SP9880 Series

8 Ports VoIP Gateway

- RFC 3261 SIP protocol VoIP IAD
- 8 FXO, 8 FXS, and 5 Ethernet ports
- 1 WAN + 4 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, call parking and more
- WEB based configuration (HTTP/ HTTPs)
- TR-069, TR-104, DHCP Auto Provision
- SNMP V3/ V2c/ V1
- Busy Tone candence auto learning/ detection

SP9880 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. SP9880 VoIP Gateway empowers service providers delivering carrier- class IP Centrex service over a Total-IP NGN/ 3GPP IMS infrastructure in a more secured way.

SP9880 VoIP Gateway provides a connection platform between traditional POTS lines and the Internet. Over various broadband technologies including xDSL, HFC, wireless and fiber, the SP9880 carries toll quality voice, fax and data traffic simultaneously in a cost-effective way.

In addition, SP9880 VoIP Gateway 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

- SP9880-8S: 8 FXS, 4 LAN, 1 WAN
- SP9880-8O: 8 FXO, 4 LAN, 1 WAN
- SP9880-8S8O: 8 FXS, 8 FXO, 4 LAN, 1 WAN
- SP9880-8S8P: 8 FXS, 8 PSTN, 4 LAN, 1 WAN

Voice Feature

- G.722, G.711a/μ-law, G.729A/B, G.726, G.723.1
- 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)
- Auto or 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 (SIP Server Required)
- 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 four SIP servers
- SIP Registration Automatic 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 Support
- 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 x10/100/1000M 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
- 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/ Detection:
  - 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 Generation (FXS)
- T.30 FAX bypass, T.38 Real-Time FAX Relay
- FXS Line test and diagnostics with visual alarm indication
  - Inward Self Test:
    Loopback – codec
    Loopback – analogue
    SLIC DC power voltage
    Tip/ Ring DC feed
    Ringer
  - Outward Test (GR909 Standard):
    REN
    Phone Line disconnected
    H.F. DC Voltage (Hazardous and foreign DC Voltage)
    H.F. AC Voltage (Hazardous and foreign AC Voltage)
    Tip/ Ring Short
- Failsafe mechanism: FXS auto or manual relay to PSTN through hardware relay or internal PCM Bus while Network, Service or power failure occurs
- Emergency Number Table (PSTN)
- Modem over IP up to 14,400 bps
- ROH Tone (Receiver Off-Hook Tone @ 480 Hz)
- Loop Current Suppression

LED Indicators

- Power, Provision/Alarm, Register, WAN, LAN 1 ~ 4, Phone off-hook 1 ~ 8/ Phone Ch Alarm 1 ~ 8, Line 1 ~ 8

Accessories

- RJ11 cables
- RJ45 cables
- Power adaptor

General Information

- Dimensions: W 302 x D 179 x H 45 mm
- Weight: 1200g
- Power Source: 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
- It’s available to be installed on 19” shelf

NETWORK FEATURES AND MANAGEMENT

IP Network Specification

- WAN: Static IP, PPPoE, DHCP, PPTP
- Network Protocol Support:
  IP, TCP, UDP, TFTP, FTP, RTP, RTCP, XR, ARP, RARP, ICMP, NTP, SNTP, HTTP, HTTPS, DNS, DNS SRV, Telnet, DHCP Server, DHCP Client, SNTU Client, UPnP, IGMP, IGMP snooping, IGMP proxy, RTSP ALG, SIP ALG

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- NAT Functions
  - Support up to 255 clients
  - Port Forwarding (Virtual Servers)
  - DMZ
  - Port Triggering
- Support IPv4, IPv6 future upgradeable
- 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)
- Route/ Bridge mode support

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 (option)
- SNMP V3/ V2c /V1
- Two level (Admin/User) Web authority login
- System Information (Port /Registration status)
- System Log Client (General/ CDR/ SIP)

Firewall

- Port filter
- IP filter
- MAC filter
- URL filter
- IP White List

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)