DATA SHEET

Polycom® SoundPoint® IP 560

Cutting-edge SIP and GigE meet Polycom® HD Voice™

Unrivaled voice experience and ease of provisioning and upgrade in a four-line SIP phone

The Polycom® SoundPoint® IP 560 desktop phone with GigE, a four-line SIP phone that delivers calls of unprecedented richness and clarity and supports a comprehensive range of cutting-edge features to future-proof your investment in network infrastructure. It is ideal for professionals and managers with demanding collaborative communication needs.

Higher productivity with less fatigue

The SoundPoint IP 560 desktop phone features revolutionary Polycom® HD Voice™ technology, which brings life-like richness and clarity to every call[1,2]. Polycom HD Voice technology incorporates wideband audio for over twice the voice clarity and Polycom's patented Acoustic Clarity Technology for crystal-clear, noise- and echo-free sound plus a best-in-class system design for high-fidelity, faithful voice reproduction.

The SoundPoint IP 560 desktop phone is engineered to make installation, configuration, and upgrades as simple and efficient as possible. The phone's built-in IEEE 802.3af PoE circuitry and a dual-port Gigabit Ethernet switch enable flexible deployment options and savings on cabling expenses. It supports remote, zero-touch provisioning and upgrades from a variety of servers as well as boot and call server redundancy to ensure reliable, uninterrupted performance.

Investment protection

Powered by state of the art Gigabit Ethernet IP telephony technology, the SoundPoint IP 560 desktop phone features a dual-port Gigabit Ethernet switch for seamless integration with a PC or desktop server. For organizations with existing GigE deployment, the SoundPoint IP 560 desktop phone delivers unobstructed, high-speed access to productivity-boosting applications. For organizations with plans to migrate to GigE, the SoundPoint IP 560 desktop phone protects investment in network infrastructure.

Benefits

- **Unparalleled sound quality**—Polycom HD Voice technology enables rich, clear, life-like voice communications[1]
- **Advanced features & applications**[2]
  - Four lines
  - Backlit 320 x 160-pixel graphical grayscale LCD
  - Shared call/bridged line appearance
  - Busy lamp field (BLF)
  - Presence, buddy lists
  - XHTML micro-browser for productivity-boosting Web applications
  - Gigabit Ethernet support
- **Ease of provisioning and upgrade**
  - Integrated IEEE 802.3af Power over Ethernet (PoE) support
  - Remote, zero-touch provisioning with support of a variety of protocols including FTP, TFTP, HTTP, or HTTPS
  - Dual-port Gigabit Ethernet switch for flexible deployment options and lower cabling expenses
- **Broad and deep interoperability**—Certified to support a comprehensive set of features with a variety of leading SIP-based IP PBX and Softswitch platforms. For a complete list of certified platforms see *www.polycom.com/vipmatrix*

Product specifications

Lines (Directory numbers)

Up to 4 lines with up to 24 concurrent calls per line

Display

- 320 x 160 pixel backlit gray scale graphical LCD
- LED backlight with custom intensity control
- Message Waiting Indicator (MWI) LED

Feature keys

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

Headset compatibility

- Dedicated RJ-9 headset port
  - Amplified headsets are recommended

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

- Polycom® HD Voice™ technology delivers life-like voice quality for each audio path—the handset, the hands-free speakerphone, and the optional headset
- Full-duplex hands-free speakerphone
  - Type 1 compliant with IEEE 1329 full duplex standards
- Frequency response—100 Hz–7 kHz for handset, optional headset and hands-free speakerphone modes
- Codecs
  - G.722 (wideband)
  - G.711 µ/A
  - G.729A (Annex B)
  - iLBC
- 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

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

- Transport Layer Security (TLS)
- Secure Real-time Transport Protocol (SRTP)
- Shipped with X.509 certificate installed
- 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
- External Source 48V DC @ 380mA

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
- Anatel

Safety

- CE Mark
- EN 60950-1
- IEC 60950-1
- AS/NZS 60950-1
- UL/C-UL

Operating conditions

- Temperature—32–104° F (0–40° C)
- Relative Humidity: 5–95%, non-condensing

Storage temperature

-40–160° F (-40–70° C)

Polycom® SoundPoint® IP 560 ships with

- SoundPoint IP 560 console
- Handset with handset cord
- Base stand
- Network (LAN) cable
- Quick Start Guide
- Product registration card

Size (W x H x D x T)

10.5 x 6 x 7.5 x 2.5 in
(26.5 x 15 x 19 x 6.5 cm)

Weight

Phone weight—2.4 lb (1.1 kg)

Unit box dimensions/weight

- 12.5 x 13.25 x 3.5 in (31.75 x 33.66 x 8.89 cm)
- 4.0 lbs (1.8 kg)

Master carton quantity

Five

Country of origin

Thailand

Warranty

1 year

1. In some calling scenarios, such as IP to PSTN, Polycom HD Voice will not be available and the call will progress in narrowband instead.
2. Some of these features need to be supported by an appropriate call/applications server.

About Polycom

Polycom is the global leader in open standards-based unified communications and collaboration (UC&C) solutions for voice and video collaboration, trusted by more than 415,000 customers around the world. Polycom solutions are powered by the Polycom® RealPresence® Platform, comprehensive software infrastructure and rich APIs that interoperate with the broadest set of communication, business, mobile and cloud applications and devices to deliver secure face-to-face video collaboration in any environment.

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