TELEPRIME™

"Connecting Technology to People Worldwide"™

POLYCOM®

A POLYCOM DATA SHEET

Polycom® SoundStation® IP 6000

SIP-Based IP Conference Phone

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

HDvoice

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

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

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

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

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

Learn More

To find out how Polycom solutions can help your organization, visit us at www.polycom.com or speak with a Polycom Account Representative.

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

SoundStation IP 6000 Conference Phone Specifications

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

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

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

Audio Features
- Loudspeaker
- Frequency: 220-14,000 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.711 (A-law and Mu-law)
- G.729a (Annex B)
- G.722, G.722.1
- G.722.1C
- Siren 14

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

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

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

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

Safety
- 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

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

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

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