POLYCOM®

A POLYCOM DATA SHEET

Polycom® SoundStation® IP 6000
IP Conference Phone

Next-generation IP conference phone designed for midsize rooms

Delivering superior performance for midsize conference rooms, the Polycom® SoundStation® IP 6000 conference phone offers a price to performance breakthrough for SIP environments with advanced features such as broad interoperability and remarkable voice quality. 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 that sound as natural as being there.

From full-duplex audio that eliminates distracting drop-outs to the latest echo cancellation advancements, only Polycom can deliver a conference phone experience with no compromises. Automatic Gain Control intelligently adjusts the microphone sensitivity based on where participants are seated in the room. The SoundStation IP 6000 also resists interference from mobile phones and other wireless devices, delivering clear communications without distractions. Optional expansion microphones and lapel microphone support make the SoundSound IP 6000 the optimal choice in applications where flexible room coverage is required.

The SoundStation IP 6000 shares the same UC software as Polycom's award-winning SoundPoint IP products to deliver the most robust and feature-rich standards-based SIP interoperability in the industry. Robust provisioning, management and security features make Polycom's IP conference phones the only choice for meeting rooms in SIP-based environments. A high-resolution backlit display with multi-language support offers call information and context sensitive call functions.

About Polycom

Polycom is the global leader in standards-based unified communications (UC) solutions for telepresence, video, and voice powered by the Polycom® RealPresence® Platform. The RealPresence Platform interoperates with the broadest range of business, mobile, and social applications and devices. More than 400,000 organizations trust Polycom solutions to collaborate and meet face-to-face from any location for more productive and effective engagement with colleagues, partners, customers, specialists, and prospects. Polycom, together with its broad partner ecosystem, provides customers with the best TCO, scalability, and security for video collaboration, whether on-premises, hosted, or cloud-delivered. Visit www.polycom.com or connect with Polycom on Twitter, Facebook, and LinkedIn.

Benefits

Polycom HD Voice – unparalleled clarity to make your conference calls more efficient and productive

Polycom's patented Acoustic Clarity Technology – deliver the best conference phone experience with no compromises

12-foot microphone pickup – combined with Automatic 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 maximize 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
Features and Specifications

Power
- IEEE 802.3af Power over Ethernet (Class 0)
- 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, menu, navigation keys

Audio Features
- 3 cardioid microphones 200 to 14,000 Hz
- Loudspeaker frequency response: 220-14,000 Hz
- 12ft (3.6m) microphone pickup
- Volume: Adjustable to 86 dB at 0.5 meter peak volume
- Full-duplex: Type 1 compliant with IEEE 1329
- Individual volume settings with visual feedback for each audio path
- Automatic Gain Control
- 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, G722.1
  -G.722.1C
  -Siren 14
  -Siren 22 (receive audio)
  -iLBC 13.33 and 15.2kbps

SIP Call Handling Features
- Call transfer, hold, divert (forward) and pickup
- Distinctive incoming call treatment/call waiting
- Advanced Local three-way conferencing (conference, join, split, hold, resume)
- One-touch speed dial, redial
- Remote missed call notification
- Automatic off-hook call placement
- SIP URI dialing
- Do not disturb function
- Shared call/bridged line appearance
- Busy Lamp Field (BLF)
- Multicast Group Paging and Push-to-Talk

Other Features
- SIP Server Redundancy
- Time and date display/call timer
- User-configurable contact directory and call history (missed, placed, and received)
- Corporate Directory (LDAP) support
- User selectable ringer tones
- Wave file support for call progress tones
- Unicode UTF-8 character support
- Multilingual user interface encompassing Simplified Chinese, Traditional Chinese, Danish, Dutch, English (Canada /US/UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, Swedish
- Called, calling, connected party information

Interfaces
- Ethernet 10/100 Base-T
- 2.5 mm connection port (audio in) e.g. for lapel pin microphone
- 2 RJ-9 ports for wired/wireless expansion microphones

Network and Provisioning
- IP Address Configuration: DHCP and Static IP
- Time synchronization with SNTP server
- FTP/TFTP/FTPS/HTTP/HTTPS server-based central provisioning for mass deployments. Provisioning server redundancy supported.
- Web portal for individual unit configuration and online software upgrade
- QoS Support: IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS and DSCP
- Network Address Translation (NAT) support - static
- RTCP support (RFC 1889)
- Configuration import/export
- Local digit map (dialing plan)
- Hardware diagnostics
- Status and statistics
- Reset to factory settings

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
- IEEE 802.1x Network Access Control

Safety
- UL60950-1
- IEC60950-1
- EN60960-1
- CE Mark
- CAN/CSA C22.2, No. 60950-1-03
- AS/NZS60950-1

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
- EN55024
- RoHS Compliant

Telecom
- AS/ACIF S004
- Telepermit
- KC
- TRA
- ANATEL
- GOST-R

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

Power over Ethernet version ships with
- Conference Phone Console
- 25 foot (7.6m) Ethernet cable
- Quick Start Guide

AC Power version ships with
- Conference Phone Console
- 25 foot (7.6m) Ethernet cable
- Universal Power Supply 48V, 0.4A
- 7 foot (2.1m) region-specific power cord
- Power Insertion Cable
- Quick Start Guide

Accessories
- 2 wired or wireless expansion microphones
- Lapel pin microphone (limited availability)

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

Warranty
- 1 year

Country of Origin
- Thailand

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

Phone Console Weight
- 1.75 lb (0.8 kg)

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

Box Weight
- 5.1 lb (2.32 kg)

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