Polycom® SoundStation ® 7000

Astounding voice quality and clarity from the world's most advanced IP conference phone.

DATA SHEET

The Polycom® SoundStation® IP 7000 is a breakthrough conference phone that delivers outstanding performance and a robust feature set for SIP-based VoIP platforms. It is the most advanced conference phone ever developed, and is ideal for executive offices, conference rooms, and boardrooms.

Polycom® HD Voice
Unparalleled clarity to make your conference calls more efficient and productive.

Polycom® Acoustic Clarity™ technology
Delivers the best conference phone experience with no compromises.

Flexible Configuration Options
Multi-unit connectivity, expansion microphones and integration with Polycom HDX room telepresence solutions to meet the needs of many different types of rooms.

Strong, robust SIP software
Leverages 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.

Large high-resolution display with XHTML microbrowser
Enables new applications that make conference calling easier and more functional.

DATA SHEET Polycom SoundStation 7000 Specifications

Features and Specifications

Power

- IEEE 802.3af Power over Ethernet (built in)
- Optional external universal AC power supply kit: 100-240v, 1.3A, 48V/50W

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
- Directional navigation wheel

Audio Features

- Loudspeaker
- Frequency: 160-22,000 Hz
- Volume: Adjustable to peak volume 88 dB at ½ meter peak volume
- Full-duplex: Type 1 compliant with IEEE 1329 full duplex standards
- 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 µ-law)
  - G.729a (Annex B)
  - G.722, G.722.1
  - G.722.1C
  - Polycom® Siren™ 14
  - Polycom® Siren™ 22

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
- USB ports: Mini an regular USB 1.1
- EX mic ports: Two Walta 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- 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
- Field upgradeable

Security

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

Safety

- CE Mark
- EN60950-1
- IEC60950-1
- UL60950-1
- CSA C22.2 No.60950-1-03
- AS/NZS60950-1

EMC

- FCC (47 CFR Part 15) Class A
- ICES-003 Class A
- EN55022 Class A
- CISPR22 Class B
- AS/NZS CISPR22 Class A
- VCCI Class A
- EN55024
- RoHS compliant

Protocol support

- IETF SIP (RFC 3261 and companion RFCs)
- IEEE 802.3af Power over Ethernet version

AC Power version ships with

- Telephone console
- Universal power supply
- 7 ft. region-specific power cord
- Power insertion cable
- Conference Phone Console
- 25 foot Ethernet 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

- 15.5 x 14.6 x 2.9 in (39.4 x 37.2 x 7.3 cm) (L x W x H)
- Box- 19.1 x 17.0 x 5.1 in (48.4 x 43.3 x 13 cm)

Phone Console Weight

5.4 lb (2.43 kg)