Polycom® SoundStation® IP 6000

SIP-Based IP Conference Phone

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

The Polycom SoundStation IP 6000 is an advanced IP conference phone designed for small to medium- sized rooms. It includes advanced features and high voice quality that ensures excellent performance.

The Polycom SoundStation IP 6000 is HD voice-enabled, a technology that delivers high quality audio that ensures clear conversations. Ordinary conference calls are transformed into interactive conversations that sound as natural as face to face conversations. With an audio range from 220 Hz to 14 KHz, the Polycom SoundStation IP 6000 is able to capture the deep lows and high frequencies of the human voice.

It also employs Acoustic Clarity Technology, a patented technology that enables users to experience conference calls without any compromises. Drop-outs and echo are virtually eliminated, far exceeding previous generations of conference phones.

The Polycom SoundStation IP 6000 improves the user experience with its 12-foot microphone pick-up combined with Automatic Gain Control. You may even add up to two optional expansion microphones for larger rooms. It also employs technology that resists interference from cell phones and other wireless devices.

The high resolution phone display with multi-language support enables users to see call information clearly.

Automatic provisioning and management features make the Polycom SoundStation IP 6000 the ideal choice for conference calls. Use it with the integrated Power over Ethernet (PoE) feature or with an AC adapter for non-PoE environments.

Features and Specifications

Power

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

Display

- Size (pixels): 255 x 128 (W x H)
- 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

Audio Features

- Loudspeaker
  - Frequency: 220-14,000 Hz
  - Volume: Adjustable to 85 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

Call Handling Features

- Distinctive incoming call treatment / call waiting
- Call timer
- Call transfer, hold, pickup
- Called, calling, connected party information
- Three-way conferencing
- One-touch speed dial, redial
- Call waiting
- 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
- 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, SwedishOne-touch speed dial, redial

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
- Web portal for individual unit configuration
- QoS Support -- IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS and DSCPSpanish, SwedishOne-touch speed dial, redial
- 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

Safety

- UL1950
- CE Mark
- CSA C22.2, No 60950
- EN60950
- IEC60950
- AS/NZSS3260

Approvals

- FCC (47 CFR Part 15) Class B
- ICES-003 Class
- EN55022 Class B
- CISPR22 Class B
- AS/NZS 3548 Class B
- VCCI Class B
- EN61000-3-2; EN61000-3-3
- EN55024
- ROHS compliant

Protocol Support

- IETF SIP (RFC 3261 and companion RFCs)

Included

- Telephone Console
- Ethernet cable
- Universal Power Supply
- Power Insertion Cable
- User Guide

Features and Specifications continued

Environmental Conditions

- Operating temperature: 32 - 104 degrees F (0 - 40 degrees C)
- Relative humidity: 20% - 85% (non-condensing)
- 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)