

## Cisco SPA 301 1-Line IP Phone

### Cisco Small Business

Basic, Affordable, IP Phone for Business or Home Office

#### Highlights

- Basic 1-line business-class IP phone
- Connects directly to an Internet telephone service provider or to an IP private branch exchange (PBX)
- Easy installation and secure remote provisioning, as well as web-based configuration
- Supports both Session Initiation Protocol (SIP) and Smart Phone Control Protocol (SPCP) with the Cisco® Unified Communications 500 Series

**Figure 1.** Cisco SPA 301 1-Line IP Phone



#### Comprehensive Interoperability and SIP-Based Feature Set

Based on SIP, the Cisco SPA 301 1-Line IP Phone has been tested to help 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 SPA 301 addresses the requirements of traditional business users while building on the advantages of IP telephony. Features such as easy station moves and shared line appearance (across local and geographically dispersed locations) are just some of the many advantages of the SPA 301.

## Carrier-Grade Security, Provisioning, and Management

The Cisco SPA 301 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.

### Telephony Features

- One voice line
- Music on hold\*
- Call waiting
- Outbound caller ID blocking\*\*
- Call transfer: attended and blind\*\* ( SIP Only )
- Three-way call conferencing with local mixing
- Multiparty conferencing via external conference bridge
- Automatic redial of last calling and last called numbers
- Call pickup: selective and group\*
- Call swap
- Call back on busy\*\*
- Call blocking: anonymous and selective\*\*
- Call forwarding: unconditional, no answer, and on busy\*\*
- Hot line and warm line automatic calling
- Digits dialed with number auto-completion
- Anonymous caller blocking\*\*
- Support for Uniform Resource Identifier (URI) (IP) dialing (vanity numbers)
- Multiple ring tones with selectable ring tone per line
- Call duration and start time stored in call logs in web GUI
- 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 Serial Tunnel (STUN) support
- DNS SRV and multiple A records for proxy lookup and proxy redundancy
- Syslog, debug, report generation, and event logging
- Support for highly secure encrypted voice communications
- 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.

\*\*Feature activated via feature code

### Hardware Features

- Voicemail message waiting indicator light
- Voicemail message retrieval button
- Volume control
- Redial Button
- Flash Button
- Standard 12-button dialing pad
- High-quality handset
- One Ethernet WAN port 10/100BASE-T RJ-45
- 5 VDC universal (100–240V) switching power adaptor included

### Regulatory Compliance

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

### 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

### Documentation

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

### Package Contents

- Cisco SPA 301 IP phone, handset
- Handset cord
- RJ-45 Ethernet cable
- Power adapter
- Quick installation guide
- CD

## Specifications

Table 1 provides specifications for the Cisco SPA 301 1-Line IP Phone.

**Table 1.** Specifications for the Cisco SPA 301 1-Line IP Phone  
**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 SPA 301 at the time of provisioning.

Description	Specification
<b>Data networking</b>	<ul style="list-style-type: none"> <li>• MAC address (IEEE 802.3)</li> <li>• IPv4 (RFC 791)</li> <li>• Address Resolution Protocol (ARP)</li> <li>• DNS: A record (RFC 1706), SRV record (RFC 2782)</li> <li>• Dynamic Host Configuration Protocol (DHCP) client (RFC 2131)</li> <li>• Internet Control Message Protocol (ICMP) (RFC 792)</li> <li>• TCP (RFC 793)</li> <li>• User Datagram Protocol (UDP) (RFC 768)</li> <li>• Real Time Protocol (RTP) (RFC 1889, 1890)</li> <li>• Real Time Control Protocol (RTCP) (RFC 1889)</li> <li>• Real Time Control Protocol – Extended Report (RFC 3611)</li> <li>• Differentiated Services (DiffServ) (RFC 2475)</li> <li>• Type of service (ToS) (RFC 791, 1349)</li> <li>• VLAN tagging 802.1p/Q: Layer 2 quality of service (QoS)</li> <li>• Simple Network Time Protocol (SNTP) (RFC 2030)</li> </ul>
<b>Voice gateway</b>	<ul style="list-style-type: none"> <li>• SIP version 2 (RFC 3261, 3262, 3263, 3264)</li> <li>• SPCP with the Cisco Unified Communications 500 Series</li> <li>• SIP proxy redundancy: dynamic via DNS SRV, A records</li> <li>• Re-registration with primary SIP proxy server</li> <li>• SIP support in NAT networks (including STUN)</li> <li>• SIPFrag (RFC 3420)</li> <li>• Highly secure (encrypted) calling via Secure Real-Time Transport Protocol (SRTP)</li> <li>• SIP/TLS</li> <li>• Codec name assignment</li> <li>• Voice algorithms: <ul style="list-style-type: none"> <li>◦ G.711 (A-law and <math>\mu</math>-law)</li> <li>◦ G.726 (16/24/32/40 kbps)</li> <li>◦ G.729 AB</li> <li>◦ G.722</li> </ul> </li> <li>• Dynamic payload support</li> <li>• Adjustable audio frames per packet</li> <li>• Dual-tone multifrequency (DTMF), in-band and out-of-band (RFC 2833) (SIP INFO)</li> <li>• Flexible dial plan support with interdigit timers</li> <li>• IP address/URI dialing support</li> <li>• Call progress tone generation</li> <li>• Jitter buffer: adaptive</li> <li>• Frame loss concealment</li> <li>• Voice activity detection (VAD) with silence suppression</li> <li>• Attenuation/gain adjustments</li> <li>• Message waiting indicator (MWI) tones</li> <li>• Voicemail waiting indicator (VMWI), via NOTIFY, SUBSCRIBE</li> <li>• Caller ID support (name and number)</li> <li>• Third-party call control (RFC 3725)</li> </ul>
<b>Provisioning, administration, and maintenance</b>	<ul style="list-style-type: none"> <li>• Integrated web server provides web-based administration and configuration</li> <li>• Automated provisioning and upgrade via TFTP, HTTP or HTTPS</li> <li>• Asynchronous notification of upgrade availability via NOTIFY</li> <li>• Nonintrusive in-service upgrades</li> <li>• Report generation and event logging</li> <li>• Statistics transmitted in BYE message</li> <li>• RTCP-XR</li> <li>• Syslog and debug server records: configurable per line</li> </ul>

Description	Specification
<b>Power supply</b>	<ul style="list-style-type: none"> <li>Switching type (100–240V) automatic</li> <li>DC input voltage: +5 VDC at 1.0A maximum</li> </ul>
<b>Physical interfaces</b>	<ul style="list-style-type: none"> <li>One 10/100BASE-T RJ-45 Ethernet port (IEEE 802.3)</li> <li>Handset: RJ-9 connector</li> </ul>
<b>Indicator lights/LED</b>	<ul style="list-style-type: none"> <li>Message waiting indicator LED</li> </ul>
<b>Dimensions (W x H x D)</b>	7.76 x 3.82 x 1.77 inches (194 x 97 x 45 mm)
<b>Unit weight</b>	1.7 lb ( 0.7711 kg)
<b>Operating temperature</b>	32° ~ 113°F (0° ~ 40°C)
<b>Storage temperature</b>	–13° ~ 185°F (–20° ~ 70°C)
<b>Operating humidity</b>	5% to 95% noncondensing
<b>Storage humidity</b>	5% to 95% noncondensing

Table 2 compares the SPA 301 with other Cisco Small Business 500 Series IP Phones.

**Table 2.** Cisco Small Business 500 and 300 Series IP Phones Comparison

Model	Voice Lines	Ethernet Ports	High-Resolution Graphical Display	Power over Ethernet (PoE) Support
SPA 301G	1	1	No	No
SPA 303G	3	2	Yes	No
SPA 501G	8	2	No	Yes
SPA 502G	1	2	Yes	Yes
SPA 504G	4	2	Yes	Yes
SPA 508G	8	2	Yes	Yes
SPA 509G	12	2	Yes	Yes
SPA 525G	5	2	Color	Yes

Tables 3 and 4 provide ordering information for the Cisco SPA 301 and accessories.

**Table 3.** Ordering Information

Part Number	Description
SPA301-G1	Cisco SPA 301G, North America power adapter
SPA301-G2	Cisco SPA 301G, Europe power adapter
SPA301-G3	Cisco SPA 301G, UK power adapter
SPA301-G4	Cisco SPA 301G, Australia power adapter
CON-SBS-SVC1	3-year Cisco Small Business Support Service

**Table 4.** Optional Accessories

Part Number	Description
MB100	Wall-mount brackets for SPA 300, SPA 500, CP 500, and SPA 900 Series