Cisco SPA941 4-Line IP Phone with 1-Port Ethernet
Cisco Small Business IP Phones

Advanced, Affordable, Feature-Rich IP Phone for the Home Office and Business

Highlights

- Full-featured four-line business-class IP phone with display
- Connects directly to an Internet telephone service provider or to an IP PBX
- Speakerphone, caller ID, call hold, transfer, conferencing, and more
- Easy installation with secure remote provisioning, as well as menu-based and web-based configuration

Comprehensive Interoperability and SIP-Based Feature Set

Based on the Session Initiation Protocol (SIP), the Cisco® SPA941 4-Line IP Phone (Figure 1) has been tested to ensure comprehensive interoperability with equipment from voice over IP (VoIP) infrastructure leaders, enabling service providers to quickly roll out competitive, feature-rich services to their customers.

With hundreds of features and configurable service parameters, the Cisco SPA941 addresses the requirements of traditional business users while building on the advantages of IP telephony. Features such as easy station moves and shared line appearances (across local and geographically dispersed locations) are just some of the many advantages of the SPA941.

Carrier-Grade Security, Provisioning, and Management

The Cisco SPA941 uses standard encryption protocols to perform secure remote provisioning and unobtrusive in-service software upgrades. Secure remote provisioning tools include detailed performance measurement and troubleshooting features, enabling network providers to deliver high-quality support to their subscribers. Remote provisioning also saves service providers the time and expense of managing, preloading, and reconfiguring customer premises equipment.

Figure 1.   Cisco SPA941 4-Line IP Phone with 1-Port Ethernet

Telephony Features

- Four voice lines and up to eight call appearances
- Pixel-based display: 128 x 64 monochrome graphical liquid crystal display (LCD)
- Line status: active line indication, name and number
- Menu-driven user interface
- Shared line appearance**
- Speakerphone
- Call hold
- Music on hold**
- Call waiting
- Caller ID name and number
- Outbound caller ID blocking
- Call transfer: attended and blind
- Three-way call conferencing with local mixing
- Multiparty conferencing via external conference bridge**
- Automatic redial of last calling and last called numbers
- On-hook dialing
- Call pickup: selective and group**
- Call park and unpark**
- Call swap
- Call back on busy
- Call blocking: anonymous and selective
- Call forwarding: unconditional, no answer, on busy
- Hot line and warm line automatic calling
- Call logs (60 entries each): made, answered, and missed calls
- Redial from call logs
- Personal directory with auto-dial (100 entries)
- Do not disturb (callers hear line busy tone)
- Digits dialed with number auto-completion
- Uniform Resource Identifier (URI) (IP) dialing support (vanity numbers)
- On-hook default audio configuration (speakerphone and headset)
- Multiple ring tones with selectable ring tone per line
- Ability to call number using name: directory matching or via caller ID
- Subsequent incoming calls with calling name and number
- Date and time with intelligent daylight savings support
- Call duration and start time stored in call logs
- Call timer
- Name and identity (text) displayed at startup
- Distinctive ringing based on calling and called number

- 10 user-downloadable ring tones

- Speed dialing, eight entries

- Configurable dial/numbering plan support

- Intercom**

- Group paging**

- Network Address Translation (NAT) Traversal, including STUN support

- DNS SRV and multiple A records for proxy lookup and proxy redundancy

- Syslog, debug, report generation, and event logging

- Secure call encrypted voice communication support

- Built-in web server for administration and configuration with multiple security levels

- Automated remote provisioning, multiple methods; up to 256-bit encryption: (HTTP, HTTPS, Trivial File Transfer Protocol [TFTP])

- Option to require administrator password to reset unit to factory defaults

** Feature requires support by call server.

Hardware Features

- Pixel-based display: 128 x 64 monochrome graphical LCD

- Dedicated illuminated buttons for:
  - Audio mute on/off
  - Headset on/off
  - Speakerphone on/off

- Four soft-key buttons

- Four-way rocking directional knob for menu navigation

- Voicemail message waiting indicator light

- Voicemail message retrieval button

- Dedicated hold button

- Settings button for access to feature, setup, and configuration menus

- Volume control rocking up/down knob to control handset, headset, speaker, and ringer

- Standard 12-button dialing pad

- High-quality handset and cradle

- Built-in high-quality microphone and speaker

- Headset jack: 2.5 mm

- Ethernet LAN: 10BASE-T RJ-45

- 5 VDC universal (100–240V) switching power adapter

- LED test function

Regulatory Compliance

- FCC (Part 15, Class B) , CE Mark, A-Tick

Security Features

- Password-protected system, preset to factory default
- Password-protected access to administrator and user-level features
- HTTPS with factory-installed client certificate
- HTTP digest: encrypted authentication via MD5 (RFC 1321)
- Up to 256-bit Advanced Encryption Standard (AES) encryption

Documentation

- Quick-start installation and configuration guide
- User guide
- Administration guide
- Provisioning guide (for service providers only)

Package Contents

- Cisco SPA941 IP Phone, handset, and stand
- Handset cord
- 5V power adapter
- RJ-45 Ethernet cable
- Quick-start installation and configuration guide

Specifications

Table 1 gives specifications for the Cisco SPA941 4-Line IP Phone with 1-Port Ethernet. Table 2 compares the SPA941 with other Cisco Small Business IP Phones.

Table 1.   Specifications for the Cisco SPA941 4-Line IP Phone with 1-Port Ethernet

Note: Many features are programmable within a defined range or list of options. Please see the SPA Administration Guide for details. The target configuration profile is uploaded to the SPA941 at the time of provisioning.	
Data networking	- MAC address (IEEE 802.3) - IPv4 - Internet Protocol v4 (RFC 791) - ARP - Address Resolution Protocol - DNS - A record (RFC 1706), SRV record (RFC 2782) - DHCP client - Dynamic Host Configuration Protocol (RFC 2131) - ICMP - Internet Control Message Protocol (RFC 792) - TCP - Transmission Control Protocol (RFC 793) - UDP - User Datagram Protocol (RFC 768) - RTP - Real Time Protocol (RFC 1889, 1890) - RTCP - Real Time Control Protocol (RFC 1889) - DiffServ - Differentiated Services (RFC 2475) - ToS - Type of service (RFC 791, 1349) - VLAN tagging 802.1p/Q - Layer 2 quality of service (QoS) - SNTP - Simple Network Time Protocol (RFC 2030)

Voice gateway	• SIP version 2 (RFC 3261, 3262, 3263, 3264)
	• SIP proxy redundancy: Dynamic via DNS SRV, A records
	• Reregistration with primary SIP proxy server
	• SIP support in NAT networks (including STUN)
	• SIPFrag (RFC 3420)
	• Secure (encrypted) calling via prestandard implementation of Secure RTP
	• Codec name assignment
	• Voice algorithms:
	  ◦ G.711 (A-law and μ-law)
	  ◦ G.726 (16/24/32/40 kbps)
	  ◦ G.729 A
	  ◦ G.723.1 (6.3 kbps, 5.3 kbps)
	• Dynamic payload support
	• Adjustable audio frames per packet
	• Dual-tone multifrequency (DTMF), in-band and out-of-band (RFC 2833) (SIP INFO)
	• Flexible dial plan support with interdigit timers
	• IP address/URI dialing support
	• Call progress tone generation
	• Jitter buffer: adaptive
	• Frame loss concealment
	• Voice activity detection (VAD) with silence suppression
	• Attenuation/gain adjustments
	• Voicemail waiting indicator (VMWI) via NOTIFY, SUBSCRIBE
	• Caller ID support (name and number)
	• Third-party call control (RFC 3725)

Provisioning, administration, and maintenance	• Integrated web server for web-based administration and configuration • Telephone keypad configuration via display menu/navigation • Automated provisioning and upgrade via HTTPS, HTTP, TFTP • Asynchronous notification of upgrade availability via NOTIFY • Nonintrusive in-service upgrades • Report generation and event logging • Statistics transmitted in BYE message • Syslog and debug server records: configurable per line
Physical Interfaces	• One 10BASE-T RJ-45 Ethernet port (IEEE 802.3) • Handset: RJ-9 connector • Built-in speakerphone and microphone • Headset 2.5-mm port
Power supply	• Switching type (100–240V) automatic • DC input voltage: +5 VDC at 2.0A maximum • Power adapter: 100–240V 50–60 Hz (26–34 VA) AC input
Indicator lights/LED	• 4 line key buttons with LED • Speakerphone on/off button with LED • Headset on/off button with LED • Mute button with LED • Message waiting indicator LED
Dimensions (W x H x D)	7.68 x 6.30. x 7.09 in. (195 x 160 x 180 mm)
Unit weight	2.15 lb (0.9752 kg)
Operating temperature	41º ~ 113ºF (5º ~ 45ºC)
Storage temperature	−13º ~ 185ºF (−25º ~ 85ºC)
Operating humidity	10% to 90% noncondensing
Storage humidity	10% to 90% noncondensing

Table 2.   Cisco Small Business IP Phone Comparison Chart

Model	Voice Lines	Ethernet Ports	High-Resolution Graphical Display	PoE Support
SPA901	1	1	No	No
SPA921	1	1	Yes	No
SPA922	1	2	Yes	Yes
SPA941	4	1	Yes	No
SPA942	4	2	Yes	Yes
SPA962	6	2	Color	Yes

Cisco Limited Warranty for Cisco Small Business Series Products

This Cisco Small Business product comes with a 2-year limited hardware warranty with return to factory replacement. In addition, Cisco offers software application updates for bug fixes and telephone technical support at no charge for the first 12 months following the date of purchase. To download software updates, go to: http://www.cisco.com/go/smallbiz.

Product warranty terms and other information applicable to Cisco products are available at http://www.cisco.com/go/warranty.

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