



Product Name: Cisco SPA508G IP Phone

Manufacturer: Cisco Systems Model Number: SPA-508G

Please Note: The Cisco SPA508G IP Phone is PoE (Powered-Over-Ethernet) and does not include a mains Power Supply. If mains power is required, this can be added via the drop down menu above.

Cisco SPA 508G IP Phone

The Cisco SPA508G is an 8-line IP Phone with PoE, a 2-Port Switch and a Full Duplex Speakerphone. The SPA508G has been rigorously tested to make certain of full compatibility with equipment from VoIP infrastructure leaders and means service providers to quickly deliver competitive, feature-rich services to their customers.

Cisco SPA508G IP Phone Key Features

- آذًا Full-featured 8-line business-class IP phone supporting Power over Ethernet (PoE)
- ī¿½ Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- � Cisco HD Voice sound
- าั¿½ Monochrome backlit display for ease of use, aesthetics, and on-screen applications
- � Full Duplex Speakerphone
- ï¿1/2 Headset port
- � Call transfer, call waiting, call conferencing
- าั¿½ Supports up to two Cisco SP500S Expansion Module, adding up to 64 additional buttons

Carrier-Grade Security, Provisioning, and Management

The Cisco SPA 508G uses standard encryption protocols to perform highly secure remote provisioning and unobtrusive in-service software upgrades. 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.

Comprehensive Interoperability and SIP-Based Feature Set

With hundreds of features and configurable service parameters, the Cisco SPA508G 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 SPA508G. The SPA 508G IP Phone 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.

Note: Please refer to the Cisco VoIP Detailed Comparison Table for more information. Cisco SPA508G IP Phone Technical Specifications

Telephony Features

- � Eight voice lines
- ï¿1/2 Eight independent SIP Registrations
- า๊¿½ Line status: active line indication, with name and number
- ï¿1/2 Menu-driven user interface
- � Shared line appearance
- ï¿1/2 Speakerphone
- � Call hold
- ï¿1/2 Music on hold
- � Call waiting



- � Caller ID name and number
- ï¿1/2 Outbound caller ID blocking
- ï¿1/2 Call transfer: attended and blind
- า๊¿½ Three-way call conferencing with local mixing
- īപ് Multiparty conferencing via external conference bridge
- ī¿½ Automatic redial of last calling and last called numbers
- ï¿⅓ On-hook dialing
- � Call pickup: selective and group
- � Call park and unpark
- � Call swap
- � Call back on busy
- � Call blocking: anonymous and selective
- تز½ Call forwarding: unconditional, no answer, on busy
- า๊¿1/2 Hot line and warm line automatic calling
- าั¿1/2 Call logs (60 entries each): made, answered, and missed calls
- � Redial from call logs
- آذِ Personal directory with auto-dial (100 entries)
- ï¿⅓ Do not disturb
- � Digits dialled with number auto-completion
- � Anonymous caller blocking
- آذِيًّ Uniform Resource Identifier (URI) (IP) dialling support (vanity numbers)
- تزير On-hook default audio configuration (speakerphone and headset)
- า๊¿½ Multiple ring tones with selectable ring tone per line
- เช่น Called number with directory name matching
- าั¿½ Ability to call number using name: directory matching or via caller ID
- ī¿½ Subsequent incoming calls show calling name and number
- าั¿½ Date and time with support for intelligent daylight savings
- ï¿1/2 Call start time stored in call logs
- � Call timer
- � Name and identity (text) displayed at startup
- ī స్ట్రి Distinctive ringing based on calling and called number
- � 10 user-downloadable ring tones
- � Speed dialling, eight entries
- � Configurable dial/numbering plan support
- ï¿⅓ Intercom
- ï¿⅓ Group Paging
- าั¿½ Network Address Translation (NAT) Traversal, including Simple Traversal of UDP Through NATs (STUN) support
- � DNS SRV and multiple A records for proxy lookup and proxy redundancy
- تزار Syslog, debug, report generation, and event logging
- า๊¿½ Highly secure call encrypted voice communications support
- از الله 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])
- ī¿½ Option to require administrator password to reset unit to factory defaults

Hardware Features

- ī¿1/2 Pixel-based display: 128 x 64 monochrome LCD graphical display with backlight
- เว้า Dedicated illuminated buttons for: Audio mute on/off, Headset on/off, Speakerphone on/off
- ī సైస్ట్ 4-way rocking directional knob for menu navigation
- า๊¿½ Voicemail message waiting indicator (VMWI) light
- ï¿1/2 Voicemail message retrieval button
- � Dedicated hold button
- ī¿1/2 Settings button for access to feature, setup, and configuration menus



- 12.1/2 Volume control rocking up/down knob controls handset, headset, speaker, ringer
- ï¿1/2 Standard 12-button dialling pad
- ï¿1/2 High-quality handset and cradle
- า๊¿1/2 Built-in high-quality microphone and speaker
- ï¿1/2 Headset jack: 2.5 mm
- ï¿1/2 LED test function
- ī¿1/2 Two Ethernet ports with integrated Ethernet switch: 10/100BASE-T RJ-45
- ï¿1/2 802.3af-compliant PoE
- าั¿½ Optional 5 VDC universal (100-240V) switching; power supply is ordered separately (Cisco PA100)

Data Networking

- ï¿1/2 MAC address (IEEE 802.3)
- � IPv4 (RFC 791)
- � Address Resolution Protocol (ARP)
- آذِر DNS: A record (RFC 1706), SRV record (RFC 2782)
- าั¿½ Dynamic Host Configuration Protocol (DHCP) client (RFC 2131)
- تزير Internet Control Message Protocol (ICMP) (RFC 792)
- � TCP (RFC 793)
- ï¿1/2 User Datagram Protocol (UDP) (RFC 768)
- تزير Real-Time Transport Protocol (RTP) (RFC 1889, 1890)
- i¿½ 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)
- آذِ1/2 Simple Network Time Protocol (SNTP) (RFC 2030)
- � Cisco Discovery Protocol (CDP)
- � Link Layer Discovery Protocol (LLDP)

Voice Gateway

- ï¿1/2 SIP version 2 (RFC 3261, 3262, 3263, 3264)
- าั¿½ SPCP with the Cisco Unified Communications 500 Series
- īస్ట్ SIP proxy redundancy: dynamic via DNS SRV, A records
- � Reregistration with primary SIP proxy server
- تزار SIP support in NAT networks (including STUN)
- � SIPFrag (RFC 3420)
- ï¿1/2 Secure (encrypted) calling via SRTP
- � Codec name assignment
- ϊ¿½ 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
- آذًا Dual-tone multifrequency (DTMF), in-band and out-of-band (RFC 2833) (SIP INFO)
- 17.1/2 Flexible dial plan support with interdigit timers
- ï¿1/2 IP address/URI dialling support
- � Call progress tone generation
- تزير Jitter buffer: adaptive
- � Frame loss concealment
- � Comfort Noise Generation (CNG)
- า๊ะ่½ Voice activity detection (VAD) with silence suppression
- � Attenuation/gain adjustments
- า๊¿½ VMWIVoicemail waiting indicator, via NOTIFY, SUBSCRIBE
- � Caller ID support (name and number)
- � Third-party call control (RFC 3725)



Provisioning, Administration, and Maintenance

- าั¿½ Integrated web server provides web-based administration and configuration
- � Telephone keypad configuration via display menu/navigation
- า๊ะ่น Automated provisioning and upgrade via HTTPS, HTTP, TFTP
- า๊¿½ Asynchronous notification of upgrade availability via NOTIFY
- � Nonintrusive in-service upgrades
- � Report generation and event logging
- ï¿⅓ Statistics transmitted in BYE message
- า๊¿½ Syslog and debug server records: configurable per line

Power Supply

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

Physical Interfaces

- Ti21/2 Two 10/100BASE-T RJ-45 Ethernet ports (IEEE 802.3)
- ï¿1/2 Handset: RJ-9 connector
- า๊¿½ Built-in speakerphone and microphone
- ï¿1/2 Headset 2.5mm jack

Security Features

- � Password-protected system, preset to factory default
- រ៉េ¿½ Password-protected access to administrator and user-level features
- تز½ HTTPS with factory-installed client certificate
- าั¿½ HTTP digest: encrypted authentication via MD5 (RFC 1321)
- าั¿½ Up to 256-bit Advanced Encryption Standard (AES) encryption
- � SIP over Transport Layer Security (TLS)
- า๊¿½ Secure Real-Time Transport Protocol (SRTP)

Package Contents

- تزارد Cisco SPA 508G IP Phone, handset, and stand
- ï¿1/2 Handset Cord
- � RJ-45 Ethernet cable
- � Quick-Start Guide
- CD ½زï

Physical

- � Body Dimensions (W x H x D): 8.42 x 8.35. x 1.73in. (214 x 212 x 44 mm)
- � Unit Weight: 2.00 lb (0.9 kg)
- آذِنا Operating Temperature: 32º ~ 104ºF (0º ~ 40ºC)
- تزير Storage Temperature: -4º ~ 158ºF (-20º ~ 70ºC)
- า๊¿½ Operating Humidity: 5% to 95% noncondensing
- آزٰ 1/2 Storage Humidity: 5% to 95% noncondensing

Regulatory Compliance



� FCC (Part 15, Class B)
� CE Mark
� A-Tick
� C-Tick
� Telepermit
� UL
� CB

Price: £120.00

Options available for Cisco SPA508G IP Phone :

Power Supply Required

Not Required, Cisco PSU PA100 (+£13.00).