SP5250+ Series

4 Ports VoIP Gateway

- RFC 3261 SIP protocol VoIP IAD
- 2 FXS + 2 FXO/ 4FXO and 5 Ethernet ports
- 1 WAN + 2 LAN, RJ-45 10/100/1000 Ethernet
- Interactive Voice Response (IVR)
- T.30 and T.38 compliant
- Line reversal and metering tone (12K/ 16KHz)
- Ethernet switch function with QoS and VLAN
- IGMP Proxy/ Snooping
- Advanced calling features, 3-Way Conference with/ without media server, call parking and more
- WEB based configuration (HTTP/ HTTPs)
- TR-069, TR-104, DHCP Auto Provision
- SNMP V3/ V2c/ V1
- IPv4, IPv6 support

The SP5250+ Series analog VoIP gateways are feature-rich and cost-effective products designated for telecom operators to provide telephony services with security-enhanced capabilities in the modern complex networks.

The SP5250+ Series VoIP Gateway empowers service providers delivering carrier-class IP Centrex service over a Total-IP NGN/ 3GPP IMS infrastructure in a more secured way.

The SP5250+ gateway provides a connection platform between traditional POTS lines and the Internet. Over various broadband technologies including xDSL, HFC, wireless and fiber, the SP5250+ Series carries toll quality voice, fax and data traffic simultaneously in a cost-effective way. In addition, the SP5250+ Series supports intelligent features like long loop, line testing, polarity reversal, caller ID, call transfer, call waiting and 3-way calling. The ideal applications include MTU/ MDU, virtual PBX, IP Centrex, PBX extension and hosted telephony services.

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Model

- SP5250SO+: 2 FXS, 2 FXO, 2 LAN, 1 WAN
- SP5250O+: 4 FXO, 2 LAN, 1 WAN

Voice Feature

- G.722, G.711 a/μ-law, G.729, G.726, G.723.1, GSM 6.10 Full Rate
- DTMF Detection and Generation
- Silence Suppression & Detection
- Comfort Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation (G.165/ G.168)
- Adaptive (Dynamic) Jitter Buffer
- Call progress tone detection (FXO) and generation (FXS)
- Programmable Gain Control
- Inbuilt Local Mixer
- ITU-T V.152 Voice-band Data over IP Networks

SIP Method Support

ACK, BYE, CANCEL, INFO, INVITE, MESSAGE, NOTIFY, OPTIONS, PING, PRACK, PUBLISH, REFER, REGISTER, SUBSCRIBE, UPDATE

SIP Call Features

- Peer to Peer Call
- Call Hold/ Retrieve
- Call Waiting
- Call Pick Up
- Call Park / Retrieve (with SIP Server)
- Call Forward – unconditional, busy, no answer
- Call Transfer – attended, unattended
- Do not Disturb
- Speed Dialing
- Repeat Dialing
- Three-way Calling
- MWI (RFC-3842)
- Hot Line and Warm Line

SIP Call Management

- Support Outbound Proxy
- Support up to three SIP servers
- SIP Registration Failover
- Group Hunting
- Privacy Mechanism /Private Extension to SIP
- Session Timers (Update / Re-invite)
- DNS SRV Support
- Call Types: Voice/ Modem/ FAX
- Call Routing by Prefix Number
- User Programmable Dial Plan
- Toll-Free Support (FXO)
- Automatic Calling Number Manipulation (VoIP & FXO)
- CDR Client
- Manual Peer Table (for P2P calls)
- E.164 Numbering, ENUM support

SIP Account Management

- By port registration
- By device registration (share account)
- Mixed mode (Hunt number for inbound, by port number for outbound)
- Invite with Challenge
- Register by SIP Server IP Address or Domain Name
- Support RFC3986 SIP URI format

Physical Interface

- WAN: 1 x 10/100/1000M Ethernet, auto cross-over, auto speed negotiation, RJ-45
- LAN: 2 x 10/100/1000M Ethernet, auto cross-over, auto speed negotiation, RJ-45
- RJ-11 telephony connectors
- Power jack, Power switch
- Reset button

Telephony Specification

- In-Band DTMF, Out-of-Band DTMF Relay (RFC2833 or SIP INFO)
- DTMF/ PULSE Dial Support
- Caller ID Generation (FXS) and Detection (FXO):
  - DTMF
  - FSK-Bellcore Type 1 & 2
  - FSK-ETSI Type 1 & 2
  - FSK-NTT
  - FSK: Calling Name, Number, Date and Time, VMWI
- FXS metering pulse:
  - Polarity Reversal
  - 12kHZ calling tone
  - 16kHZ calling tone
- Polarity Reverse Detection (FXO) and Generation (FXS)
- T.30 FAX bypass, T.38 Real Time FAX Relay
- Fax and Modem over IP (up to 14,400bps)
- ROH Tone (Receiver Off-Hook Tone @ 480Hz)
- Loop Current Suppression

LED Indicators

- Power, Alarm, Register, Provision, WAN, LAN 1 ~ 2, Phone 1 ~ 2 or Line 1 ~ 4

Accessories

- RJ11 cables
- RJ45 cables
- Power adaptor

General Information

- Dimensions: W 222 x D 145 x H 33 mm (excl. Stand)
- Weight: 450g
- Power: AC 100~240V 50/60Hz input, DC 12V/2A output
- Operating temperature: 0°C ~ 45°C
- Storage temperature: -25°C ~ 75°C
- Operating Humidity: Up to 90% RH, non-condensing

NETWORK FEATURES AND MANAGEMENT

IP Network Specification

- WAN: Static IP, PPPoE, DHCP, PPTP
- Network Protocol Support:
  IP, TCP, UDP, TFTP, FTP, RTP, RTCP, ARP, RARP, ICMP, NTP, SNTP, HTTP, HTTPS, DNS, DNS SRV, Telnet, DHCP Server, DHCP Client, STUN Client, UPnP, IGMP, IGMP snooping, IGMP proxy, RTSP ALG
- NAT Functions
  - Support up to 255 clients
  - Port Forwarding (Virtual Servers)
  - DMZ
  - Port Triggering
- Support IPv4, IPv6
- QoS Support:
  - WAN: DiffServ, IP Precedence
    Priority Queue
    Rate Control
    802.1Q (VLAN Tagging), 802.1P (Priority Tag)
  - LAN: Rate Limit
- DDNS Support
  - Dyndns.org (Dynamic and Custom)

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Network Security Specifications

- PPTP Client
- DIGEST Authentication
- MD5 Encryption
- DoS Protection

Management

- Web Based Configuration
- Auto-provisioning (HTTP/ HTTPS/ TFTP)
- Telnet
- IVR
- FTP/ TFTP/ HTTP Software Upgrade
- Configuration Backup and Restore
- Reset to Default Button
- TR-069, TR104 (optional)
- SNMP V3/ V2c /V1

STANDARD COMPLIANCE

SIP, Voice and FAX Related Standard

- RFC1889 RTP: A Transport Protocol for Real-Time Applications
- RFC2543 SIP: Session Initiation Protocol
- RFC2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC2880 Internet Fax T.30 Feature Mapping
- RFC2976 The SIP INFO Method
- RFC3261 SIP: Session Initiation Protocol
- RFC3262 Reliability of Provisional Responses in Session Initiation Protocol (SIP)
- RFC3263 Session Initiation Protocol (SIP): Locating SIP Servers
- RFC3264 An Offer/Answer Model with Session Description Protocol (SDP)
- RFC3265 Session Initiation Protocol (SIP) - Specific Event Notification
- RFC3311 The Session Initiation Protocol (SIP) UPDATE Method
- RFC3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC3325 Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
- RFC3362 Real-time Facsimile (T.38) - image/ t38 MIME Sub-type
- Registration
- RFC3515 The Session Initiation Protocol (SIP) Refer Method
- RFC3550 RTP: A Transport Protocol for Real-Time Applications.
- July 2003
- RFC3665 Session Initiation Protocol (SIP) Basic Call Flow Examples
- RFC3824 Using E.164 numbers with the Session Initiation Protocol (SIP)
- RFC3841 Caller Preferences for the Session Initiation Protocol (SIP)
- RFC3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
- RFC3891 The Session Initiation Protocol (SIP) "Replaces" Header
- RFC3892 The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC3960 Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- RFC3986 Uniform Resource Identifier (URI): Generic Syntax
- RFC4028 Session Timers in the Session Initiation Protocol (SIP)
- Draft-ietf-sipping-service-examples-08 for call features

Network Related Standard

- RFC318 Telnet Protocols
- RFC791 Internet Protocol
- RFC792 Internet Control Message Protocol
- RFC793 Transmission Control Protocol
- RFC768 User Datagram Protocol
- RFC826 Ethernet Address Resolution Protocol
- RFC959 File Transfer Protocol
- RFC1034 Domain Names - concepts and facilities
- RFC1035 Domain Names - implementation and specification
- RFC1058 Routing Information Protocol
- RFC1157 Simple Network Management Protocol (SNMP)
- RFC1305 Network Time Protocol (NTP)
- RFC1321 The MD5 Message- Digest Algorithm
- RFC1349 Type of Service in the Internet Protocol Suite
- RFC1350 The TFTP Protocol (Revision 2)
- RFC1661 The Point-to-Point Protocol (PPP)
- RFC1738 Uniform Resource Locators (URL)
- RFC2854 The 'text/ html' Media Type
- RFC2131 Dynamic Host Configuration Protocol
- RFC2136 Dynamic Updates in the Domain Name System (DNS UPDATE)
- RFC2327 SDP: Session Description Protocol
- RFC2474 Definition of the Differentiated Services Field (DS Field)
- RFC2516 A Method for Transmitting PPP Over Ethernet
- RFC2616 Hypertext Transfer Protocol - HTTP/1.1
- RFC2617 HTTP Authentication: Basic and Digest Access Authentication
- RFC2637 Point-to-Point Tunneling Protocol
- RFC2766 Network Address Translation - Protocol Translation (NAT-PT)
- RFC2782 A DNS RR for Specifying the location of Services (DNS UPDATE)
- RFC2818 HTTP Over TLS (HTTPS)
- RFC2916 E.164 Number and DNS
- RFC3022 Traditional IP Network Address Translator
- RFC3489 STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)