

Universal Service Node Product Information Folder

Release 5.0



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 Conferencing, Desktop Video Conferencing, Enhanced Firebar/IP Ring down conferencing, Command & Control Conferencing, Hoot-n-Holler Conferencing, Mass Notification, Group SMS, Enhanced Voicemail and Voice SMS applications. All of these applications are accessible over legacy TDM and next generation VoIP networks. Our applications find their uses in many vertical markets including Service Providers, Independent Telcos, Military establishments, City/County governments and small, medium large Enterprises.

Major Customers include:











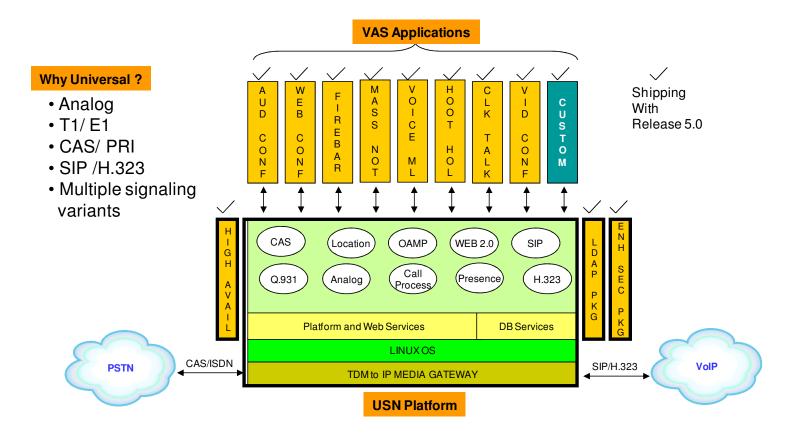






Universal Service Node - Architecture

The Universal Service Node (USN) enables improved productivity by providing multiple value added service applications on a single platform. The primary benefit of this multiapplication 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 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 organization's migration from the legacy circuit switched environment to next generation packet switched environment.



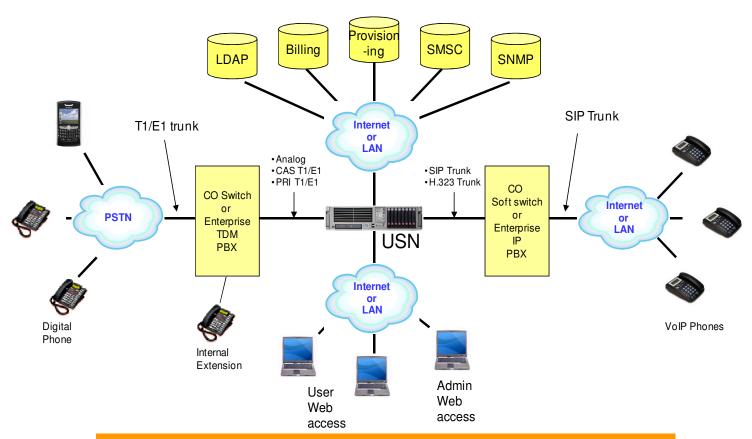
The USN Platform, Platform Extensions, and VAS Applications are licensed separately. TDM and VoIP ports can be shared dynamically across multiple applications or segmented per application.

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 external 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 Internet Explorer, Firefox 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

Improving Productivity Through Effective Communications





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
- 120 VoIP ports



2U server

- 8 16 analog ports
- 24/30 192/240 T1/E1ports
- 240 VoIP ports



4U server

- 48/60 960/1200 T1/E1 DS0 ports
- 1000 VoIP ports



8U server

- 672- 2016 TDM/DS3/E3 DS0 ports
- 2016 VoIP ports

Industrial grade servers

- 1, 2, 4 or 8 Rack Units high, 19" wide rack mountable
- Scalable port capacity
 - 4 ports to 2016 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
- l:l Hot and warm standby configuration
- 1:1 Load shared configuration

Robust

- RAID-1 Mirrored Hard drives
- Redundant power supplies
- Two Quad core CPUs
- Linux Operating System
- NEBS/CE compliant (optional)

Platform Specifications

Port Configurations

The number of ports can be any mix of TDM and VoIP channels, 8—16 analog ports, 24/30 -384/480 T1/E1 ports, 672 trial grade chassis or NEBS complaint chassis SCSI/SATA RAID 1 Mirrored Disks, 36 to 144

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, 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 Requirements

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

Warranty and Support

Hardware Warranty: One year included Software Maintenance: One year included Extended warranty/support available.

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 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 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 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 op- tion)	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.

Web Conferencing Application





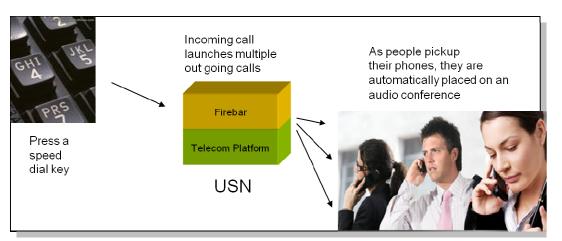
- Get rid of monthly Web conferencing billing.
- Cut down on unnecessary travel.
- Make your Web conference secure.
- Make your meeting more productive by combining with Audio Conferencing.
- Record web and audio conference session to create your own Webinars.

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 results 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 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 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 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.)

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 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.

Enhanced Voicemail Application







 Share a common Voicemail platform across multiple TDM and VoIP switches/PBXs.

XOPNetworks

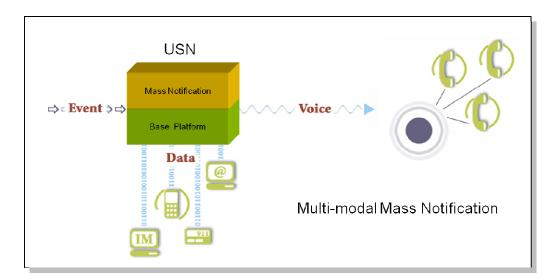
- Use it as 'Voicemail Central'. Share the same mailbox across multiple subscriber phones.
- Get Voicemails using 'patented' nonsequential access approach. Access voicemails through popular PDAs.
- No change to end user experience due to emulation of legacy Voicemail IVRs.

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 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.				

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.

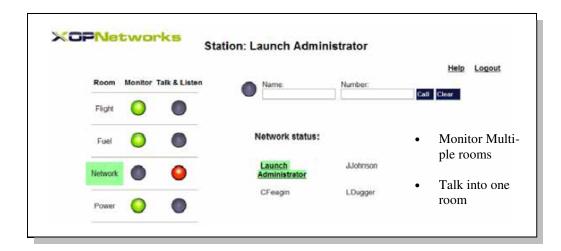
Mass Notification Application for the Universal Service Node is designed to send multi-modal messages to thousands of people during emergency and non-emergency situations.

- 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 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. 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.	Great way to inform people during changing emergency situations such as hurricanes, blackouts 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.

Hoot and Holler Conferencing Application





- Enter a conference room simply by going offhook on a squawk box.
- Enter a conference room by pressing a speed dial key on your digital hand set or a soft phone.
- Web portal for emulating legacy Key sets/ Turret devices.
- Monitor audio from multiple rooms and Talk and Listen into one conference.

Hoot and Holler conferences are typically used for managing 'many to many' communications between stock traders, military command and control personnel etc. A user simply 'hoots' into a squawk box or a phone and waits until a distant party responds with a 'holler'. This process goes on throughout the duration of the conference which may last for an entire day of trading or for a multiday mission.

- Supports Hoot and Holler conferencing across TDM and VoIP networks
- Supports legacy 'Squawk box' and next generation IP phones and soft phones
- Provides a web portal that emulates the functions of legacy Key Sets and Turrets
- Using the web portal a Supervisor can

See the 'presence' status of each participant in a given conference room

Monitor audio from different rooms

Barge into a given conference room

Dial out and pull in additional participants into a given conference room

- 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
Web portal based management of mission critical conferences	A Supervisor 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 Supervisors	System administrator sets up H&H rooms, stations and their associations via USN web portal.	Supervisor 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	Supervisor 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









- 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.

Cryptographic Protocols

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

Programming quality control

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



Service Provider Package

The XOP Networks' Service Provider Package (SPP) provides a set of capabilities that permit the Service Provider to offer a number of hosted Value Added Services including Audio Conferencing, Web Conferencing, Voicemail, Firebar Emergency Conferencing, Mass Notification and other services from 1+1 or N+1 redundant USN platforms.

XOPNetworks Address Line 1: 5508 W Plano Parkway Address Line 2: Suite B City: Plano ntact Name: Sudhir Gupta ing Contact Phone: (972) 590-0201 illing Contact Email: sgupta@xopn 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 Service Provider Package also adds the capability to provider 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

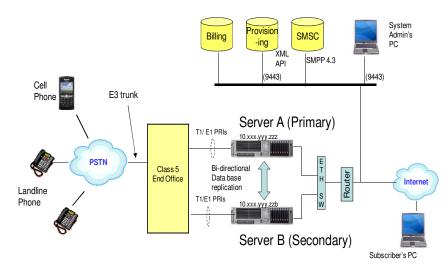
- XML API for Provisioning customer accounts through External CRM System
- Synchronization / Authentication via LDAP
- Flexible CDR Formatting
- **Automated CDR uploading to External** Billing Systems
- **Daily Cumulative Sign-up Report**
- **Historical Usage Trending**
- Per Conference Logs/ Analytics

U	Created-at	Company	Moderator	Moderator	oderator-pric	Service	SOUNICE	Service-type	Service-subjec	Service	SELVICE	Service	Destination	Call	Caled-at	Joined-Service-a	Disconnected-at	FUIL
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34	2/25/2010 15:52		7	1_Iname, subscrit	407	-11	9	FB	conference on	2/25/2010 15:51	2/25/2010 15:52	N	764	N	2/25/2010 15:51	2/25/2010 15:51	2/25/2010 15:52	6
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Typical System Billing Record (CDR)

The Service Provider Management Package adds several 'High Availability' features that significantly boost the availability and reliability of the USN platform.

- 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.
- SNMP based monitoring using industry



Hot Standby Configuration