IP Voice Services
Device Overview

Polycom SoundPoint IP 331

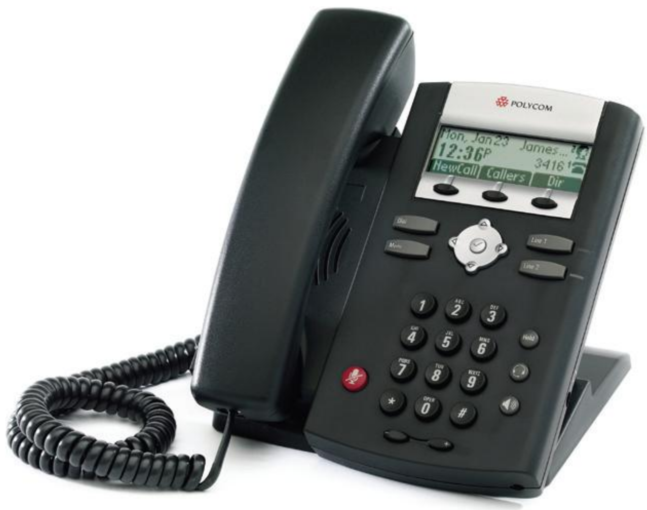


Figure 1: Polycom SoundPoint IP 331 2-Line

Highlights

- Enterprise-grade entry-level IP Phone
- HTTPS secure provisioning
- Full-featured 2-line business-class IP phone
- Integrated PoE support (IEEE 802.af)
- Full-duplex speakerphone with Acoustic Clarity Technology
- 102 x 33 pixel graphical LCD
- 15 dedicated Hard keys, 3 context-sensitive soft keys
- Certified SIP call control

Overview

The SoundPoint IP 331 is a two-line SIP phone that delivers superb sound quality as well as a wide range of supported business telephony features. The SoundPoint IP 331 phone, with its dual-port 10/100 Ethernet switch for LAN and PC connection, presents a cost-effective solution for cubicle workers as well as call centre operators.

Features Overview

Superb Sound Quality

The SoundPoint 331 phone features a full-duplex IEEE 1329 Type 1-compliant speakerphone with Polycom's legendary Acoustic Clarity Technology that delivers excellent sound quality and enables noise-free and echo-free conversations that are as natural as being there.

Efficient Installation

The SoundPoint IP 331 phones are engineered to make installation, configuration, and upgrades as simple and efficient as possible. Built-in IEEE 802.3af PoE circuitry and a dual-port Ethernet switch enable flexible deployment options and savings on cabling expenses.

Enterprise-Grade Feature Set

The SoundPoint IP 331 phone delivers through an intuitive user interface a full feature set encompassing both traditional business telephony features such as call hold, park, pick-up, transfer, and three-way local conferencing, and more advanced capabilities such as shared call appearance and call control toolbar integration.

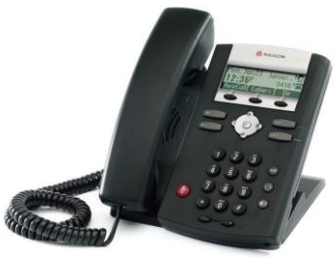
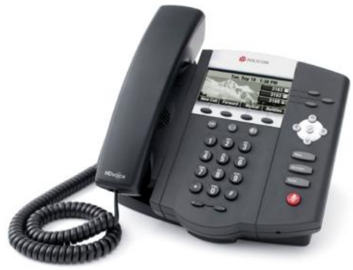
Secure Provisioning

The SoundPoint IP 331 phone provides encrypted provisioning information using HTTPS to secure communication between the user and the service provider. The Polycom secure remote provisioning software provides a highly secure mechanism for configuration and the device software upgrades.

Availability

The Polycom SoundPoint IP 331 is available to order pre-configured from TheVoipShop Products Shop at : www.thevoipshop.co.uk

IP Voice Services
Polycom Device Comparison Sheet

Polycom SoundPoint IP Phones	Polycom SoundPoint IP 331	Polycom SoundPoint IP 450	Polycom SoundPoint IP 650
			
Application Target	Cubicle / Common Area	Cubicle	Executive Office
Telephony Keys	2	3	6 (12 with Exp. Module)
Line/Speed Dial Keys	2	3	6 (48 with Exp. Module)
Display	102 X 33 pixel graphical	256 X 116 pixel backlit graphical	320 X 160 pixel backlit graphical
HTTPS Secure Provisioning	Yes	Yes	Yes
Mute, Handset, and Hands Free	Yes	Yes	Yes
Context-Sensitive Soft Keys	3	4	4
Display / Menu Navigation Keys	5	5	4
Busy Lamp Field (BLF) Functionality	Yes	Yes	Yes
Polycom HD Voice Technology	No	Yes	Yes
Full Duplex Speakerphone with Acoustic Clarity Technology	Yes	Yes	Yes
Expansion Module	No	No	Up to 3 Backlit Expansion modules
Power Over Ethernet	Yes	Yes	Yes
Two Port 10/100 Mbps Ethernet switch	Yes	Yes	Yes
SIP (RFC 3261)	Yes	Yes	Yes

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Polycom® SoundPoint IP® 321 and 331

Enterprise-grade SIP phones for remarkable value Entry-level IP phones with excellent sound quality, an enterprise-grade feature set and business telephony features.

The SoundPoint IP 321 and 331 are two-line SIP phones that deliver superb sound quality as well as a wide range of supported business telephony features. The SoundPoint IP 321 phone has a single 10/100 Ethernet port and is designed for common areas, such as lobbies, hallways and break rooms, as well as various wall-mounted deployments. The SoundPoint IP 331 phone, with its dual-port 10/100 Ethernet switch for LAN and PC connection, presents a cost-effective solution for cubicle workers as well as call centre operators who use a "hard" phone in conjunction with a "soft" client running on the PC.

Superb Sound Quality
The SoundPoint IP 321 and 331 phones feature a full-duplex IEEE 1329 Type 1-compliant speakerphone with Polycom's legendary Acoustic Clarity Technology that delivers excellent sound quality and enables noise-free and echo-free conversations that are as natural as being there.

Enterprise-Grade Feature Set[1]
The SoundPoint IP 321 and 331 phones deliver through an intuitive user interface a full feature set encompassing both traditional business telephony features such as call hold, park, pick-up, transfer and three-way local conferencing, and more advanced capabilities such as shared call/bridged line appearance, hosts built-in XML microbrowser and distinctive call treatment.

Efficient installation and provisioning
The SoundPoint IP 321 and 331 phones are engineered to make installation, configuration and upgrades as simple and efficient as possible. The phones' standard base stand can be reversed to become a wall mount, eliminating the need for a separate accessory. Built-in IEEE 802.3af PoE circuitry and a dual-port Ethernet switch (SoundPoint IP 331 phone only) enable flexible deployment options and savings on cabling expenses.

Make Great Things Happen with Polycom SoundPoint IP 321 and 331
In today's Internet driven world, the ability to conduct real time communication and collaboration has become critical to an organisation's survival. As the market leader in voice, video, data and Web solutions, our award-winning IP telephony and conference technology makes it easy for people to interact and maximise productivity over any network, in just about any environment, anywhere around the globe. That's why more organisations worldwide use and prefer Polycom IP telephony and application solutions. Because when people work together, great things happen. See how you too can achieve great things with the Polycom SoundPoint IP 321 and 331 desktop phones.

Benefits

- **Excellent Sound Quality**
  Polycom Acoustic Clarity Technology enables crystal-clear simultaneous hands-free conversations as natural as being there
- **Enterprise-Grade Feature Set[1]**
  Two lines, support of shared line presence, 3-way local conferencing and built-in XML microbrowser
- **Efficient Installation and Provisioning**
  Remote, zero-touch provisioning with support of a variety of servers
- **Broad and Robust Interoperability**
  Certified to interoperate with a broad array of SIP call control platforms to enable open choices and innovations while simplifying provisioning, management and support.

Polycom® SoundPoint IP® 321 and 331 Specifications

Lines (Directory Numbers)
- Up to 2 lines with up to 2 calls per line

Display
- 102 x 33 pixel graphical LCD

Feature Keys
- 3 context-sensitive "soft" keys
- 2 line keys with bi-colour (red/green) LED
- 2 feature keys ("Menu" and "Dial")
- 4-way navigation key cluster with centre "Select" key
- 2 volume control keys
- Dedicated hold key
- Dedicated headset key
- Dedicated hands-free speakerphone key
- Dedicated microphone mute key
- Headset compatibility
- Dedicated 2.5-mm headset port compatible with most monaural mobile phone headsets

Hearing aid compatibility
- Compliant with ADA Section 508 Recommendations: Subpart B 1194.23 (all)
- Hearing aid compatible (HAC) handset for
- magnetic coupling to approved HAC hearing aids
- Compatible with commercially-available TTY adapter equipment

Audio Features
- Full-duplex hands-free speakerphone with Polycom
- Acoustic Clarity Technology
- Type 1 compliant with IEEE 1329 full duplex
- standards
- Frequency response - 300 Hz - 3300 Hz for handset, headset and hands-free speakerphone modes
- Codecs: G.711 µ/A and G.729A (Annex B)
- Individual volume settings with visual feedback for
- each audio path
- Voice activity detection
- Comfort noise fill
- DTMF tone generation / DTMF event RTP payload
- Low-delay audio packet transmission
- Adaptive jitter buffers
- Packet loss concealment
- Acoustic echo cancellation
- Background noise suppression

Call handling features[1]
- Shared call / bridged line appearance
- Flexible line appearance (one or more line keys
- can be assigned for each line extension)
- Distinctive incoming call treatment / call waiting
- Call timer
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Local three-way conferencing
- One-touch speed dial, redial
- Call waiting
- Remote missed call notification
- Intercom
- Automatic off-hook call placement
- Do not disturb function

Other Features
- Interoperability with Microsoft LCS 2005 for telephony and presence[3]
  - Compatibility with Microsoft Office Communicator and Windows® Messenger 5.1 Clients
- Enabled for Polycom Productivity Suite
- Local feature-rich GUI
- Time and date display
- User-configurable contact directory and call history (missed, placed and received)
- Wave file support for call progress tones
- Unicode UTF-8 character support. Multilingual user interface encompassing Chinese, Danish, Dutch, English (Canada/US/UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, Swedish

Protocol Support
- IETF SIP (RFC 3261 and companion RFCs)

Network and provisioning
- SoundPoint IP 331 – two-port 10/100 Mbps Ethernet switch
- SoundPoint IP 321 – single 10/100 Mbps Ethernet port
- Manual or dynamic host configuration protocol (DHCP) network setup
- Time and date synchronisation using SNTP
- FTP / TFTP / HTTP / HTTPS server-based central
- provisioning for mass deployments
- Provisioning and call server redundancy supported
- Web portal for individual unit configuration
- QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3
- TOS and DSCP
- Network Address Translation (NAT) support for static configuration and "Keep-Alive" SIP signalling
- RTCP support (RFC 1889)
- Event logging
- Syslog
- Local digit map
- Hardware diagnostics
- Status and statistics reporting

Security[1]
- Transport Layer Security (TLS)
- Encrypted configuration files
- Digest authentication
- Password login
- Support for URL syntax with password for boot server
- HTTPS secure provisioning
- Support for signed software executables
- Power
- Built-in, auto-sensing IEEE 802.3af Power over Ethernet (Class 1)
- External universal input AC adapter (optional4;24V DC @ 500mA)

Approvals
- FCC Part 15 (CFR 47) Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- AS/NZS CISPR 22 Class
- VCCI Class B
- EN55024 Class B
- EN61000-3-2; EN61000-3-3; EN-61000-6-1
- ROHS compliant
- Anatel
- GOST
- C-Tick
- CCC

Safety
- CE Mark
- EN 60950-1
- IEC 60950-1
- NRTL
- CAB/CSA-C22.2 No. 60950-1-03
- AS/NZS 60950-1

Operating Conditions
- Temperature: +10 to 40°C (+50 to 104°F)
- Relative humidity: 20% to 85%, non-condensing

Storage Temperature
- -40 to +70°C (-40 to +160°F)

SoundPoint IP 331/321 comes with:
- SoundPoint IP 331/321 console
- Handset with handset cord
- Base stand
- Network (LAN) cable
- Quick start guide
- Product registration card

Size
- 6.7 in x 5.7 in x 6.9 in x 1.4 in
- (17 cm x 14.5 cm x 17.5 cm x 3.5 cm)

Weight

Phone weight: 1.37 lb (0.625 kg)

Part Numbers / UPC Codes
- SoundPoint IP 331
- 2200-12365-025/610807694694 for all markets
- SoundPoint IP 321
- 2200-12360-025/610807690276 for all markets

Unit Box Dimensions / Weight
- 10 in x 4.2 in x 11.6 in (W x H x D)
- (25 cm x 10.5 cm x 29.5 cm)
- 3 lb 4 oz (1.49 kg)

Master Carton Quantity
- Ten

Country of Origin
- Thailand

Warranty
- 1 year

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Polycom SoundPoint IP 650

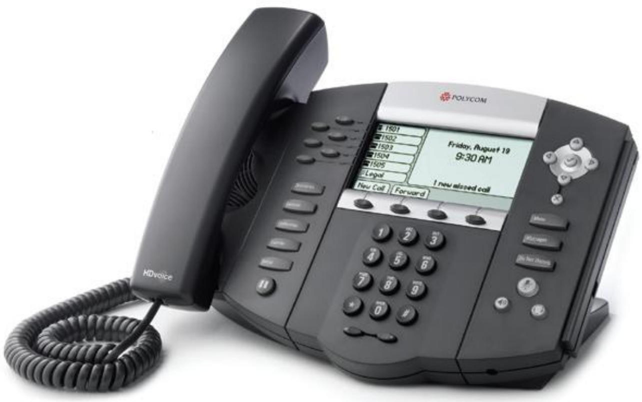

Figure 1: Polycom SoundPoint IP 650 6-Line

Highlights

- Enterprise-grade IP Phone
- HTTPS secure provisioning
- Full-featured 6-line business-class IP phone
- Integrated PoE support (IEEE 802.af)
- Polycom HD Voice*
- 320 x 160 pixel backlit grayscale graphical LCD
- 26 dedicated Hard keys, 4 context-sensitive soft keys
- Certified SIP call control
- Dedicated RJ-9 headset port

Overview

Designed to appeal to both executive users who require advanced features and who need multiple line support, the Polycom SoundPoint IP 650 sets the standard for high-performance IP phones. With its high-resolution, graphical backlit display, dual-port 10/100 Ethernet Switch, and Polycom HD Voice, the SoundPoint IP 650 sets new standards for the executive-range SIP desktop phone.

Features Overview

Unsurpassed Voice Quality and Clarity

The SoundPoint IP 650 features Polycom HD Voice*, bringing life-like richness and clarity to every call. Polycom HD Voice incorporates wideband audio for over twice the voice clarity, Polycom's patented Acoustic Clarity Technology for crystal-clear, noise and echo-free sound, plus best-in-class system design for high-fidelity voice reproduction.

Efficient Installation

The SoundPoint IP 650 phones are engineered to make installation, automatic configuration, and upgrades as simple and efficient as possible. Built-in IEEE 802.3af PoE circuitry and a dual-port Ethernet switch enable flexible deployment options and savings on cabling expenses.

Enterprise-Grade Feature Set

The SoundPoint IP 650 phone deliver through an intuitive user interface a full feature set encompassing both traditional business telephony features such as call hold, park, pick-up, transfer, and three-way local conferencing, and more advanced capabilities such as shared call appearance and call control toolbar integration.

Secure Provisioning

The SoundPoint IP 650 phone provides encrypted provisioning information using HTTPS to secure communication between the user and the service provider. The Polycom secure remote provisioning software provides a highly secure mechanism for configuration and device software upgrades.

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IP Voice Services
Device Overview

Polycom SoundPoint IP 450

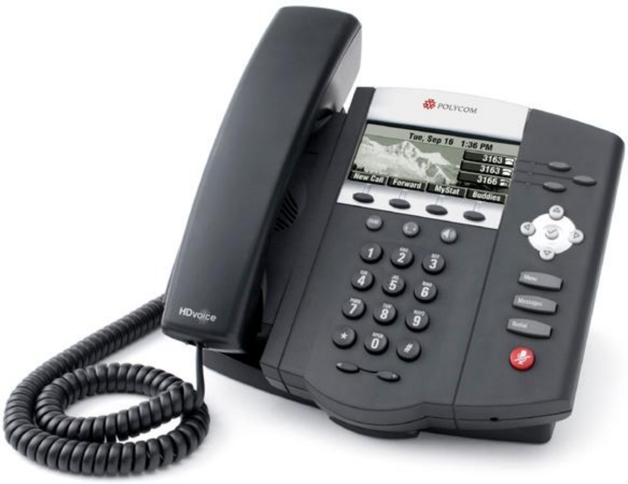


Figure 1: Polycom SoundPoint IP 450 3-Line

Highlights

- Enterprise-grade IP Phone
- HTTPS secure provisioning
- Full-featured 3-line business-class IP phone
- Integrated PoE support (IEEE 802.af)
- Polycom HD Voice*
- 256 x 116 pixel backlit grayscale graphical LCD
- 17 dedicated Hard keys, 3 context-sensitive soft keys
- Certified SIP call control
- Dedicated RJ-9 headset port

Overview

The SoundPoint IP 450 desktop phone is designed to bring advanced telephony features and applications to cubicle/office workers handling a moderate volume of calls. With its high-resolution, graphical backlit display, dual-port 10/100 Ethernet Switch, and Polycom HD Voice, the SoundPoint IP 450 sets new standards for the mid-range SIP desktop phone.

Features Overview

Unsurpassed Voice Quality and Clarity

The SoundPoint IP 450 features Polycom HD Voice*, bringing life-like richness and clarity to every call. Polycom HD Voice incorporates wideband audio for over twice the voice clarity, Polycom's patented Acoustic Clarity Technology for crystal-clear, noise and echo-free sound, plus best-in-class system design for high-fidelity voice reproduction.

Efficient Installation

The SoundPoint IP 450 phones are engineered to make installation, automatic configuration, and upgrades as simple and efficient as possible. Built-in IEEE 802.3af PoE circuitry and a dual-port Ethernet switch enable flexible deployment options and savings on cabling expenses.

Enterprise-Grade Feature Set

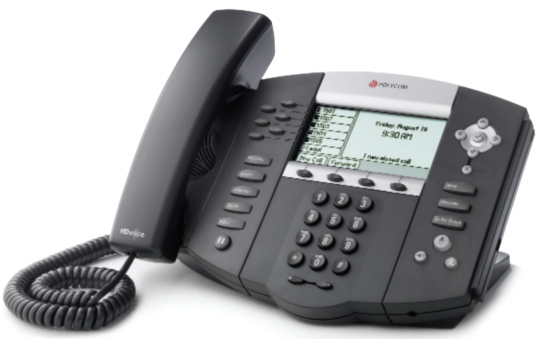
The SoundPoint IP 450 phone delivers through an intuitive user interface a full feature set encompassing both traditional business telephony features such as call hold, park, pick-up, transfer, and three-way local conferencing, and more advanced capabilities such as shared call appearance and call control toolbar integration.

Secure Provisioning

The SoundPoint IP 450 phone provides encrypted provisioning information using HTTPS to secure communication between the user and the service provider. The Polycom secure remote provisioning software provides a highly secure mechanism for configuration and device software upgrades.

Polycom® SoundPoint® IP 650

High Performance IP Phone with Polycom HD Voice



Benefits

Revolutionary Voice Quality – Polycom HD Voice enables life-like interactivity, richness, and clarity of voice communications

Advanced Features & Applications –

- Integration with Microsoft LCS 200 and Microsoft Office Communicator
- USB port for future application
- XHTML microbrowse
- Backlit 320x160 graphical grayscale LC
- Integrated PoE suppor

Advanced Call Handling Capabilities –

- Six lines (standalone) / 12 lines wit Expansion Module(s)
- Shared call / bridged line appearance [2]
- Busy lamp field (BLF) [2]

Expandability – Supports up to three SoundPoint IP Expansion Modules for an attendant console application

Proven – Polycom is the leading independent supplier of standards-based IP phones that are fully interoperable with key IP PBX and Softswitch platforms

Delivering revolutionary voice quality, an advanced feature set, and the expandability to support SoundPoint IP Expansion Modules

Designed to appeal to both executive users who require advanced features and applications, and telephone attendants who need multiple line support, the Polycom SoundPoint IP 650 sets a new standard for high-performance IP phones.

Revolutionary Voice Quality

The SoundPoint IP 650 is the first IP phone to use Polycom's revolutionary HD Voice technology to bring life-like richness and clarity to voice communications. Polycom HD Voice incorporates wideband audio for over twice the voice clarity, Polycom's patented Acoustic Clarity Technology 2, as well as best-in-class system design to deliver unprecedented voice quality.

Advanced Features and Applications[2]

The phone supports Microsoft® Live Communications Server 2005 for telephony and presence, and interoperates with Microsoft Office Communicator. The SoundPoint IP 650 also features a USB port for future applications.

Enhanced Call Handling Capabilities

The SoundPoint IP 650 accommodates 6 lines in standalone mode, and 12 lines as an attendant console, when equipped with SoundPoint IP Expansion Modules. The phone supports shared call/bridged line appearance[2], an essential feature for effective phone interaction between executives and their assistants. The phone's busy lamp field (BLF)[2] functionality enables phone attendants to monitor the on-hook / off-hook status of key contacts, and dispatch incoming calls for those contacts more efficiently.

Expandability

When equipped with up to three Expansion Modules, the SoundPoint IP 650 delivers the advanced call handling capabilities and enhanced user interface of a high-performance attendant console. Designed to improve productivity of telephone attendants, the SoundPoint IP attendant console allows effective and efficient management and monitoring of up to 24 simultaneous calls on up to 12 lines.

Intuitive User Interface

The SoundPoint IP 650 delivers all of its capabilities through an intuitive user interface, featuring a backlit 320x160 graphical grayscale LCD display, easy-to-navigate menu, and a combination of 26 dedicated hard keys and 4 context-sensitive soft keys for easy access to essential telephony features.

Efficient Installation and Provisioning

Designed to make installation, configuration, and upgrade as simple and efficient as possible, the SoundPoint IP 650 boasts a two-port Ethernet switch and integrated Power over Ethernet circuitry. The SoundPoint IP 650 can be centrally configured and upgraded in the field from an FTP, TFTP, HTTP[4], or HTTPS[4] server and supports provisioning server redundancy.

Make Great Things Happen with Polycom SoundPoint IP 650

In today's Internet driven world, the ability to conduct real time communication and collaboration has become critical to an organisation's survival. As the market leader in voice, video, data and Web solutions, our award-winning conference technology makes it easy for people to interact and maximise productivity — over any network, in just about any environment, anywhere around the globe. That's why more organisations worldwide use and prefer Polycom conferencing solutions. Because when people work together, great things happen. See how you, too, can achieve great things with Polycom SoundPoint IP 650.

SPECIFICATIONS

Lines (Direct Numbers)
- Up to 6 lines (standalone mode)
- Up to 12 lines with Expansion Module(s)

SoundPoint IP Expansion Module Support
- Supports up to three Expansion Modules

Display
- 320 x 160 backlit greyscale graphical LCD
- White LED backlight with custom intensity control

Feature Keys
- 4 context-sensitive "soft" keys
- 26 dedicated "hard" keys
  - 6 line keys with bi-color (red/green) LED
  - 8 feature keys
  - 6 display/menu navigation keys
  - 2 volume control keys
  - Illuminated mute key
  - Illuminated headset key
  - Illuminated hands-free speakerphone key
  - Dedicated hold key

Headset and Hearing Aid Compatibility
- Dedicated RJ-9 headset port
  - Amplified headsets are recommended
- Compliant with ADA Section 508 Recommendations: Subpart B 1194.23 (all)
- Hearing Aid Compatible (HAC) handset for magnetic coupling to approved HAC hearing aids
- Compatibility with commercially-available TTY adapter equipment

Audio Features
- Polycom HD Voice technology delivers life-like voice quality for each audio path - the handset, the hands-free speakerphone, and the headset[1]
- Full-duplex hands-free speakerphone
  - Type 1 compliant with IEEE 1329 full duplex standards
- Frequency response - 150Hz - 7kHz for handset, headset[1] and hands-free speakerphone modes
- Codecs: G.722 (wideband), G.711 µ/A, and G.729A (Annex B)
- Individual volume settings with visual feedback for each audio path
- Voice activity detection
- Comfort noise fill
- DTMF tone generation / DTMF event RTP payload
- Low-delay audio packet transmission
- Adaptive jitter buffers
- Packet loss concealment
- Acoustic echo cancellation
- Background noise suppression

Call Handling Features[2]
- Shared call / bridged line appearance
- Flexible line appearance (one or more line keys can be assigned for each line extension)
- Busy Lamp Field (BLF)
- Distinctive incoming call treatment / call waiting
- Call timer
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Local three-way conferencing
- One-touch speed dial, redial
- Call waiting
- Remote missed call notification
- Intercom
- Automatic off-hook call placement
- Do not disturb function

Other Features
- Integration with Microsoft LCS 2005 for telephony and presence[3]
  - Compatibility with Microsoft Office Communicator and Windows® Messenger 5.1 Clients
- Universal Serial Bus (USB)
  - Full Host Controller
  - Compliant with OHCI 1.1 specification
  - Support for Full-speed and Low-speed peripherals[5]
  - Type-A receptacle interface
- Local feature-rich GUI
- Time and date display
- User-configurable contact directory and call history (missed, placed, and received)
- Customisable call progress tones
- Wave file support for call progress tones
- Unicode UTF-8 character support. Multilingual user interface encompassing Chinese, Danish, Dutch, English (Canada / US / UK), French, German, Italian, Japanese, Korean, Norwegian, Portuguese, Russian, Spanish, Swedish

Protocol Support
- IETF SIP (RFC 3261 and companion RFCs)

Network and Provisioning
- Two-port 10/100 Mbps Ethernet switch
- Manual or dynamic host configuration protocol (DHCP) network setup
- Time and date synchronization using SNTP
- FTP / TFTP / HTTP / HTTPS[4] server-based central provisioning for mass deployments. Provisioning server redundancy supported
- Web portal for individual unit configuration
- QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS, and DSCP
- Network Address Translation (NAT) support – static
- RTCP support (RFC 1889)
- Event logging
- Local digit map
- Hardware diagnostics
- Status and statistics

Security[2]
- Transport Layer Security (TLS)[3]
- Encrypted configuration files[3]
- Digest authentication
- Password login
- Support for URL syntax with password for boot server[4]
- HTTPS secure provisioning[4]
- Support for signed software executables[4]

Power
- Built-in, auto-sensing IEEE 802.3af Power over Ethernet
- External Universal AC adapter (included; 24V DC @ 500mA min.)

Approvals
- FCC Part 15 (CFR 47) Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- AS/NZS CISPR 22 Class B
- VCCI Class B
- EN55024
- EN61000-3-2; EN61000-3-3
- ROHS compliant

Safety
- UL 60950
- CE Mark
- CAN/CSA-C22.2 No. 60950
- EN 60950-1
- IEC 60950-1
- AS/NZS 60950

Operating Conditions
- Temperature: +10 to 40 °C (+50 to 104 °F)
- Relative Humidity: 20% to 85%, non-condensing

Storage Temperature
- -40 to +70 °C (-40 to +160 °F)

SoundPoint IP 650 Comes With:
- SoundPoint IP 650 console
- Handset with handset cord
- Base stand
- Network (LAN) cable
- Universal power adapter (including country-specific cord kit)
- Quick Start Guide
- Product registration card

Size
- 26.5 cm x 15 cm x 19 cm x 6.5 cm (10.5 in x 6 in x 7.5 in x 2.5 in) (W x H x D x T)

Weight
- Shipping: 1.26 kg (2.75 lb)

Part Numbers / UPC Codes
- 2200-12651-001 / 610807522959 for NA, TWN
- 2200-12651-002 / 610807522966 for Japan
- 2200-12651-012 / 610807522973 for AU, NZ
- 2200-12651-015 / 610807522980 for UK, HK, Singapore, Malaysia
- 2200-12651-016 / 610807522997 for Korea
- 2200-12651-022 / 610807523000 for China
- 2200-12651-122 / 610807523017 for ROE

Box Dimensions / Weight
- 32 cm x 345 cm x 9 cm (12.5 in x 13 in x 3.5 in)
- 2045g (for all countries)

Master Carton Quantity
- Five

Country of Origin
- Thailand

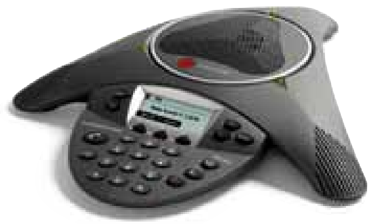
Warranty
- 1 year

1 To enjoy the benefits of Polycom HD Voice when using the phone in the headset mode, you must use a wideband headset. Please contact headset manufacturers to identify appropriate headset models.
2 Most software-enabled features and capabilities must be supported by the server. Please contact your IP PBX/Softswitch vendor or service provider for a list of supported features.
3 Requires SIP version 2.0.1 or higher.
4 Requires BootROM version 3.2.x or higher.
5 Please contact Polycom for current device driver support

Polycom® SoundStation® IP 6000

SIP-Based IP Conference Phone

Next-generation IP conference phone designed for small to midsize rooms



The SoundStation IP 6000 is an advanced IP conference phone that delivers superior performance for small to midsize conference rooms. With advanced features, broad SIP interoperability and remarkable voice quality, the SoundStation IP 6000 is a price/performance breakthrough for SIP-enabled IP environments.

The SoundStation IP 6000 features Polycom HD Voice technology, boosting productivity and reducing listener fatigue by turning ordinary conference calls into crystal-clear interactive conversations. It delivers high-fidelity audio from 220 Hz to 14 kHz, capturing both the deeper lows and higher frequencies of the human voice for conference calls that sound as natural as being there.

For all conference calls, the SoundStation IP 6000 delivers advanced audio performance that far exceeds previous generations of conference phones. From full-duplex technology that eliminates distracting drop-outs to the latest echo cancellation advancements, only Polycom can deliver a conference phone experience with no compromises. Plus, Automatic Gain Control intelligently adjusts the microphone sensitivity based on where participants are seated in the conference room, making the conversations clearer for all participants. It also features technology that resists interference from mobile phones and other wireless devices, delivering clear communications without distractions.

The SoundStation IP 6000 leverages Polycom's strong history in both conference phone and VoIP technology to deliver the most robust standards-based SIP interoperability in the industry. It shares the same SIP phone software base with Polycom's award-winning SoundPoint IP products – the most comprehensive, reliable and feature-rich SIP products in the industry – with proven interoperability with a broad array of IP PBX and hosted platforms.

Robust provisioning, management and security features make Polycom's family of IP conference phones the only choice for meeting rooms in SIP-based environments. Integrated Power over Ethernet (PoE) simplifies setup, with an AC power kit available for non-PoE environments. Plus, the SoundStation IP 6000 includes a high-resolution backlit display for vital call information and multi-language support.

Make Great Things Happen with the Polycom SoundStation IP 6000 Conference Phone
In today's Internet driven world, the ability to conduct real time communication and collaboration has become critical to an organisation's survival. As the market leader in voice, video and content collaboration and communications, our award-winning conference technology makes it easy for people to interact and maximise productivity over any network, in just about any environment, anywhere around the globe. That's why more organisations worldwide use and prefer Polycom conferencing solutions. Because when people work together, great things happen. See how you too, can achieve great things with the Polycom SoundStation IP 6000 conference phone.

- HD Voice – Unparalleled clarity to make your conference calls more efficient and productive
- Polycom's patented Acoustic Clarity Technology – Delivering the best conference phone experience with no compromises
- 12-foot microphone pickup – Combined with Automati Gain Control for performance far beyond older SoundStation IP conference phones. Add up to two optional expansion microphones for even greater coverage.
- Industry-leading SIP software – Leveraging the most advanced SIP endpoint software in the industry, with advanced call handing, security, and provisioning features
- Robust interoperability – Compatible with a broad array of SIP call platforms to maximise voice quality and feature availability while simplifying management and administration
- High-resolution display – Enables robust call information and multi-language support

Polycom® SoundStation IP 6000 Conference Phone Specifications

Power
- IEEE 802.3af Power over Ethernet (built in)
- Optional external universal AC power supply: 100-240V, 0.4A, 48V/19W

Display
- Size (pixels): 248 x 68 (W x H)
- White LED backlight with custom intensity control

Keypad
- Standard 12-key keypad
- Context-dependent soft keys: 3
- On-hook/Off-hook, redial, mute, volume up/down

Audio Features
- Loudspeaker
  - Frequency: 220-14000 Hz
  - Volume: Adjustable to 85 dB at 1/2 meter peak volume
- Individual volume settings with visual feedback for each audio path
- Voice activity detection
- Comfort noise fill
- DTMF tone generation / DTMF event RTP payload
- Low-delay audio packet transmission
- Adaptive jitter buffers
- Packet loss concealment
- Acoustic echo cancellation
- Background noise suppression
- Supported Codecs
  - G.711 (A-law and Mu-law)
  - G.729a (Annex B)
  - G.722, G.722.1
  - G.722.1C
  - Siren 14

Call Handling Features
- Shared call / bridged line appearance
- Busy Lamp Field (BLF)
- Distinctive incoming call treatment / call waiting
- Call timer
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Local three-way conferencing
- One-touch speed dial, redial
- Call waiting
- Remote missed call notification
- Automatic off-hook call placement
- Do not disturb function

Other Features
- Local feature-rich GUI
- Time and date display
- User-configurable contact directory and call history (missed, placed, and received)
- Customisable call progress tones
- Wave file support for call progress tones
- Unicode UTF-8 character support. Multilingual user interface encompassing Chinese, Danish, Dutch, English (Canada / US / UK), French, German, Italian, Japanese, Korean, Norwegian, Portuguese, Russian, Spanish, Swedish

Network and Provisioning
- Ethernet 10/100 Base-T
- 2.5mm connection port
- EX mic ports: Two RJ-9 ports
- IP Address Configuration: DHCP and Static IP
- Time synchronisation with SNTP server
- FTP / TFTP / HTTP / HTTPS server-based central provisioning for mass deployments. Provisioning server redundancy supported.
- Web portal for individual unit configuration
- QoS Support -- IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS and DSCP
- Network Address Translation (NAT) support - static
- RTCP support (RFC 1889)
- Event logging
- Local digit map
- Hardware diagnostics
- Status and statistics
- User selectable ringer tones
- Convenient volume adjustment keys
- Field upgradeable

Security
- Transport Layer Security (TLS)
- Encrypted configuration files
- Digest authentication
- Password login
- Support for URL syntax with password for boot server
- HTTPS secure provisioning
- Support for signed software executables

Safety
- UL1950
- CE Mark
- CSA C22.2, No 60950
- EN60950
- IEC60950
- AS/NZSS3260

EMC
- FCC (47 CFR Part 15) Class B
- ICES-003 Class
- EN55022 Class B
- CISPR22 Class B
- AS/NZS 3548 Class B
- VCCI Class B
- EN61000-3-2; EN61000-3-3
- EN55024
- ROHS compliant

Protocol Support
- IETF SIP (RFC 3261 and companion RFCs)

IEEE 802.3af Power over Ethernet version ships with
- Telephone Console
- 25 foot Ethernet cable
- Quick Start Guide
- Quick User Guide

AC Power version ships with
- Telephone Console
- 25 foot Ethernet cable
- Universal Power Supply
- 7 foot region-specific power cord
- Power Insertion Cable
- Quick Start Guide
- Quick User Guide

Environmental Conditions
- Operating temperature: 0 - 40° C (32 - 104° F)
- Relative humidity: 20%-85% (noncondensing)
- Storage temperature: -30 - 55° C (-22 - 131° F)

Warranty
- 1 year

Country of Origin
- China

Phone Dimensions
- 36.8 x 31.1 x 6.4 cm (14.5 x 12.25 x 2.5 in)
  (L x W x H)

Phone Console Weight
- 0.8 kg (1.75 lb)

Box Dimensions
- 33 x 39.5 x 15 cm (13.0 x 15.5 x 6.0 in) (L x W x H)

Box Weight
- 2.32 kg (5.1 lb)

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Polycom® VVX® 500

A performance business media phone delivering best-in-class desktop productivity and unified communications for busy professionals

The Polycom® VVX® 500 performance business media phone unifies superior voice capabilities and applications into a simple-to-use, yet high performance unified communications (UC) solution. It is the ideal, all-in-one knowledge worker productivity tool, built to integrate seamlessly into a wide range of UC environments.

Simplicity and Ease of Use

The Polycom VVX 500 phone is built for today's busy managers and knowledge workers who need a powerful, yet intuitive, expandable office phone that keeps up with their multitasking and schedule-juggling. Building on the behavior common to mobile phones, the intuitive, multi-touch, gesture-based user interface of the VVX 500 phone makes navigation easy and requires no learning curve.

Maximize Productivity

Designed for a broad range of environments from small and medium businesses to large enterprises, the VVX 500 performance business media phone improves productivity, by complimenting the workplace applications on the user's computer. Users benefit from such capabilities as viewing and managing their Exchange calendars and contacts on the phone and receive meeting reminders while still having access to their corporate directory—and all while waiting for their PCs to boot. Users can also extend their PC desktop to include the VVX 500 phone's screen, enabling simplified interactions and dialing using their PC's mouse and keyboard. Training and multipoint communication applications are complemented by the VVX 500 video playback capability for streaming content.

Best-in-Class Deployment and Administration

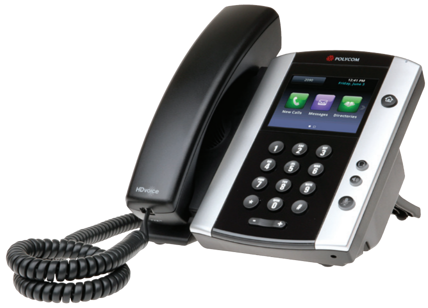
The VVX 500 phone is easy to deploy and simple to manage. Its enterprise-grade, Web-based, intuitive configuration method gives administrators the ability to easily provision and maintain a large number of phones throughout the entire enterprise. The built-in, broad interoperability capabilities allow IT departments to leverage previous IT infrastructure investments and achieve easy integration with third-party UC and productivity applications.

Highly Customizable and Expandable

The VVX 500 phone provides personalized information at a glance, through built-in Web applications and even a Digital Photo Frame. Polycom's "My Info Portal" serves up stocks, news, sports, weather, and other streamed content directly to the phone's screen. The VVX 500 phone also comes ready for future expansion modules and accessories for applications such as video conferencing and even wireless networking.

Market -leading Open Standards Interoperability

Designed for enhanced interoperability, leveraging and complementing the other existing IT investments of any enterprise, the VVX 500 phone delivers HD voice, a superior UC experience bundled with business applications. With the broadest call server interoperability in the industry, the Polycom VVX 500 business media phone can become the flexible and future-proof foundation for any organization's unified communications strategy.



Benefits

- Improves knowledge-worker productivity
- Reduces training time through superior calling features in a sleek design and simple-to-use phone
- Reduces telephony administration and maintenance costs
- Leverages previous IT infrastructure investments
- Is simple to deploy and easy to administer, upgrade, and maintain
- Delivers easy integration with third-party UC and productivity applications

Polycom VVX 500 Specifications

User Interface Features
- Gesture-based, multitouch-capable, capacitive touchscreen
- 3.5-in TFT LCD display at QVGA (320x240 pixel) resolution, 4:3 aspect ratio
- Screensaver and digital picture frame mode
- On-screen virtual keyboard
- Voicemail and videomail support[1]
- Dual USB ports (2.0 compliant) for media and storage applications
- WebKit-based Browser
- Adjustable base height
- Unicode UTF-8 character support. Multilingual user interface including Chinese, Danish, Dutch, English (Canada/US/UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, and Swedish

Audio Features
- Polycom HD Voice technology delivers life-like voice quality for each audio path-handset, the hands-free speakerphone, and the optional headset
- Polycom Acoustic Clarity™ technology providing full-duplex conversations, acoustic echo cancellation and background noise suppression
  - Type 1 compliant (IEEE 1329 full duplex)
- Frequency response – 100 Hz – 20 kHz for handset, optional headset and hands-free speakerphone modes
- Codecs: G.711 (A-law and μ-law), G.729AB, G.722, G.722.1, G.722.1C
- Individual volume settings with visual feedback for each audio path
- Voice activity detection
- Comfort noise generation
- DTMF tone generation (RFC 2833 and in-band)
- Low-delay audio packet transmission
- Adaptive jitter buffers
- Packet loss concealment

Headset and Handset Compatibility
- Dedicated RJ-9 headset port
- Hearing aid compatibility ITU-T P.370 and TIA 504A standards
- Compliant with ADA Section 508 Subpart B 1194.23 (all)
- Hearing aid compatible (HAC) handset for magnetic coupling to hearing aids
- Compatible with commercially-available TTY Adapter equipment
- Support USB Headsets (see TB37477 for list of compatible headsets)

Call Handling Features[1]
- 12 lines (registrations)
- Up to 24 simultaneous calls
- Shared call/bridged line appearance
- Flexible line appearance (one or more line keys can be assigned for each line extension)
- Distinctive incoming call treatment/call waiting
- Call timer and call waiting
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Local three-way audio conferencing
- One-touch speed dial, redial
- Remote missed call notification
- Do not disturb function
- Electronic hook switch capable
- Local configurable digit map/dial plan

Open Application Platform
- WebKit enabled full browser that supports HTML5, CSS, SSL security, and JavaScript
- Supports Polycom Apps SDK and API for third-party business and personal applications
- Bundled with Polycom Productivity Suite:
  - Corporate Directory Access using LDAP
  - Local Voice Call Recording on USB flash drive
  - Visual Conference Management

Network and Provisioning
- SIP Protocol Support
- SDP
- IETF SIP (RFC 3261 and companion RFCs)
- Two-port Gigabit Ethernet switch
  - 10/100/1000Base-TX across LAN and PC ports
  - Conforms to IEEE802.3-2005 (Clause 40) for Physical Media Attachment
  - Conforms to IEEE802.3-2002 (Clause 28) for Link Partner Auto-Negotiation
- Manual or dynamic host configuration protocol
- (DHCP) network setup
- Time and date synchronization using SNTP
- FTP/TFTP/HTTP/HTTPS[4] server-based central provisioning for mass deployments
- Provisioning and call server redundancy supported[1]
- QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS, and DSCP
- VLAN - CDP, DHCP VLAN discovery, LLDP-MED for VLAN discovery
- Network Address Translation (NAT) – support for static configuration and "Keep-Alive" SIP signaling
- RTCP and RTP support
- Event logging
- Syslog
- Local configurable digit map/dial plan
- Hardware diagnostics
- Status and statistics reporting
- IPv4
- TCP
- UDP
- DNS-SRV

Security
- 802.1X Authentication and EAPOL
- Media encryption via SRTP
- Transport Layer Security (TLS)[3]
- Encrypted configuration files[3]
- Digest authentication
- Password login
- Support for URL syntax with password for boot server address[3]
- HTTPS secure provisioning[3]
- Support for signed software executables[3]

Power
- Built-in auto sensing IEEE 802.3 at Power over Ethernet (Class 4)
- Energy-saving smart motion detector enables the screen to go into power-save mode when no one is in the office.
- External Universal AC Adaptor (optional, 48V 380mA DC)

Approvals
- FCC Part 15 (CFR 47) Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- VCCI Class B
- EN55024
- EN61000-3-2; EN61000-3-3
- NZ Telepermit
- Korea KC
- China CCC
- ROHS compliant
- UAE TRA
- Russia GOST-R
- Brazil ANATEL
- Australia A & C Tick

Safety
- UL 60950-1
- CE Mark
- CAN/CSA-C22.2 No. 60950-1-03
- EN 60950-1
- IEC 60950-1
- AS/NZS 60950-1

Operating Conditions
- Temperature: (+32 to 104°F (0 to 40°C)
- Relative Humidity: 5% to 95%, noncondensing

Storage Temperature
- -40 to +160°F (-40 to +70°C)

Polycom VVX 500 Comes With:
- VVX 500 console
- Handset with handset cord
- Network (LAN) cable
- Quick Start Guide
- Product registration card

Size
- 7.5 x 6 x 7 in (19 x 15 x 18 cm) (W x H x D)

Part Numbers
- 2200-44500-025 – WW PoE

Weight
- Unit weight: 2.0 lbs (0.9 kg)

Unit Box Dimensions / Weight
- 12 x 9 x5 in
- 3.1 lbs (1.4 kg)

Master Carton Quantity
- Five (5)

Country of Origin
- China

Warranty
- One (1) year

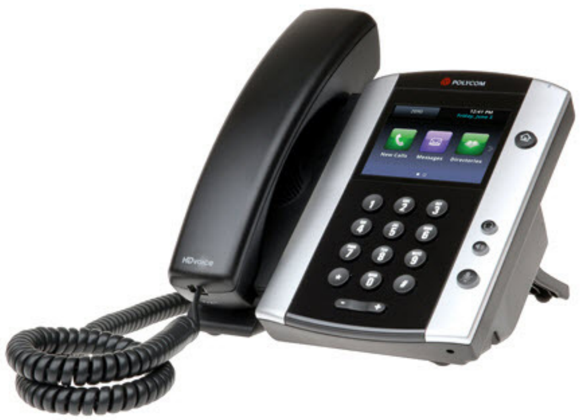
[1] Most software-enabled features and capabilities must be supported by the server. Please contact your IP PBX/Softswitch vendor or service provider for a list of supported features.
[2] To enjoy all the benefits of Polycom HD Voice when using the phone in the headset mode, you must use a wideband headset.
[3] Requires UCS SW version 4.0.1 or higher.

About Polycom
About Polycom Polycom, Inc. (Nasdaq: PLCM) is a global leader in unified communications solutions with industry-leading telepresence, video, voice, and infrastructure solutions built on open standards. Polycom powers smarter conversations, transforming lives and businesses worldwide.

Learn what Polycom solutions can do for your organization. Visit us at www.polycom.com or call 1-800-POLYCOM to speak with a Polycom representative.

IP Voice Services
Device Overview

Polycom VVX 500 IP Phone



Overview

The Polycom VVX 500 unifies IPVS voice, video and applications into a simple-to-use business media phone. With its unique touch screen interface, the VVX 500 is an easy to use high quality affordable IP handset.

What makes this handset unique is that it has a touchscreen that enables users too quickly and simply make calls and access phone features. Smartphones have led the way with their simple touchscreen controls and now Polycom have brought this technology to the fixed handset.

Later in the year, October 2012, Polycom will be shipping a USB camera that can be added to the device which will then allow IPVS users to use the recently launched video service. However for now don't forget uses can purchase a Polycom 1500 Video phone or use a MobileOffice soft client if they wish to use the Video service.

Highlights and Key Features

- Up to 12 accounts (registrations) with IPVS Application Integration
- 3.5 inch colour, touch screen LCD display
- **Plug and Play** - HTTPS secure, remote automatic provisioning
- Supports **HD Voice** (G722), G711 and G729 codecs
- Easy to use **business telephony features** such as extension dialling, call hold, transfer, pickup, park and retrieve, conference, speakerphone, mute, RJ-9 headset port and voicemail integration
- Automatic Busy Lamp Field integration
- 2 x 10/100/1000Base-TX ports for LAN and PC, Poe

Plug and Play Secure Provisioning

The Polycom VVX 500 is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly secure mechanism for the service provider to remotely manage the phone/user configuration.

The VVX 500 **can be deployed as a User's primary phone, an additional line or as an additional device** using Shared Call Appearance. This includes the option for a SIP Trunk Mobility User to use the Polycom VVX 500 as an additional device for video calling once this feature is supported.

Cisco SPA508G IP Phone

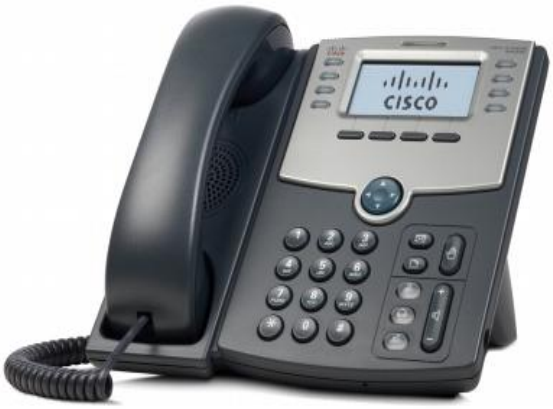


Figure 1: Cisco SPA508-G 8 Line IP Phone

Highlights

- Full-featured 8-line business-class IP phone supporting Power over Ethernet (PoE)
- Monochrome backlit display for ease of use, aesthetics, and on-screen applications
- G722 HD Voice audio support for unsurpassed voice clarity and enhanced speaker quality*
- HTTPS secure automatic provisioning and software control
- Supports up to two Cisco® SPA 500S Expansion Module, adding up to 64 additional buttons for IPVS Busy Lamp Field feature integration
- IPVS Application Integration
- 4 context-sensitive soft keys

Overview

With hundreds of features and configurable service parameters, the Cisco SPA 508G addresses the requirements of traditional business users while building on the advantages of IP telephony.

Standard features on the Cisco SPA508G include eight active lines, VLAN-capable dual switched Ethernet ports, a full-duplex, high-quality speakerphone, and 802.3af PoE support.

The Cisco SPA508G is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly secure mechanism for the service provider to remotely manage the phone/user configuration and the device software upgrades.

Features

The Cisco SPA508G IP phone can easily grow with your business. New features can be added to the phone over time via self service portals and software updates. New employees or employees who need to move locations can simply plug in their preconfigured phones anywhere on the network, and the network will recognise the change.

Handset features:

- Eight voice lines
- Line status: active line indication, with name and number
- Menu-driven user interface
- Speakerphone
- Call hold
- Call Pickup
- Call Park and Retrieve
- Caller ID name and number
- Outbound caller ID blocking
- Call transfer: attended and blind
- Three-way call conferencing with local mixing
- Automatic redial of last calling and last called numbers
- On-hook dialing
- Call back on busy
- Call blocking: anonymous and selective
- Call logs (60 entries each): made, answered, and missed calls
- Redial from call logs
- Personal directory with auto-dial (100 entries)
- Do not disturb
- Digits dialed with number auto-completion
- On-hook default audio configuration (speakerphone and headset)
- Multiple ring tones
- 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
- Distinctive ringing based on calling and called number
- 10 user-downloadable ring tones
- Speed dialing, eight entries
- Configurable dial/numbering plan support

Cisco SPAP501G IP Phone

Highlights

- Full-featured 8-line business-class IP phone supporting Power over Ethernet (PoE)
- 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
- Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- Supports up to two Cisco® SPA500S Expansion Module, adding an additional 64 buttons
- 2 x 10/100BASE-T RJ-45 Ethernet ports
- Monochrome Display

Overview

With hundreds of features and configurable service parameters, the Cisco SPA 501G addresses the requirements of traditional business users while building on the advantages of IP telephony. Connects directly to an Internet telephoney.

The Cisco SPA501G is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly
secure mechanism for the service provider to remotely manage the phone/user configuration and the device software upgrades.

Features

The Cisco SPA501G IP phone can easily grow with your business. New features can be added to the phone over time via self service portals and software updates. New employees or employees who need to move locations can simply plug in their preconfigured phones anywhere on the network.

Handset features

- Eight voice lines
- Line status: active line indication
- Shared line appearance
- Speakerphone
- Call hold
- Music on hold
- Call waiting
- Outbound caller ID blocking
- Call transfer: attended and blind
- Three-way call conferencing with local mixing
- 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
- Personal directory with auto-dial (100 entries)
- Do not disturb
- Digits dialed with number auto-completion
- Anonymous caller blocking
- On-hook default audio configuration (speakerphone and headset)
- Multiple ring tones with selectable ring tone per line
- Date and time with support for intelligent daylight savings
- Call start time stored in call logs
- Distinctive ringing based on calling and called number
- 10 user-downloadable ring tones
- Speed dialing, eight entries

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COMPANY REGISTRATION NUMBER: 05693707
www.thevoipshop.co.uk

IP Voice Services

Device Overview

Cisco SPAP504G IP Phone

Highlights

- Full-featured 4-line business-class IP phone supporting Power over Ethernet (PoE)
- 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
- Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- Supports up to two Cisco® SPA500S Expansion Module, adding an additional 64 buttons
- 2 x 10/100BASE-T RJ-45 Ethernet ports
- Pixel-based display: 128 x 64 monochrome LCD graphical display with backlight

Overview

With hundreds of features and configurable service parameters, the Cisco SPA 504G addresses the requirements of traditional business users while building on the advantages of IP telephony. Connects directly to an Internet telephoney.

The Cisco SPA504G is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly secure mechanism for the service provider to remotely manage the phone/user configuration and the device software upgrades.

Features

The Cisco SPA504G IP phone can easily grow with your business. New features can be added to the phone over time via self service portals and software updates. New employees or employees who need to move locations can simply plug in their preconfigured phones anywhere on the network.

Handset features

- Four Voice lines
- Line status: active line indication
- Shared line appearance
- Speakerphone
- Call hold
- Music on hold
- Call waiting
- Outbound caller ID blocking
- Call transfer: attended and blind
- Three-way call conferencing with local mixing
- 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
- Personal directory with auto-dial (100 entries)
- Do not disturb
- Digits dialed with number auto-completion
- Anonymous caller blocking
- On-hook default audio configuration (speakerphone and headset)
- Multiple ring tones with selectable ring tone per line
- Date and time with support for intelligent daylight savings
- Call start time stored in call logs
- Distinctive ringing based on calling and called number
- 10 user-downloadable ring tones
- Speed dialing, eight entries

IP Voice Services
Device Overview

the voip shop
BUSINESS COMMUNICATION SPECIALISTS

Cisco SPAP509G IP Phone

Highlights

- Full-featured 12-line business-class IP phone supporting Power over Ethernet (PoE)
- 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
- Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- Supports up to two Cisco® SPA500S Expansion Module, adding an additional 64 buttons
- 2 x 10/100BASE-T RJ-45 Ethernet ports
- Pixel-based display: 128 x 64 monochrome LCD graphical display with backlight

Overview

With hundreds of features and configurable service parameters, the Cisco SPA 509G addresses the requirements of traditional business users while building on the advantages of IP telephony. Connects directly to an Internet telephoney.

The Cisco SPA509G is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly secure mechanism for the service provider to remotely manage the phone/user configuration and the device software upgrades.

Features

The Cisco SPA509G IP phone can easily grow with your business. New features can be added to the phone over time via self service portals and software updates. New employees or employees who need to move locations can simply plug in their preconfigured phones anywhere on the network.

Handset features

- Four Voice lines
- Line status: active line indication
- Shared line appearance
- Speakerphone
- Call hold
- Music on hold
- Call waiting
- Outbound caller ID blocking
- Call transfer: attended and blind
- Three-way call conferencing with local mixing
- 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
- Personal directory with auto-dial (100 entries)
- Do not disturb
- Digits dialed with number auto-completion
- Anonymous caller blocking
- On-hook default audio configuration (speakerphone and headset)
- Multiple ring tones with selectable ring tone per line
- Date and time with support for intelligent daylight savings
- Call start time stored in call logs
- Distinctive ringing based on calling and called number
- 10 user-downloadable ring tones
- Speed dialing, eight entries

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Cisco SPA525G IP Phone

5-Line Business IP Phone with Colour Display and Enhanced Connectivity

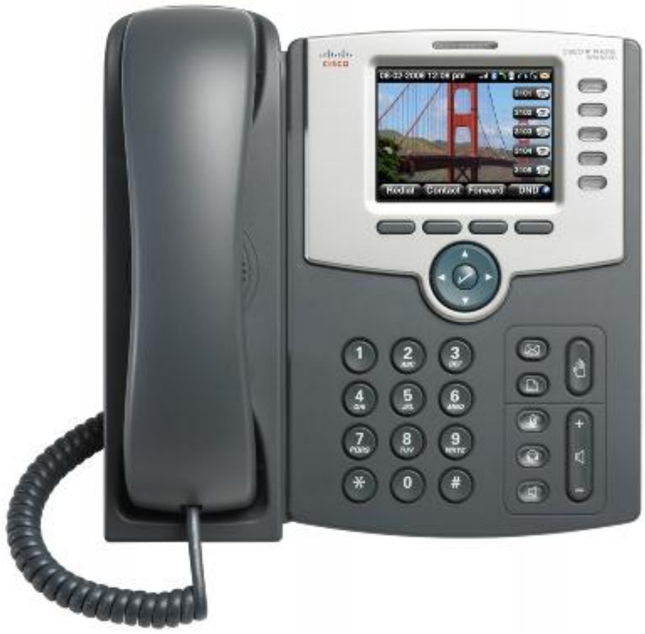

Figure 1: Cisco SPA525G 5-Line IP Phone with Colour Display

Highlights

- Feature Rich and stylish 5-line business IP phone
- Enhanced connectivity, with Power over Ethernet, 802.11g Wi-Fi client mode and Bluetooth headset support
- Graphic-rich, high-resolution 3.2-inch QVGA 320 x 240 colour screen
- Support for multimedia functions, such as playing MP3's, displaying digital photos, and viewing RSS feeds
- G722 HD Voice audio support for unsurpassed voice clarity and enhanced speaker quality*
- HTTPS secure automatic provisioning and software control
- Supports up to two Cisco® SPA 500S Expansion Module, adding up to 64 additional buttons for IPVS Busy Lamp Field feature integration
- IPVS Application Integration

Product Overview

The Cisco SPA525G 5-Line IP Phone with Colour Display (Figure 1) is an excellent choice for businesses that require an enhanced user experience with TheVoipShop Wholesale's IP Voice Services. The SPA525G uses industry-leading technology from Cisco, with high-quality hardware providing Bluetooth for headset connectivity, Power over Ethernet (PoE) (802.3af), or a Wireless-G client (802.11g).

The Cisco SPA525G is an IPVS Auto-Provisioning Device, using HTTPS/SSL to secure configuration information between the user and service provider. This provides a highly secure mechanism for the service provider to remotely manage the phone/user configuration and the device software upgrades.

Features

Standard features on the Cisco SPA525G include five lines, VLAN-capable dual switched Ethernet ports, 802.3af PoE support, a 3.2-inch QVGA colour display, a full-duplex, high-quality speakerphone, a Bluetooth interface for headset connectivity, a Wireless-G (802.11g) client, a 2.5-mm stereo headset port, and a USB 2.0 host port. Each line can be configured independently to use a unique phone number (or extension) or can use a shared number that is assigned to multiple phones. The power supply for the SPA525G is sold separately.

The Cisco SPA525G IP phone can easily grow with your business. New features can be added to the phone over time via self service portals and firmware updates. New employees or employees who need to move locations can simply plug in their preconfigured phones anywhere on the network, and the network will recognize the change. The phone also provides the option for wireless network connectivity, providing unrestricted placement.

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