Polycom® SoundStation® IP 6000

Polycom™

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

The Polycom® 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 offers a price/performance breakthrough for SIP-enabled IP environments.

- **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 compro¬mises
- **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
- **High-resolution display**— enables robust call information and multi-language support

Power

- IEEE 802.3af Power over Ethernet (built in)
- Optional external universal AC power supply: 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
- 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

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
- 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)
- Customizable call progress tones
- Wave 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

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 synchronization with SNTP server
- 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

- 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

- 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

- 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

HDX room telepresence systems ready version ships with

- Telephone console
- 25 ft (7.6 m) Ethernet cable
- 15 ft (4.6) C-link cable for connection to HDX group video systems
- Quick Start Guide

Environmental conditions

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

Warranty

- 1 year

Country of origin

- Thailand

Phone dimensions (L x W x H)

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

Phone console weight

- 1.75 lb (0.8 kg)

Box dimensions (L x W x H)

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

Box weight

- 5.1 lb (2.32 kg)