Cisco SPA509G IP Phone

Product Name: Cisco SPA509G IP Phone
Manufacturer: Cisco Systems
Model Number: SPA-509G

Cisco SPA 509G IP Phone
The Cisco SPA509G is part of Cisco's Small Business Pro series and has been fully tested to ensure complete interoperability with other VoIP equipment. The successful testing means that service providers are able to launch feature rich services to customers. The SPA509G is a 12-Line IP Phone with a 2-Port Switch, Power over Ethernet and an LCD Display.
Cisco SPA 509G Key Features

- Cisco HD Voice for unsurpassed voice clarity and enhanced speaker quality
- Monochrome backlit display for ease of use, aesthetics, and on-screen applications
- Connects directly to an Internet telephone service provider or to an IP private branch exchange (PBX)
- Easy installation and highly secure remote provisioning, as well as menu-based and web-based configuration
- Full-featured 12-line business-class IP phone supporting Power over Ethernet (PoE)
- Supports up to two Cisco SPA500S Expansion Module, adding up to 64 additional buttons
- Supports both Session Initiation Protocol (SIP) and Smart Phone Control Protocol (SPCP) with the Cisco Unified Communications 500 Series for Small Business

Comprehensive Interoperability and SIP-Based Feature Set
With hundreds of features and configurable service parameters, the Cisco SPA509G IP Phone 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 Cisco SPA 509G IP Phone.
The Cisco SPA 509G IP Phone with Programmable Keys also supports productivity-enhancing features such as VoiceView Express and Cisco XML applications when used with the Cisco Unified Communications 500 Series in SPCP mode.
Carrier-Grade Security, Provisioning, and Management

Cisco SPA 509G IP Phone Technical Specifications
Telephony Features

- 12 voice lines
- Twelve Independent SIP Registrations
- Line status: active line indication, with 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
• Anonymous caller blocking
• 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
• Called number with directory name matching
• Ability to call number using name: directory matching or via caller ID
• Subsequent incoming calls show calling name and number
• Date and time with support for intelligent daylight savings
• Call 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, Simple Traversal of UDP Through NATs (STUN) support
• DNS SRV and multiple A records for proxy lookup and proxy redundancy
• Syslog, debug, report generation, and event logging
• Highly secure call encrypted voice communications 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

Hardware Features

• Pixel-based display: 128 x 64 monochrome LCD graphical display with backlight
• Dedicated illuminated buttons for: Audio mute on/off, Headset on/off, Speakerphone on/off
• 4-way rocking directional knob for menu navigation
• Voicemail message waiting indicator (VMWI) light
• Voicemail message retrieval button
• Dedicated hold button
• Settings button for access to feature, setup, and configuration menus
• Volume control rocking up/down knob controls handset, headset, speaker, ringer

• Standard 12-button dialing pad
• High-quality handset and cradle
• Built-in high-quality microphone and speaker
• Headset jack: 2.5 mm
• LED test function
• Two Ethernet LAN ports with integrated Ethernet switch: 10/100BASE-T RJ-45
• 802.3af-compliant PoE
• Optional 5 VDC universal (100-240V) switching; power supply is ordered separately (Cisco PA100)

Data Networking

• MAC address (IEEE 802.3)
• IPv4 (RFC 791)
• Address Resolution Protocol (ARP)
• DNS: A record (RFC 1706), SRV record (RFC 2782)
• Dynamic Host Configuration Protocol (DHCP) client (RFC 2131)
• Internet Control Message Protocol (ICMP) (RFC 792)
• TCP (RFC 793)
• User Datagram Protocol (UDP) (RFC 768)
• Real-Time Transport Protocol (RTP) (RFC 1889, 1890)
• Real-Time Control Protocol (RTCP) (RFC 1889)
• Differentiated Services (DiffServ) (RFC 2475)
• Type of service (ToS) (RFC 791, 1349)
• VLAN tagging 802.1p/Q: Layer 2 quality of service (QoS)
• Simple Network Time Protocol (SNTP) (RFC 2030)
• Cisco Discovery Protocol (CDP)
• Link Layer Discovery Protocol (LLDP)

Voice Gateway

• SIP version 2 (RFC 3261, 3262, 3263, 3264)
• SPCP with the Cisco Unified Communications 500 Series
• 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 SRTP
• Codec name assignment
• Voice algorithms: G.711 (A-law and µ-law), G.726 (16/24/32/40 kbps), G.729 A, G.722
• 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
• Comfort Noise Generation (CNG)
• Voice activity detection (VAD) with silence suppression
• Attenuation/gain adjustments
• VMWIVoicemail waiting indicator, via NOTIFY, SUBSCRIBE
• Caller ID support (name and number)
• Third-party call control (RFC 3725)

Provisioning, Administration and Maintenance

- Integrated web server provides 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

Power Supply

- Power supply is optional and is purchased separately
- Models: Cisco PA100-NA, PA100-EU, PA100-UK, PA100-AU
- DC output voltage: +5 VDC at 2.0A maximum
- Switching power adapter: 100-240V 50-60 Hz AC input

Physical Interfaces

- Two 10/100BASE-T RJ-45 Ethernet ports (IEEE 802.3)
- Handset: RJ-9 connector
- Built-in speakerphone and microphone
- Headset 2.5mm jack

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
- SIP over Transport Layer Security (TLS)
- Secure Real-Time Transport Protocol (SRTP)

Package Contents

- Cisco SPA 509G IP Phone with Programmable Keys, handset, and stand
- Handset cord
- RJ-45 Ethernet cable
- Quick-Start Guide
- CD

Physical

- Body Dimensions (W x H x D): 8.42 x 8.35. x 1.73 in. (214 x 212 x 44 mm)
- Unit Weight: 2.00 lb (0.9 kg)
- Operating Temperature: 32º ~ 104ºF (0º ~ 40ºC)
- Storage Temperature: -4º ~ 158ºF (-20º ~ 70ºC)
- Operating Humidity: 5% to 95% noncondensing
- Storage Humidity: 5% to 95% noncondensing

Regulatory Compliance

- FCC (Part 15, Class B)

- CE Mark
- A-Tick
- C-Tick
- Telepermit
- UL
- CB

Price: £103.90

Options available for Cisco SPA509G IP Phone :

Power Supply Required

Required (+£9.00), Not Required.