Cloud PBX Phones

Polycom

CISCO

Yealink
EASY VoIP

Cloud PBX Phones

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Polycom®

Polycom® VVX® 201

Entry-level Two-line IP phone with HD sound quality and 2 Ethernet ports.

The Polycom® VVX® 201 is a simple, yet reliable, two-line IP phone, with two 10/100 Ethernet ports, that delivers enterprise grade sound quality. The Polycom VVX 201 phone is a stylish, cost effective telephony solution, ideal for retail environments, call centers or shared/common areas, such as lobbies, hallways and break rooms or anywhere needing simple and reliable connectivity.

Unsurpassed voice quality and clarity

The VVX 201 features full duplex Type 1-compliant speakerphone with legendary Polycom® HD VoiceTM and Polycom® Acoustic FenceTM technology that delivers superior sound quality and enables noise- and echo-free conversations that are as natural as being there.

Simplicity and ease-of-use

The Polycom VVX 201 comes with a familiar, intuitive user interface with multi-language support that you can use without having to think about the "how to". The phone features a backlit LCD for improved readability.

Benefits

- Ideal for call centers, retail environments and for shared/ common-areas
- Make more efficient and productive calls with Polycom's HD Voice technology

Polycom® VVX® 201

USER INTERFACE FEATURES

- 2.5 in Graphical Backlit LCD (132 x 64) resolution
- Voicemail support
- Reversible deskstand/wallmount
- 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

FEATURE KEYS

- 4 context-sensitive "soft" keys
- 2 line keys with bi-color (red/green) LED
- "Home" feature key
- 4-way navigation key cluster with center "Select" key
- 2 volume control keys
- Dedicated hold key
- Dedicated headset key
- Dedicated hands-free speakerphone key
- Dedicated microphone mute key

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 ClarityTM technology providing full-duplex conversations, acoustic echo cancellation and background noise suppression
- Type 1 compliant (IEEE 1329 full duplex)
- Frequency response – 150 Hz – 7 kHz for hands-free speakerphone, handset and optional headset mode
- Codecs: G.711 (μ-law), G.729AB
- Individual volume settings with visual feedback for each audio path
- Voice activity detection
- 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 to 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

CALL HANDLING FEATURES[1]

- 2 SIP identities (registrations)
- 2 programmable line keys
- Shared call/bridged line appearance • Flexible line appearance (one or two line keys can be assigned for each registration)
- 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

NETWORK

- SIP Protocol Support
- SDP
- IETF SIP (RFC 3261 and companion RFCs
- Two-port Ethernet switch
- 10/100Base-TX across LAN and PC Ports
- Manual or dynamic host configuration protocol (DHCP) network setup
- Time and date synchronization using SNTP
- QoS Support–IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS, and DHCP
- 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
- Hardware diagnostics
- Status and statistics reporting • TCP
- UDP
- DNS-SRV

SECURITY

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

POWER

- Built-in auto sensing IEEE 802.3af Power over
- Ethernet (Class 2)
- External Universal AC Adapter (optional, 12V 6W 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
- UAE TRA
- Australia RCM
- ROHS compliant
- ICASA
- CITC
- ANATEL[3]
- Customs Union[3]
- KCC[3]
- TAA[3]
- CCC[3]

SAFETY

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

OPERATING CONDITIONS

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

STORAGE TEMPERATURE

- -40 to +70°C (-40 to +160°F)

POLYCOM VVX 201 COMES WITH:

- VVX 201 console
- Handset with handset cord
- Network (LAN) cable - CAT-5E • Quick Start Guide

SIZE

- 6.5 x 6 x 7 in (17 x 15 x 18 cm)(W X H X D) Part Numbers
- 2200-40450-025 – VVX201 WW PoE 2200-40450-019 – VVX 201, Skype for Business, POE

UNIT WEIGHT

- 1.8 lbs (0.8 kg)

MASTER CARTON QUANTITY

- Ten (10)

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] Available in future UC Software release
[3] Planned compliance and localization

Polycom® VVX® 300-310

Powerful entry-level business media phones for today'scubicle workers handling a low to moderate volume of calls delivering crystal clear communications

The Polycom® VVX® 300 is an expandable business media phone that delivers crystal clear communications, enhanced collaboration and personal productivity.

Simplicity and ease-of-use

The VVX 300 phone brings high-quality, cost effective solutions to any environment through advanced UC features. The intuitive user interface of the VVX 300 makes usability and navigation easy and requires minimal training.

Unsurpassed voice quality and clarity

The VVX 300 delivers breakthrough Polycom® HD Voice™ quality for life-like conversations while minimizing fatigue making calls more efficient and productive.

Maximize productivity

Give your front line workers the best experience with this high quality six-line business media phone.

Customizable and expandable

The VVX 300 phone provides personalized information at a glance, through built-in web applications and custom backgrounds. The VVX 300 phone also comes ready for future expansion modules as your users' needs and business grows.

Market-leading open standards interoperability

Designed for enhanced interoperability, the VVX 300 leverages and complements the other existing IT investments in your business. With the broadest call server interoperability in the industry, the Polycom VVX 300 entry level business media phone can become the flexible and future-proof foundation for any organization's unified communications strategy.

Benefits

- Improve productivity for cubicle workers and call center operators through an intuitive easy to use user interface
- Make more efficient and productive calls with the unparalleled voice clarity of Polycom® HD Voice™
- Leverage previous IT infrastructure investments—deploy VVX 300 business media phones on your existing network without needing to upgrade your call control platform

Polycom® VVX® 300-310 Series

USER INTERFACE FEATURES

- Backlit 8-level Grayscale graphical LCD (208 x 104) resolution
- Voicemail support
- 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 pathhandset, 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 – 150 Hz – 7 kHz for handset, optional headset and handsfree speakerphone modes
- Individual volume settings with visual feedback for each audio path
- DTMF tone generation (RFC 2833 and in-band)

HEADSET AND HEARING AID COMPATIBILITY

- Dedicated RJ-9 headset port
- Hearing aid compatibility to 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

CALL HANDLING FEATURES

- 6 lines (programmable line keys)
- Busy Lamp Field (BLF)
- 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
- Do not disturb function

POWER

- External Universal AC Adapter 48VDC; 12W

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[3]
- UAE TRA
- Russia GOST-R[3]
- Brazil ANATEL[3]
- Australia A & C Tick

SAFETY

- UL 60950-1
- CE Mark
- CAN/CSA C22.2 No 60950-1
- EN 60950-1
- IEC 60950-1
- AS/NZS 60950-1
- ICASA (add)
- CITC (add)

OPERATING CONDITIONS

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

STORAGE TEMPERATURE

- -40 – +70°C (-40 – +160°F)

POLYCOM VVX 400 COMES WITH:

- VVX 300 console
- Handset with handset cord
- Network (LAN) Cable - CAT-5E

SIZE

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

PART NUMBERS

- 2200-46135-025 – VVX 400 WW PoE
- 2200-46161-025 – VVX 410 WW PoE

WEIGHT

- 2.0 lbs (0.9 kg)

UNIT BOX DIMENSIONS

- 12 x 8.25 x 4.25 in
- 3.1 lbs (1.4 kg)

COUNTRY OF ORIGIN

- China

WARRANTY

- One (1) year

Polycom® VVX® 410 12-Line

A color mid-range business media phone for today's office workers and call attendants delivering crystal clear communications

The Polycom® VVX® 400 is an expandable color business media phone that delivers crystal clear communications enhanced collaboration and personal productivity.

Simplicity and ease-of-use

The VVX 400 phone brings high-quality, cost effective solution to front line staff handling moderate volume of calls through advanced UC telephony features. The intuitive color user interface of the VVX 400 makes navigation easy and requires minimal training.

Unsurpassed voice quality and clarity

The VVX 400 delivers breakthrough Polycom® HD Voice™ quality for life-like conversations, while minimizing fatigue making calls more efficient and productive.

Maximize productivity

Give your staff the best experience with this high quality twelve line color business media phone.

Customizable and expandable

The VVX 400 phone provides personalized information at a glance, through built-in web applications and custom backgrounds. The VVX 400 phone also comes ready for future expansion modules as your users' need and business grows.

Market-leading open standards interoperability

Designed for enhanced interoperability, the VVX 400 leverages and complements the other existing IT investments in your business. With the broadest call server interoperability in the industry, the Polycom VVX 400 mid-range business media phone can become the flexible and future-proof foundation for any organization's unified communications strategy.

Benefits

- Improve productivity for office staff and knowledge worker's via an intuitive larger, color display and easy to use line appearances
- Make more efficient and productive calls with the unparalleled voice clarity of Polycom® HD Voice™
- Leverage previous IT infrastructure investments—deploy VVX 400 business media phones on your existing network without needing to upgrade your call control platform

Polycom® VVX® 400 Series

USER INTERFACE FEATURES

- Backlit 3.5" color LCD (320 x 240) resolution
- Voicemail support
- 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 pathhandset, 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 – 150 Hz – 7 kHz for handset, optional headset and handsfree speakerphone modes
- Individual volume settings with visual feedback for each audio path
- DTMF tone generation (RFC 2833 and in-band)

FEATURE KEYS

- 4 context-sensitive "soft" keys
- context-sensitive "soft" keys
- Menu Key
- 2 volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

HEADSET AND HEARING AID COMPATIBILITY

- Dedicated RJ-9 headset port
- Hearing aid compatibility to 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

CALL HANDLING FEATURES

- 12 lines (programmable line keys)
- Busy Lamp Field (BLF)
- 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
- Do not disturb function
- Electronic hook switch capable

POWER

- Energy-saving after hours mode
- External Universal AC Adapter 48VDC; 12W

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[3]
- UAE TRA
- Russia GOST-R[3]
- Brazil ANATEL[3]
- Australia A & C Tick
- ROHS compliant

SAFETY

- UL 60950-1
- CE Mark
- CAN/CSA C22.2 No 60950-1
- EN 60950-1
- IEC 60950-1
- AS/NZS 60950-1
- ICASA (add)
- CITC (add)

OPERATING CONDITIONS

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

STORAGE TEMPERATURE

- -40 – +70°C (-40 – +160°F)

POLYCOM VVX 400 COMES WITH:

- VVX 400 console
- Handset with handset cord
- Network (LAN) Cable - CAT-5E

SIZE

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

PART NUMBERS

- 2200-46157-025 – VVX 400 WW PoE
- 2200-46162-025 – VVX 410 WW PoE

WEIGHT

- 2.0 lbs (0.9 kg)

UNIT BOX DIMENSIONS

- 12 x 8.25 x 4.25 in
- 3.1 lbs (1.4 kg)

COUNTRY OF ORIGIN

- China

WARRANTY

- One (1) year

Polycom®

Polycom® VVX 500 16-Line

Superior Voice Quality, and an Intuitive Touch Screen.

Unsurpassed Voice Quality

The Polycom VVX 500 brings life-like richness and clarity to voice communications. Polycom's patented Acoustic Clarity Technology 2, as well as best-in-class system design deliver unprecedented voice quality.

Enhanced Call Handling Capabilities

With the swipe of a finger or a quick touch to the screen, the VVX 500 provides quick and easy access to the features you need such as call hold, call transfer, call park, 3 way conferencing, call transfer and many other time saving features.

Touch Screen User Interface

The VVX 500 delivers all of its capabilities through a simple, touch-driven interface. Its bright color screen makes navigating the features and options quick and easy. Access to missed calls, voicemail and speed dials is as simple as a brief touch or swipe.

Simple Installation

Designed to make installation as simple and efficient as possible, the VVX 500 offers a two-port Gigabit Ethernet switch so you don't have to run separate cables for phone and PC. Once connected to your network, the phone will automatically configure itself and display its assigned extension and phone number.

Polycom® VVX 500 16-Line

LINES

- 16 lines

DISPLAY

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
- Adjustable base height

FEATURE KEYS

- 4 context-sensitive “soft” keys
- context-sensitive “soft” keys
- Menu Key
- 2 volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

HEADSET AND HEARING AID 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)

AUDIO FEATURES

- 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.729AB,
- 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

CALL HANDLING FEATURES

- 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
- Intercom
- Call park
- One-touch speed dial, redial missed call notification
- Do not disturb function
- Electronic hook switch capable

VOICEMAIL

- Message waiting light
- Voicemail notification via email
- Voicemail to email (WAV file)
- 5 min per message
- 90 min message max
- Remote access to voicemail
- Web-based access to voicemail
- Voicemail forwarding via email
- Group voicemail distribution

OTHER FEATURES

- Local feature-rich GUI
- Time and date display
- User-configurable contact directory and call history (missed, placed and received)
- 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
- Two-port 10/100/1000 Mbps Ethernet switch
- Dynamic host configuration protocol (DHCP) network setup
- Time and date synchronization using SNTP
- 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

- Transport Layer Security (TLS)
- Encrypted configuration files3
- Digest authentication

POWER

- External Universal AC adapter (included 48V DC)

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: 0 – 40°C (+32 – 104°F)
- Relative humidity: 5% to 95% (noncondensing)

STORAGE TEMPERATURE

- -40 – +70°C (-40 – 160°F)

POLYCOM VVX 500 COMES WITH:

- VVX 500 console
- Handset with handset cord Power cable
- Network (LAN) cable

SIZE

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

PART NUMBERS

- 2200-44500-025 – WW PoE

WEIGHT

- 2.0 lbs (0.9 kg)

BOX DIMENSIONS/WEIGHT

- 12 x 9 x5 in
- 3.1 lbs (1.4 kg)

WARRANTY

- One year

Polycom®

Polycom®
VVX® 600 20-Line

A premium business media phone delivering best-in-class desktop productivity for busy corporate executives and managers.

The Polycom® VVX® 600 phone is a premium business media phone designed to enhance collaboration and personal productivity.

Simplicity and ease of use

The Polycom VVX 600 phone is built for executives and managers who need a powerful, yet intuitive, expandable office phone that helps them stay connected to lead your organization. Founded on the behavior common to smartphones and tablets, the intuitive gesture-based, multi-touch user interface of the Polycom VVX 600 phone makes navigation easy and requires minimal training. With its combined ergonomic design, Polycom® HD Voice™ quality and a large, high resolution color, multi-touch screen, the Polycom VVX 600 business media phone is ideal.

Maximize productivity

Give your executives and managers the best unified communications (UC) experience and the industry's highest quality business media phone. Designed for a broad range of environments from small and medium businesses to large enterprises, the Polycom VVX 600 improves personal productivity.

Highly customizable and expandable

The Polycom VVX 600 phone provides personalized information at a glance, through built-in web applications and a digital photo frame. Polycom VVX 600 users access streaming content using the included video playback feature.

Benefits

- Improve productivity for executives and managers through larger, color multi touch display and more line appearances
- Make more efficient and productive calls with the unparalleled voice clarity of Polycom® HD Voice™
- Improve work space mobility through Bluetooth headset integration
- Leverage previous IT infrastructure investments deploy Polycom VVX 600 business media phones on your existing network without needing to upgrade your call control platform

Polycom® VVX 600

USER INTERFACE FEATURES

- Gesture based, multi-touch capable capacitive touchscreen
- 4.3in LCD (480x272 pixel) resolution
- 16:9 aspect ratio
- Screen saver and digital picture frame mode
- On-screen virtual keyboard
- Voicemail support
- Dual USB ports (2.0 compliant) for media and storage applications
- Integrated Bluetooth 2.1 EDR
- 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™ [1]
- Polycom® Acoustic Clarity™ technology providing full-duplex conversations, acoustic echo cancellation and background noise suppression—Type 1 compliant (IEEE 1329 full duplex)
- Individual volume settings with visual feedback for each audio path
- DTMF tone generation (RFC 2833 and in-band)

HEADSET AND HEARING AID COMPATIBILITY

- Bluetooth headset pairing (HFP/HSP)
- Dedicated RJ-9 headset port
- Hearing aid compatibility to 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 USB headsets are supported. (See Support site for list of compatible headsets.)

CALL HANDLING FEATURES

- 20 lines
- Busy Lamp Field (BLF)
- 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
- Do not disturb function

POWER

- Built-in auto sensing IEEE 802.3at Power over Ethernet (Class 4). Backwards compatibility with IEEE 802.3af

APPROVALS

- Argentina CNC
- South Africa ICASA
- Saudi Arabia CITC
- India TEC
- Japan MIC/VCCI Class B
- Malaysia SIRIM
- Israel MOC
- Singapore IDA
- Taiwan NCC3
- Mexico NOM-121
- 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
- UAE TRA
- Russia GOST-R
- Brazil ANATEL[3]
- Australia A&C Tick
- ROHS compliant
- China CCC[3]

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

STORAGE TEMPERATURE

- -40 – +70°C (-40 – 160°F)

POLYCOM VVX 600 COMES WITH:

- Polycom VVX 600 console
- Handset with handset cord
- Network (LAN) cable
- External Universal AC Adaptor (optional, 48V 380mA DC)

SIZE

- 21 x 15 x 18 cm (8.5 x 6 x 7 in) (W x H x D)

PART NUMBERS

- 2200-44600-025—WW PoE

WEIGHT

- 2.0 lbs (0.9 kg)

UNIT BOX DIMENSIONS

- 14 x 10 x 5 in
- 3.1 lbs (1.4 kg)

COUNTRY OF ORIGIN

- China

WARRANTY

- One (1) year

Polycom®

SoundStation® IP 5000

Conference Phone

Advanced IP conference phone with Polycom HD Voice™ clarity, designed for small conference rooms and executive offices

The Polycom® SoundStation® IP 5000 conference phone delivers remarkably clear conference calls for small conference rooms and executive offices. It features Polycom HD Voice™ technology, broad SIP interoperability, and a modern design that is ideal for smaller rooms—all at an affordable price.

With Polycom HD Voice technology, the SoundStation IP 5000 conference phone boosts productivity and reduces listener fatigue by turning ordinary conference calls into crystal-clear, interactive conversations. It captures both the deeper lows and higher frequencies of the human voice for conference calls that sound as natural as being there.

For all calls, the SoundStation IP 5000 conference phone delivers advanced audio performance that is designed for executive offices and smaller conference rooms with up to 6 participants. 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. Conference calls are made more productive and efficient by three sensitive microphones with 360° coverage that allow users to speak in a normal voice and be heard clearly from up to 7 feet away. The phone also features technology that resists interference from mobile phones and other wireless devices, delivering clear communications without distractions.

The SoundStation IP 5000 leverages Polycom's strong history in both conference phone and VoIP technology to deliver the most robust standards-based SIP interoperability in the industry.

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 optional AC power kit available for non-PoE environments. Plus, the SoundStation IP 5000 conference phone includes a high-resolution backlit display for vital call information and multilanguage support.

Benefits

- **Unparalleled clarity** – Polycom HD Voice makes your conference calls sound amazingly clear and life-like
- **More productive conference calls** – Patented Polycom Acoustic Clarity™ technology delivers the best conference phone experience with no compromises
- **Ideal for smaller rooms** – 7-foot microphone pickup and a small footprint designed for executive offices and smaller conference rooms with up to 6 participants
- **Advanced IP feature support** – the most feature-rich family of IP conference phones available, with advanced call handling, security, and provisioning features

SoundStation® IP 5000

POWER

- IEEE 802.3af Power over Ethernet (built in)
- Optional external universal AC power supply kit: 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: 4
- On-hook/Off-hook, redial, mute, volume up/down
- 5-way navigation
- Menu

AUDIO FEATURES

- Loudspeaker
  -- Frequency: 250 – 7,000 HZ
  -- Volume: Adjustable to peak volume
- 84 dB at 1/2 meter distance
- Acoustic echo cancellation
- Background noise suppression

CALL HANDLING FEATURES

- Busy Lamp Field (BLF)
- Distinctive incoming call treatment / call waiting
- Call timer
- Call transfer, hold, divert (forward), pickup
- Called, calling, connected party information
- Advanced Local three-way conferencing (conference, join, split, hold, resume)
- One-touch speed dial, redial
- Call waiting
- Automatic off-hook call placement
- Do not disturb function

OTHER FEATURES

- Local feature-rich GUI
- Time and date display
- Corporate Directory Access (search, dial, save to local directory)
- Convenient volume adjustment keys
- User-configurable contact directory and call history (missed, placed, and received)
- Customizable call progress tones
- Wav file support for call progress tones
- Unicode UTF-8 character support. Multilingual user interface encompassing Simplified Chinese, Danish, Dutch, English (Canada / US / UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, Swedish

SAFETY

- CE Mark
- EN60950-1
- IEC60950-1
- UL60950-1
- CAN/CSA C22.2 No.60950-1-03
- AS/NZS60950-1
- RoHS Compliant

EMC

- FCC Part 15 (CFR 47) Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- AS/NZS CISPR22 Class B
- VCCI Class B
- EN22024

PRODUCT SHIPS WITH

- Conference Phone Console
- 25 foot Ethernet cable

ENVIRONMENTAL CONDITIONS

- Operating temperature: 5 – 40°C (+41 – 104°F)
- Relative humidity: 20% – 85% (noncondensing)
- Storage temperature: -30 – +55°C (-22 – 131°F)

PHONE DIMENSIONS

- 10.6 x 2.6 x 11.4 in (26.5x 6.5 x 28.5 cm) (W x H x D)

PHONE CONSOLE WEIGHT

- 1.14 lb (0.52 kg)

BOX DIMENSIONS

- 29.4 x 9.5 x 37.2 cm (11.76 x 3.8 x 14.88 in) (W x H x D)

BOX WEIGHT

- 2.99 lb (1.36 kg)

COUNTRY OF ORIGIN

- China

WARRANTY

- One (1) year

SoundStation® IP 6000
Conference Phone

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

Next-generation IP Conference Phone

The SoundStation IP 6000 is an advanced IP conference phone that delivers superior performance for small to midsize conference rooms. With advanced features and remarkable voice quality, the SoundStation IP 6000 offers a price/ performance breakthrough for the small business.

The SoundStation IP 6000 boosts productivity and reduces 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.

SoundStation® IP 6000

DISPLAY

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

AUDIO FEATURES

- Loudspeaker
- Frequency: 220-14,000 Hz
- Volume: Adjustable to 86 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.729a (Annex B)

CALL HANDLING FEATURES

- 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

VOICEMAIL

- Message waiting light
- Voicemail notification via email
- Voicemail to email (WAV file)
- 5 min per message
- 90 min message max
- Remote access to voicemail
- Web-based access to voicemail
- Voicemail forwarding via email
- Group voicemail distribution

OTHER FEATURES

- Local feature-rich GUI
- Time and date display
- User-configurable contact directory and call history (missed, placed, and received)
- Customizable 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
- Time synchronization with SNTP server
- 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

KEYPAD

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

SECURITY

- Transport Layer Security (TLS)
- Encrypted configuration files
- Digest authentication

SAFETY

- CE Mark
- EN60950-1
- IEC60950-1
- UL60950-1
- CAN/CSA C22.2 No.60950-1-03
- AS/NZS60950-1
- RoHS Compliant

EMC

- FCC Part 15 (CFR 47) Class B
- ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- AS/NZS CISPR22 Class B
- VCCI Class B
- EN22024

TELECOM

- AS/ACIF S004
- Telepermit
- KCC
- GOST-R
- TRA

PROTOCOL SUPPORT

- IETF SIP (RFC 3261 and companion RFCs
- version ships with
- Telephone Console
- 25 foot Ethernet cable

POWER

- External universal AC power supply: 100-240V, 0.4A, 48V/19W

AC POWER VERSION SHIPS WITH:

- Telephone Console
- 25 foot Ethernet cable
- Universal Power Supply
- 7 foot region-specific power cord
- Power Insertion Cable

ENVIRONMENTAL CONDITIONS

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

PHONE DIMENSIONS

- 31.1 x 6.4 x 36.8 cm (12.25 x 2.5 x 14.5 in) (W x H x D)

PHONE CONSOLE WEIGHT

- 1.75 lb (0.8 kg)

BOX DIMENSIONS

- 15.5 x 6.0 x 13.0 in (39.5 x 15 x 33 cm) (W x H x D)

BOX WEIGHT

- 5.1 lb (2.32 kg)

WARRANTY

- One (1) year

SoundStation® IP 7000

SIP-Based IP Conference Phone

Astounding voice quality and clarity from the world's most advanced IP conference phone

The Polycom® SoundStation® IP 7000 is a breakthrough conference phone that delivers outstanding performance and a robust feature set for SIP-based VoIP platforms. It is the most advanced conference phone ever developed, and is ideal for executive offices, conference rooms, and boardrooms.

The SoundStation IP 7000 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 160 Hz to 22 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 7000 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.

The SoundStation IP 7000 is the most flexible and expandable conference phone ever developed. Connect two units together for increased loudness and microphone pickup, as well as multiple call control interfaces in the conference room. Connect up to two optional expansion microphones to a single phone to ensure close proximity for everyone in the room.

In the SoundStation IP 7000, Polycom has combined its rich history in voice conferencing and VoIP technology to develop a groundbreaking new conference phone that is the clear choice for SIP-enabled environments. It shares the same SIP phone software with Polycom's award-winning Polycom® SoundPoint® IP desktop phones—the most comprehensive, reliable and feature-rich SIP products in the industry.

Benefits

- **Polycom® HD Voice™** unparalleled clarity to make your conference calls more efficient and productive
- **Polycom® Acoustic Clarity™ technology**—delivers the best conference phone experience with no compromises
- **Flexible configuration options**—multi-unit connectivity, expansion microphones and integration with Polycom HDX room telepresence solutions to meet the needs of many different types of rooms
- **Strong, robust SIP software**—leverages the most advanced SIP endpoint software in the industry, with advanced call handing, security, and provisioning features

Additional Polycom SoundStation IP 7000 features/benefits

- Equipped with built-in Power over Ethernet (PoE). An optional A/C power kit also available.
- 20 ft (6.1 m) microphone pickup, and even more with optional expansion microphones or multi-unit connectivity, reaching all corners of the room.
- Automatic Gain Control intelligently adjusts the microphone sensitivity based on where participants are seated in the conference room.
- Features technology that resists interference from mobile phones and other wireless devices, delivering clear communications without distractions.

SoundStation® IP 7000

POWER

- IEEE 802.3af Power over Ethernet (built in)
- Optional external universal AC power supply kit: 100-240V, 1.3A, 48V/50W

DISPLAY

- Size (W x H): 255 x 128 pixels
- White LED backlight with custom intensity control

KEYPAD

- Standard 12-key keypad
- Context-dependent soft keys: 4
- On-hook/Off-hook, redial, mute, volume up/down
- Directional navigation wheel

AUDIO FEATURES

- Loudspeaker
  -- Frequency: 160–22,000 Hz
  -- Volume: Adjustable to 88 dB at 1/2 meter peak volume
- Full-duplex: Type 1 compliant with IEEE 1329 full duplex standards
- Individual volume settings with visual feedback for each audio path
- Acoustic echo cancellation
- Background noise suppression

CALL HANDLING FEATURES

- 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
- 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)
- Customizable 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

SAFETY

- UL60950-1
- IEC60950-1
- EN60950-1
- CE Mark
- CSA C22.2, No. 60950-1-03
- AS/NZS60950-1

EMC

- FCC (47 CFR Part 15) Class A
- ICES-003 Class A
- EN55022 Class A
- CISPR22 Class A
- AS/NZS CISPR22 Class A
- VCCI Class A
- EN55024
- RoHS compliant

AC POWER VERSION SHIPS WITH:

- Telephone console
- 25 ft (7.6 m) Ethernet cable
- Universal power supply
- 7 ft (2.1 m) region-specific power cord
- Power insertion cable

ENVIRONMENTAL CONDITIONS

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

PHONE DIMENSIONS

- 37.2 x 7.3 x 39.4 cm (14.6 x 2.9 x 15.5) (W x H x D)

PHONE CONSOLE WEIGHT

- 2.4 lb (1.08 kg)

BOX DIMENSIONS

- 43.3 x 13 x 48.4 cm (17.0 x 5.1 x 19.1 in) (W x H x D)

BOX WEIGHT

- 5.4 lb (2.43 kg)

COUNTRY OF ORIGIN

- Thailand

WARRANTY

- One (1) year

Cisco SPA 303 3-Line IP Phone

Basic and Affordable IP Phone for Business or Home Office

Highlights

- 3 line phone business-class IP phone
- Connects directly to an Internet telephone service provider
- Dual switched Ethernet ports, speakerphone, caller ID, call hold, conferencing, and more
- Easy installation and highly secure remote provisioning, as well as menu-based configuration
- Supports Session Initiation Protocol (SIP)

Telephony Features

- Three voice lines
- Pixel-based display: 128 x 64 monochrome graphical liquid crystal display (LCD)
- Line status: active line indication, name and number
- Menu-driven user interface
- Speakerphone
- Call hold
- Music on hold
- Call waiting
- Caller ID name and number
- Call transfer: attended and blind
- Three-way call conferencing
- 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 forwarding: unconditional, no answer, and on busy
- 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
- Anonymous caller blocking
- 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 duration and start time stored in call logs
- Call timer
- Speed dialing (8 entries)
- Intercom
- Group paging

Cisco SPA 303 3-Line IP Phone

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 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 ports with integrated Ethernet switch: 10/100BASE-T RJ-45
- 5 VDC universal (100–240V) switching included

Regulatory Compliance

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

Documentation

- Quick-Start Installation and Configuration Guide
- User Guide

Package Contents

- Cisco SPA 303 IP phone, handset, and stand
- Handset cord
- RJ-45 Ethernet cable
- Power adapter

POWER SUPPLY

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

PHYSICAL INTERFACES

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

INDICATOR LIGHTS/LEDS

- Speakerphone on/off button with LED
- Headset on/off button with LED
- Mute button with LED
- Message waiting indicator LED
- LED test function

BODY DIMENSIONS

- 220 x 198 x 30 mm (8.66 x 7.80. x 1.18 in.) (W x H x D)

UNIT WEIGHT

- 1.50 lb (0.68kg)

OPERATING TEMPERATURE

- 0º – 40ºC (32º – 104ºF)

STORAGE TEMPERATURE

- -20º – 70ºC (-13º – 185ºF)

OPERATING HUMIDITY

- 5% to 95% noncondensing

STORAGE HUMIDITY

- 5% to 95% noncondensing

Cisco SPA504G 4-Line IP Phone

4-Line Business IP Phone with Enhanced Connectivity and Media for a New Level of User Experience

Highlights

- For business or home office use
- Full-featured 4-line business-class IP phone
- Monochrome backlit display for ease of use, aesthetics, and on-screen applications
- Connects directly to an Internet telephone service provider
- Dual switched Ethernet ports for connecting a computer behind the phone, reducing cabling costs
- Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- Easy installation and highly secure remote provisioning
- Supports Session Initiation Protocol (SIP)

Telephony Features

- Four voice lines
- Line status: active line indication, with name and number
- Menu-driven user interface
- 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
- Automatic redial of last calling and last called numbers
- On-hook dialing
- Call pickup: selective and group
- Call park and unpark
- Call blocking: anonymous
- Call forwarding: unconditional, no answer, on busy
- 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
- Anonymous caller blocking
- On-hook default audio configuration (speakerphone and headset)
- 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
- Speed dialing, eight entries
- Intercom
- Group paging

Cisco SPA504G 4-Line IP Phone

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 ports with integrated Ethernet switch: 10/100BASE-T RJ-45
- 802.3af-compliant PoE

Regulatory Compliance

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

Documentation

- Quick-Start Installation and Configuration Guide
- User Guide

Package Contents

- Cisco SPA504G 4-Line IP phone, handset, and stand
- Handset cord
- RJ-45 Ethernet cable

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 2.5-mm port

INDICATOR LIGHTS/LEDS

- Speakerphone on/off button with LED
- Headset on/off button with LED
- Mute button with LED
- Message waiting LED

BODY DIMENSIONS

- 214 x 212 x 44 mm (8.42 x 8.35. x 1.73 in.) (W x H x D)

UNIT WEIGHT

- 2.00 lb (0.9kg)

OPERATING TEMPERATURE

- 0º – 40ºC (32º – 104ºF)

STORAGE TEMPERATURE

- -20º – 70ºC (-4º – 158ºF)

OPERATING HUMIDITY

- 5% to 95% noncondensing

STORAGE HUMIDITY

- 5% to 95% noncondensing

Cisco SPA525G2 5-Line IP Phone

5-Line Business IP Phone with Enhanced Connectivity and Media for a New Level of User Experience

The Cisco SPA525G2 5-Line IP Phone with Color Display is an excellent choice for businesses that require an enhanced user experience. Part of the Cisco Small Business Series, the SPA525G2 uses industry-leading SPA voice over IP (VoIP) technology from Cisco, with high-quality hardware providing additional connectivity via Bluetooth, PoE (802.3af), or a Wireless-G client (802.11g).

The Cisco SPA525G2 IP phone can easily grow with your business. New employees or employees who need to move to another location can simply plug in their preconfigured phones anywhere on the network, and the network will recognize the change. The phone also supports wireless network connectivity, providing unrestricted placement without the cost of running network cabling.

Standard Cisco SPA525G2 features include five active lines, dual switched Ethernet ports, 802.3af PoE support, a 3.2-inch QVGA color display, a full-duplex, high-quality speakerphone, a Bluetooth interface, a Wireless-G (802.11g) client, a 2.5-mm stereo headset port, and a USB 2.0 host port.

Highlights

- Full-featured and stylish business IP phone
- Cisco Mobile Link: Bluetooth enhanced integration with mobile phones to make and receive calls, import your personal contacts, and charge your mobile phone
- Enhanced network connectivity with Power over Ethernet (PoE), 802.11g Wi-Fi client with Wi-Fi Protected Setup (WPS), and Bluetooth headset support
- Graphics-rich, high-resolution 3.2-inch QVGA 320 x 240 color screen
- Cisco AnyConnect VPN Client: Highly secure Internet phone connection for remote users that is simple and easy to set up
- Cisco XML services framework: Support for productivity applications directly on your phone
- Support for multimedia functions, such as playing MP3s, displaying digital photos, and viewing RSS feeds
- Wideband audio for unsurpassed voice clarity and enhanced speaker quality
- Support for Session Initiation Protocol (SIP)

Cisco SPA525G2 5-Line IP Phone

LIGHTED LINE KEYS

- 5 illuminated line buttons with tricolor LEDs

HOLD KEY

- Puts current call on hold

VOICEMAIL KEY

- 1-button access to voicemail

MENU KEY

- Accesses call history, directory, speed dials, MP3 player, web applications, user preferences, network configuration, device administration, and status

LIGHTED MUTE KEY

- Lights up red when the call is on mute, and turns off when mute is removed. Also lights red in the event of a network failure

LIGHTED HEADSET KEY

- Lights up green when pressed and using a Bluetooth or 2.5-mm headset for hands-free calling

LIGHTED SPEAKERPHONE KEY

- Activates full-duplex speakerphone; stays lit while speakerphone is on

LIGHTED MESSAGE WAITING

- Lights when there is new voicemail; visible on the phone chassis above the LCD screen; stays lit until the new

INDICATOR

- Voicemail has been processed by the user

GRAPHICAL DISPLAY

- Color 3.2-in. QVGA (320 x 240 pixels) backlit LCD graphical display

5-WAY NAVIGATIONAL BUTTONS

- Navigating menus and multimedia applications

4 SOFT-KEY BUTTONS

- Dynamically present calling options to the user

NETWORK FEATURES

- Cisco Discovery Protocol, IEEE 802.1p/Q

WI-FI

- 802.11b/g, Wi-Fi Multimedia (WMM) (802.11e)

WI-FI SECURITY

- Wired Equivalent Privacy (WEP), 64 or 128 bit
- Wi-Fi Protected Access (WPA), Personal and Enterprise
- WPA2, Personal and Enterprise
- Wi-Fi Protected Setup (WPS)

ETHERNET SWITCH

- 10/100 PC switch port enables LAN connectivity to a co-located PC. 802.3af PoE WAN port; disabled when phone is used in Wi-Fi mode

VOLUME CONTROL

- Volume-control toggle provides easy decibel-level adjustments of the handset, monitor speaker, and ringer

APPLICATIONS

- Customizable screen saver on phone display (Photo Album)
- Music player (MP3)

CALL CONTROL AND AUDIO FEATURES

- Call hold
- Music on hold
- Call waiting
- Caller ID name and number and outbound caller ID blocking
- Outbound caller ID blocking
- Call transfer - attended or blind
- Call conferencing - hosted (N-party) or local (3-party)
- Call forwarding - unconditional, no answer, on busy
- Visual voice message waiting indicator (VMWI)
- Call pickup - selective and group
- Call park and unpark
- Call blocking - anonymous
- Do not disturb
- Intercom
- Group paging
- Extension mobility
- Individual volume setting per audio path (headset/handset/speaker)

WEIGHT

- 2.0 lbs (0.9 kg)

BOX DIMENSIONS/WEIGHT

- 12 x 9 x5 in
- 3.1 lbs (1.4 kg)

DIMENSIONS

- 212 x 214 x 44 mm (8.3 x 8.4 x 1.7 in) (W x H x D)

WEIGHT

- 0.9 kg
- 2.0 lb
- 32.8 oz

PHONE CASING COMPOSITION

- Acrylonitrile butadiene styrene (ABS) plastic in textured dark gray with silver-colored bezel

OPERATING TEMPERATURE

- 0° – 45°C (32° – 113°F)

RELATIVE HUMIDITY

- 5% to 95% noncondensing, operating and nonoperating

STORAGE TEMPERATURE

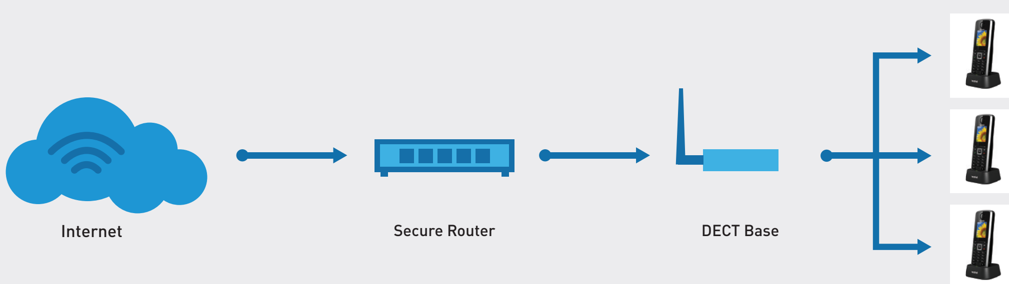
- -25° – 80°C (-13° – 176°F)

IP DECT Phone W52

Flexible VoIP solution for small businesses

- Exceptional sound quality
- Up to 4 simultaneous external calls
- Up to 5 DECT cordless handsets
- 1.8'' color display with intuitive user interface
- 5~10 hours talk time, 100 hours standby time
- Integrated PoE (Class 1)
- Desktop or wall mountable
- OTA (Over-The-Air) update

Yealink W52 is a SIP Cordless Phone System designed for small business and home offices looking for immediate cost savings from a scalable SIP-based mobile communications system. Combining the benefits of wireless communication with rich business features of Voice over IP telephony, the user can benefit from freedom of movement, lifelike voice communications, multi-tasking convenience, professional features like intercom, transfer, call forward, 3-way conferencing, PoE etc.



- Up to 5 DECT Cordless Handsets Per base depending on your needs
- DECT radio coverage up to 150 ft. indoors and 900 ft. outdoors
- Energy-saving ECO features

Yealink IP DECT Phone W52

PHONE FEATURES

- Up to 4 simultaneous calls
- Up to 5 handsets
- Handset select for receiving calls
- Handset and Number select for making calls
- Call hold, call transfer, 3-way conferencing
- Switching between calls
- Call waiting, mute, DND
- Caller ID display, redial
- Anonymous call, anonymous call rejection
- Call forward (always/busy/no answer)
- Speed dial, voicemail
- Message Waiting Indication (MWI)
- Local phonebook for up to 500 entries (store in the base)
- Phonebook search / import / export
- Call history (outgoing/missed/accepted)
- Keypad lock, emergency call
- Music on hold

PERSONALIZATION

- 9 ringer melodies
- Screen saver, two kind of colour schemes
- Multiple languages

VOICE AND CODECS FEATURES

- Full-duplex speakerphone
- Receiver volume control: 5 steps
- Ringer volume control: 5 steps+off
- Multiple advisory tones
- Acoustic warning for low battery status
- DTMF
- Narrowband codec: G.711μ/, G.729

NETWORK FEATURES

- SIP v1 (RFC2543), v2 (RFC3261)
- SNTP/NTP
- 802.1x, LLDP, PPPoE
- STUN Client (NAT Traversal)
- UDP, TCP

CONNECTORS

- CAT-iq2.0
- 1 x RJ45 10/100M Ethernet port
- Power over Ethernet (IEEE 802.3af)
- Headset jack (2.5 mm)
- A mini USB Port

PHYSICAL FEATURES

- Indoor Range: 150 ft.
- Outdoor Range: 900 ft. (In ideal conditions)
- Standby Time: 100h (In ideal conditions)
- Talk Time: 5~10h (The ideal talk time is 10h)
- 1.8'' 128x160 pixels color display
- Desktop or wall mountable
- LCD backlit, key backlit
- Energy-saving ECO mode/ECO Mode+
- 12 key numerical keypad, 5 navigation keys 2 softkeys, 6 function keys, 6 shortcut keys
- 3 LEDs on Base: 1 x power, 1 x Network, 1 x Call
- Base station: DC 5V / 600mA Output
- Charger: DC 5V / 600mA Output
- Phone size: 144mm x 50mm x 24mm
- Base station size: 153mm x 108mm x 45mm
- Operating humidity: 10 ~ 95%
- Operating temp: 14-122°F

PACKAGE FEATURES

- 1 x handset, 1 x base station, 1 x belt clip 2 x rechargeable battery, 1 x charger cradle, 2 x power adapter 1 x ethernet cable, 2 x guide, 1 x CD-ROM
- Qty/CNT: 10pcs
- Giftbox size: 213mm*198mm*97mm
- Carton meas: 495mm*400mm*219mm
- N.W: 16.9 lbs
- G.W: 17.4 lbs

SPECIAL FEATURES

- Increase range with up to 6 repeaters

CERTIFICATIONS

Yealink
EASY VoIP

SIP-T42G

Standard and Affordable SIP Phone for Business

The SIP-T42G is a feature-rich sip phone for business. The 12-Line IP Phone has been designed by pursuing ease of use in even the tiniest details. Delivering a superb sound quality as well as rich visual experience. Supports seamless migration to GigE-based network infrastructure. With programmable Keys, the IP Phone supports vast productivity-enhancing features. Using standard encryption protocols to perform highly secure remote provisioning and software upgrades.

Gigabit

Paperless

Revolutionary new design

Yealink's SIP Phones continue to evolve, the T4 Series have been designed by pursuing ease of use in even the tiniest details, these new designs include a paper label free design, new foot stand allows two positions for the device, non-slip rubber feet, ergonomic recessed buttons etc.

Enhanced Call Management

The SIP-T42G supports vast productivity-enhancing features such as SCA, BLF List, call forward, call transfer, 3-way conference. Three pages of 6 flexible buttons are shown on the display can be programmed up to 15 various features. Support for Yealink YHS32, with EHS36 the user can control phones through a wireless headset.

Efficient Installation and Provisioning

Dual-port Gigabit Ethernet is designed for flexible deployment options and lower cabling expenses. Integrated IEEE 802.3af Power-over-Ethernet allows easy deployment with centralized powering and backup.

Benefits

- Revolutionarily new design
- Dual-port Gigabit Ethernet
- 2.7» 192x64-pixel graphical LCD with backlight
- Paper label free design
- PoE support
- Headset, EHS support
- Integrated stand with 2 adjustable angles
- Wall mountable
- Simple, flexible and secure provisioning options

Yealink SIP-T42G

AUDIO FEATURES

- Narrowband codec: G.711(μ), G.729AB
- DTMF: In-band, Out-of-band(RFC 2833) and SIP INFO
- Full-duplex hands-free speakerphone with AEC
- VAD, CNG, AEC, PLC, AJB, AGC

PHONE FEATURES

- One-touch speed dial, redial
- Call forward, call waiting
- Call transfer, call hold
- Call return
- Mute, auto answer, DND
- 3-way conference call
- Ring tone selection/import/delete
- Hotline, emergency call
- Dial plan

DIRECTORY

- Local phonebook up to 1000 entries
- Black list
- Phonebook search/import/export
- Call history: dialed/received/missed/ forwarded

IP-PBX FEATURES

- Busy Lamp Field (BLF)
- Bridged Line Apperance(BLA)
- Anonymous call, anonymous call rejection
- Hot-desking
- Message Waiting Indicator (MWI)
- Voice mail
- Call park, call pickup
- Intercom, paging,
- Music on hold

DISPLAY AND INDICATOR

- 2.7» 192x64-pixel graphical LCD with backlight
- LED for call and message waiting indication
- Dual-color (red or green) illuminated LEDs for line status information
- Intuitive user interface with icons and soft keys
- National language selection
- Caller ID with name, number

FEATURE KEYS

- 6 line keys with LED
- 6 line keys can be programmed up to 15 various features (3-page view)
- 5 features keys: message, headset, mute, redial, hands-free speakerphone
- 4 context-sensitive "soft" keys
- 6 navigation keys
- 2 volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

INTERFACE

- Dual-port Gigabit Ethernet
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 1XRJ12 (6P6C) EHS port
- Power over Ethernet (IEEE 802.3af), Class 2

OTHER PHYSICAL FEATURES

- Stand with 2 adjustable angles
- Wall mountable
- External universal AC adapter (optional) AC 100~240V input and DC 5V/1.2A output
- Power consumption (PSU): 1.4-3.9W
- Power consumption (PoE): 2.1-5.9W
- Dimension(W*D*H*T): 212mm*189mm*175mm*54mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C

PACKAGE FEATURES

- Qty/CTN: 5 PCS
- N.W/CTN: 6.9 kg
- G.W/CTN: 7.6 kg
- Giftbox size: 241mm*236mm*138mm
- Carton Meas: 707mm*253mm*244mm

MANAGEMENT

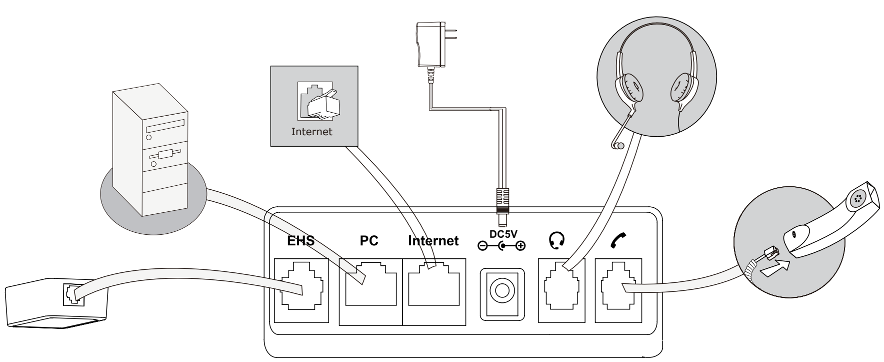
- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

NETWORK AND SECURITY

- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT transverse: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS DSCP
- HTTPS certificate manager
- AES encryption for configuration file
- Digest authentication using MD5/ MD5-sess
- OpenVPN, IEEE802.1X
- IPv6

CERTIFICATIONS

REACH ISO 9001



Yealink
EASY VoIP

SIP-T46G

Revolutionary SIP phone with a multitude of professional features

The SIP-T46G is a revolutionary 16-line IP Phone for executive users and busy professionals. New designs with high-resolution TFT color display, delivering a rich visual experience. Supports Gigabit Ethernet and a variety of device connections, including EHS headset and USB.
With programmable keys, the IP Phone supports vast productivity- enhancing features.

1000 M Gigabit

Paperless

Support Bluetooth Earphone

Revolutionary new design

Yealink's SIP Phones continue to evolve the SIP-T4 Series have been designed by pursuing ease of use in even the tiniest details. The new design includes a paper label free design, new foot stand allowing two positions for the device, non-slip rubber feet, ergonomic recessed buttons etc.

A rich visual experience for applications

The SIP-T46G supports vast productivity enhancing features such as SCA, BLF List, call forward, call transfer, 3-way conferencing, the 4.3 inch back-lit color display enables rich visual presentation and easier navigation of the menu. Three pages of 10 flexible buttons are shown on the display can be programmed up to 27 various features.

Advanced connectivity and expandability

For network connectivity, the SIP-T46G comes with two Gigabit Ethernet ports, one of them suitable for Power over Ethernet. Supports a variety device connections, including EHS headset and USB. User enabled Bluetooth headset through USB Dongle.

Supports up to 6 Expansion Modules, adding up to 240 additional buttons with a screen based LCD display and LED system.

Benefits

- Revolutionarily new design
- Dual-port Gigabit Ethernet
- 4.3» 480 x 272-pixel color display with backlight
- Built-in a USB port, support Bluetooth headset (Through USB Dongle)
- Paper label free design
- PoE support
- Headset, EHS support
- Supports expansion modules
- Stand with 2 adjustable angles
- Wall mountable
- Simple, flexible and secure provisioning options

Yealink SIP-T46G

AUDIO FEATURES

- Codecs: G.711(μ), G.729AB
- DTMF: In-band, Out-of-band(RFC 2833) and SIP INFO
- Full-duplex hands-free speakerphone with AEC
- VAD, CNG, AEC, PLC, AJB, AGC

PHONE FEATURES

- Call hold, mute, DND
- One-touch speed dial, hotline
- Call forward, call waiting, call transfer
- Emergency call
- Redial, call return, auto answer
- 3-way conferencing
- Ring tone selection/import/delete
- Dial plan

DIRECTORY

- Local phonebook (up to 1000 entries)
- Black list
- Phonebook search/import/export
- Call history: dialed/received/missed/ forwarded

IP-PBX FEATURES

- Busy Lamp Field (BLF)
- Bridged Line Apperance (BLA)
- Anonymous call, anonymous call rejection
- Hot-desking
- Message Waiting Indicator (MWI)
- Voice mail
- Call park, call pickup
- Intercom, paging
- Music on hold
- Call completion

DISPLAY AND INDICATOR

- 4.3» 480 x 272-pixel color display with backlight
- 16 bit depth color
- LED for call and message waiting indication
- Dual-color (red or green) illuminated LEDs for line status information
- Wallpaper
- Intuitive user interface with icons and soft keys
- National language selection
- Caller ID with name, number and photo

FEATURE KEYS

- 10 line keys with LED
- 10 line keys can be programmed up to 27 various features (3-page view)
- 7 features keys: message, headset, hold, mute, transfer, redial, hands-free speakerphone
- 4 context-sensitive "soft" keys
- 6 navigation keys
- Volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

INTERFACE

- Dual-port Gigabit Ethernet
- Support Bluetooth headset through USB Dongle
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 1XRJ12 (6P6C) EXT port:
- Supports up to 6 Expansion Modules for an attendant console application
- Power over Ethernet (IEEE 802.3af), class 3

OTHER PHYSICAL FEATURES

- Stand with 2 adjustable angles
- Wall mountable
- External universal AC adapter (optional): AC 100~240V input and DC 5V/2A output
- Power consumption (PSU): 1.8-5.4W Power consumption (PoE): 2.1-8.0W
- Dimension(W*D*H*T): 244mm*213mm*185mm*54mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C

PACKAGE FEATURES

- Qty/CTN: 5 PCS
- N.W/CTN: 8.1KG
- G.W/CTN: 9.16K
- Giftbox size: 274mm*258mm*142mm
- Carton Meas: 732mm*286mm*266mm

MANAGEMENT

- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

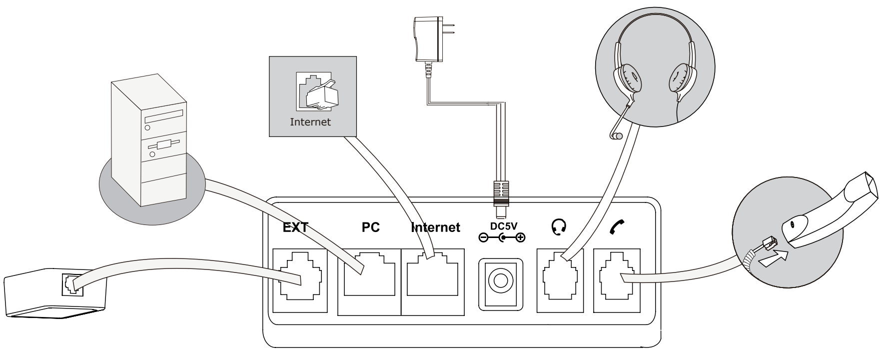
NETWORK AND SECURITY

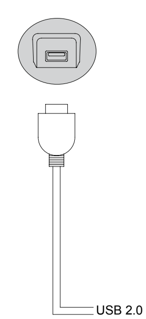
- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT transverse: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP/PPPoE
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS DSCP
- HTTPS certificate manager
- AES encryption for configuration file
- Digest authentication using MD5/ MD5-sess
- OpenVPN, IEEE802.1X
- IPv6

CERTIFICATIONS

CE FC RoHS REACH ISO 9001

3CX Asterisk AVAYA BROADSOFT Genesys





SIP-T48G

Revolutionary SIP phone with a 7-inch touch-panel function

The SIP-T48G is Yealink's most recent innovative 20-line IP Phone for a fast-changing world. Designed specifically for both local and international use by business, industry and commerce, it incorporates a large touch panel that makes switching between different screens and applications swift, easy and convenient. The T48G is also built for the Gigabit Ethernet and facilitates very rapid call handling and the application of accessories such as a Bluetooth USB Dongle, plus wired and wireless headsets. The new IP Phone supports impressive productivity-enhancing features too which make it the natural and obvious efficiency tool for today's busy executives and professionals.

- 1000M Gigabit
- 7" 7 inch touch screen
- Paperless
- Support Bluetooth Earphone

Revolutionary new design

Yealink's SIP Phones continue to evolve. The SIP-T4 Series has been designed by pursuing ease of use in even the tiniest details. The new design includes a 7 inch touch screen, scratch-resistant surface, metallic texture, non-slip rubber feet, ergonomic recessed buttons, etc.

A rich visual experience for applications

SIP-T48G Supports vast productivity-enhancing feature such as call forward, call transfer, 3-way conferencing and a superior UC experience. the 7" 800 x 480-pixel color touch screen with back-light enables rich visual presentation and easier navigation of the menu.

Advanced connectivity

For network connectivity, the SIP-T48G comes with two Gigabit Ethernet ports, one of them suitable for Power over Ethernet. Built-in a USB port, and supports Bluetooth headset through USB Dongle.

Benefits

- Revolutionary new design
- Dual-port Gigabit Ethernet
- PoE support
- 7" 800 x 480-pixel color touch screen with back-light
- Supports Bluetooth headset through USB Dongle
- Paper label free design
- Headset, EHS support
- Supports expansion modules
- Wall mountable
- Simple, flexible and secure provisioning options

Yealink SIP-T48G

AUDIO FEATURES

- Codecs: G.711(μ), G.729AB
- DTMF: In-band, Out-of-band(RFC 2833) and SIP INFO
- Full-duplex hands-free speakerphone with AEC
- VAD, CNG, AEC, PLC, AJB, AGC

PHONE FEATURES

- Call hold, mute, DND
- One-touch speed dial, hotline
- Call forward, call waiting, call transfer
- Emergency call
- Redial, call return, auto answer
- 3-way conferencing
- Ring tone selection/import/delete
- Set date time manually or automatically
- Dial plan

DIRECTORY

- Local phonebook (up to 1000 entries)
- Black list
- Phonebook search/import/export
- Call history: dialed/received/missed/ forwarded

IP-PBX FEATURES

- Busy Lamp Field (BLF)
- Bridged Line Apperance(BLA)
- Anonymous call, anonymous call rejection
- Remote office, hot-desking
- Message Waiting Indicator (MWI)
- Voice mail, call park, call pickup
- Intercom, paging, music on hold
- Call completion

DISPLAY AND INDICATOR

- 7» 800 x 480-pixel color touch screen with backlight
- 24 bit depth color
- LED for call and message waiting indication
- Wallpaper
- Intuitive user interface with icons and soft keys
- National language selection
- Caller ID with name, number and photo

FEATURE KEYS

- 29 one-touch DSS keys
- 7 features keys: message, headset, hold, mute, transfer, redial, hands-free speakerphone
- 6 navigation keys
- Volume control keys
- Illuminated mute key
- Illuminated headset key
- Illuminated hands-free speakerphone key

INTERFACE

- Dual-port Gigabit Ethernet
- Power over Ethernet (IEEE 802.3af), class 0
- Built-in a USB port, support Bluetooth headset (Through USB Dongle)
- 1xRJ9 (4P4C) handset port
- 1xRJ9 (4P4C) headset port
- 1XRJ12 (6P6C) EXT port: Supports up to 6 Expansion Modules for an attendant console application

OTHER PHYSICAL FEATURES

- Wall mountable (optional)
- External universal AC adapter (optional): AC 100~240V input and DC 5V/2A output
- Power consumption (PSU): 2.0-6.4W
- Power consumption (PoE): 2.4-10.5W
- Dimension(W*D*H*T): 266mm*226mm*185mm*54mm
- Operating humidity: 10~95%
- Operating temperature: -10~50°C

PACKAGE FEATURES

- Qty/CTN: 5 PCS
- N.W/CTN: 8.2 kg
- G.W/CTN: 9 kg
- Giftbox size: 324mm*263mm*128mm
- Carton Meas: 660mm*338mm*273mm

MANAGEMENT

- Phone lock for personal privacy protection
- Reset to factory, reboot
- Package tracing export, system log

NETWORK AND SECURITY

- SIP v1 (RFC2543), v2 (RFC3261)
- Call server redundancy supported
- NAT transverse: STUN mode
- Proxy mode and peer-to-peer SIP link mode
- IP assignment: static/DHCP/PPPoE
- HTTP/HTTPS web server
- Time and date synchronization using SNTP
- UDP/TCP/DNS-SRV(RFC 3263)
- QoS: 802.1p/Q tagging (VLAN), Layer 3 ToS DSCP
- HTTPS certificate manager
- AES encryption for configuration file
- Digest authentication using MD5/MD5-sess
- OpenVPN, IEEE802.1X
- IPv6

CERTIFICATIONS

CE FC RoHS REACH ISO 9001

3CX Asterisk Compatible BROADSOFT

Genesys