



DATA SHEET

Polycom® OBi302

2 line VoIP Adapter connects home and business analog phones to the digital voice communications world

Polycom OBi302 expands your service portfolio by enhancing the communication possibilities of home offices with flexibility in voice, fax and modem applications as they transition to the digital communications world. Home and business offices can maximize their current analog investment by keeping up to two analog phones or fax machines subscribed to up to four independently configurable VoIP services, one OBiTALK peer-to-peer voice service, and one POTS (Plain Old Telephone Service) subscription*. This dedicated device prioritizes phone calls above other internet traffic coming through the LAN port to ensure users clearly hear every important word. Alternatively, with support of the T.38 fax standard, home and business office users can send and/or receive reliable facsimile calls over the Internet.

Provide your customers new wireless capabilities with WiFi and Bluetooth

Polycom OBi302 is never tied to any inconvenient Ethernet port location. It can be placed anywhere within range of a 2.4 GHz or 5 GHz, 802.11ac wireless access point when used with the Polycom USB WiFi adapter. Similarly, the USB Bluetooth adapter pairs a user's mobile phone or Bluetooth audio headset so they can place and answer calls using their mobile service via analog phones.

Simplify deployment and on-going support with OBiTALK

Polycom's OBiTALK device management platform will help you save time and hassle by being able to access the device without the need to get behind the customer's on-site network. With support for the OBi302, Polycom OBiTALK lets you provision the device to meet your customer specifications or update them from any place with a connection to the Internet. Provide best in class support to your customers by quickly identifying issues and remotely troubleshooting devices.

The Polycom OBiTALK is a reliable, secure, cloud-based, out-of-band management interface designed for massive scale zero-touch device bootstrapping, configuration customization, and provisioning. Polycom OBiTALK also provides for in-service device management and troubleshooting, with functionality including device logging, packet capture, call QoS and statistics reporting.



Key benefits

- Expand your service portfolio by offering flexible voice and fax applications to home and business offices
- Easily configure and roll-out services and provide on-going fast support with the Polycom OBiTALK cloud-based management platform
- The only VoIP adapter that supports WiFi capabilities through optional adapter for voice and voice-band data applications
- Connect up to a combination of 2 analog phones or fax machines to transition voice communications to the digital world

*Requires Polycom OBILINE USB FXO adapter

Product requirements

- Active Internet Connection
- Analog Touch Tone Phone
- Access to Internet Via a Switched Ethernet Port on Home or Office Router
- (Optional) Active Internet Phone Service Subscription with All Required SIP Credentials to Make & Receive Calls

Polycom OBi302 ships with

- OBi302 Voice Service Bridge and Telephone Adapter
- Power Adapter
- 1 x RJ-45 Ethernet Cable (80 inches/203 centimeters)
- Quick Start/Installation Guide

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)
- Phone: 2 x RJ-11 FXS Analog Phone Port
- USB: USB 2.0
- Reset Button: Located on Bottom of Case
- LEDs: Power/Status, Ethernet Activity (WAN), Phone Port 1, Phone Port 2
- LED indications: Power On, Status, upgrade in Progress Status, Packet RX/TX, Phone Port 1 and 2 Status

Telephony features

- Call Routing Rules
- Automated Attendant with Configurable Answer Delay
- PIN Access Control to AA (Up to 4 PINs)
- Recursive Digit Map for Call Routing (AA, Phone, Voice Gateways, Trunk Groups)
- AA Configurable Outbound Call Routing Rules
- SIP Service Configurable Inbound Call Routing Rules
- Fax Pass Through (G.711)
- T.38 Fax Relay for Fax over IP
- Modem Pass Through (G.711)
- 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 SP and OBiTALK Service
- Ring Profile per SIP SP and OBiTALK Service
- Star Code Profile per SIP SP and OBiTALK Service
- Full Duplex Audio
- G.165, 168 Echo Cancellation
- VAD—Voice Activity Detection
- Silence Suppression
- Comfort Noise Generation
- Three Way Conference Calling with Local Mixing
- Hook Flash Event Signaling
- Flash Hook Timer
- Caller ID—Name and Number per Bellcore, ETSI, DTMF, and NTT
- MWI—Message Waiting Indicator
- Visual Message Waiting Indication (VMWI)
- Daylight Savings Time Support—Worldwide
- Caller ID Enable/Disable
- Caller ID Number
- Caller ID Name (Alphanumeric)
- Caller ID Spoofing
- Call Waiting
- Maximum Session Control
- Call Forward—Unconditional
- Call Forward on Busy
- Call Forward on No Answer (Ring Count Configurable)
- Call Transfer Enable/Disable
- Anonymous Call Block
- Anonymous Call
- Do Not Disturb
- Call Return
- Repeat Dialing

Data networking

- 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

Security

- Local Access Interface: IVR Password
- 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

- 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 OBiTALK Service 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
- 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 OBiTALK Service
- Dynamic Audio Payload
- Packet Loss Concealment
- Jitter Buffer (Adaptive)
- STUN
- ICE
- SUBSCRIBE/NOTIFY Framework (RFC 3265)
- NOTIFY Dialog, Line Status
- SUBSCRIBE Message Summary

- VoIP NAT Interworking
- DATE Header Support
- Remote-Party-ID (RPID)
- P-Asserted-Identity (PAID)
- RTP Statistics in BYE Message and SIP PUBLISH with MOS Score

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: 0° to 45° C (32° to 113° F)
- Relative humidity: 10% to 90% Non-condensing
- Storage temperature: –25° to 85° C (–13° to 185° F)

Management—Configuration

- Local Access Interface: IVR, Web Page—Password Protected (Admin and User Level Log-in)
- Remote Access Interface: Syslog (Multi-Level Granularity), Invokable via SIP Notify, Web, Provisioning
- Device Web Page Standard: HTTP v1.1, XML v1.0
- 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 OBi Web Page, Export to XML
- LED Indications: Power, Device Status, Upgrade Progress Status, Ethernet Activity, PHONE Status

- Session Information: SIP Session Status, OBiTALK Status, Phone Port 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 %
- Jitter Buffer Length—ms
- Received Interarrival Jitter—ms
- Jitter Buffer Underruns
- Jitter Buffer Overruns

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)

Star Code Configuration

- Configurable Star Codes
- Star Code Profiles (2)
- Redial

- Call Return
- Activate Block Caller ID
- Deactivate Block Caller ID
- Block Caller ID Once
- Unblock Caller ID Once
- Activate Call Forwarding (All Calls)
- Deactivate Call Forwarding (All Calls)
- Activate Call Forward on Busy
- Deactivate Call Forward on Busy
- Activate Call Forward on No Answer
- Deactivate Call Forward on No Answer
- Activate Block Anonymous Calls
- Deactivate Block Anonymous Calls
- Activate Call Waiting
- Deactivate Call Waiting
- Activate Do Not Disturb
- Deactivate Do Not Disturb
- Activate Repeat Dial
- Deactivate Repeat Dial

FXS SLIC (Subscriber Line Integrated Circuit): Phone Port

- Ringer Specifications: Ring Frequency: 14Hz–68Hz, Ring Waveform: Trapezoidal, Sinusoidal, Ring Voltage: 55v–85v
- Maximum Ring Load: 5 REN (Ringer Equivalence Number) per Phone Port

FXS (PHONE Port) Configuration Settings

- Recursive Digit Map and Associated Outbound Call Routing
- On-Hook Tip Ring Voltage: 30v–52v
- Off-Hook Current Max: 15mA–45mA
- Impedance: 12 Independent Settings
- DTMF Playback Level: –90 dBm–3 dBm
- Caller ID Method: Bellcore, ETSI (FSK or DTMF)
- Caller ID Trigger (Before/After First Ring, Polarity Reversal)
- Channel Tx Gain: –12dB to 6 dB at 1 dB Resolution
- Channel Rx Gain: 12dB to 6 dB at 1 dB Resolution
- Silence Detect Sensitivity
- Hook Flash Time Max
- Hook Flash Time Min
- CPC Delay Time
- CPC Duration
- Idle Polarity
- Connect Polarity

Learn more

Visit www.polycom.com/voip to learn more about our VoIP products.

About Polycom

Polycom helps organizations unleash the power of human collaboration. More than 400,000 companies and institutions worldwide defy distance with video, voice and content solutions from Polycom. Polycom and its global partner ecosystem provide flexible collaboration solutions for any environment that deliver the best user experience and unmatched investment protection.

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