Q-SIPIQ™

A softswitch with SIP Proxy Server (Q-SIPPS) functionality supporting IP Centrex functions when used with full-featured SIP-compliant IP Phones

A carrier-grade Softswitch for deployment by small and mid-sized ITSP in handling VoIP call traffic originating from SIP-compliant or H323-compliant* endpoints (VoIP gateways or IP Phones or softphone) and Gatekeeper or softswitch for call termination to/through SIP-compliant or H323-compliant* endpoint

Essentially a SIP Proxy Server, SIP-to-H323 Translator*, H323-to-SIP Translator*, RADIUS Client and IP-based IVR incorporated with all the features required for Call Routing, Call Control, Gateways and Port Monitoring, CDR Generation, and Network Security functions to support an ITSP operation with multiple levels of resellers in gateway termination and/or prepaid calling card operation

When integrated with Q-Bill-R and Q-CallShop Servers, it provides the ASP functions for Q-CallShop service at www.qcallshop.com

When integrated with Q-Bill-R and Q-Commerce Servers, it becomes the eCommerce platform for Ecocall (PC to Phone), Ecocard (Prepaid Calling Card) and C2ToIP (Click to Talk over IP) services at www.ecocarrier.net

Together with Q-Bill-R and Q-Manager Servers, it forms a complete solution excellent for building a rapidly growing ITSP business operation

* = optional

Outstanding Attributes

(1) Q-SIPIQ is designed with architecture and functionality that support

- ITSP who wants to offer IP Centrex service to subscribers using SIP-compliant IP Phones or SIP-compliant VoIP gateways to enable them to have PBX functions without having to have a PBX; these functions include Call-Hold, Call Transfer, Call-Forward, Follow-Me, Call Display, and the optional features for Call Conference and Voice Mail; similar to the services offered by www.vonage.com or www.skype.com
- ITSP selling call termination services through multiple-levels of resellers either facility-based or non-facility-based
- ITSP specialized in selling to Residential Markets – using Calling Line Number (CLI) for authentication
- ITSP specialized in selling to Enterprise Markets (corporations) – using CLI for authentication and working with private IP addresses
- VoIP-based Call Back Call Operation
- In-country access through Local Access Number or Toll-Free Access Number – working with DNIS (Dialed Number Identification Service)

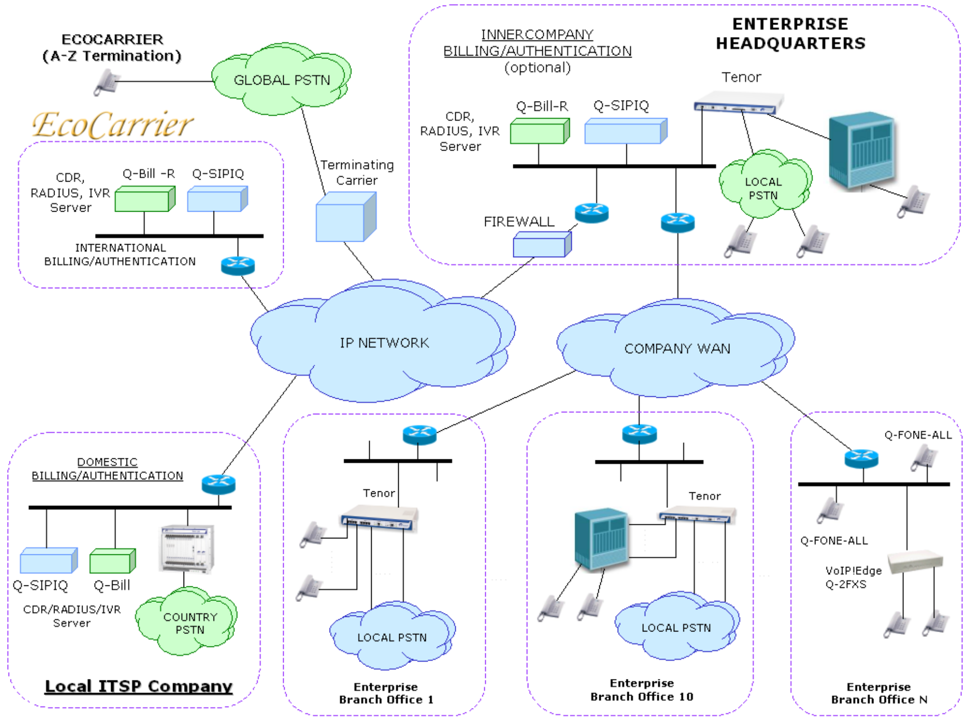
(2) Q-SIPIQ is designed with architecture and functionality that supports multiple carriers for call routing to a city/country dial code with selection by Least Cost Routing according to certain rules such as Day and Time, Network Condition Parameters, and Service Level Agreement

(3) Q-SIPIQ is designed with architecture and functionality that support real-time monitoring of the operation of gateways and their port status

(4) Q-SIPIQ has proven reliable, stable performance in peak-load operation

(5) Q-SIPIQ is priced to be affordable to all ITSP large and small with small threshold/start-up cost and low license fee per incremental concurrent VoIP call session (conversation) for capacity

Other Functional Features



H323-to-SIP Translator (an optional feature) provides a facility to enable Q-SIPIQ to interoperate with H323-compliant endpoints, Gatekeeper or softswitch from which the VoIP call traffic originates.

SIP-to-H323 Translator (an optional feature) provides a functional capability to enable Q-SIPIQ to interoperate with H323-compliant terminating IP Phones, VoIP gateways, Gatekeeper and softswitch.

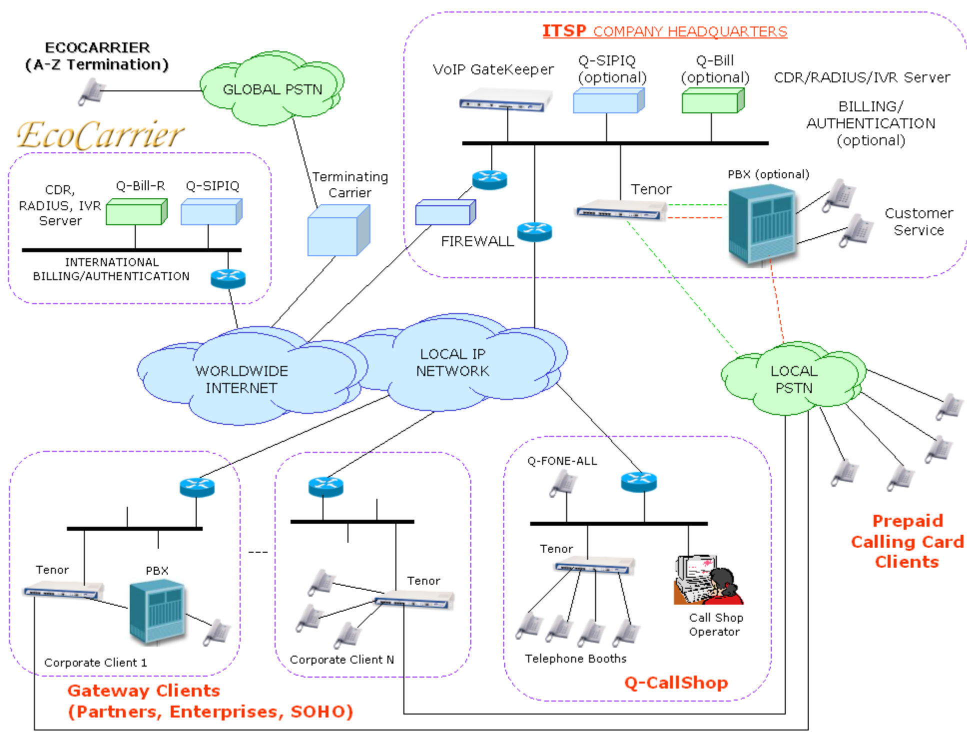
H.323 Proxy included as part of SIP-to-H323 Translator works as a H.323 traffic concentrator passing H.323 traffic through. The signaling (H.225 including FastStart, H.245) is always tunneled. Depending on a selected call destination, it may optionally let all media traffic (RTP/RTCP) for particular calls pass through. The network administrator can control the priority sequence of tunneled media traffic by setting an appropriate value of precedence field within outgoing IP packets.

Address Translation service translates phone numbers (or aliases) to network addresses of destination devices so that calls can be routed across networks.

Admissions Control service authorizes registered endpoints to make calls. The service authorizes/denies network access on a basis of the following criteria: call authorization, bandwidth and gateway availability, matching supported codecs, source IP address, etc.

Bandwidth Control service controls and monitors SIP bandwidth usage in order to ensure optimal performance and provide superior QoS. This service helps to protect other network applications from SIP traffic.

Bandwidth Management controls and limits the number of devices that are allowed to simultaneously use the network. Q-Regime works with Tenor VoIP gateways equipped with Packetsaver option to send traffic to terminating gateways that are not equipped with Packetsaver capability (optional).



Zone Control and Management provides all defined call control services to all registered endpoints in the gatekeeper's zone.

Intelligent Call Routing Mechanism routes a call to the best available gateway. The call routing is based on the following criteria: caller/callee IDs, gateway availability, link characteristics (cost, QoS, available bandwidth).

Call Authorization Mechanism authorizes access to a particular H.323 terminal using H.225.0 signaling. The network administrator can define a flexible security policy for controlling access to/from particular terminals or gateways based on type of service, gateway restrictions, and restricted access during certain periods of time. (applicable where SIP-to-H323 Translator option is in use)

Call Management maintains a list of ongoing H.323 calls and provides an effective mechanism for controlling those calls (displaying call statistics, call termination, call simulation, etc.). (applicable where SIP-to-H323 Translator option is in use)

Billing Information. Q-SIPIQ generates detailed billing records for each call and synchronously stores them in various file formats including text file. Thus, you are not limited to a particular billing system and may use virtually any solution of your choice for processing billing information.

SIP Proxy Server processes SIP requests and responses and provides the primary capabilities for call session management, including SIP proxy or redirect functionality (either stateful or stateless), registrar and location services. The server is compliant with the standards described in RFC 3261. It provides the configuration and control facility for a SIP-compliant IP telephone and multimedia communications system. It supports most of the common features found in an enterprise private branch exchange (PBX).

Dialed Number Translation feature lets the network administrator create and maintain flexible dial plans to suit your application.

Easy Configuration. Q-SIPIQ is equipped with a fast and easy-to-use administration and configuration interface, which facilitates updating system configuration without interruption of service.

Reliable and Fault-Tolerant Service. Q-SIPIQ is designed and implemented as a reliable and fault-tolerant service to ensure a high level of availability. It incorporates local and remote monitors that perform continuous monitoring and take appropriate actions in case of service failure. Additionally, Q-SIPIQ utilizes different messaging facilities so as to notify a system administrator of real or possible service failures.

Supported Protocols
H.323 v.3, including Annex F Fast Start
RTP/RTCP
H.245 v.7
H.225 v.4
T.38 Real Time Fax
SIP RFC 3261
SIP RFC 2833
SIP to H.323 automatic translation (optional)
H323 to SIP automatic translation (optional)

Supported Codecs
G.723.1, G.729AB, G.711 with automatic codec negotiation

Compatibility
EcoFone (QiiQ's softphone)
VoIP!Edge Q-2FXS, Q-4FXS
VoIP!Edge Q-1FXS-1FXO, Q-2FXS-2FXO
VoIP!Edge Q-1FXS+56K
VoIP!Edge Q-4FXO, Q-6FXO
VoIP!Phone Q-WiFi-Fone, Q-FONE-ALL, Q-FONE-XUV, Q-FONE-DU-X, Q-FONE-USB
Tenor Analog ASG200/400, ASM200/400
Tenor Analog AXT800/1600/2400, AXG800/1600/2400, AXE800/1600/2400, AXM800/1600/2400
Tenor Digital DX 2008/2016/2024/2030/4048/4060/6120/8120/2048/2060/4096/4120
Tenor CMS 240, CMS 960
Cisco ATA-186, AS5300, AS5350, AS5400, AS5600 series, AS3600 series
Microsoft NetMeeting
Other H323 v.3-compliant VoIP gateways
Other SIP-compliant or H323-compliant IP Phones

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