Tenor
AX Series

Pulse
www.pulsesupply.com

net™
Network Equipment Technologies

Tenor AX Series VoIP MultiPath Switch

- 8, 16, 24 or 48 Analog line and trunk interfaces
- Up to 48 simultaneous VoIP calls
- Available in MultiPath or Gateway configurations
- Auto-Provisionable
- Survivable
- Unified Communications Proxy (UCP)
- Supported by Remote Management Tools*
- Perfect for small to medium sized businesses, SOHOs and branch offices

The NET Tenor AX Series gives small to medium sized businesses, SOHOs and branch offices with analog voice infrastructures an easy, cost-effective way to capitalize on the power of Voice over IP (VoIP). Tenor offers a complete survivable branch office solution providing support for integrating analog endpoints and the PSTN, while offering backup call processing support for all SIP and analog calls.

ALL NET TENORS INCLUDE THE FEATURES THAT MADE NET THE VOIP MARKET VALUE LEADER

- MultiPath architecture for easy integration with existing voice and data infrastructure, meaning little or no re-programming of the PBX, or upgrades are required and no need for special dialing plans.
- Transparent MultiPath Call Routing to intelligently route calls between the PBX, the PSTN, and the IP network to achieve the best combination of cost and quality. The Tenor can also route calls over IP to reduce costs, and then transparently "hop off" to the PSTN, to reach off-net locations.
- Auto Provisioning interface allows the Tenor Series solution to be deployed worldwide, and configured automatically by acquiring the configuration information from a central server, upon installation. The same auto-provisioning capability allows the Tenor Series to be configured from a VoIP application, thus supporting a single user interface between the VoIP application and the Tenor configuration.
- Unified Communications Proxy (UCP) provides 'Any to Any' connectivity: SIP - SIP, TDM - TDM and SIP - TDM, for easy integration with any network.

THE TENOR SERIES VOIP MULTIPATH SWITCH OFFERS A ROBUST VOIP SWITCHING SOLUTION FOR A VARIETY OF APPLICATIONS

NET's Tenor Series switching/gateway solutions are deployed worldwide to provide highly reliable, easy-to-implement VoIP connectivity in a full range of enterprise and service provider environments. These highly adaptable solutions work with traditional PBXs, legacy analog equipment, IP PBXs and IP phones, and SIP-based communications environments. They are thus ideal for a wide variety of applications including interoffice trunking, integration of IP PBXs with legacy infrastructure, service provider VoIP trunking, call centers, calling card services, call shops and VoIP termination.

THE NET TENOR OFFERS A SURVIVABLE BRANCH OFFICE VOIP ACCESS SOLUTION FOR IP PBXS AND IP CENTREX

All NET Tenors offer a complete SIP-based VoIP solution for the branch office or for service provider customer premise equipment, providing full survivability in the event that access to the central and/or hosted IP PBX is lost. Because it delivers a complete solution in a single device - and because it offers such easy, scalable implementation and management - the Tenor significantly reduces the total lifecycle cost of VoIP deployment, while optimizing service reliability. Tenor also offers PSTN connectivity and legacy equipment integration for both analog and modem based devices.

* Available for an additional cost

- Universal Dial Plan that provides a programmable dial plan so the Tenor Series can be integrated into any network environment.
- PacketSaver™ Technology multiplexing to reduce bandwidth consumption by up to 57% by combining voice packets from multiple calls into a single packet.
- NATAccess™ to allow Tenor Series to operate behind NAT firewalls to translate internal IP addresses into public addresses when a VoIP call is established with an outside party.
- Remote Management for anywhere, anytime remote management even behind NAT firewalls with NET's Remote Management Session Server.

CALL MANAGEMENT FEATURES

- Automatic call type detection: Voice/Modem/Fax
- Answer and Disconnect Supervision
- Trunk group support
- Public and private dial plan support
- User programmable dial plan support
- Forced IP routing and IP port mapping
- Pass-through support for calls to Toll-free, local and special service numbers (ie. emergency services, etc.)
- Automatic appending and stripping of digits to dialed numbers
- Call Detail Records
- Least Cost Routing with external NET VoIP Call Routing Server*
- Type 1 Caller ID/Name Support (Telcordia, ETSI, NTT and DTMF)

TECHNICAL SPECIFICATIONS:

Telephony Specifications:

- Voice algorithms: G.723.1a, G.729ab, G.711
- Auto codec negotiation
- Fax Support: Group III at 2.4, 4.8, 7.2, 9.6, 14.4 Kbps, using industry standard T.38; Super G3 compatible up to 33.6 Kbps
- Modem support with G.711
- Choice of 8, 16, or 24 FXS and/or FXO configurations or a 48 FXO only configuration
- Standard 50 pin RJ-21 Telco connectors
- Coding: A-law, Ì-law
- Enhanced (Carrier Grade) Echo Cancellation: ITU Rec. G168, up to 128 msec tailsize
- Loop Start, Reverse Battery, Battery Disconnect
- Tandem/TDM switching
- Maximum Call Rate: 1,800 calls/hour

IP Network Specifications:

- LAN Interface: 1 Fast Ethernet port (10/100Base-T)
- Standard RJ-45 Interface (IEEE 802.3) for 10Base-T or 100Base-TX connections
- DHCP Client
- QoS Support: IP TOS, DiffServ

VoIP Network Specifications:

- Support for multiple SIP User Agents (RFC3261 compliant endpoint)
- DTMF Signaling via RFC2833 or SIP Info
- SIP Supplementary Services
- SIP Refer Method Support
- Message Waiting Indicator
- IVR/RADIUS server support for AAA with integrated multi-lingual IVR+
- Adaptive Voice Activity Detection (VAD) with Comfort Noise Generation (CNG)
- Adaptive Jitter Buffer
- Packet Loss Concealment
- NATAccess™ - NAT/Firewall
- Security: IP Filtering
- Up to 48 simultaneous VoIP calls
- Support for DNS-SRV
- Survivability - (Option only available in AXM and AXT Series)
  - SIP Outbound Proxy
  - Supports up to 100 SIP endpoints

Configuration / Management:

- Auto-Provisionable
- NET Tenor Series Configuration Manager (GUI) for configuration of remote individual Tenors
- NET Tenor Series Monitor (GUI) for alarm monitoring, call monitoring and CDR monitoring
- Remote access with external Remote Management Session Server
- SNMPv2 Agent
- Command Line Interface

GENERAL SPECIFICATIONS:

Dimensions: 1U High Chassis

W 17 3/8" x H 1 3/4" x D 10 3/4"

W 44.5cm. x H 4.5cm. x D 27.6cm.

- Maximum weight: 10 lbs. (4.55kg.)
- AC Power: 100-240 Volts AC, 50/60 Hz, 60 watts
- Operating temperature: 40° - 104° F (5° - 40° c.)
- Operating humidity: 20% - 80% non-condensing
- Telco: FCC Part 68, TS-016, TBR4, TS-038, CS03
- EMC: FCC Part 15 EN55022, EN55024, EN61000-2-3, EN61000-3-3, AS/NZS3260
- Safety: UL60950, EN60950, AS/NZS60950

Tenor AX Series	MultiPath AXM Series	Station AXG Series	Trunk AXT Series
8 Simultaneous VoIP Calls	8 FXS/8 FXO	8 FXS/0 FXO	0 FXS/8 FXO
16 Simultaneous VoIP Calls	16 FXS/16 FXO	16 FXS/0 FXO	0 FXS/16 FXO
24 Simultaneous VoIP Calls	24 FXS/24 FXO	24 FXS/0 FXO	0 FXS/24 FXO
48 Simultaneous VoIP Calls	N/A	N/A	0 FXS/48 FXO

LEADERS IN NETWORK EVOLUTION