



Product Name: Polycom SoundStation IP6000 IP Conference Phone Manufacturer: -Model Number: 2200-15600-001

Please Note: This product has been discontinued. For an alternative, please see the other Poly Products.

Polycom SoundStation IP6000 IP Conference Phone The Polycom SoundStation IP6000 is an advanced IP conference phone that delivers superior performance, for small to medium-sized conference rooms. The SoundStation IP6000 also delivers advanced HD audio performance, whilst also eliminating distracting call drop-outs.Polycom SoundStation IP6000 Key Features

- ï¿<sup>1</sup>/<sub>2</sub> Polycom HD Voice
- ï¿1/2 Acoustic Clarity Technology

i¿1/2 12-foot microphone pickup – Add two optional expansion microphones for even greater coverage

- ï¿1/2 Robust interoperability Compatible with most SIP call platforms
- ï¿1/2 High-resolution display
- ï¿1/2 Integrated Power over Ethernet (PoE enabled)

Intelligent ConferencingAutomatic Gain Control adjusts the microphone sensitivity based on where participants are seated in the conference room, making the conversations clearer without distortion. It also features technology that resists interference from mobile phones and other wireless devices, delivering clear communications without distractions. Polycom SoundStation IP6000 - Technical 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

ï¿1/2 Standard 12-key keypad

- ï¿1/2 Context-dependent soft keys: 3
- ï¿1/2 On-hook/Off-hook, redial, mute, volume up/down

#### Audio Features

- ï¿1/2 Loudspeaker
- ï¿1/2 Frequency: 220-14,000 Hz
- ï¿1/2 Volume: Adjustable to 86 dB at 1/2 meter peak volume
- i¿1/2 Individual volume settings with visual feedback for each audio path
- ï¿1/2 Voice activity detection
- ï¿1/2 Comfort noise fill
- i¿½ DTMF tone generation / DTMF event RTP payload
- ï¿1/2 Low-delay audio packet transmission
- ï¿1/2 Adaptive jitter buffers



ï¿1/2 Packet loss concealment

� Acoustic echo cancellation

ï¿1/2 Background noise suppression

ï¿1/2 Supported Codecs

ï¿1/2 G.711 (A-law and Mu-law)

� G.729a (Annex B)

� G.722, G.722.1

� G.722.1C � Siren 14

Call Handling Features

ï¿1/2 Shared call / bridged line appearance

ï¿1/2 Busy Lamp Field (BLF)

ï¿1/2 Distinctive incoming call treatment/ call waiting

ï¿1/2 Call timer

ï¿1/2 Call transfer, hold, divert (forward), pickup

ï¿1/2 Called, calling, connected party information

ï¿1/2 Local three-way conferencing

ï¿1/2 One-touch speed dial, redial

ï¿1/2 Call waiting

ï¿<sup>1</sup>/<sub>2</sub> Remote missed call notification

ï¿1/2 Automatic off-hook call placement

ï¿1/2 Do not disturb function

**Other Features** 

ï¿1/2 Local feature-rich GUI

ï¿1/2 Time and date display

i¿1/2 User-configurable contact directory and call history (missed, placed, and received)

i¿1/2 Customisable call progress tones

ï¿1/2 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

ï¿1/2 Ethernet 10/100 Base-T

ï¿<sup>1</sup>/<sub>2</sub> 2.5mm connection port

ï¿1/2 EX mic ports: Two RJ-9 ports

ï¿1/2 IP Address Configuration: DHCP and Static IP

ï¿1/2 Time synchronisation with SNTP server

ï¿1/2 FTP / TFTP / HTTP / HTTPS serverbased central provisioning for mass deployments.

Provisioning server redundancy supported.

ï¿1/2 Web portal for individual unit configuration

ı̈¿ $\frac{1}{2}$  QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS and DSCP

i¿1/2 Network Address Translation (NAT) support – static

ï¿<sup>1</sup>/<sub>2</sub> RTCP support (RFC 1889)

ï¿1∕2 Event logging

� Local digit map

ï¿1/2 Hardware diagnostics

ï¿<sup>1</sup>/<sub>2</sub> Status and statistics

ï¿1/2 User selectable ringer tones

ï¿1/2 Convenient volume adjustment keys



#### ï¿1/2 Field upgradeable

Security

i¿½ Transport Layer Security (TLS)
i¿½ Encrypted configuration files
i¿½ Digest authentication
i¿½ Password login
i¿½ Support for URL syntax with password for boot server
i¿½ HTTPS secure provisioning
i¿½ Support for signed software executables

Safety

i¿½ CE Mark i¿½ EN60950-1 i¿½ IEC60950-1 i¿½ UL60950-1 i¿½ CAN/CSA C22.2 No.60950-1-03 i¿½ AS/NZS60950-1 i¿½ 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

i¿1/2 IETF SIP (RFC 3261 and companion RFCs

**Environmental Conditions** 

ï¿<sup>1</sup>/<sub>2</sub> Operating temperature: 32 - 104 degrees F (0 - 40 degrees C)

- ï¿1/2 Relative humidity: 20%-85% (noncondensing)
- ï¿1/2 Storage temperature: -22 131 degrees F (-30 55 degrees C)

Weight and Dimensions

i¿½ 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) i¿½ Phone console weight: 1.75 lb (0.8 kg) i¿½ Box dimensions: 13.0 x 15.5 x 6.0 in (33 x 39.5 x 15 cm) (L x W x H)



ï¿1/2 Box weight: 5.1 lb (2.32 kg

#### **Please Enquire**

Options available for Polycom SoundStation IP6000 IP Conference Phone :

Power Supply Required? IP6000 PSU (+£75.00), - Not Required -.