

Universal Service Node Product Information Folder Release 5.0

Improving Productivity
Through Effective Communications



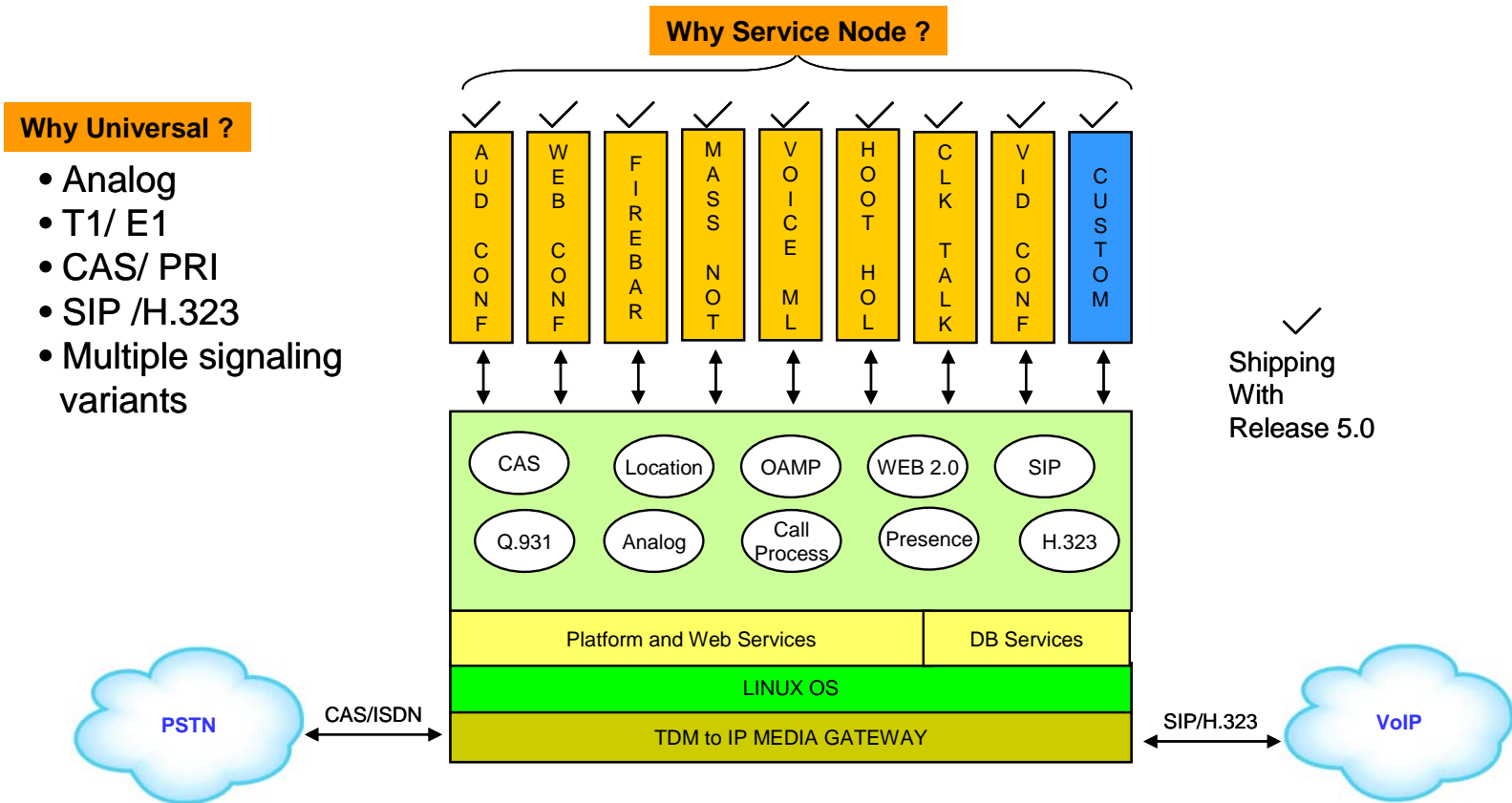
We develop and market next generation value added service platforms for small, medium and large enterprise and service providers—worldwide. Our Universal Service Node product support Enhanced Voicemail, Audio Conferencing, Web Conferencing, Desktop Video Conferencing, Enhanced Firebar/IP Ring down conferencing, Hoot-n-Holler conferencing, Mass Notification, Group SMS and Voice SMS applications. All of these applications are accessible over legacy TDM and next generation VoIP networks. Our applications find their use in many vertical markets including Service Providers, Independent Telcos, Military establishments and small, medium, large Enterprises and City/County governments.

Major Customers include:



Universal Service Node - Architecture

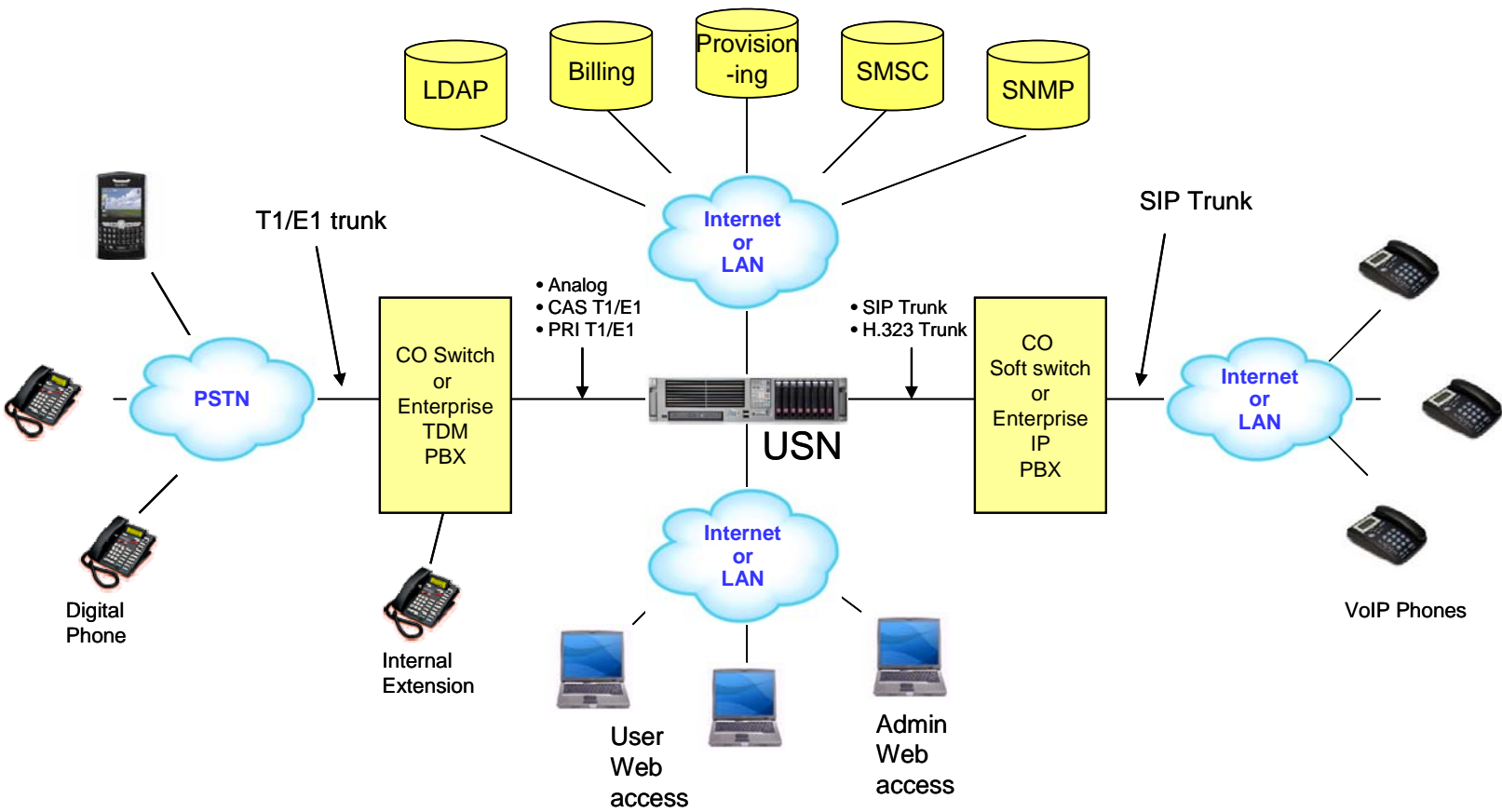
The Universal Service Node (USN) enables improved productivity by providing multiple value added service applications on one platform. The primary benefit of this multi-application platform approach being savings of 75% on CapEx and 50% on OpEx for end users of our customers. The USNs are based on Linux operating system and hence extremely robust and resilient. The server hardware is based on industrial grade chassis with redundant RAID-1 disk mirrored hard drives, redundant power, redundant Ethernet ports etc. The USN can be shipped with TDM only, VoIP only or TDM and VoIP mixed mode configurations. This facilitates an organizations migration from the legacy circuit switched environment to next generation packet switched environment.



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.

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 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 Billing and Provisioning systems, SMPP 4.3 specification for integration with external SMSC gateways and SNMP protocol for integration with external Network Management Systems. The product is typically deployed behind an organizations Firewall and can be accessed via any computer that support Internet Explorer, Firefox or other browser over a secure HTTPS data connection.

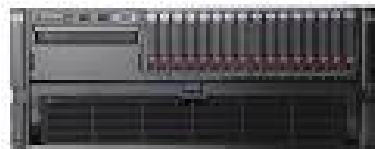




1U server
8 – 16 analog ports
24/30 – 96/120 T1/E1



2U server
24/30 – 192/240 T1/



4U server
48/60 – 384/480 T1/



8U server
384- 3840 T1/E1/DS3

- Industrial grade servers
 - 1, 2, 4 or 8 Rack Units high, 19" wide rack mountable
- Distributed or Centralized configuration
 - Scalable port capacity
 - Up to 14,000 ports in one cluster
- Flexible product options
 - TDM only mode
 - TDM and SIP/VoIP hybrid mode
 - SIP/VoIP only mode
- High Availability
 - 1:1 server level redundancy
 - Hot and warm standby configuration
- Robust
 - RAID-1 Mirrored Hard drives
 - 110/220 or -48V Redundant power supplies
- Linux Operating System
- NEBS compliant (optional)

Platform Specifications

Port Configurations

The number of ports can be any mix of TDM and VoIP channels , 8—16 analog FXO ports, 24/30 -384/480 T1/E1 ports, 672-2016 DS3/E3 ports.

TDM Interfaces

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

IP Interfaces

VoIP Interface 100BaseT Ethernet, RJ-45
Encoding formats G.711, G.729a/b, G.723
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 power supply (optional)
Weight range: 40 - 60 Kg

Network Management

SNMP MIBs for external NMS integration
Automated health check and reporting

Usage Measurements

AMA, CDRs over TCP/IP
Customizable Billing Interface

Operating Requirements

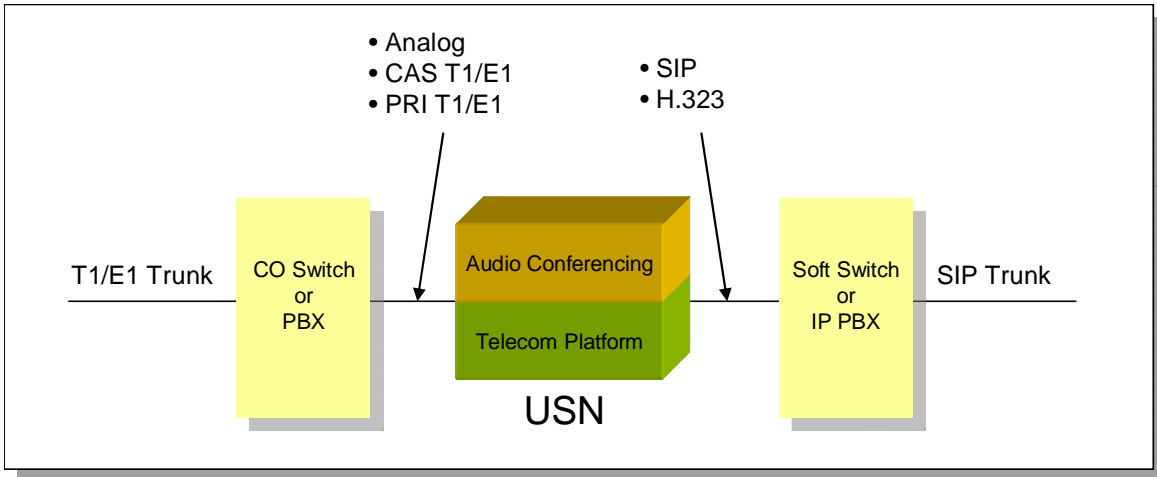
Operating Temperature + 0 deg Celsius to 50 deg Celsius
Storage Temperature -20 deg Celsius to 70 deg Celsius
Humidity 8% to 80% non-condensing

Warranty and Support

Hardware Warranty: One year included
Software Maintenance: Basic and advanced packages available.



Audio Conferencing Application



- Get rid of monthly Audio Conferencing bill.
- Cut down unnecessary travel.
- Make your meeting more productive by combining with Web Conferencing.
- Record Audio and Web conference sessions to create your own Webinars.

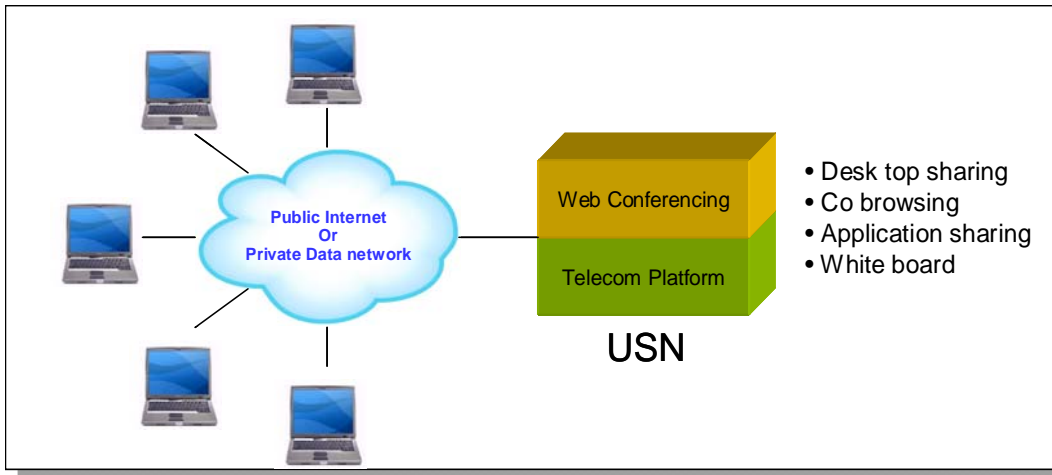
Audio Conferencing Application for the Universal Service Node provides rich conferencing experience for demanding users.

- Support both reservation less and reservation based conferences.
- Customize conference rooms per your requirements, e.g., select entry tones, select memorable vanity PINs, turn recording on/off, select auto-call back on/off 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. Loudest speaker display allows the identification and muting of a participant who may be inadvertently injecting noise into the conference.
- 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.

| FEATURE | HOW IT WORKS | BENEFIT |
|--|--|--|
| Ad-hoc 'Meet Me' conference. | Moderator and participants agree upon a start time and PIN to use. When people dial in and enter their PIN, they are placed on the conference. | Simple to use. No/little training required |
| Scheduled 'Meet Me' with PIN conference. | Schedule from familiar Microsoft Outlook Calendar, after checking for participant availability. | 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 calls trigger 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. |



Web Conferencing Application



- Desk top sharing
- Co browsing
- Application sharing
- White board

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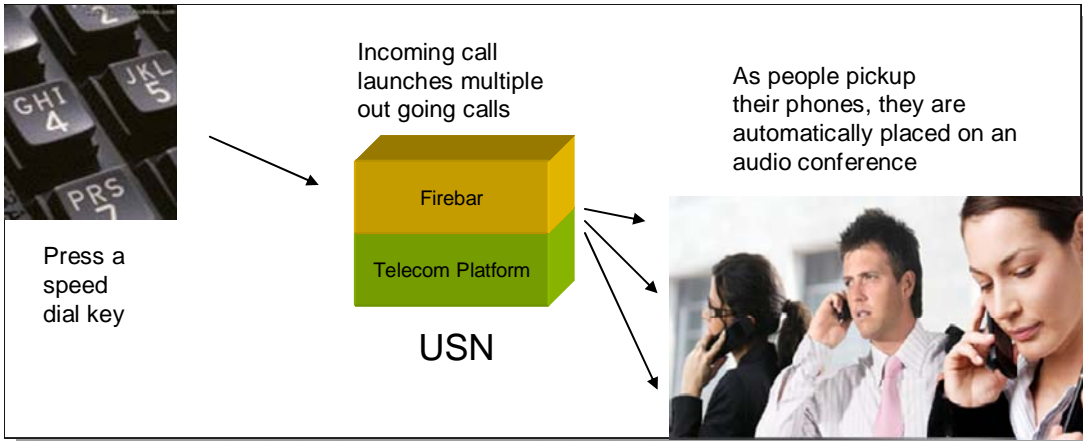
Web Conferencing Application for the Universal Service Node is designed to significantly boost the productivity of your meeting.

- Web based application, no software download required to your PC.
- Designed to run across the Internet or a private data network without requiring any changes to firewalls. Running behind a secure private data network insures that your web conference content cannot be compromised.
- Can be used standalone or in conjunction with XOP Network's Audio Conferencing application. When used together participants can collaborate in real time and achieve desired end result quickly without exchanging multiple emails.

| FEATURE | HOW IT WORKS | BENEFIT |
|-------------------------------|---|---|
| Desk Top Sharing Mode | 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. |
| Presentation Sharing Mode | Upload PowerPoint & PDF documents. Use annotation tools to edit in a collaborative session. | Significantly reduce number of edits/versions to produce final version. |
| White Boarding Mode | Create diagrams/visuals with fellow participants in a collaborative session. | Ideal for brainstorming. |
| Public & private Chat Room | Moderator can respond to questions in public or privately. | Makes Web conferencing more productive. |
| Multiple Presenters | Moderator can allow another participant to take control and share his/her desktop. | Multiple points of view on one conference. |
| Record a Web Conference | Moderator can record the conference using Flash Player. | Conference can be replayed on any PC. |
| Record Audio & Web Conference | When used with the XOP Digital Collaboration Bridge a combined recording can be made. | Can be used for in-house Webinars. Ideal for product training, seminars, legal reviews, etc. |



Enhanced Firebar Application



- Establish an audio conference with press of a 'speed dial' key.
- Allow first responders to be reached over their land line 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.

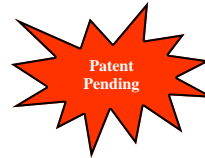
Traditional Firebar or out dialed conferencing is used by emergency dispatch personnel to inform and bring a group of first responders into an audio conference quickly. These systems require dedicated phone lines that terminate on the central office switch of a phone company. XOP Networks' Firebar application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

- Send calls to any landline or cellular phone instead of just dedicated 'red' or emergency phones.
- Select communications 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 and a progress bar on a Web Portal.
- Provide summary and detailed reports on call completions (Busy, No Answer, Answering machine etc.)

| FEATURE | HOW IT WORKS | BENEFITS |
|------------------------------------|--|--|
| Emergency specific groups | Set-up via Web Portal. Use CSV file upload to create multiple groups quickly. | 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, or upon detection of an off-hook (for analog or SIP phones). | Multiple first responders are called in parallel - reduces the overall time required to contact. |
| 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. |



Enhanced Voicemail Application



- Share a common Voicemail platform across multiple TDM and VoIP switches/PBXs.
- Use it as 'Voicemail Central'. Share 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

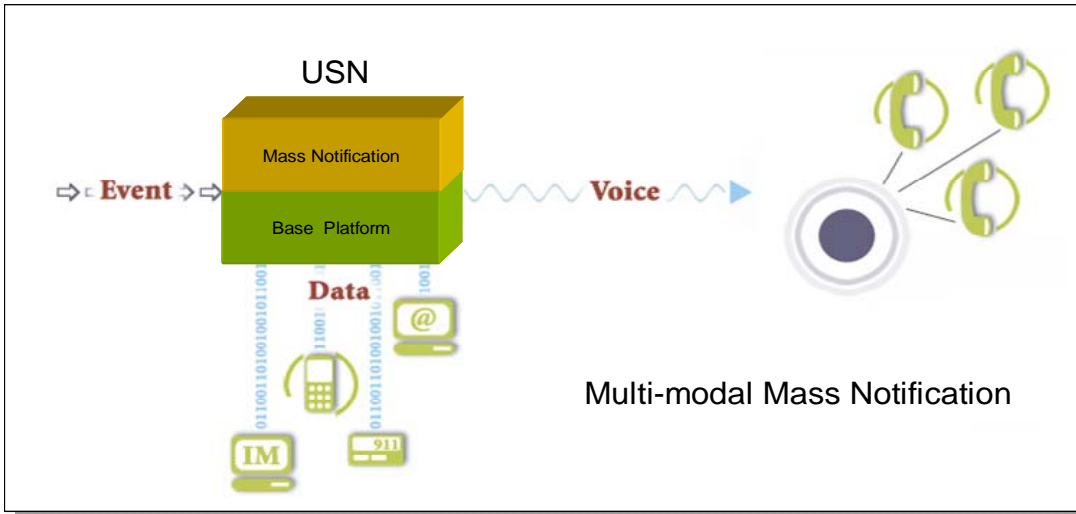
Current voicemail systems only allow sequential access to voicemails. XOP Networks Enhanced Voicemail Application allows stored voicemail messages to be accessed in random order from a Web portal. The Web portal can be accessed via a PC or via a number of PDAs including Blackberry and iPhone. By a simple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

- From 100 to 10,000 Voicemail boxes per chassis
- Subscriber Web portal for managing the Voicemail configuration, 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
- Voicemail usage reporting

| FEATURE | HOW IT WORKS | BENEFITS |
|--|---|---|
| PDA accessible Voicemail portal | Login to a web portal through 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. Pull voicemails over data bandwidth without burning cellular minutes. |
| Sub mail boxes with auto attendant | Up to 9 sub mail boxes per subscriber. Different pin with different greeting for each sub box | Each family/company can have his/her own mail box. |
| Voicemail to email forwarding | Automatic .wav file sent to email | Save and play messages on your PC. |
| 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. |



Mass Notification Application



- Get rid of your out-sourced messaging service bill.
- 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.

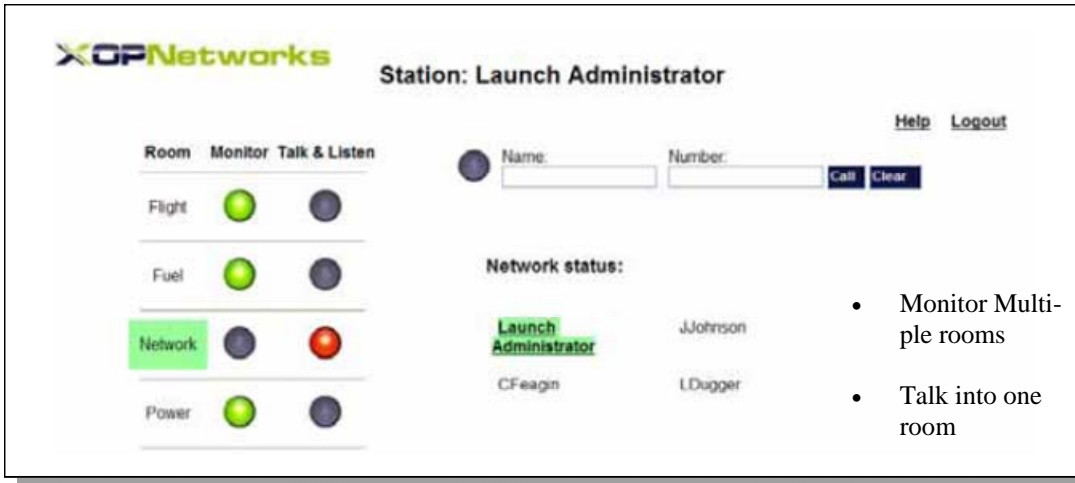
Mass Notification Application for the Universal Service Node is designed to send multi-modal messages to thousands of people during emergency and non-emergency situations.

- Select communications 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 delivering a message.
- 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.
- Provide summary and 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 and 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 message. |
| Announcement Box capability | Moderator periodically dials in and records a message in an announcement box. People can call in and hear the updated message. | Great way to inform people during changing emergency situations such as hurricanes, blackouts etc. |
| Re-iteratively contact the un-contacted | Set up Group Alert with 'un-contacted' option. Trigger same Group Alert multiple times until message is received by desired percentage of recipients. | No wasted calls. Iteratively build up the percentage of people who received calls. |
| 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. |



Hoot-n-Holler Application



- Enter a conference room simply by going off-hook on a SIP enabled squawk box.
- Enter a conference room by pressing a speed dial key on your digital hand set or a soft phone.
- Mission Controller Web portal for emulating legacy CPE devices.
- Monitor audio from multiple rooms and Talk and Listen into one.

- Monitor Multiple rooms
- Talk into one room

Hoot and Holler conferences are typically used for managing 'many to many' communications in scenarios such as stock trading floors and junk yards etc. To facilitate such communication, multiple locations are connected via leased lines into a centrally located Audio Conference bridge. As corporations establish their own data networks, they desire to run all of their voice and data applications on the same data network. The XOP Networks' USN supports the Hoot and Holler application across both TDM and VoIP networks. This facilitates the transition of the application from leased line infrastructure to the VoIP network infrastructure.

- Support Mission Critical conferencing across TDM and VoIP networks
- Support legacy 'Squawk box' and next generation IP phones and soft phones
- Built in 'presence' manager indicates availability of the participants connected to VoIP network
- Mission controller can 'monitor' audio from multiple conference rooms simultaneously via a web portal
- Mission controller can 'talk and listen' into a given conference room via a web portal
- Allow Mission Controller to dial out and pull in additional participants
- Ability to make separate audio recordings - at the individual room level and at the controller level
- Usage reporting and call logs.

| FEATURE | HOW IT WORKS | BENEFITS |
|--|--|---|
| Web portal based management of mission critical conferences | Mission controller can monitor audio from one or more rooms and talk and listen to one room simply by a click on a web portal. Can dial out to external parties and bring them into a conference room. | Immune from mechanical failures - typical of legacy CPE equipment Can be accessed remotely over the Internet |
| Separate web portals for USN administrator and Mission Controllers | System administrator sets up H&H rooms, stations and their associations via USN web portal. | Mission controllers only operate the Hoot-n-holler web portal. Easy to learn, train and use. |
| Visual indicator showing status of participants in a given room | Participant status in a given room is displayed with different fonts. Regular text -> Associated Italicized text -> Registered SIP end point Bold text -> actively participating Bold text with green color -> loudest speaker | Mission controllers can quickly ascertain the status of a meeting in a given room. |
| Dial out with 'Find-you' capability | Upon initiation of a dial out the system will dial up to 4 phone numbers for the individual as part of the 'find you' process. | Improves the probability of reaching an out dialed participant |



Enhanced Security Package



- Prevent 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

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

Intrusion Prevention

- Lock account after multiple login failures (optional, configurable)
- Temporary freeze access from IP address upon multiple login failures
- Lock unused accounts
- Prevent multiple logins from the same accounts or bump 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
- Detect and lock upon automated PIN attacks on dial-in lines

Authorization restrictions

- Administrator accounts optionally allowed access to user accounts
- Administrator may lock/unlock user accounts
- Auditor accounts to review system alerts, but not allowed user rights

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)