Advanced VoIP Router

SP5250 Series

Protocol: SIP (RFC3261)

Ethernet: 1WAN 4LAN

Telephony Ports: 4

Telephony Connector: RJ-11

GENERAL FEATURES AND SPECIFICATIONS

Voice Features
- G.722, G.711 a/μ-law, G.729, G.726, G.723.1, GSM 6.10 Full Rate, iLBC 13.3 kbps
- 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
- Local Mixer
- ITU-T V.152 Voice-band Data over IP Networks

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

Telephony Specifications
- In-Band DTMF, Out-of-Band DTMF Relay (RFC2833 or SIP INFO)
- DTMF / PULSE Dialing
- 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
- 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 @ 480 Hz)
- Loop Current Suppression

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
- Server contact format: IP Address or Domain Name
- Support RFC3986 SIP URI format

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

SIP Call Management
- Support Outbound Proxy
- Support up to 3 SIP servers
- SIP Registration Automatic Failover
- Group Hunting
- Privacy Mechanism / Private Extensions to SIP
- Session Timers (Update / Re-invite)
- DNS SRV Support
- Call Types: Voice / Modem / FAX
- Call Routing by Prefix Number
- User Programmable Dial Plan
- By routing Calling Number Manipulation
- Radius CDR Client
- Manual Peer Table (for P2P calls)
- E.164 Numbering, ENUM support

Physical Interface
- WAN : 1 x 10/100/1000M Ethernet, auto cross-over, auto speed negotiation, RJ-45
- LAN : 4 x 10/100/1000M Ethernet, auto cross-over, auto speed negotiation, RJ-45
- RJ-11 telephony connectors
- AC power jack, power switch
- Reset button

LED Indicators
- Power, Provision/Alarm, Register, WAN, LAN1~4, Phone 1~4(or Line1~4 for FXO)

Accessories
- RJ11 cables
- RJ45 cables
- AC Power adaptor
- User Manual in CD

FC CE

Advanced VoIP Router

Ordering Information

Model	Description				
	WAN	LAN	FXS	FXO	PSTN
SP5250S	1	4	4		
SP5250O	1	4		4	
SP5250SO-2S2O	1	4	2	2	
SP5250SO-2S2P	1	4	2		2
SP5250SP-4S4O	1	4	4	4	
SP5250SP-4S4P	1	4	4		4

General Information

Dimensions: W 22.2 cm x H 3.3 cm x D 14.5 cm (excl. Stand)

Weight: 450 g

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

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  document to refer to either the entities claiming the marks and names or their products.

* Specifications, availability and terms of offers may change without notice.

NETWORK FEATURES AND MANAGEMENT

IP Network Specifications

- 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
- 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)

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-069/104 (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 SRV)
- 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)