Technical Note

Quintum SIP Feature List

Release P104

for

Tenor AS, AF, AX, BX, DX Series, Call Relay, and CMS Products

Technical Note

Quintum SIP Support for Tenor Products

Tenor AS, AF, AX, BX, DX Series, CMS240 and CMS960 Products

Quintum is continuing to enhance its advanced SIP support for all current models of the Tenor range of Voice over IP (VoIP) Gateways and Multipath switches.

SIP protocol is rapidly becoming the dominant protocol in the VoIP marketplace. Many manufacturers of VoIP endpoints and application devices such as IP PBXs or softswitches have released SIP-based products and there is a growing demand for VoIP gateways and switches to support such devices. In response to this demand, Quintum has added advanced SIP support to its range of Tenor products.

All new Tenor products will now support both H.323 and SIP for VoIP communication. The Tenor is now software upgradeable and configurable to support the long established H.323 protocol and the increasingly popular SIP protocol. SIP-enabled Tenors will be able to accept both H.323- and SIP-based incoming calls and handle them on a call-by-call basis. Users will be able to configure the call signaling preference for outbound calls. The Tenor will then send out calls with signaling configured based on the stated preference.

SIP-enabled Tenors will provide the same rich feature set synonymous with the Tenor name. Features such as the unique MultiPath architecture with intelligent call routing, SelectNet auto-switching PSTN back-up, NATAccess for secure deployment behind NAT Firewalls, and PacketSaver packet multiplexing for bandwidth conservation all remain unchanged. Cost-effective migration, ease of installation, and guaranteed VoIP quality and connectivity continue to be the cornerstone of Tenor installations.

When configured for SIP, the Tenor will act as a SIP User Agent (Endpoint) as defined in IETF RFC3261. Multiple user agents allow for separate agents to be allocated to each SIP call. The Tenor will be able to gateway calls to and from the IP network, and Customer Premise Equipment (CPE) such as phones, PBXs, and FAX machines, or the Public Switched Telephone Network (PSTN). Quintum has tested and verified interoperability with standard compliant CPE devices including SIP phones, VoIP gateways, and terminal adaptors from many other vendors. Quintum's SIP User Agent is also fully compatible with SIP softphone products such as Microsoft (Windows XP) Messenger.

The Tenor SIP User Agent will work in conjunction with an external SIP proxy or redirect server to route and connect calls over SIP-based networks. Quintum has tested and verified interoperability with various commercially available, standard-compliant, SIP proxy servers such as those provided by DynamicSoft, Vovida, and Asterisk. In larger networks, calls can also be routed by using SIP Tenors in conjunction with Quintum's high capacity Call Routing Server.

Recent enhancements include support for supplementary services, registration of local directory numbers (LDN) and hop-off numbers (LAM) to two SIP registrar servers, SIP proxy registration using UDP and TCP, and also call redirection on receipt of a redirect response from external proxy or redirect servers.

A great deal of interoperability testing has also been carried out in association with other vendors to insure seamless operation with softswitches, IP PBX's, SIP servers, and CPE devices.

Quintum's P104 SIP implementation is certified interoperable with the Avaya S8300 Media Server and G350 Media Gateway and 4620 (SIP and H.323) IP phones, BroadSoft BroadWorks softswitch (Access Device), and with Nortel CS1000 (SIP and H.323) IP PBX systems.

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Tenor SIP Feature List (Release P104)

Feature
Software Upgradeable
Configurable for both H.323 **and** SIP Protocols
Configurable Outbound Call Preference (SIP or H.323)
Multiple SIP User Agent Support
IETF RFC3261 Compliant Endpoint
IETF RFC2833 In-band Signaling Support
IETF RFC2976 INFO Method Support
IETF RFC3262 Reliability of Provisional Responses Support
IETF RFC3264 Offer/Answer Model Support
IETF RFC3265 SIP Specific Event Notification Support
IETF RFC3311 UPDATE Method Support
IETF RFC3515 REFER Method Support
IETF RFC3842 Message Waiting Indicator Event Package
IETF RFC3891 "REPLACES" Header Support
IETF RFC3892 REFERRED_BY Mechanism
IETF RFC4028 Session Timers
Nortel Proprietary out-of-band DTMF implementation (SIP INFO Based)
Supplementary Services - Blind Call Transfer
Supplementary Services - Consultative Call Transfer
Supplementary Services - Call Hold/Drop
Supplementary Services - Call Waiting
Call Routing Using DNS or IP Address
Outbound Proxy Support
Security – Accept Calls from Selected Proxies Only
Caller ID and Caller Name Support
Enhanced DN Channel Mapping (Maps DN's to UA's for DID)
Primary/Secondary Proxy Server Support
Intelligent Call Routing via External Proxy
Intelligent Call Routing via Call Routing Server
SIP Proxy Server Connection via UDP (TCP under development)
Primary/Secondary Redirect Server Support
Primary/Secondary Registrar Server Support
MultiPath Switching Architecture
SelectNet Auto-switching PSTN Back-up
PacketSaver Packet Multiplexing
NATAccess NAT/Firewall Traversal
Remote NAT Support
Public, Private, User-Definable, Hop-on, and Hop-off Number Plans
SIP-Specific Debugging and Diagnostic Tools

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SIP Interoperability

Quintum has carried out extensive testing with partners throughout the world to ensure seamless interoperability with as many commercial applications as possible. The list below shows specific devices that have been certified or tested for interoperability. Some products have been certified interoperable by the vendors indicated and others have been tested for interoperability either by Quintum or by other entities. We continuously test interoperability so if you do not see a specific device of interest listed please contact us.

Vendor	Product	Device	Status
Avaya	S8300 Media Server/G350 MGW Communications Manager (R3.1)	IP PBX	Certified
Avaya	4620 (R2.2.2)	SIP Phone	Certified
Avaya	4620 (R2.3.0)	H.323 Phone	Certified
BroadSoft (Access Device)	BroadWorks (Release 13)	Softswitch	Certified
BroadSoft (Network Device)	Broadworks (Release 13)	SoftSwitch	In Progress
Nortel Networks	CS1000 (SIP Release 4.50)	IP PBX	Certified
Nortel Networks	CS1000 (H.323 Release 4.0)	IP PBX	Certified
Aastra/Sansay	480i	SIP Phone	Tested
Cisco Systems	7940 Series	SIP Phone	Tested
Dialexia	Dial-Office	IP PBX	Tested
Dialexia	Dial-Gate	Hosted IP PBX	Tested
Digium	Asterisk	IP PBX	Tested
DynamicSoft		SIP Proxy	Tested
Intel	HMP	SIP Media Server	Tested
IPTel	SER	SIP Proxy	Tested
PingTel	Xpressa	SIP Phone	Tested
PolyCom	Soundpoint IP 301/601	SIP Phone	Tested
SNOM	100/320/360	SIP Phone	Tested
Swissvoice	IP110	SIP Phone	Tested
Telco Systems	AC201N/AC211N	SIP Adaptors	Tested
3Com	3C10402A	SIP Phone	Tested
Vonage		VoIP Service	Tested
Vovida		SIP Proxy	Tested