Polycom® VVX® 201

Entry-level Two-line IP phone with HD sound quality and 2 Ethernet ports.

The Polycom® VVX® 201 is a simple, yet reliable, two-line IP phone, with two 10/100 Ethernet ports, that delivers enterprise grade sound quality. The Polycom VVX 201 phone is a stylish, cost effective telephony solution, ideal for retail environments, call centers or shared/common areas, such as lobbies, hallways and break rooms or anywhere needing simple and reliable connectivity.

Unsurpassed voice quality and clarity

The VVX 201 features full duplex Type 1-compliant speakerphone with legendary Polycom® HD VoiceTM and Polycom® Acoustic FenceTM technology that delivers superior sound quality and enables noise- and echo-free conversations that are as natural as being there.

Simplicity and ease-of-use

The Polycom VVX 201 comes with a familiar, intuitive user interface with multi-language support that you can use without having to think about the "how to". The phone features a backlit LCD for improved readability.

Benefits

- Ideal for call centers, retail environments and for shared/ common-areas
- Make more efficient and productive calls with Polycom's HD Voice technology

Polycom® VVX® 201

USER INTERFACE FEATURES

- 2.5 in Graphical Backlit LCD (132 x 64) resolution
- Voicemail support
- Reversible deskstand/wallmount
- Unicode UTF-8 character support. Multilingual user interface including Chinese, Danish, Dutch, English (Canada/US/UK), French, German, Italian, Japanese, Korean, Norwegian, Polish, Portuguese, Russian, Slovenian, Spanish, and Swedish

FEATURE KEYS

- 4 context-sensitive "soft" keys
- 2 line keys with bi-color (red/green) LED
- "Home" feature key
- 4-way navigation key cluster with center "Select" key
- 2 volume control keys
- Dedicated hold key
- Dedicated headset key
- Dedicated hands-free speakerphone key
- Dedicated microphone mute key

AUDIO FEATURES

- Polycom HD Voice technology delivers life-like voice quality for each audio path-handset, the hands-free speakerphone, and the optional headset
- Polycom® Acoustic ClarityTM technology providing full-duplex conversations, acoustic echo cancellation and background noise suppression
- Type 1 compliant (IEEE 1329 full duplex)
- Frequency response – 150 Hz – 7 kHz for hands-free speakerphone, handset and optional headset mode
- Codecs: G.711 (μ-law), G.729AB
- Individual volume settings with visual feedback for each audio path
- Voice activity detection
- DTMF tone generation (RFC 2833 and in-band)
- Low-delay audio packet transmission
- Adaptive jitter buffers
- Packet loss concealment

HEADSET AND HANDSET COMPATIBILITY

- Dedicated RJ-9 headset port
- Hearing aid compatibility to ITU-T P.370 and TIA 504A standards
- Compliant with ADA Section 508 Subpart B 1194.23 (all)
- Hearing aid compatible (HAC) handset for magnetic coupling to hearing aids
- Compatible with commercially-available TTY adapter equipment

CALL HANDLING FEATURES[1]

- 2 SIP identities (registrations)
- 2 programmable line keys
- Shared call/bridged line appearance • Flexible line appearance (one or two line keys can be assigned for each registration)
- Distinctive incoming call treatment/call waiting
- Call timer and call waiting
- Call transfer, hold, divert (forward), pickup Called, calling, connected party information
- Local three-way audio conferencing • One-touch speed dial, redial
- Remote missed call notification
- Do not disturb function
- Electronic hook switch capable
- Local configurable digit map/dial plan

NETWORK

- SIP Protocol Support
- SDP
- IETF SIP (RFC 3261 and companion RFCs
- Two-port Ethernet switch
- 10/100Base-TX across LAN and PC Ports
- Manual or dynamic host configuration protocol (DHCP) network setup
- Time and date synchronization using SNTP
- QoS Support–IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS, and DHCP
- VLAN - CDP, DHCP VLAN discovery, LLDP-MED for VLAN discovery
- Network Address Translation (NAT) – support for static configuration and "Keep-Alive" SIP signaling
- RTCP and RTP support
- Hardware diagnostics
- Status and statistics reporting • TCP
- UDP
- DNS-SRV

SECURITY

- 802.1X Authentication and EAPOL
- Media encryption via SRTP
- 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

POWER

- Built-in auto sensing IEEE 802.3af Power over
- Ethernet (Class 2)
- External Universal AC Adapter (optional, 12V 6W DC)

APPROVALS

- FCC Part 15 (CFR 47) Class B • ICES-003 Class B
- EN55022 Class B
- CISPR22 Class B
- VCCI Class B
- EN55024
- EN61000-3-2; EN61000-3-3 • NZ Telepermit
- UAE TRA
- Australia RCM
- ROHS compliant
- ICASA
- CITC
- ANATEL[3]
- Customs Union[3]
- KCC[3]
- TAA[3]
- CCC[3]

SAFETY

- UL 60950-1
- CE Mark
- CAN/CSA C22.2 No 60950-1 • EN 60950-1
- IEC 60950-1
- AS/NZS 60950-1

OPERATING CONDITIONS

- Temperature: 0 to 40°C (+32 to 104°F)
- Relative Humidity: 5% to 95%, noncondensing

STORAGE TEMPERATURE

- -40 to +70°C (-40 to +160°F)

POLYCOM VVX 201 COMES WITH:

- VVX 201 console
- Handset with handset cord
- Network (LAN) cable - CAT-5E • Quick Start Guide

SIZE

- 6.5 x 6 x 7 in (17 x 15 x 18 cm)(W X H X D) Part Numbers
- 2200-40450-025 – VVX201 WW PoE 2200-40450-019 – VVX 201, Skype for Business, POE

UNIT WEIGHT

- 1.8 lbs (0.8 kg)

MASTER CARTON QUANTITY

- Ten (10)

COUNTRY OF ORIGIN

- China

WARRANTY

- One (1) year

[1] Most software-enabled features and capabilities must be supported by the server. Please contact your IP PBX/Softswitch vendor or service provider for a list of supported features.
[2] Available in future UC Software release
[3] Planned compliance and localization