Quintum AFT Tenor AFT200 2 port FXO VoIP Trunking Gateway

Product Name: Quintum AFT Tenor AFT200 2 port FXO VoIP Trunking Gateway
Manufacturer: Quintum Technologies
Model Number: 501-1199

Tenor S (Survivable MultiPath Switch) Offers a Survivable Branch Office VoIP Access Solution for IP PBXs and IP Centrex Quintum's Tenor S line is 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 offer such easy, scalable implementation and management - the Tenor S line significantly reduces the total lifecycle cost of VoIP deployment, while optimizing service reliability. Tenor S also offers PSTN connectivity and legacy equipment integration for both analog and modem based devices. Both versions of Tenor still include the features that made Quintum 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 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 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.
• 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 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 Quintum'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 Quintum 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

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• Fax support: Industry standard T.38 and Group III at 2.4, 4.8, 7.2, 9.6, 14.4 Kbps
• Modem over IP
• Choice of 2, 4, 6 or 8 FXS and/or FXO configurations
• Standard RJ-11 connectors
• Coding: A law, µ 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: 900 calls/hour

IP Network Specifications

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

VoIP Network Specifications

• H.323 v.3 Gateway and Integrated Gatekeeper+
• SIP User Agent (RFC3261 compliant endpoint)
• SIP RFC2833 In-band DTMF signaling
• SIP Supplementary Services
• Message Waiting Indicator
• SIP Refer Method support
• 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 Compensation
• NATAccess™
• Security: IP Filtering
• Up to 8 simultaneous VoIP calls
• Support for DNS Addressing

Price: £405.00