About Sonus Networks

Sonus enables and secures real-time communications so the world's leading service providers and enterprises can embrace the next generation of SIP and 4G/LTE solutions including VoIP, video, instant messaging and online collaboration. With customers in more than 50 countries and nearly two decades of experience, Sonus offers a complete portfolio of hardware-based and virtualized Session Border Controllers (SBCs), Diameter Signaling Controllers (DSCs), policy/routing servers and media and signaling gateways. For more information, visit www.sonus.net or call 1-855-GO-SONUS. Sonus is a registered trademark of Sonus Networks, Inc. All other company and product names may be trademarks of the respective companies with which they are associated.

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Microsoft Partner
Gold Communications

Voice
Unified Communications
Business Productivity Solutions
Midmarket Solution Provider

VX 900

VX 900 Secure Voice Exchange

- Bandwidth-efficient secure voice platform
- Robust, secure Voice relay over IP
- Purpose-built voice and data solution for government and military networks
- Integrated signaling, gateway and media server functions
- Dynamic flow control technology to keep secure call connected in a degraded environment
- Supports 200 simultaneous Future Narrowband Digital Terminal (FNBDT) calls over IP

Sonus SBC VX 900

The award-winning VX 900 platform provides the most powerful, most complete, Voice over Internet Protocol (VoIP) solution for the government market. The VX 900 offers an efficient, robust, and cost-effective way to provide high-quality voice and secure voice services over the diverse and challenging transmission environments in fixed/mobile government and military networks.

Powerful, Cost-Effective Solution

The VX 900 platform supports analog, digital, and native IP voice, as well as port and trunk side serial data, making it a compelling solution for any government agency looking to deploy IP-based voice and data solutions. The product line combines the functionality of a media gateway, signaling control point, H.323/SIP inter-working device, media server and voice/data mux in a single chassis. As a port side interface, the optional serial data card enables legacy equipment to take advantage of the IP backbone, realizing full convergence.

The serial data card allows users to connect serial data circuits to the VX 900 at rates from 300Bps to 8.192Mbps, making the VX 900 a versatile voice and data multiplexing platform. Data from the serial interface is encapsulated into IP packets using the Structure Agnostic TDM over Packet (SAToP) standard, allowing it to inter-work with the Promina IP trunk solution. As a trunk side interface, the serial data card can interface with any serial based transport, including legacy satellite systems, HF radio equipment and serial crypto devices.

Robust Implementation

The VX 900 uses dynamic flow control technology to overcome the challenges of maintaining secure VoIP calls in a degraded environment. The platform will keep secure calls connected, even when the network experiences slow traffic flow and significant packet loss. Other implementations "relay" secure traffic as a normal (high bit rate) compressed audio call. Any lost, late or corrupt packets often result in the modem carrier slipping, causing the modem to retrain which typically results in a dropped call. The VX 900 can withstand significant packet loss and jitter, and still keep the calls connected. SCIP calls through the VX 900 can withstand complete network failure for up to 10 seconds without failing the call. SCIP calls can sustain up to 5% packet loss mid-call with no significant degradation of audio quality.

Comprehensive Protocol Support

With support for SIP, H.323, our own bandwidth optimized Bestflow Signaling Protocol (BSP), V.150 Universal modem relay, many ISDN variants, both TDM and IP Multi-Level Precedence and Preemption (MLPP) and SCIP Relay, the VX 900 platform can be set up in multiple deployment scenarios and in multi-vendor environments. Our comprehensive voice protocol support allows a site to inter-connect to any system in the network, including IP phones, soft switches, IP PBXs and traditional TDM PBXs, providing true any-to-any protocol translation.

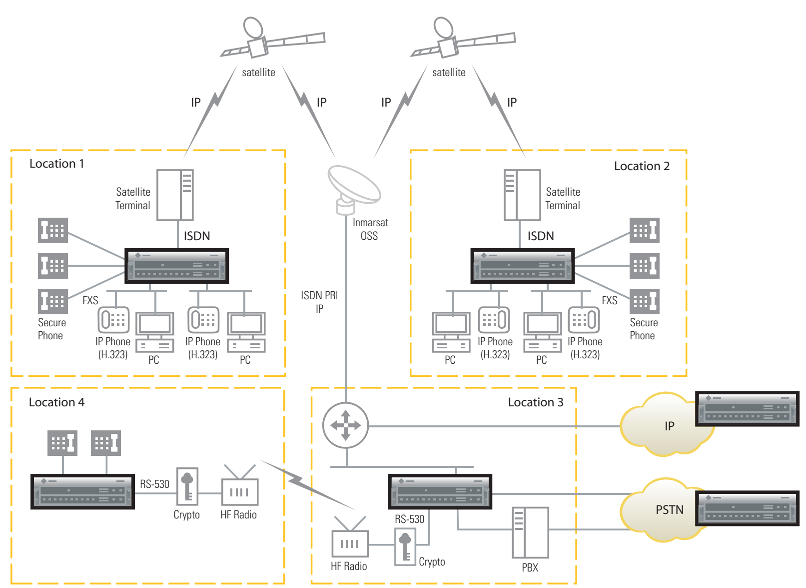
Our standards-based V.150.1 UMR implementation is ideal for carrying IP-based Type 1 encrypted phone traffic, such as the General Dynamics vIPer phone, across an IP network. The technology reliably carries modem traffic over IP and is the latest extension of what NET has been doing for years with the VX Series secure voice gateway platform in transporting secure SCIP and FNBT phones across IP networks.

Distributed Architecture

The VX 900 integrates the key elements of a VoIP network into a single platform. By implementing a distributed architecture, each VX 900 node acts as an independent intelligent processing element. There is no reliance on centralized servers for routing and management decisions, so a VX 900 network has no single point of failure. A distributed architecture also results in lower call set-up times and allows the network to grow to any size without a detrimental effect on call performance. Each node essentially adds processing power to the network.

Support System

The VX 900 system incorporates comprehensive provisioning and monitoring tools that ensure successful deployment and smooth operation of IP-based voice services. VXbuilder™ is a graphical user interface (GUI) that allows network operators to configure every aspect of the VX 900 network in real time. VXwatch™ provides GUI-based, real-time monitoring of all channel and alarm activity on VX 900 nodes throughout a network.



- Secure VoIP Appliance enabling non-disruptive migration from a TDM infrastructure to a converged IP network
- Integrated platform with voice router, media gateway, signaling and protocol conversion capabilities
- Efficient FNBDT/SCIP relay over IP backbone
- Advanced call routing and trunk group concepts
- Industry-leading frame packing solution
- Flexible any-to-any signaling protocol conversion
- Comprehensive topology and link quality management (LQM)
- Secure NAT traversal and firewall traversal
- Unique SIP user agent proxy interface for non-SIP devices (TDM and H.323)

VX 900 Specifications

Interface Cards

- 1, 2 and 4 port T1/E1
- 8 port FXS
- 4 port sync serial (both port and network sides)

Secure Call Support

- V.150 UMR call detection, relay and/or bypass
- FNBDT/SCIP call detection and relay
- Up to 10 secure calls over a 64k IP link (2.4kbps per call)
- Up to 200 simultaneous compressed secure calls per node

Tested Secure Phones

- General Dynamics vIPer, both SIP and PSTN modes
- General Dynamics Sectera (Terminal Adapter)
- L3 STE Telephone Unit in FNBDT mode
- L3 Omni Terminal Adapter

Serial Interface

- IP over serial (HDLC and Cisco HDLC)
- SAToP (Structure-Agnostic TDM over Packet)
- DTE and DCE modes
- Crypto re-synch
- Both access and network side support
- V.35, RS-449 and RS-530 interfaces
- Proven data interworking with Promina® and Promina BBS® platform families

VoIP Signaling

- SIP
- H.323
- BSP (BESTflow Signaling Protocol)

SIP

- Session Border Controller
- Transport Layer Security (TLS) encryption
- MD5 digest authentication
- SIP registrar for basic SIP IP-PBX
- Call Admission control with Static Local session control and management
- RFC2833 and in-band DTMF signaling over RTP
- Extends central SIP services to the non-SIP devices

TDM Signaling

- T1 CAS (E&M, DTMF, MF, Loopstart, R1)
- E1 CAS (MFCR2)
- ISDN – AT&T 4ESS, Nortel DMS-100, Euro ISDN (ETSI 300-102)
- QSIG, NTT (Japan), Harris 20/20, AT&T 5ESS, CoreNet
- National ISDN-2 (NI-2)

Voice Features

- Voice compression: G.723 (5.3 or 6.3 Kbps), G.729A, mG.729AB, G.711, G.726 (16, 24 or 32 Kbps), G.727 (16, 24, or 32 Kbps)
- RTCP
- Automatic call type detection/pass through formvoice/modem/fax
- A-law and u-law encoding
- G.168 echo cancellation (128ms tail size, full duplex)
- Fax: group 3 at 2.4 to 14.4 Kbps
- T.38 real time fax relay
- Voice activity detection, silence suppression, comfort noise generation
- Call progress tone generation – dial tone, busy, ring-back, and congestion

Routing

- Dynamic call routing/link quality management (LQM)
- Multiple routing tables per channel, per port
- Load balancing
- Static IP routing
- DiffServ-based IP QoS support (marking, queuing and scheduling)
- Least cost routing
- Hair-pinning – route backup over PSTN or secondary IP network

Security

- Secured RTP (SRTP)
- Built-in firewall
- SSH
- Smart card authentication
- IPsec
- Extensive ACL support
- Hardened operating system
- Denial of service (DOS) attack mitigation and protection
- Advanced remote access control
- Anti-spoofing
- DNIS, CLID, call type pre-authentication

Configuration/Network Management

- SNMPv2/SNMPv3 – traps, alarms and statistics
- VXbuilder (GUI-based provisioning tool)
- VXwatch (real-time alarm monitoring tool)
- Network logging (syslog)

Interactive Voice Response (IVR)

- Full platform integration
- No utilization of network bandwidth – all .wav files are played locally
- IVR support on all TDM and IP interfaces
- Fully customizable using VXscript

Service Creation via VX 900script

- Fully integrated open-source programming and scripting languag
- Playback and recording of audio to and from a call
- DTMF digit collection
- Complete access to call routing and CDR parameters
- Enables the creation of unique customer applications

Network Timing

- Network Time Protocol
- Local stratum 3 clock as reference
- Redundant timing sources
- Synchronized between serial and T1/E1

Chassis Data

- Rack mount: 19in. (483mm) rack mountable with EIA and telco rack support
- Normal operating conditions: 40F to 104F (5C to 40C)
- AC power: 100 – 240 Volt, 50/60 Hz, 200 Watt
- Humidity: 20% to 80% non-condensing

EMC & Safety Compliance

- Meets EN 300 386, EN55022, EN55024, EN60950
- UL60950, CB

Model	VX 900
T1/E1 Capacity	1-8 T1/E1
FXS Capacity	16
Compressed Call Capacity	192 (T1) / 240 (E1)
Rack Units	1
Dimensions (HxWxD)	1.75 x 17 x 17 (in.) 44.45 x 431.8 x 432 (mm)