



Product Name: Cisco SPA303G IP Phone Manufacturer: -Model Number: SPA-303G

Please Note: This product has been discontinued. Please see the Cisco SPA501G IP Phone.

Please Note: The Cisco SPA303G 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 SPA303G IP Phone

The Cisco SPA303G is an IP phone that is perfect for users who require simplicity and affordability, for example, both business and home office use. The Cisco SPA 303 features 3-way conferencing, call waiting and call transfer and is compatible with most IP systems. Cisco SPA-303G Key Features

ï¿1/2 3-line IP phone

- i¿1/2 Pixel-based display: 128 x 64 monochrome graphical liquid crystal display (LCD)
- ï¿1/2 Compatible with most IP systems
- ï¿1/2 2 Ethernet ports
- ï¿1/2 3-way conferencing, call waiting, call transfer

Comprehensive Interoperability and SIP-Based Feature Set

With hundreds of features and configurable service parameters, the Cisco SPA 303 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 303 IP Phone.

The Cisco SPA 303 IP phone can also be used with productivity-enhancing features such as VoiceView Express, and Cisco XML applications when interfacing with the Cisco Unified Communications 500 Series in SPCP mode.

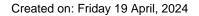
Carrier-Grade Security, Provisioning, and Management

The Cisco SPA 303 IP Phone 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.Note: Please refer to the Cisco VoIP Detailed Comparison Table for more information. Cisco SPA 303 IP Phone Technical Specifications

Telephony Features

ï¿1/2 Three voice lines

- ï¿1/2 Pixel-based display: 128 x 64 monochrome graphical liquid crystal display (LCD)
- ï¿1/2 Line status: active line indication, name and number
- ï¿1/2 Menu-driven user interface
- ï¿1/2 Shared line appearance
- ï¿1/2 Speakerphone
- � Call hold
- � Music on hold
- ï¿1∕₂ Call waiting
- ï¿1/2 Caller ID name and number
- ï¿<sup>1</sup>/<sub>2</sub> Outbound caller ID blocking





- ï¿1/2 Call transfer: attended and blind
- ï¿1/2 Three-way call conferencing with local mixing
- i¿½ Multiparty conferencing via external conference bridge
- ï¿1/2 Automatic redial of last calling and last called numbers
- ï¿1/2 On-hook dialing
- $\ddot{\imath} ¿ \frac{1}{2}$  Call pickup: selective and group
- ï¿1/2 Call park and unpark
- � Call swap
- ï¿1/2 Call back on busy
- ï¿1/2 Call blocking: anonymous and selective
- ï¿1/2 Call forwarding: unconditional, no answer, and on busy
- ï¿1/2 Hot line and warm line automatic calling
- ï¿1/2 Call logs (60 entries each): made, answered, and missed calls
- ï¿1⁄2 Redial from call logs
- ï¿1/2 Personal directory with auto-dial (100 entries)
- � Do not disturb
- i¿1/2 Digits dialled with number auto-completion
- ï¿1/2 Anonymous caller blocking
- ï¿<sup>1</sup>/<sub>2</sub> Support for Uniform Resource Identifier (URI) (IP) dialling (vanity numbers)
- ï¿1/2 On-hook default audio configuration (speakerphone and headset)
- $\ddot{\iota}_{2}^{1/2}$  Multiple ring tones with selectable ring tone per line
- ï¿1/2 Called number with directory name matching
- ï¿1/2 Ability to call number using name: directory matching or via caller ID
- ï¿1/2 Subsequent incoming calls show calling name and number
- ï¿1/2 Date and time with support for intelligent daylight savings
- ï¿1/2 Call duration and start time stored in call logs
- ï¿1∕₂ Call timer
- ï¿1/2 Name and identity (text) displayed at startup
- ïزئ Distinctive ringing based on calling and called number
- ï¿1/2 10 user-downloadable ring tones
- ï¿1/2 Speed dialling, eight entries
- ï¿1/2 Configurable dial/numbering plan support
- � Intercom
- ï¿1∕₂ Group Paging
- تزائر Network Address Translation (NAT) traversal, including Serial Tunnel (STUN) support
- $\ddot{\imath}_{2} ^{1\!\!2}$  DNS SRV and multiple A records for proxy lookup and proxy redundancy
- � Syslog, debug, report generation, and event logging
- i¿1/2 Support for highly secure encrypted voice communications
- ï¿1/2 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])
- i¿1/2 Option to require administrator password to reset unit to factory defaults
- Hardware Features
- ï¿1/2 Pixel-based display: 128 x 64 monochrome LCD graphical display
- ïزائ Dedicated illuminated buttons for: Audio mute on/off, Headset on/off, Speakerphone on/off
- i¿1/2 Four-way rocking directional knob for menu navigation
- ï¿1/2 Voicemail message waiting indicator light
- ï¿1/2 Voicemail message retrieval button
- ï¿1/2 Dedicated hold button
- ï¿1/2 Settings button for access to feature, setup, and configuration menus

**Data Networking** 



ï¿1/2 MAC address (IEEE 802.3)

iزئ IPv4 (RFC 791)

- ï¿1/2 Address Resolution Protocol (ARP)
- ï¿1/2 DNS: A record (RFC 1706), SRV record (RFC 2782)
- ï¿1/2 Dynamic Host Configuration Protocol (DHCP) client (RFC 2131)
- ï¿1/2 Internet Control Message Protocol (ICMP) (RFC 792)
- � TCP (RFC 793)
- ï¿1/2 User Datagram Protocol UDP (RFC 768)
- � Real Time Protocol RTP (RFC 1889, 1890)
- ï¿1/2 Real Time Control Protocol (RTCP) (RFC 1889)
- ï¿1/2 Real Time Control Protocol Extended Report (RTCP-XR) (RFC 3611)
- ï¿1/2 Differentiated Services (DiffServ) (RFC 2475)
- ï¿1/2 Type of service (ToS) (RFC 791, 1349)
- ï¿1/2 VLAN tagging 802.1p/Q: Layer 2 quality of service (QoS)
- تزن<sup>1</sup>/<sub>2</sub> Simple Network Time Protocol (SNTP) (RFC 2030)

Voice Gateway

ï¿1/2 SIP version 2 (RFC 3261, 3262, 3263, 3264)

- $\ddot{i}_{2}$  SPCP with the Cisco Unified Communications 500 Series
- ïزئ SIP proxy redundancy: dynamic via DNS SRV, A records
- ï¿1/2 Re-registration with primary SIP proxy server
- ï¿1/2 SIP support in NAT networks (including STUN)
- ï¿1/2 SIPFrag (RFC 3420)
- i¿½ Highly secure (encrypted) calling via Secure Real-Time Transport Protocol (SRTP)
- � SIP/TLS
- ï¿1/2 Codec name assignment
- � Voice algorithms: G.711 (A-law and µ-law), G.726 (16/24/32/40 kbps), G.729 AB, G.722
- $\ddot{i}_{2}$  Dynamic payload support
- ï¿1/2 Adjustable audio frames per packet
- ïز1/2 Dual-tone multifrequency (DTMF), in-band and out-of-band (RFC 2833) (SIP INFO)
- ï¿1/2 Flexible dial plan support with interdigit timers
- ï¿1/2 IP address/URI dialling support
- ï¿1/2 Call progress tone generation
- ï¿1/2 Jitter buffer: adaptive
- ï¿1/2 Frame loss concealment
- i¿1/2 Voice activity detection (VAD) with silence suppression
- ï¿1/2 Attenuation/gain adjustments
- ï¿1/2 Message waiting indicator (MWI) tones
- ï¿1/2 Voicemail waiting indicator (VMWI), via NOTIFY, SUBSCRIBE
- ï¿<sup>1</sup>/<sub>2</sub> Caller ID support (name and number)
- ï¿1/2 Third-party call control (RFC 3725)

Provisioning, Administration and Maintenance

ï¿1/2 Integrated web server provides web-based administration and configuration

- ï¿1/2 Telephone keypad configuration via display menu/navigation
- ïز1/2 Automated provisioning and upgrade via HTTPS, HTTP, TFTP
- i¿1/2 Asynchronous notification of upgrade availability via NOTIFY
- ï¿<sup>1</sup>/<sub>2</sub> Nonintrusive in-service upgrades
- ï¿1/2 Report generation and event logging
- ï¿1/2 Statistics transmitted in BYE message
- ï¿⅓ RTCP-XR
- ïزئ Syslog and debug server records: configurable per line



#### Power Supply

� Switching type (100-240V) automatic � DC input voltage: +5 VDC at 1.0A maximum

**Physical Interfaces** 

i¿½ Two 10/100BASE-T RJ-45 Ethernet ports (IEEE 802.3)
i¿½ Handset: RJ-9 connector
i¿½ Built-in speakerphone and microphone
i¿½ Headset 2.5-mm port

**Security Features** 

� Password-protected system, preset to factory defaults
� 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

Package Contents

� Cisco SPA 303 IP phone, handset, and stand
� Handset Cord
� RJ-45 Ethernet cable
� Power adapter
� Quick installation guide
� CD
Physical

� Dimensions (W x H x D): 8.66 x 7.80. x 1.18 in. (220 x 198 x 30 mm)
� Unit Weight: 1.50 lb ( 0.68kg)
� Operating Temperature: 32º ~ 113ºF (0º ~ 40ºC)
� Storage Temperature: -13º ~ 185ºF (-20º ~ 70ºC)
� Operating Humidity: 5% to 95% noncondensing
� Storage Humidity: 5% to 95% noncondensing
Regulatory Compliance

i¿½ FCC (Part 15, Class B) i¿½ UL i¿½ CE Mark i¿½ A-Tick

Price: £52.33

Options available for Cisco SPA303G IP Phone :

#### **Power Supply Required**



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