PLANET
Networking & Communication

SIP-50

SIP Proxy Server

As an increasing number of carrier deployment and enterprise solutions, the need for direct interconnection between these VoIP networks is becoming more demanding. The PLANET SIP-50 gives enterprises, system integrators and service providers a standalone SIP 2.0 (RFC3261) compliant proxy server for VoIP networks.

Based on the Session Initiation Protocol (SIP), the SIP-50 SIP Proxy provides a smart call authentication, call routing capabilities to maximize network performance in both corporation and Internet voice networks.
The SIP-50 provides the SIP client registration, authentication, and administration for all the SIP compliant endpoints in your network, including gateways, IP phones and PC clients.

The PLANET SIP-50 provides the NAT transversal capability to serve both the public and private network environments. The product detects and configures the network address information of calling parties. It also supports call routing to provide connectivity to calling parties not in the same VoIP network. This advanced call routing algorithm enables the SIP-50 providing load balancing and more flexibility in the VoIP deployment.

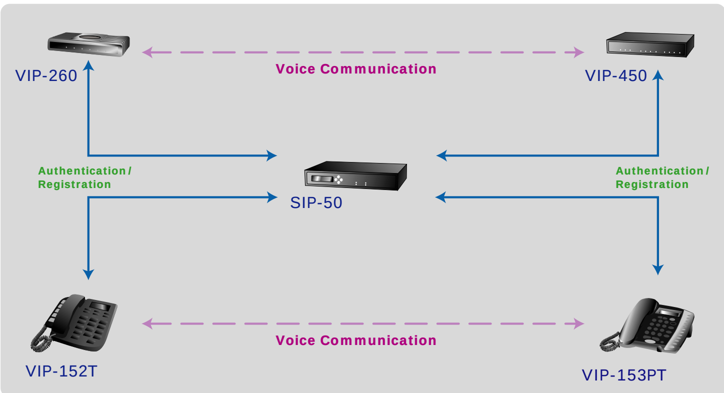
Key Feature

- Compliant with SIP 2.0 standard (RFC3261)
- SIP supplemental service support- Call Waiting, Call Transfer, Call Conferencing support
- Built-in calling prefix routing for inter SIP proxy communications
- Auto NAT user detection, stateful RTP proxy
- External RADIUS Authentication, Authorization and Accounting support
- CDR (Call Detail Record) export
- Real time monitor for active calls/online users
- Built-in internal user authentication for various VoIP applications

Specification

Product	SIP Proxy Server
Model	SIP-50
Ports LAN	1 x 10/100Base-T Ethernet ports
Protocol and standards	
Protocols	RFC 3261, RFC 2976, RFC 3263, RFC 2327, RFC 2833, RFC 3581
SIP Registrar Dynamic Register	SIP Registrar Dynamic Register Registrar Authentication
SIP Outbound Proxy Server	Stateful Proxy Server Support Call-based Authentication MD5 Sequential & Parallel Call Forking Support Inter-Proxy Auto NAT User Detect
Telephony features	
Call processing	Call Transfer (client-based), Call Waiting, Call Hold, 3-Way Call Conference, Call forward (Unconditional, No Answer, Busy), Prefix Call Route
Network	
NAT transversal	Outbound and Inbound NAT traversal support Automatically NAT detection and RTP Proxy NAT Partition support Intelligent RTP Proxy Resource Management
Management	
Interface	Console, SNMP, LCD and Web-based Management Browser-based Real Time Monitor Password Security User Account Manager NTP time synchronization (SNTP V4) Time Zone Support
AAA (Authentication / Authorization / Accounting)	Local Daily CDR, CDR export to FTP server Registrar Authentication via external RADIUS Server
Max system capacity	Max Subscribers Support: 200 Max Concurrent Call: 50 Call Attempt per Seconds: 50 cps Max Concurrent RTP Support for NAT user: 50
Environmental	
Environmental	Temperature: 5~40 Degree C (operating) Humidity: 20 to 90% (non-condensing)
Emission	EMI: FCC part 15, CE / PTT: FCC part 68

Applications



Ordering Information

SIP-50	SIP Proxy Server (1 x RJ-45)

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