Product Spec Sheet from Rentex

Polycom SoundStation Duo Dual-Mode

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Mfr Part No. 2200-19000-001

The Polycom SoundStation Duo dual-mode analog/IP conference phone provides exceptional deployment flexibility and best-in-class investment protection. Designed for small to midsize rooms, SoundStation Duo is easy to set up and use. And, it delivers Polycom's legendary audio performance. In VoIP environments, the SoundStation Duo conference phone delivers the most robust, standards-based interoperability in the industry.

Polycom HD Voice technology. Full-duplex audio. The latest in echo cancellation and resistance to interference from mobile phones and other wireless devices. These features and more help the SoundStation Duo conference phone deliver unrivaled group-conferencing experiences without distractions.

Features

- Operates in analog telephony environments and supports the migration to VoIP
- Compatible with a broad array of SIP call platforms to maximize voice quality and feature availability while simplifying management and administration
- Strong, robust SIP software

Detailed Specifications

Display	Size (W x H): 248 x 68 pixels White LED backlight with custom intensity control
Keypad	Standard 12-key keypad; Context-dependent soft keys: 4 On-hook/Off-hook, conference, redial, mute, volume up/down, menu, 5-way navigation keys
Audio Features	3 cardioid microphones: 200–7000Hz; Loudspeaker frequency response: 220–7000Hz; 10 ft (3 m) microphone pickup; Volume: Adjustable to 86 dB at 0.5 meter peak volume Full-duplex: Type 1 compliant with IEEE 1329 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, iLBC 3.33 and 15.2kbps
Other Features	Automated failover (SIP to PSTN); SIP Server Redundancy; Time and date display/call timer; User-configurable contact directory and call history (missed, placed, and received); Corporate Directory (LDAP) support; User selectable ringer tones; Wave file support for call progress tones; Unicode UTF-8 character support; Multilingual user interface encompassing Simplified Chinese, Traditional Chinese, Danish, Dutch, English (Canada /US/UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, Swedish; Called, connected party information; Support for multiple Caller ID standards: Bellcore Type 1, ETSI, DTMF
SIP Call Handling Features	Call hold, Call transfer, divert (forward) and pickup; Distinctive incoming call treatment/call waiting; Advanced Local three-way conferencing (conference, join, split, hold, resume); One-touch speed dial, redial; Remote missed call notification; Automatic off-hook call placement; SIP URI dialing; Do not disturb function; Shared call/bridged line appearance; Busy Lamp Field (BLF); Multicast Group Paging and Push-to-Talk
Interfaces	Ethernet 10/100 Base-T; Two-wire RJ-11 analog PBX or PSTN interface; 2.5 mm connection port; 2 RJ9 ports for wired expansion microphones
Power	IEEE 802.3af Power over Ethernet External universal AC power supply: 100–240 V, 24 V, 0.5 A, 2.5 mm DC plug
Weight	1.62 lb (0.74 kg)
Dimensions	15.6 x 12.9 x 2.5 in (34.6 x 32.7 x 6.4 cm)