Polycom® SoundStation ® 6000

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

DATA SHEET

The Polycom® SoundStation® 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
Polycom HD Voice makes your conference calls sound amazingly clear and life-like.

Patented Polycom 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 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 handling, 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 mutli-language support.

DATA SHEET Polycom SoundStation 6000 Specifications

Features and Specifications

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: 3
- On-hook/Off-hook, redial, mute, volume up/down

Audio Features

- Loudspeaker
- Frequency: 250-14,000 HZ
- Volume: Adjustable to peak volume 86dB at ½ 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
- Support Codecs
  - G.711 (A law and Mµ-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
- Advanced Local three-way conferencing (conference, join, split, hold, resume)
- 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 GUY
- 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 support
- IP Address Configuration: DHCP and Static IP
- EX mic ports: Two RJ-9 ports
- 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- IEEEE 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[3]

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
- EN55024

Telecom

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

Protocol support

- IETF SIP (RFC 3261 and companion RFCs)

IEEE 802.3af POE 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:
- 41-104 degrees F (5-40 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)
- Box- 14.88 x 11.76 x 3.8 in (37.2 x 29.4 x 9.5 cm)

Phone Console Weight

1.75lb (0.8 kg)