



# **Replacing Spectel Conference Bridge with Next Generation XOP Networks' Universal Service Node (USN)**

*An XOP Networks White Paper*

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**EXECUTIVE SUMMARY**

Many of the older generation Spectel audio conference bridges have been deployed in Mission Critical applications at large companies around the world. These bridges, although highly successful in their time, have become expensive to maintain, and in some cases completely unsupportable and unreliable.

Spectel's current owner (Avaya, CS-7000) has issued "Manufacturer Discontinued" notices to all of the major Spectel bridge owners, informing them that "all support will cease at the end of 2010". As a result, replacement has become of utmost urgency for mission critical situations, and for anyone who relies on conferencing as an essential business component.

The dilemma facing current Spectel bridge owners can be summarized as follows:

- How to deploy an economical replacement that can interface to the legacy TDM network now and prepare it for migration towards next generation VoIP/SIP network.
- Employees & End-users are very familiar with the use the Spectel system. Any new replacement should emulate the features and preserve the end user experience of the old system.
- How to add new capabilities such as Web and Video conferencing to my network without having the risk of fork-lift again?

This white paper discusses the details of how a Spectel CS7000 can be successfully replaced with a XOP Networks' Universal Service Node (USN) that can support TDM and VoIP/SIP trunks, and provide the foundation for new services such as Web, Video, and Multimedia conferencing.

## ***FACTORS DRIVING THE CHANGE***

### ***NETWORK EVOLUTION***

The network provider and/or commercial business is being driven by competition and lowering operating costs to evolve the network from TDM to VoIP, that supports VoIP phone, Internet, Video and Multi-media. For medium and large business with multi-sites over a wide geographical area the new network will typically employ their own fiber network or lease circuits from a MPLS carrier.

However, no company implements such a change over night. This kind of change will occur over a period of time. The service provider will typically deploy a soft switch in addition to the current legacy switches. During this phase, the company will cap any new stations/subscriber additions to the legacy network and add new stations/subscribers to the IP Network. When they have reached critical mass, they will move their remaining stations/customers from legacy switches to the soft switch and phase out the older legacy switches.

This transition period could last for several years depending upon the number of switches involved. During this period, the company will need to provide Conferencing/Firebar/Emergency Alerting to sets of stations/customers on the TDM and VoIP networks.

Since the old legacy conferencing systems cannot support VoIP, the option is to add VoIP gateways or replace the entire system with a new system that can support multiple domains.

### ***MISSION CRITICAL SITUATIONS***

Mission Critical Conferencing is a specialized application of audio conference bridges. Such conferencing is routinely used for FAA Air Traffic Control, NASA Rocket Launches, commercial Satellite Launch Control, Transfer Orbit Support Services etc. It can also allow multiple teams of people located in different parts of the world to quickly and easily interact with each other via 'always on' conferences. From their command and control centers, the conference operators can go in and out of one or more conferences simply by pressing key on a keyset or web click/ web touch (with touch sensitive screen) on a web portal. Legacy mission critical conference bridges use monolithic architecture and are based on TDM technology. What is needed is a system that is equally at home in a legacy TDM network and a VoIP next generation network.

## *FEATURE EMULATION*

In large multi-site companies, as well as service providers, asking people to change the methods they are familiar with is very difficult, if not impossible. In a service provider situation it might well result in significant loss of business and customers. In private industry situations, especially in where mission critical conferencing is critical to handling emergency situations (e.g., Oil, Gas, and Power industries) retraining employees is both time consuming, expensive, and often less than adequate. However, when a company is forced to upgrade its existing equipment (e.g. fork-lift), it is very important to replace the existing system with one that can emulate the features and user experience in such a way as to be immediately acceptable by management, employees and customers with very little (if any) retraining.

## *NETWORK/SERVICE EVOLUTION*

When asking any company to spend capital, one of the most important factors to consider is “how long will the new equipment last ... will it evolve to meet my future requirements?”

Typical of other/future features and services as they relate to conferencing are:

- Network Vendor agnostic – must work with all vendor’s PBXs, Switches and Soft Switches
- Web Conferencing – should promote collaborative work environment via integrated desktop sharing, application sharing, webinars etc.
- Video Conferencing – allow high quality desk-top video conferencing with efficient use of bandwidth instead of requiring expensive dedicated video conference rooms
- Multi-media conferencing, Audio, Web, Video in a tightly integrated package.
- Integration with other vendor’s product such as MS Exchange, MS Active Directory (LDAP), and SNMP based NMS etc.
- Internet Appliance – allow access to all system OAMP and conference scheduling/monitoring functions via a Web Portal.

## *SUPPORT SERVICES*

When selecting a new vendor to supply the conferencing equipment, it is important to consider the following factors:

- Vendors who specialize in conferencing typically offer better value for money as their products are state-of-the-art and must be network vendor agnostic to survive.
- Vendor's product's architecture should allow quick changes to match the features and environment of the legacy bridge. The Vendor should be prepared to make such changes quickly, efficiently and with minimum expense.

## ***THE SPECTEL CS7000***

The CS7000<sup>1</sup> is first generation conferencing equipment and its usual configuration consists of the following sub systems:

- LAN Switch
- Spectel CS7000 Shelf (19 card slots)
- Shelf Processor Card configured for redundant operation
- N x CPU
- Shelf Power Supply Unit configured for redundant operation
- N x Dual Primary Rate Interface Card
- N x Dual Audio Conference Card (62 ports each)
- N x Messaging System Interface
- Operator Telephone Interface



Figure 1: CS7000 Interface Shelf

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<sup>1</sup> CS7000 information derived from Internet search

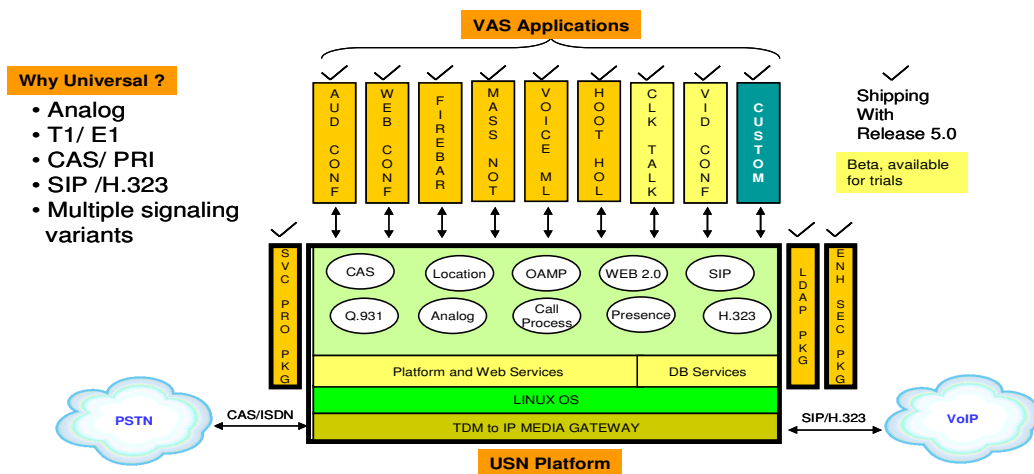
The Spectel Bridge and its various subsystems usually consist of two (2) complete racks of equipment, and it is by today's standards, power, cooling and real-estate hungry, as well as utilizes outdated technology.

In terms of features, the Spectel Bridge includes support for Audio Conferencing, Alerting/Mass Notification and Emergency Conferencing/Enhanced Firebar applications, and Operator Control Positions (see Appendix A for comparison between CS7000 and the USN).

## THE SOLUTION: XOP NETWORKS' UNIVERSAL SERVICE NODE

### USN ARCHITECTURE

The XOP Networks' Universal Service Node (USN) is a state-of-the-art next generation multi service platform that supports E1/T1 & VoIP interfaces and is functionally equivalent to the Spectel CS7000 conference bridge. The USN also adds several new capabilities, including an easy to use web portal for system administration, and hot standby high availability configuration in which USNs are deployed in different geographic locations with real time database replication between these locations. In terms of reliability, the USN far out-performs the old Spectel Bridge. Its Linux operating system is extremely robust and reliable compared to the Windows operating system. The USNs are built using industrial grade servers (typically HP Proliant or Dell PowerEdge) with all critical components duplicated (CPUs, Memory, Disks, PSUs).

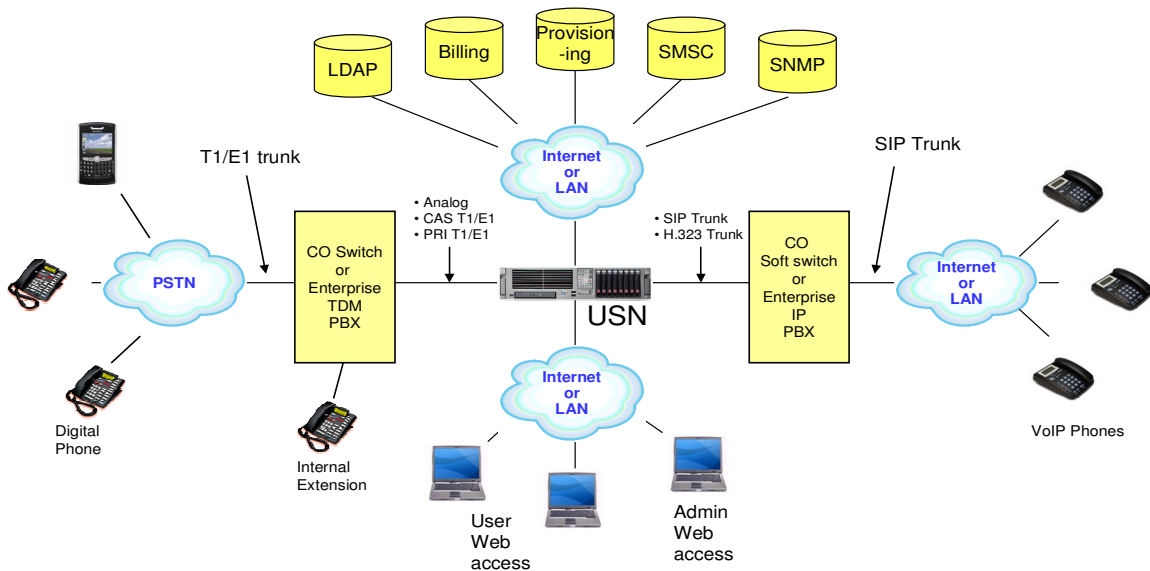


The USN Platform, Platform Extensions, and VAS Applications are licensed separately. TDM and VoIP ports can be shared dynamically across multiple applications or segmented per application.

Figure 2: USN System Architecture

## USN NETWORK DEPLOYMENT

The diagram following shows a typical deployment for the USN. The system can be directly connected to a VoIP soft-switch via SIP trunks or TDM via T1/E1 PRI trunks. Its unique architecture allows multiple value added services to be offered over TDM interface or VoIP/SIP based packet interface or both. The USN can also support a LDAP interface to an external provisioning system, perform a scheduled FTP upload of Call Detail Recordings, and supports SMPP3.4 based interface to external SMSC Gateway for supporting SMS messaging to and from the server.



**Multiple signaling interface types allow product to be deployed in TDM or VoIP or hybrid networks**

Figure 3: USN Deployment Options

## USN PHYSICAL LAYOUT



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/E1 ports  
 • 240 VoIP ports



4U server  
 • 48/60 – 384/480 T1/E1 DS0 ports  
 • 480 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
  - 8 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
  - 1:1 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)

**Product purpose built to suit the needs of medium to large enterprises and Independent Telephone Companies.**

Figure 4: USN Physical Layout and Size Options



## **CONCLUSION**

Change need not be painful! Given that all major manufactures do at some time chose to manufacturing discontinue their product lines (typically due to acquisition and resulting high product support costs); the choice to replace the existing system must take into account the following factors:

- Economical & efficient deployment in both legacy TDM and VoIP networks.
- Feature Emulation; New system must mimic the old as far as features and feature operation is concerned.
- New platform must be able to support new features and services without fork-lift, i.e., Web, Video, Multi-media.

The XOP Networks' USN conferencing Platform answers all of these demands in a most economical and elegant fashion, providing high reliability, low operating costs and service longevity. In addition, XOP Networks stands committed to adapt its standard product to suit its customer's environment.

## **APPENDIX A - CS7000 vs USN FEATURE COMPARISON**

	<b>Spectel CS7000</b>	<b>XOP Networks' USN</b>
<b>SYSTEM LEVEL FEATURES</b>		
Operating System	Proprietary	Linux - highly reliable
Industrial Grade Hardware in compact chassis	First generation, many subsystems	HP ProLiant - 8U can support 2K Ports
Military Grade Security	No	Yes - JTIC Compliant
Hot Standby/Load sharing	No	Yes
Real Time Databas Replication	No	Yes
Support TDM (T1/E1) and VoIP (SIP & H323)	No	Yes - built in Gateway function
LDAP to MS Active Directory	No	Yes - can authenticate user accounts from external HR database.
SMS Interface	No	Yes - messages can be sent to and from SMSC Gateway
Billing output/CDRs	Yes, but limited	CDR via FTP
Multi Media, including Web, Video and Multimedia	No not possible	Yes - USN can support all
<b>CONFERENCING FEATURE SET</b>		
<b>Dial-in and dial-out conferencing</b>	Yes	Yes
<b>Click-to-connect Conferencing</b>	No	Yes - driven from web portal

<b>SMS Initiated Conferencing</b>	No	Yes - conferencing to pre-set groups can be initiated by sending SMS message. USN will call-out to all in predefined group, and as they answer join them to the multiparty conference
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**BASIC CONFERENCE TYPES**

<b>Reservation Based</b> Modertor is able to reserve ports and time on the bridge	Yes	Yes
<b>On Demand</b> Conferences can be set up to provide on demand conferencing	Yes	Yes
<b>Meet-me Direct Level 1</b> Participant dial pre-determined number, system plays message and prompt for PIN, then joins conference	Yes	Yes
<b>Meet-me Direct Level 2</b> User dial DDI number, system plays message, user enters PIN, then joins conference	Yes	Yes
<b>Meet-me Secure Level 3</b> User dials DDI number, enters participant PIN, enters Conference PIN, and joins conference. If caller enters wrong PIN 3 time, call is diverted to Operator. Operator can transfer caller back into conference.	Yes	Yes
<b>Meet-me Operator Assisted</b> Same as Level 1 but caller is screen by operator after entering PIN	Yes	Yes
<b>Meet-me Secure Conference</b> PIN based and CLI based.	Yes	Yes
PIN Based, Dial DDI, enter User PIN, enter Conference PIN	Yes	Yes
CLI Conference (Direct) Based on CLI, No PIN required	Yes	Yes
<b>Management Conference</b> Open Conference with Large Groups	Yes	Yes

Reporting

- System Alarms and Logs	Yes	Yes
- Call Detail Records	Yes	Yes
- Conference Detail Records	Yes	Yes
- Operator transaction logs	Yes	Yes
- DTMF User transaction logs	Yes	Yes
- Audio Conference recorded names (roll call audio file)	Yes	Yes
- Post conference reporting and billing	Yes	Yes

ALERTING

- Different Alerting Groups	Yes	Yes - unlimited
- Crisis Alert	Yes	Yes
- Volunteer Alert	Yes	Yes
- Management Alert	Yes	Yes

Crisis Alert Conference

User triggers call, system calls pre-selected group, moderator/chairman, CLI, Recording, end when moderator clears, 2nd 3rd and 4th calls initiate separate conference	Yes	Yes
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Volunteer Alert Conferences

Pre-recorded messages sent to Pre-determined List, As they answer message is played, Support for multiple messages	Yes	Yes - list uploaded via LDAP or CSV File
Caller prompts blast dial by calling predetermined number and entering a PIN	Yes	Yes
Caller dials pre-determined number, optionally prompted for PIN, then played pre-recorded message	Yes	Yes

Management Alert Conference

Management Alert Conference	Yes	Yes
Team Communication Service	Yes	Yes

User dials pre-determined number, enters PIN, enters as moderator, system dials out to pre-determined list, as they answer joined into conference, moderator can specify calling number.	Yes	Yes
Automatic take down upon moderator exit	Yes	Yes

### *Want to Learn More?*

For more information, please visit our Web site <http://www.xopnetworks.com> or send an email to [marketing@xopnetworks.com](mailto:marketing@xopnetworks.com)

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#### **About XOP Networks**

Headquartered in Dallas, Texas, XOP Networks was founded in 2002 and is backed by a seasoned management team. Deployed at multiple Fortune 100 companies, US defense organizations, Mobile operators and CLEC/IOC customers, XOP Networks' products allow customers to improve employee productivity, promote business continuity and generate new revenue streams. Having both legacy and VoIP interfaces, XOP products allow customers to seamlessly transition their value added services from legacy circuit switched networks to VoIP based packet switched networks.

In 2007 XOP Networks launched its Universal Service Node (USN) which offers multiple value added services on one platform. In year 2008 IP based Hoot and Holler conferencing capability was added to the platform. In 2009 XOP Networks introduced Ring down Firebar Conference Server (RFCS) that allows a conference to be set up simply by lifting a handset on an analog or IP phone. Several other enhancements were introduced including support for Secure conferencing, SMS driven mobile conferencing, Group SMS and Voice SMS capabilities. The product line was also hardened for use in defense networks and has received the coveted JITC certification. Flexibility of the product's architecture allows XOP Networks to quickly customize its solutions to fit the needs of its customers.

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