The HDA50, a simple open SIP headset adapter, delivers the optimal solution for contact centers: desk phone call quality while using the soft client specifically for easy call management. The small desktop device reduces workspace clutter and provides clear, reliable audio. Combined with a Plantronics USB headset, customer service representatives can always count on hearing and being heard. The HDA50 just makes it simple: easy-to-use device for CSRs plus remote provisioning, troubleshooting and device management for IT.

**HDA50**

- Simple open SIP device
- Optimal solution: soft client for call management, HDA50 for voice
- Simple, remote provisioning and management
- Small desktop footprint for less workspace clutter

**BENEFITS**

- Delivers desk phone call quality for assurance of maintaining critical conversations while using 3rd party soft client for call management
- Optimized for Plantronics headsets, compatible with most USB headsets.
- Troubleshoot devices remotely with device logs, packet capture and QoS data to immediately understand call performance
- Set-and-forget automated software updates means reduced support costs and consistent workflow experiences

MAKE THE CALL TO BOOST CLARITY AND RELIABILITY.
POLY HDA50

SPECIFICATIONS

PRODUCT REQUIREMENTS
• Access to internet via a switched ethernet port
• Active phone service subscription with all required SIP credentials to make and receive calls
• USB Type-A headset supporting HID protocol

POLY HDA50 SHIPS WITH
• HDA50 VoIP endpoint
• Power adapter
• 1 x RJ-45 ethernet cable (80 inches/203 centimeters)
• Quick start/installation guide
• Velcro tape

WARRANTY
• 1 year

PRODUCT DIMENSIONS (L X W X H)
• 6.9 cm x 6.9 cm x 3.0 cm (2.7 in x 2.7 in x 1.2 in)

UNIT WEIGHT
• 198 grams/7 ounces
• Shipping weight: 340 grams/12 ounces (including power supply, ethernet cable and packaging)

INTERFACE FEATURES
• Internet (WAN): 2 x 10/100BaseT ethernet port (802.3)
• USB Type-A: USB 2.0 (headset port)
• Reset button: located on bottom of case
• LED indications: power on, status, ethernet activity (WAN), upgrade in progress status, packet RX/TX and USB headset status
• Phone: 2 x RJ-11 FXS analog phone port (optional)

TELEPHONY FEATURES
• Call routing rules
• SIP service configurable inbound call routing rules
• In-band DTMF (G.711)
• Out-of-voice band DTMF (RFC 2833)
• Out-of-voice band DTMF (SIP INFO method)
• Call progress tone generation
• Tone profile per SIP Service Provider and Polycom Device Management Service for Service Providers
• Ring profile per SIP Service Provider and Polycom Device Management Service for Service Providers
• Full duplex audio
• G.165, 168 echo cancelation
• VAD—voice activity detection
• Silence suppression
• Comfort noise generation
• Three-way conference calling with local mixing
• Daylight savings time support—worldwide
• Call waiting
• Maximum session control

DATA NETWORKING
• Ethernet Authentication (IEEE 802.1X)
• MAC address (IEEE 802.3)
• UDP (RFC 768) in SSL/TLS
• TCP (RFC 793) in SSL/TLS
• IP version 4, IPv4 (RFC 791)—static IP and DHCP support
• ICMP (RFC 792)
• ARP—address resolution protocol
• Domain name system (DNS) A records (RFC 1706) and SRV records (RFC 2782)
• RTP (RFC 1889, 1890), RFC 5966
• RTP/RTCP (RFC 1889), DHCP client (RFC 2131)
• DiffServ (RFC 2475)—independently configured: service, SIP and media
• ToS (RFC 791, 1349)—independently configured: service, SIP and media
• VLAN Tagging (802.1p)—independently configured: service, SIP and media
• SNTP (RFC 2030)—primary and secondary NTP servers
• LLDP-MED
• Traverse through web proxy server access via HTTP, TFTP—HTTPS

SECURITY
• Remote access interface: user name and password access via HTTP, TFTP—HTTPS
• Device web page standard: HTTP v1.1, XML v1.0
• Secure remote provisioning: HTTP, HTTPS

VOIP FEATURES
• Simple Certificate Enrollment Protocol (SCEP)
• Four (4) service provider configuration profile assignments (ITSP 1–4)
• Four (4) service/trunk subscription profile assignments (SP 1–4)
• SIPv2 (RFC 3261, 3262, 3263, 3264)
• SIP over UDP
• SIP over TCP
• SIP over TLS
• 4 SIP service provider service sessions—concurrent operation
• 1 Polycom Device Management Service for Service Providers session
• SIP proxy redundancy—local or DNS based SVR, primary and secondary fallback list restrict source IP address
• Maximum number of sessions—independent per service
• 4 trunk groups
• Voice gateway—direct dialing
• G.711 A-law (64 kbps)
• G.711 μ-law (64 kbps)
• G.726 (32 kbps) G.729a (8 kbps) iLBC (13.3, 15.2 kbps) codec pre-selection code
• Wideband codec (USB headset): G.722 (64 kbps), Opus
• Voice processing per SIP service—TX/RX audio gain, echo cancellation
• Adjustable audio frames per packet
• Codec name assignment
• Codec profile per SIP SP and Service Provider and Polycom Device Management Service for Service Providers
• Dynamic audio payload
• Packet loss concealment
POLY HDA50

SPECIFICATIONS

POWER
• Universal switching with fixed US, EU, UK style plug prongs (model dependent)
• AC input: 100 to 240 volts 0.3A 50–60Hz (26–34 VA)
• DC input: +12V 1.0 amp max

CERTIFICATIONS
• FCC part 15 class B
• A-tick
• CE
• ICES-003
• RoHS
• WEEE
• UL/cUL—power adapter

ENVIRONMENTAL CONDITIONS
• Operating temperature: 0º to 45º C (32º to 113º F)
• Relative humidity: 10% to 90% non-condensing
• Storage temperature: –25º to 85º C (-13º to 185º F)

MANAGEMENT—CONFIGURATION
• Remote provisioning: XML via TFTP or HTTP, (TR069/TR104 parameter naming syntax)
• Secure remote provisioning: HTTPS, encrypted XML via HTTP or TFTP—dedicated user name and password
• Secure remote firmware update: encrypted binary file via TFTP or HTTP + dedicated user name and password
• Customization: OBi-ZT: Obihai zero-touch automatic customization and configuration **
• Call history (CDRs): call detail records on Service Provider and Polycom Device Management Service for Service Providers web page, export to XML
• LED Indications: power, device status, upgrade progress status, ethernet activity, USB headset status
• Session information: SIP session status, Polycom Device Management Service for Service Providers status
• Primary SIP service set-up wizard: dedicated device web page for quick ITSP account set-up
• System settings back-up/restore: save and restore configuration via XML file to/from a local folder

RTP STATISTICS
• RTP transport type
• Audio codec type (Tx/Rx)
• RTP packetization—in multiples of 10ms (Tx/Rx)
• RTP packet count (Tx/Rx)
• RTP byte count (Tx/Rx)
• Packets out-of-order
• Packets interpolated
• Packets late (dropped)
• Packets lost
• Packet loss rate %
• Packet drop rate %

CALL PROGRESS
• Configurable call progress tone
• Call progress tone profiles (2)
• Dial tone
• Busy tone
• Ringback tone
• Reorder tone
• Confirmation tone
• Holding tone
• Second dial tone
• Stutter tone
• Howling tone
• Prompt tone
• Call forwarded tone
• Conference tone
• SIT tones (1–4)
• Ringing and call waiting tone configuration
• Ring patterns (10)—configurable
• Call waiting tone patterns (10)—configurable
• Call waiting tone pattern profiles (2)

LEARN MORE
Visit poly.com/hda50 to learn more about the Poly HDA50

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