Polycom SoundStation IP 7000 (SIP) Conference Phone

Product Name: Polycom SoundStation IP 7000 (SIP) Conference Phone

Manufacturer: -

Model Number: 2230-40600-025

Please Note: This product has been discontinued. Please see alternatives here.

Polycom SoundStation IP 7000 (SIP) Conference Phone
The 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.
 SoundStation IP 7000 Features

ï¿½ HD Voice
ï¿½ Acoustic Clarity Technology
ï¿½ Flexible Configuration
ï¿½ Industry-leading SIP Software
ï¿½ Robust Interoperability
ï¿½ High-resolution Display
ï¿½ Built-in Power over Ethernet (PoE)
ï¿½ 20-foot Microphone Pickup
ï¿½ Automatic Gain Control
ï¿½ Delivers Clear Communications
ï¿½ Built-in 2.5mm Applications Port

 SoundStation IP 7000 Technical Specification
Power

ï¿½ IEEE 802.3af Power over Ethernet (built in)
ï¿½ Optional external universal AC power supply:1
ï¿½ 00-240V, 1.5A, 48V/50W

Display

ï¿½ Size (pixels): 255 x 128 (W x H)
ï¿½ White LED backlight with custom intensity control

Keypad

ï¿½ Standard 12-key keypad
ï¿½ Context-dependent soft keys: 4
ï¿½ On-hook/Off-hook, redial, mute, volume up/down
ï¿½ Directional navigation wheel

Audio Features

ï¿½ Loudspeaker - Frequency: 160-22,000 Hz - Volume: Adjustable to 88 dB at 1/2 meter peak volume
ï¿½ Full-duplex: Type 1 compliant with IEEE 1329 full duplex standards
ï¿½ Individual volume settings with visual feedback for each audio path
ï¿½ Voice activity detection
ï¿½ Comfort noise fill
ï¿½ DTMF tone generation / DTMF event RTP payload

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ï¿½ Low-delay audio packet transmission
ï¿½ Adaptive jitter buffers
ï¿½ Packet loss concealment
ï¿½ Acoustic echo cancellation
ï¿½ Background noise suppression
ï¿½ Supported Codecs - G.711 (A-law and Mu-law) - G.729a (Annex B) - G.722, G.722.1 - G.722.1C - Siren 14 - Siren 22 - Call Handling Features
ï¿½ Shared call / bridged line appearance
ï¿½ Busy Lamp Field (BLF)
ï¿½ Distinctive incoming call treatment / call waiting
ï¿½ Call timer
ï¿½ Call transfer, hold, divert (forward), pickup
ï¿½ Called, calling, connected party information
ï¿½ Local three-way conferencing
ï¿½ One-touch speed dial, redial
ï¿½ Call waiting
ï¿½ Remote missed call notification
ï¿½ Automatic off-hook call placement
ï¿½ Do not disturb function

Other Features

ï¿½ Local feature-rich GUI
ï¿½ Time and date display
ï¿½ User-configurable contact directory and call history (missed, placed, and received)
ï¿½ Customisable call progress tones
ï¿½ Wave file support for call progress tones
ï¿½ Unicode UTF-8 character support.
ï¿½ Multilingual user interface

Network and Provisioning

ï¿½ Ethernet 10/100 Base-T
ï¿½ 2.5mm connection port
ï¿½ USB ports: Mini and regular USB 1.1 (not active at launch)
ï¿½ EX mic ports: Two Walta ports
ï¿½ IP Address Configuration: DHCP and Static IP
ï¿½ Time synchronisation with SNTP server
ï¿½ FTP / TFTP / HTTP / HTTPS serverbased central provisioning for mass deployments. provisioning server redundancy supported.
ï¿½ Web portal for individual unit configuration
ï¿½ QoS Support – IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS and DSCP
ï¿½ Network Address Translation (NAT) support - static
ï¿½ RTCP support (RFC 1889)
ï¿½ Event logging
ï¿½ Local digit map
ï¿½ Hardware diagnostics
ï¿½ Status and statistics
ï¿½ User selectable ringer tones
ï¿½ Convenient volume adjustment keys
ï¿½ Field upgradeable

Security

ï¿½ Transport Layer Security (TLS)

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ï¿½ Encrypted configuration files
ï¿½ Digest authentication
ï¿½ Password login
ï¿½ Support for URL syntax with password for boot server
ï¿½ HTTPS secure provisioning
ï¿½ Support for signed software executables

Safety

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

EMC

ï¿½ FCC (47 CFR Part 15) Class A
ï¿½ ICES-003 Class A
ï¿½ EN55022 Class A
ï¿½ CISPR22 Class A
ï¿½ AS/NZS CISPR22 Class A
ï¿½ VCCI Class A
ï¿½ EN55024
ï¿½ RoHS compliant

Protocol Support

ï¿½ IETF SIP (RFC 3261 and companion RFCs)

IEEE 802.3af Power over Ethernet version ships with

ï¿½ Telephone Console
ï¿½ 25 foot Ethernet cable
ï¿½ Quick Start Guide
ï¿½ Quick User Guide

Protocol Support

ï¿½ IETF SIP (RFC 3261 and companion RFCs

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AC Power version ships with

ï¿½ Telephone Console
ï¿½ 25 foot Ethernet cable
ï¿½ Universal Power Supply
ï¿½ 7 foot region-specific power cord

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ï¿½ Power Insertion Cable
ï¿½ Quick Start Guide
ï¿½ Quick User Guide

HDX Ready version ships with

ï¿½ Telephone Console
ï¿½ 25 foot Ethernet cable
ï¿½ 15 foot C-link cable for connection to HDX
ï¿½ Quick Start Guide
ï¿½ Quick User Guide

Environmental conditions

ï¿½ Operating temperature: 32 - 104 degrees F (0 - 40 degrees C)
ï¿½ Relative humidity: 20%-85% (noncondensing)
ï¿½ Storage temperature: -22 - 131 degrees F (-30 - 55 degrees C)

Warranty

ï¿½ 1 year

Country of origin

ï¿½ Thailand

Phone Dimensions

ï¿½ 39.4 x 37.2 x 7.3 cm (15.5 x 14.6 x 2.9 in) (L x W x H)

Phone Console Weight

ï¿½ 1.08kg (2.4lb)

Box Dimensions

ï¿½ 48.4 x 43.3 x 13 cm (19.1 x 17.0 x 5.1 in) (L x W x H)

Box Weight

ï¿½ 2.43kg (5.4lb)

Please Enquire