XML based API for integration with external CRM/Provisioning systems, SMPP 3.4 interface for integration with carrier SMSC gateways and SNMP protocol for integration with external Network Management Systems. The product is typically deployed behind an organization's Firewall and can be accessed via any computer that supports Chrome, Firefox, Microsoft Edge or other browsers over a secure HTTPS data connection. The product supports a number of Information Assurance features that make it suitable deployment in secure environments.



Multiple signaling interface types allow product to be deployed in TDM or VoIP or hybrid networks. Variety of interfaces available for Back Office Integration.



1U Mini-server
 • 30 VoIP ports
 External Analog to SIP gateways



1U server
 • 4 - 12 analog ports
 • 24/30 – 96/120 T1/E1 DS0 ports
 • 240 VoIP ports



2U server
 • 8 – 16 analog ports
 • 24/30 – 192/240 T1/E1 ports
 • 480 VoIP ports



4U server
 • 48/60 – 960/1200 T1/E1 DS0 ports
 • 960 VoIP ports



8U server
 • 8000 TDM/DS3/E3 DS0 ports
 via External Media Gateway
 • 16,000 VoIP ports
 • 1000 VoIP ports per slot

- Industrial grade servers
 - 1, 2, 4 or 8 Rack Units high, 19" wide rack mountable
- Scalable port capacity
 - 4 ports to 8000 ports in one server
- Flexible product options
 - TDM only mode
 - TDM and SIP/VoIP hybrid mode
 - SIP/VoIP only mode
- High Availability
 - Real time database replication
 - 1:1 Hot and warm standby configuration
 - 1:1 Load Shared configuration
- Robust
 - RAID-1 Mirrored Hard drives
 - Redundant AC/DC power supplies
- Linux Operating System
- NEBS/CE compliant (optional)

Product purpose built to suit the needs of Small, Medium and Large Enterprises and Service Providers.

Port Configurations

The number of ports can be any mix of TDM and VoIP channels.

TDM Interfaces

Analog FXO and FXS
 T1/E1/J1
 DS3 /E3 (N+1 redundant)

IP Interfaces

VoIP Interface 100BaseT, GigE Ethernet, RJ-45
 Encoding formats G.711, G.729a/b, G.723, G.722
 DTMF Relay: Inband, RFC2833, SIP Info

TDM Protocols

Analog: FXO/FXS Loop Start
 T1: CAS E&M (Wink Start, Immediate Start), MF, DTMF
 T1: ISDN NI-2, 4ESS, 5ESS, DMS250, INS1500, Q.Sig
 E1: CAS Many country specific MFC-R2 variants
 E1: Euro ISDN, NET5, DPNSS, DASS32, QSIG

IP Protocols

SIP: RFC2543 and RFC 3261 (partial)
 H.323 V2: H225.0, Q.931, H.245

Hardware Specifications

1U, 2U, 4U or 8U standard 19" rack mountable industrial grade chassis or NEBS complaint chassis
 SCSI/SATA RAID 1 Mirrored Disks, 36 to 144 GB
 Power: CES complaint, Redundant power
 110-240 VAC, 47-63 Hz, 600 Watts max
 -48 V DC power supply (optional)
 Weight range: 40 - 100 Kg

Network Management

SNMP MIBs for external NMS integration
 Automated health check and reporting

Usage Measurements

CDRs over TCP/IP
 Summary reports, analytics

Operating Ranges

Operating Temperature + 0 to 50 deg Celsius
 Storage Temperature -20 to 70 deg Celsius
 Humidity 8% to 80% non-condensing

Warranty and Technical Support

Hardware & Software Warranty: One Year Included
 Technical Support, M-F, 8-5 CST: One Year Included
 Extended Hardware and Software Warranty: Available
 Technical Support, 24 x 7: Available

USN Software can be deployed on VMware based Virtual Machines.



Audio Conferencing Application



- Get rid of monthly Audio Conferencing billing.
- Cut down on unnecessary travel.
- Make your meeting more productive by combining with Web Conferencing.
- Record Audio and Web conference sessions and play on any media player.

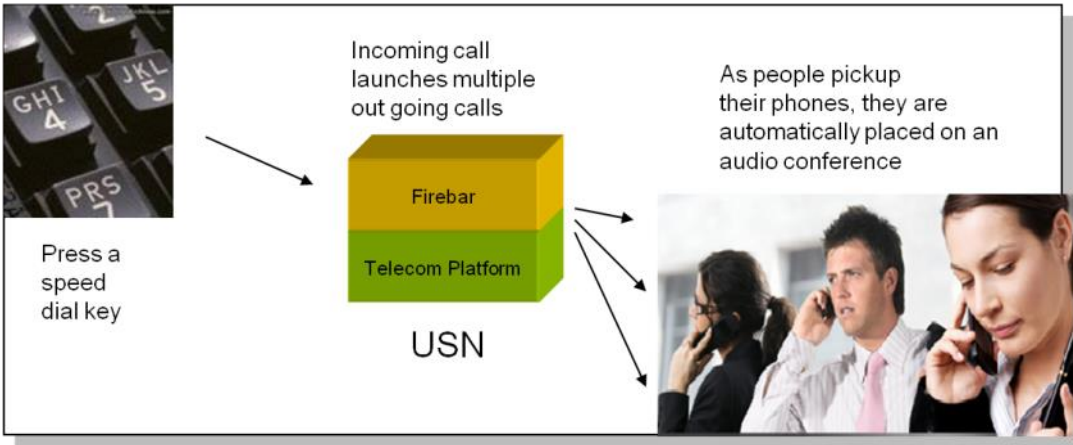
Audio Conferencing Application for the Universal Service Node provides rich conferencing experience for demanding users.

- Support both reservation less and reservation based conferences.
- Customize conference rooms per your requirements, e.g., specify entry tones, memorable vanity PINs, recording on/off, noise thresholds, participant specific security pins, etc.
- Schedule recurring audio conferences via the Web Portal. Use Microsoft Outlook iCalendar application to send invitations to desired participants.
- See real time view of a running conference via Web Portal. Participants can be seen by name or caller ID. Display Loudest speaker.
- Exercise multiple in-conference controls via phone key presses or the Web portal.
- Merge two or more conferences into one without dropping any calls. Transfer participants between conferences.
- After a conference is over, a detailed end of conference summary report is sent and conference recording is posted in the moderators account.
- Usage data and associated CDRs can be forwarded to an external billing system via TCP/IP based interface.
- Download and playback recordings on any media player

FEATURE	HOW IT WORKS	BENEFIT
Ad-hoc 'Meet Me' conference.	Moderator and participants agree upon a start time and a PIN to use. When people dial in and enter their PIN, they are placed in the conference.	Simple to use. No/little training required
Scheduled 'Meet Me' with PIN conference.	Schedule meetings using Moderator portal. Send meeting invites using iCAL.	Use your existing Outlook 'contacts'. Calendar will automatically remind participants about upcoming conference.
Dialed number (DNIS) based conference.	Multiple participants simply dial a phone number and join a conference.	No PINs to remember.
Progressive dial out conference.	Moderator can dial out from the bridge and bring participants into a conference one by one.	Impromptu conferencing, no need to inform participants ahead of time.
Instantaneous Dial out with 'Find-you' conference. (with Firebar option)	Incoming call triggers a dial out conference. Bridge will call participants at their multiple locations and connect them into a conference.	Communicate with a 'group' with a single key press.
Scheduled Dial out with 'Find-you' conference.	At a scheduled time, bridge will trigger a dial out conference.	Reduces excuses for not joining a conference.



Firebar Ring-Down Conference Application



- Establish an audio conference with press of a 'speed dial' key.
- Allow first responders to be reached over their land lines and/or cell phones.
- Increase probability of attendance based on built-in 'find you' capability.
- Besides establishing an audio conference, send emails, SMS and Pager messages to first responders.

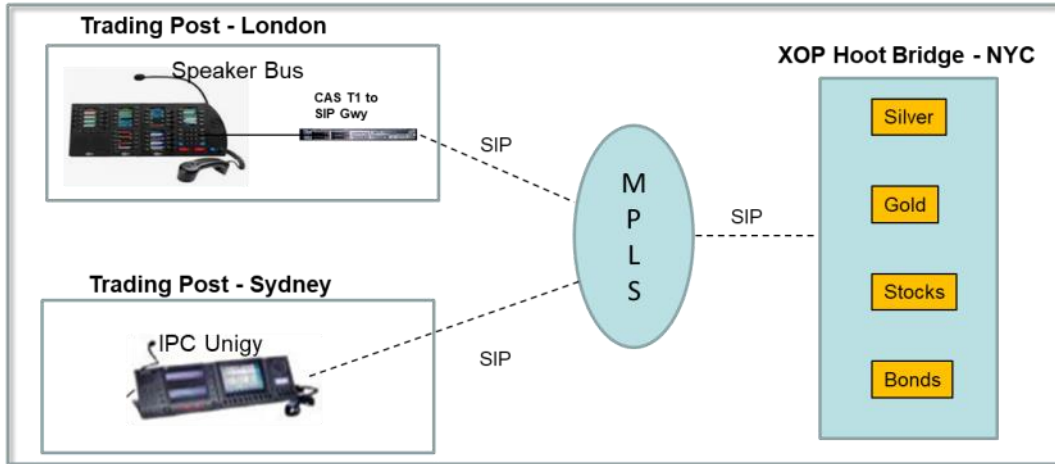
Traditional Firebar or Ring-Down dial out conferencing is used by emergency dispatch personnel to inform and bring a group of first responders into an audio conference quickly. These systems require dedicated phone lines that terminate on the central office switch of a phone company. XOP Networks' Firebar application resides on the USN that is deployed behind your on premises PBX. It provides several enhancements.

- Send calls to any landline or cellular phone instead of just dedicated 'red' or emergency phones.
- Select communication medium to be used for message delivery (Voice only, Email only, Voice and SMS, etc.)
- Use built-in 'Find-you' capability to increase the probability of finding a recipient.
- Send Caller-ID of your choice that can be used by cell phones to display associated 'caller name' (e.g., Fire Chief ') - leading to higher percentage of recipients picking up the phone.
- Display real time call activity on a Web Portal.
- Schedule recurring dial-out calls.
- Provide summary and detailed reports on call completions (Busy, No Answer, Answering machine etc.)
- Record all emergency conference calls if necessary

FEATURE	HOW IT WORKS	BENEFITS
Emergency specific groups	Set-up via Web Portal. Use CSV file upload to create multiple groups quickly. Sync your groups via LDAP based Active Directory	Pre-planned group members may belong to different organizations (Fire, Police, EMS etc.).
Blast Dial capability	Trigger the dial-out based on incoming phone call, click on a Web Portal, closure of a relay, (optional) or at a scheduled time.	Multiple first responders are called in parallel - reduces the overall time required to contact. Routine testing can be automated
Find-you capability	System dials up to 4 phone numbers and sends SMS and emails when locating an individual recipient.	Improves the probability of reaching an individual first responder
Secure audio conferencing	Allow responders to join the conference after they enter a security key.	Prevents unauthorized participants to enter an emergency conference.
Call logging	Record all Firebar calls and capture call logs with time stamps.	Useful for post-event analysis
Send SMS, Email and Pager messages	Set-up via Web Portal. Send SMS, Email and Pager blasts with or without accompanying voice calls.	Helps in disseminating emergency related information in multiple ways.



Hoot and Holler Conferencing Application



- Enter a Hoot conference room simply by pressing a button on a Turret.
- Support ARD and MRD calls between legacy and soft Turrets.
- Support signaling interworking between disparate turret types.
- Easily troubleshoot audio issues on Private Wire networks.

Hoot and Holler conferences are typically used in Trader Voice or Private Wire networks. These networks are typically used by stock traders, energy traders, auto parts traders etc. A user simply presses a button on a Turret or Squawk box for a specific 'hoot' room and joins that hoot conference room instantaneously. A hoot conference can last for an entire day of trading or for a multiple days. No phone numbers need to be dialed for using hoot and holler conferences.

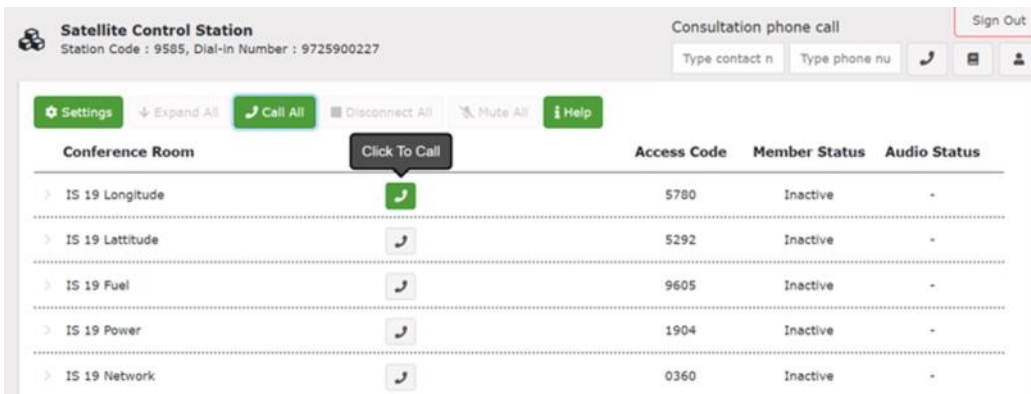
- Support Hoot and Holler conferencing across TDM and VoIP networks
- Support Automatic ring Down (ARD) and Manual Ring Down (MRD) for Financial Services
- Provide signaling interworking between Turrets made by different manufacturers (IPC, Etrali, Speakerbus, British Telecom)
- Provide interworking with on-prem IP PBXs (Cisco CUCM, Avaya Aura etc.) for hoot calls
- Record all or selected hoot rooms
- Real time view of hoot calls for troubleshooting call legs with audio issues
- Provides usage reports and call logs

FEATURE	HOW IT WORKS	BENEFITS
Hoot Conference	Hoot conference rooms are 'always on' statically configured conferences. A caller is placed in a given hoot room based on the button depressed on a Turret or off-hook on a associated DS0 channel.	No dial in is required to join a hoot conference. Traders can switch between hoot rooms simply by depressing different buttons on their Turrets. Enables quick conversation between a group of traders.
Automatic Ring Down (ARD)	Trader A depresses a Button Labeled Trader B on his/her Turret. The ARD call causes a Ring signal to be sent to Trader B. Trader B goes off-hook and the two traders can now talk.	Creates a hotline between two traders. No phone number needs to be dialed. Allows quick consultation between traders.
Manual Ring Down (MRD)	Trader A needs to draw the attention of Trader B when they are already in a call. Trader A presses 'MRD' button on his/her turret. This causes an alerting ring signal to be superimposed on top of the existing voice path between two traders.	Allows a Trader to get the attention of a far end trader during an on-going call. ARD/MRD capabilities become important during execution of stock trades on Wall Street.
Signaling Interworking between disparate Turrets	Multiple manufacturers of Turrets use signaling e.g., A/B bits on CAS T1s, SIP INFO on SIP trunks differently. XOP bridges provide necessary interworking to allow seamless voice communication between disparate Turrets.	Trader Voice/ Private Wire network operators can leverage equipment from different manufacturers.



XOP Networks

Command & Control Conferencing Appl.



- Visually see presence of people in different conference rooms.
- Send PC audio to multiple conference rooms simultaneously.
- Listen from one or more selected rooms simultaneously
- Consult with people by dialing over the PSTN
- Bring consultants into on-going conferences

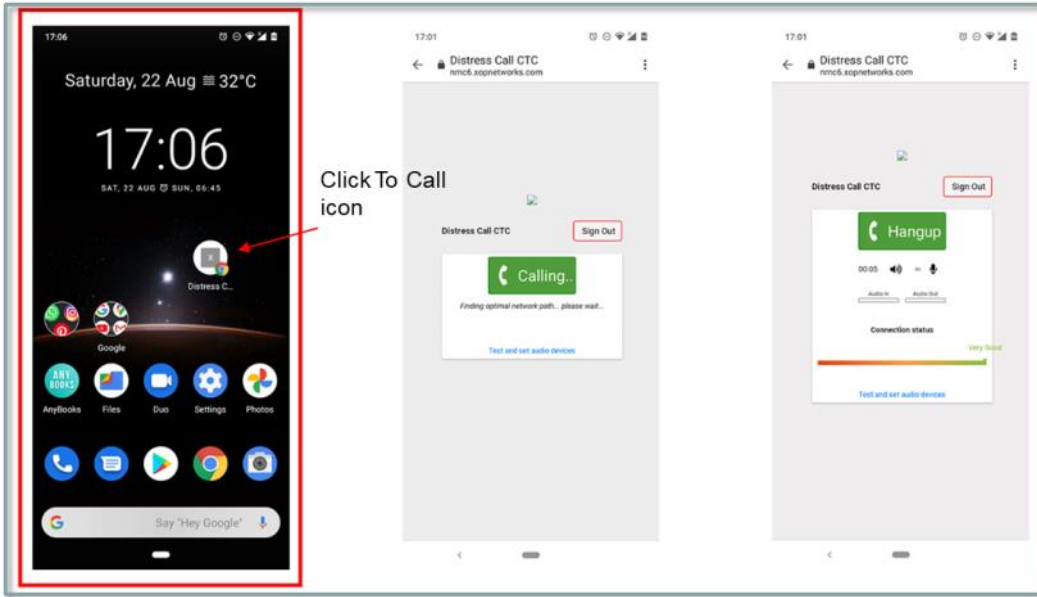
Command and Control Conferences are typically used when a Supervisor needs to manage the activities of multiple groups of people. The USN Command and Control Conferencing application provides such capabilities over a web portal. In the example shown a Satellite Launch Mission Director can simply click on the handset button and barge into one or more rooms. The manager can also selectively listen from one or more conference rooms. This application is WebRTC based and hence requires use of modern browsers (Chrome, Firefox, Edge).

- Using the web portal a Supervisor can
 - See the 'presence' status of each participant in a given conference room
 - Visually see the 'active speaker'
 - Speak into multiple conference rooms simultaneously
 - Barge into a given conference room
 - Monitor and listen to audio from one or more conference rooms simultaneously
- Dial-out to external participants for consultation and drop the called party into a given conference room if needed
- Ability to make separate audio recordings - at the individual room level and at the Supervisor level
- Provides usage reporting and call logs.

FEATURE	HOW IT WORKS	BENEFITS
WebRTC based Command and Control Portal	Moderator/Supervisor can use Chrome or Firefox based portal to access the command and control portal. Use PCs microphone and speaker to interact with people in various groups being controlled.	No download required. Can be accessed remotely over the Internet. Participants can join conferences using WebRTC/VoIP audio or dial-in over PSTN.
Multiple simultaneous portals per Moderator	A Moderator can set up multiple sessions and view them on separate tabs on the browser	Allows a Moderator to run multiple Command and Control sessions on one computer
Visual indicator showing status of participants in a given room	Moderator and Participants can detect voice activity in a given room based on loudest speaker display and then drill down for further interaction	Supervisor can quickly ascertain the status of a meeting in a given room.
Built-in Dial out capability	Moderator can dial out to PSTN number and pull the external party into a given room.	Adds flexibility to the overall operation.



Click to Call Application



- Press a button on your mobile phone and join an ongoing conference.
- Press a button on your mobile phone and launch an emergency dial-out conference with your friends and family.
- Avoid toll charges. CTC call can be placed from anywhere you have access to Internet.
- Use CTC for reaching out to people over the PSTN.

Click to Call application allows a moderator to send a WebRTC based voice call from a browser to the USN. Based on source and destination number assigned to the call, the caller can be placed into an ongoing Meet-me conference, Hoot-n-Holler conference or can be used to trigger a Ringdown Firebar conference or a trigger a Mass notification session. This capability can be used with PCs, MacBooks, Android and iOS based cell phones and tablets. The application requires use of Chrome, Firefox, and Safari browser.

The Click to Call icon can be placed on a PC's desktop or a mobile phone's icon screen. Moderator can create and download multiple icons for use with different applications. When clicked the icon automatically engages the default browser for its operation.

XOP Networks also provides Click to Call code snippet that can be used to install the CTC icon on third party portals/ websites. Once integrated, these portals can allow voice calls to be placed to designated individuals (e.g. Click [Here](#) to Call Support).

FEATURE	HOW IT WORKS	BENEFITS
Multiple Moderators mode	Multiple Moderators can simultaneously monitor audio from one or more rooms and talk and listen to selected room simply from their respective browsers.	No download required. Can be accessed remotely over the Internet. Participants can join conferences using WebRTC/VoIP audio or dial-in over PSTN.
Multiple simultaneous portals per Moderator	A Moderator can set up multiple sessions and view them on separate tabs on the browser	Allows a Moderator to run multiple Command and Control sessions on one computer
Visual indicator showing status of participants in a given room	Moderator and Participants can detect voice activity in a given room based on loudest speaker display and then drill down for further interaction	Supervisor can quickly ascertain the status of a meeting in a given room.
Built-in Dial out capability	Moderator can dial out to PSTN number and pull the external party into a given room.	Adds flexibility to the overall operation.



Web Collaboration Application



- Get rid of monthly Web conferencing bill.
- Cut down unnecessary travel, Work from Home.
- Conduct Web Collaboration in your own secure network.
- Make your meeting more productive by combining with Audio Conferencing.
- Use PC Webcams for video conferencing.

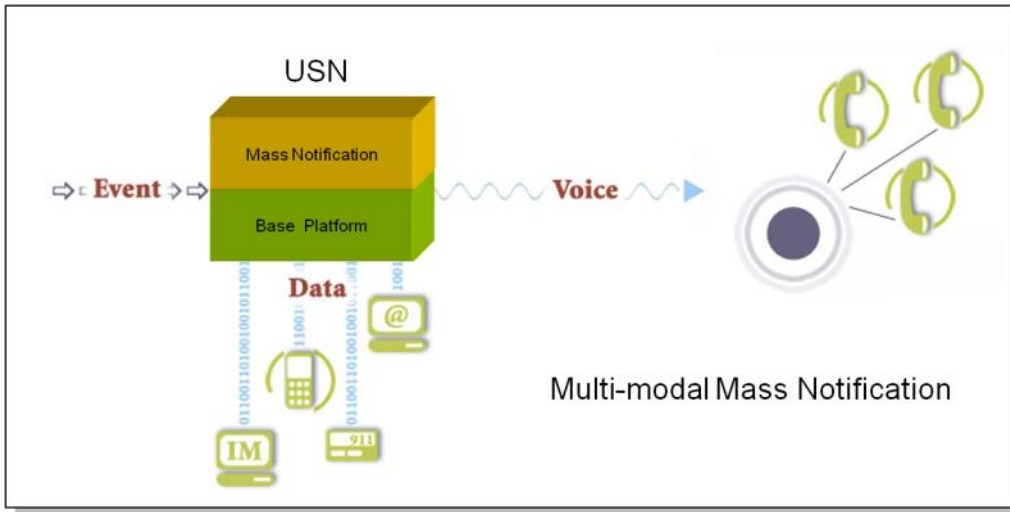
Web Collaboration Application is designed to significantly boost the productivity of your meetings.

- Brandable Portal, deploy On-prem or on the Cloud
- Web based application, no software download required to your PC.
- Browser agnostic - works with Chrome, Firefox, Microsoft Edge, Safari
- Usable across Windows PC, Apple MacBook, Android phone and iPhone
- Join Audio Conference by dialing via PSTN or using Computer Audio using WebRTC
- Call Me capability for Application to call participants
- Detachable windows based design, support two or more monitors
- Multiple Information Assurance capabilities to keep your meetings secure
- Onboarding Wizard for testing microphone, speaker and webcams

FEATURE	HOW IT WORKS	BENEFIT
Desk Top Sharing	Moderator shares his/her Desktop with fellow participants.	Show any document or co-browse the Web with fellow participants. Simple to use and ideal for product demos.
Application Sharing	Moderator can share specific application even though there may be multiple applications running on the desktop.	Participants do not see Moderator's entire desktop, also saves on bandwidth used.
Multiple Presenter Control	Moderator can allow another participant to take control and share his/ her desktop.	Multiple points of view on one conference.
Desktop Video Conferencing	Moderator and Participants can share webcam video streams. Gallery View and loudest speaker view available.	Creates more authentic meeting experience between members.
White Boarding Mode	Create diagrams/visuals with fellow participants in a collaborative session.	Ideal for brainstorming, remote consultations, distance learning
Public & private Chat Room	Moderator can respond to questions in public or privately. Participants can upload /download documents to be reviewed during meeting.	Makes Web conferencing more productive.
Webinar mode	Moderator can stream his/her video, and specific application to participants. Participants can ask questions in private.	Great tool for product introductions, training etc.
Record and Playback	Moderator record Audio + Video + desktop sharing sessions. The resulting mp4 file can be played on any media player.	Very useful for creating training videos, classroom sessions etc.



Mass Notification Application



- Get rid of your outsourced messaging service billing.
- Add Mass Notification capability to your existing USN to improve its value proposition.
- Keep large number of people well informed about unfolding events.
- Use conferencing in conjunction with Group Alerting to pull people into quick conference as needed.

Mass Notification Application for the Universal Service Node is designed to send multi-modal messages to thousands of people during emergency and non-emergency situations.

- Web portal for managing multiple call-out groups, re-usable Mass Notification sessions
- Select communication medium to be used for message delivery (Voice only, Email only, Voice and SMS, or All)
- Deliver voice message to up to 4 phone numbers using built-in 'Find-you' capability.
- Send Caller-ID of your choice that can be used by cell phones to display associated 'caller name' (e.g., Security Alert) - leading to higher percentage of people picking up a message.
- Schedule recurring dial outs.
- Control the speed of dialing out.
- Display real time call activity and a progress bar on a Web Portal.
- Use secure Announcement box for disseminating frequently changing info, e.g. related to cyclones, power ages etc.)
- Use outgoing voice calls to enable one or more dial-out conferences to pull in appropriate first responders into dial-out conferences.
- Provide detailed reports on call completions (Busy, No Answer, Answering machine etc.)

FEATURE	HOW IT WORKS	BENEFIT
Pre-recorded message delivery	Pro-actively build call out groups. Pre-record messages. Then tie groups and messages into Group Alert sessions. Trigger dial out from Web Portal or with incoming phone call.	Make messaging a planned activity. No need to search for address books at the time of actual need.
On-the-fly Message Delivery	Dial into the server, enter a PIN, record/re-record a message and send.	Quick dissemination of emergency oriented messages. No need to access a computer.
Built-in 'Find-You' capability	System captures up to four phone numbers per individual and dials them successively until making a positive contact.	Increases probability of delivering a messages.
Announcement Box capability	Moderator periodically dials in and records a message in an announcement box. People can call in and hear the updated message. Use playback controls to review long messages	Great way to inform people during changing emergency situations such as hurricanes, blackouts etc. Record and playback Podcasts, sermons etc.
Iteratively contact the un-contacted	Set up Group Alert with 'un-contacted' option. Iteratively send Group Alert message to 'un-contacted' 'n' number of times.	No wasted calls. With every pass the list of un-contacted shrivels. Reduces manual handling of Group Alerts.
Send message to 'contacted' people	Use 'swap' to convert contacted into un-contacted and send a new message.	Only people who received a previous message will get the new message. Great way to send 'all clear' message.



XOP Networks

Enhanced Voicemail Application



- Share a common Voicemail platform across multiple TDM and VoIP switches/PBXs.
- Use it as 'Voicemail Central'. Share the same mailbox across multiple subscriber phones.
- Get Voicemails using 'patented' non-sequential access approach. Access voicemails through popular PDAs.
- No change to end user experience due to emulation of legacy Voicemail IVRs.

Current voicemail systems only allow sequential access to voicemails. XOP Networks' Enhanced Voicemail Application allows stored voicemail messages to be accessed in random order from a Web portal. The Web portal can be accessed via Mobile phones (Android and iPhone), a PC or via a number of PDAs. By a simple click, a user can hear the stored message through the built in media player.

- From 100 to 10,000 Voicemail boxes per chassis.
- One Number service - specify up to 6 numbers and their order based on ToD or DoW that will be tried before a caller is taken to a voicemail box
- Subscriber Web portal for managing the Voicemail configuration, One Number Service, Greetings etc.
- Support for TDM and VoIP/SIP trunks.
- Support for multiple Message Waiting Indication types - SMDI, SIP NOTIFY and MF T1 Dial back.
- Bulk uploading of subscribers using CSV files.
- View voicemail activity in real time on a Web portal.

FEATURE	HOW IT WORKS	BENEFITS
PDA accessible Voicemail portal	Login to a web portal through a PDA based browser, and then Click and listen to voicemails in random order.	No need to listen to 15 messages before getting to the 16th. Access voicemails over data network without using cellular minutes.
Sub mail boxes with auto attendant IVR	Up to 9 sub mail boxes per subscriber. Different greeting for each sub mailbox.	Each family member/company employee can have his/her own mail box.
Voicemail to email forwarding	Stored .wav file sent to subscriber's email address.	No need to call Voicemail to retrieve a message. Play messages on your PC or PDA.
Multiple ANI (CLID) per voice mail box	Forward office, home telephones on busy/no answer to the voicemail box.	Common voicemail box across multiple subscriber phone lines i.e., 'voicemail central'.
Multiple Message Waiting Indicator types	Support traditional SMDI based MWI and MWI using SIP and MWI using dial back.	Allows one voicemail / USN to be used across hybrid TDM and VoIP network.
Flexible Voicemail IVR	Can change IVR choices by re-recording .wav files.	Emulate legacy voicemail systems.



Enhanced Security Package



- Prevents unauthorized access to the product
- Hardened security for deployment in defense networks
- Multiple voice security features
- Multiple data security features
- Configurable security features

The Enhanced Security Package includes a set of features designed to harden XOP Network's USN against various forms of network intrusion and hacking. With the entire collection of features enabled, the USN becomes a highly secure platform. However, organizations have varying security requirements and with this in mind XOP Networks has designed many of the USN's security features to be optional and configurable. Features include:

Password management

- Complex passwords enforced (optional)
- Prevent password reuse for 'n' generations (optional)
- Force password change upon first account access (optional)
- Passwords & PINs encrypted in database (optional)
- Enforce periodic password changes (configurable)
- Prevent frequent password changes
- Conference PINs encrypted in database (optional)

Alerting and Logging

- Records login attempts for both success & failure
- Email alerts for important security events

Intrusion Prevention

- Lock account after multiple login failures (optional, configurable)
- Temporarily freeze access from IP address upon multiple login failures
- Lock unused accounts
- Prevents multiple logins from the same accounts or bump-out upon second login (optional, configurable)
- Restrict administrator account logins by IP address. (optional, configurable)
- Disconnect idle sessions (optional, configurable)
- Disconnect sessions that are unable to communicate with server for 15 seconds
- Detects and locks upon automated PIN attacks on dial-in lines

Authorization restrictions

- Administrator accounts optionally allowed access to user accounts
- Administrator may lock/unlock user accounts
- Auditors accounts to review system alerts.

Programming quality control

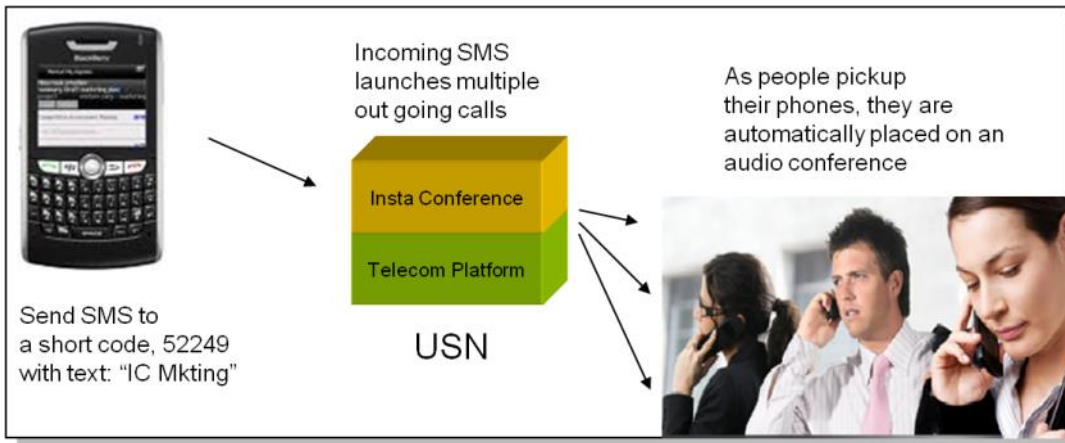
- Internal code reviews performed for XSS (Cross site scripting) attacks
- External third party security review

Cryptographic Protocols

- Only TLSv1 (SSLv3) connections allowed
- Client certificates required for access (optional)



Short Message Service Package



- Support SMPP 3.4 protocol
- Interoperates with a number of carrier SMSC gateways (e.g., Acision/Verizon)
- Enhances multiple XOP applications
- Supports both MO and MT traffic

Short Message Service (SMS) package adds SMPP 3.4 capability to the USN. Both Mobile Terminate (MT) and Mobile Originate (MO) streams are supported. This capability requires a 'bind' with an external Short Message Service Center (SMSC). The SMSC allows the traffic to be routed from/to the USN over the cellular network. Several of the XOP applications are enhanced with the addition of the SMS capability

- Meet-me conference → Send SMS based conference invitations with dial back number and PIN.
- Mobile conference -> initiate a conference by sending a text message to the USN. USN dials back to all participants and upon pickup places them into a conference.
- Voicemail → Receive a SMS with a link to the associated .wav file if someone deposits a message in your voicemail box.
- Mass Notification -> Send a Group SMS to 1000 recipients in less than 2 minutes.
- Receive Password via SMS
- Set up users and groups via SMS

Lightweight Directory Access Protocol Package

Lightweight Directory Access Protocol (LDAP) allows the USN to retrieve data stored in external databases.

This facility can be used to synchronize user profiles kept on external databases (e.g., HR database) with the corresponding profiles for users/moderators/Groups on the USN. The synchronization can be done on demand or at scheduled time. If an employee leaves or changes positions, corresponding changes are reflected on the USN automatically.

This facility can be used for implementing Single Sign-on. The USN interrogates external database using LDAP and allows a moderator to log into the USN portal with the same credentials that he/she uses to log into company's other databases.

- Support LDAP v3
- Compliant with Microsoft Active Directory
- Compliant with Novell GroupWise



Multi-Tenant Package

The XOP Networks' Multi-Tenant Package (MTP) provides a set of capabilities that permit the USN platform to be deployed as multiple virtual USNs. Each virtual USN can be used to offer a number of hosted Value Added Services including Audio Conferencing, Web Collaboration, Voicemail, Firebar Emergency Conferencing, Mass Notification and other services.



Company Set-up Page

The package provides the Service Provider the ability to create company accounts and assign Moderators to them. Each individual moderator can be given access to one or more services available on the USN. Once set-up, the individual companies can access only their own data and records, while the operating company can see all usage/details via the system admin screen. The MTP Package also adds the capability to provide Operator Assistance in the operation of one or more services. A service provider will be able to leverage the port over-subscription capability to engineer the USN resources to support a SLA at a company level.

Account Creation and Usage Reporting

- XML API for Provisioning customer accounts through External CRM System
- Interface with external Charging Gateways
- Automated CDR uploading to External FTP sites/ Billing Systems
- Flexible CDR Formatting
- Daily Cumulative Sign-up Report
- Historical Usage Trending
- Per Conference Logs/ Analytics

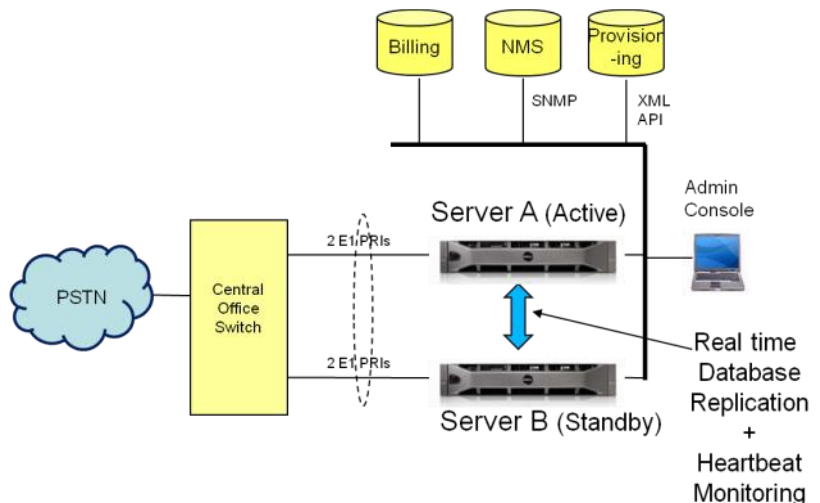
ID	Created-at	Company name	Moderator Id	Moderator Name	Moderator-Phone	Service ID	Service Results	Service Type	Service Subject	Service Started at	Service Complete at	Service Scheduled?	Destination Number	Call Direction	Call	Called-at	Joined-service-at	Disconnected-at	Port ID
24	2025/2010 15:50		7	_name, subscr	407	11	7	FB	conference one	2025/2010 15:49	2025/2010 15:50	N	403	OUT	2025/2010 15:49	2025/2010 15:49	2025/2010 15:50	2	
25	2025/2010 15:50		7	_name, subscr	407	11	7	FB	conference one	2025/2010 15:49	2025/2010 15:50	N	402	OUT	2025/2010 15:49	2025/2010 15:49	2025/2010 15:50	1	
26	2025/2010 15:50		7	_name, subscr	407	11	7	FB	conference one	2025/2010 15:49	2025/2010 15:50	N	401	OUT	2025/2010 15:49	2025/2010 15:49	2025/2010 15:50	0	
27	2025/2010 15:50		7	_name, subscr	407	11	7	FB	conference one	2025/2010 15:49	2025/2010 15:50	N	764	IN	2025/2010 15:49	2025/2010 15:49	2025/2010 15:50	4	
28	2025/2010 15:51		7	_name, subscr	407	11	8	FB	conference one	2025/2010 15:50	2025/2010 15:51	N	403	OUT	2025/2010 15:50	2025/2010 15:50	2025/2010 15:51	2	
29	2025/2010 15:51		7	_name, subscr	407	11	8	FB	conference one	2025/2010 15:50	2025/2010 15:51	N	764	IN	2025/2010 15:50	2025/2010 15:50	2025/2010 15:51	5	
30	2025/2010 15:51		7	_name, subscr	407	11	8	FB	conference one	2025/2010 15:50	2025/2010 15:51	N	401	OUT	2025/2010 15:50	2025/2010 15:50	2025/2010 15:51	0	
31	2025/2010 15:51		7	_name, subscr	407	11	8	FB	conference one	2025/2010 15:50	2025/2010 15:51	N	402	OUT	2025/2010 15:50	2025/2010 15:50	2025/2010 15:51	1	
32	2025/2010 15:52		7	_name, subscr	407	11	9	FB	conference one	2025/2010 15:51	2025/2010 15:52	N	402	OUT	2025/2010 15:51	2025/2010 15:51	2025/2010 15:52	1	
33	2025/2010 15:52		7	_name, subscr	407	11	9	FB	conference one	2025/2010 15:51	2025/2010 15:52	N	403	OUT	2025/2010 15:51	2025/2010 15:51	2025/2010 15:52	2	
34	2025/2010 15:52		7	_name, subscr	407	11	9	FB	conference one	2025/2010 15:51	2025/2010 15:52	N	764	IN	2025/2010 15:51	2025/2010 15:51	2025/2010 15:52	6	
35	2025/2010 15:52		7	_name, subscr	407	11	9	FB	conference one	2025/2010 15:51	2025/2010 15:52	N	401	OUT	2025/2010 15:51	2025/2010 15:51	2025/2010 15:52	0	

Typical System Billing Record (CDR)

High Availability Package

The High Availability Package adds a number of capabilities that allow the USN's to provide 99.999% availability.

- **Real Time Database Replication** between servers – enter user data on one server via the admin interface or XML API and it is automatically replicated on the secondary server.
- **Heart Beat Protocol** between servers allows each server to stay informed about the health of the other server.
- **1+1 Hot Standby Configuration** (Active and Standby servers) – in the event that one server becomes disabled, the secondary server automatically takes over.
- **N+1 Load Sharing Configuration** – allow multiple servers to operate in a load shared arrangement. This is possible in a VoIP/SIP as well as a TDM/T1/E1 network.
- **Common OAMP interface** across two or more servers.



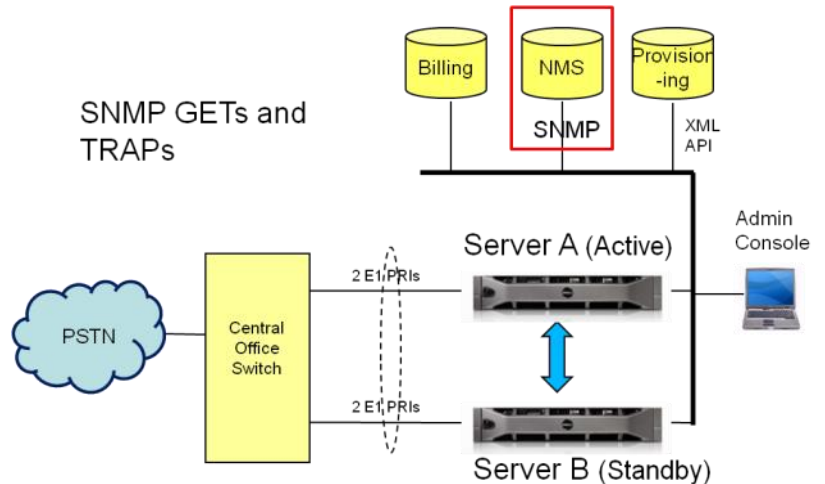
Hot Standby Configuration



Network Management Package

The Network Management Package allows the USN to be managed by an external Simple Network Management Protocol (SNMP) based Network Management system (e.g., HP Openview). The USN supports a SNMP Agent and appropriate Management Information Base (MIB). This package also adds other utilities that provide valuable information about the server.

- The USN supports SNMP based GET requests. These can be used by external NMSs for getting pertinent information from the USN.
- The USN supports the SNMP asynchronous TRAP messages. These messages are used by the USN to inform external NMSs about certain failure conditions.
- USNs typically ship as servers that support Integrated Lights Out (ILO) port or equivalent. This port allows an admin to check on the health of the server (temperature, Fans, hard drives etc.) using a GUI based interface. Additional software needed to support ILO is also included with this package.



RESTful Application Programming Interface

The USN supports a comprehensive RESTful Web Services API can be used by external systems to interrogate a USN regarding health of its major software components, remotely provision conferencing accounts, create conferences for moderators etc.

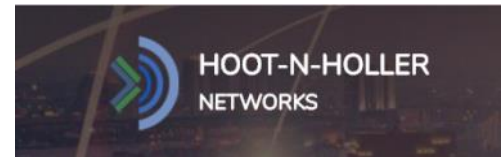
HTTP Verb	Operation	Effect on Entire Collection (e.g. /customer)	Effect on Specific Item (e.g. /customer/{id})
POST	Create	Create a new element. A 'Location' header with link to /customer/{id} containing a new (generated) ID.	Generally an error, since the {id} already exists.
GET	Read	Get a list of elements. Use pagination, sorting and filtering to navigate big lists.	Get a single element.
PUT	Replace	Replace every element's content in the entire collection. All data element fields are expected.	Replace a single element. All data element fields are expected.
PATCH	Update	Modify every element in the entire collection. A subset of the data element fields are expected.	Modify a single element. A subset of the data element fields are expected.
DELETE	Delete	Delete the entire collection	Delete the specified element.



Hosted Services

XOP Networks provides following hosted services using its own products:

- **IP based Hoot-n-Holler Service.** This service allows always-on audio conferencing using XOP Networks' Hoot-n-Holler IP Phones. Please visit <https://hootholler.net/> for more information.
- **Digital Collaboration Service.** This service allows Audio conferencing, Video conferencing, Screen sharing and Whiteboarding to promote web based digital collaboration. Please visit <https://clicktohuddle.com/> for more information.
- **Emergency Conferencing (Firebar) Service.** This service allows first responders, voluntary firemen to be placed in an on-the-fly dial-out audio conference upon receiving an inbound phone call. Please visit <https://www.xopnetworks.com> for more details.
- **Mass Notification Service.** This service is used to notify large number of people voice calls, text messages, email blasts, pager blasts etc. The out-going message can be recorded on-the-fly or pre-programmed. Please visit <https://www.xopnetworks.com> for more details.



<https://www.hootholler.net>



<https://www.clicktohuddle.com>



Consultation Services

Over number of years XOP Networks' engineers have dealt with large number of product installations worldwide. Our customer support department is well acquainted with following topics:

- **Voice Telephony: TDM, CAS, PRI, SS7 etc.**
- **VoIP Telephony: SIP, H.323 signaling, IP PBXs (Cisco Call Manager, NEC SV 9xxx, U3C etc.)**
- **IT Networking: Cisco Routers, Switches, VPN implementation, Microsoft Azure, LDAP**
- **Hosted Service Implementation: WebRTC, STUN and TURN servers etc.**
- **Crash Phone Implementation: Traditional FXS/FXO based and IP based**
- **Development of RFI/RFP etc.**