



DATA SHEET

Polycom OBi312

VoIP adapter that connects your analog phone to telco and VoIP services from a Polycom partner

Polycom OBi312 expands your voice capabilities and keeps you connected at all times by bridging your traditional telephone service and VoIP services. To get started, all you need is a phone, power, an internet connection and traditional phone service. Your voice calls are important so this dedicated device prioritizes your phone calls above other internet traffic to ensure you clearly hear every important word on up to two simultaneous calls.

Alternatively, you can send and receive reliable facsimile calls over the Internet with support of the T.38 fax standard and talk on the phone at the same time.

Simplify deployment & on-going support with PDMS-SP

Through Polycom's Device Management Services (PDMS-SP) platform, you can save time and hassle by configuring the device without laying a finger on it. Taking in form of a web portal and API, this secure service lets you provision the device to meet your customer specifications or update them from anyplace with a connection to the Internet. Provide best in class support to your customers by quickly identifying issues and remotely troubleshooting devices.

Provide your customers new capabilities with WiFi and Bluetooth

With OBi312, users are not tied to inconvenient phone jack locations. It can be placed anywhere within range of an internet router when used with the USB WiFi accessory.



Key Benefits

Expand your service portfolio by offering flexible voice and fax applications to home offices

Easily configure devices and provide better on-going support through the secure Polycom Device Management Services (PDMS-SP) portal

Put your phone anywhere in your house with the only VoIP adapter that supports optional WiFi accessory

Connect an analog phones or fax machines to transition your voice communications to the digital world and bridge with the telco world

Keep your traditional phone service and bridge VoIP services with up to one phone or fax machine



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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

Interface features

- Internet (WAN): 1 x 10/100BaseT Ethernet Port (802.3)
- Phone: 1 x RJ-11 FXS Analog Phone Port, 1 x RJ-11 FXO Analog Telco Port
- USB: USB 2.0
- Reset Button: Located on Bottom of Case
- LEDs: Power/Status, Ethernet Activity (WAN), Phone
- LED indications: Power On, Status, upgrade in Progress Status, Packet RX/TX, Phone Port Status

Telephony features

- Configurable Contact List (Inbound Call Routing)
- Automatic Attendant (English) 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 Rule
- SIP Service Configurable Inbound Call Routing Rule (4)
- Fax Pass Through (G.711)
- T.38 Fax Relay for Real-Time 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 (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 & Number per Bellcore, ETSI and DTMF
- MWI – Message Waiting Indicator
- Daylight Savings Time Support – North & South Hemispheres
- 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)
- TCP (RFC 793)
- 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) & SRV Records (RFC 2782) RTP (RFC 1889, 1890)
- RTCP (RFC 1889), DHCP Client (RFC 2131)
- DiffServ (RFC 2475) – Independently Configured: Service, SIP & Media
- ToS (RFC 791, 1349) – Independently Configured: Service, SIP & Media
- VLAN Tagging (802.1p) – Independently Configured: Service, SIP & Media
- SNTP (RFC 2030) – Primary & Secondary NTP Servers

Security

- Local Access Interface: IVR Password
- Remote Access Interface: User Name & Password Access via HTTP, TFTP – HTTPS
- Device Web Page Standard: HTTP v1.1, XMLv1.0
- Secure Remote Provisioning: TFTP, 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 TCP with TLS (v1.2)
- 4 SIP Service Provider Service Sessions – Concurrent Operation
- 1 OBiTALK Service Session
- SIP Proxy Redundancy – Local or DNS Based SVR, Primary & Secondary Fallback List Restrict Source IP Address
- Maximum Number of Sessions – Independent per Service
- Trunk Groups (4)
- Voice Gateway – Direct Dialing (8)
- 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 (4) & 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

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: +12V 1.0 Amp Max

Certifications

- FCC Part 15 Class B
- A-Tick
- WEEE
- UL/cUL - Power Adapter

Management – Configuration

- Local Access Interface: IVR, Web Page – Password Protected (Admin & 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: SSL via HTTPS, Encrypted XML via HTTP or TFTP – Dedicated User Name & Password
- Secure Remote Firmware Update: Encrypted Binary File via TFTP or HTTP + Dedicated User Name & Password
- Customization: PDMS-SP Zero-Touch Automatic Customization & 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 & Restore Configuration via XML file to / from a Local Folder

- CE
- ICES-003
- RoHS

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)

RTP Statistics

- RTP Transport Type
- Audio Codec Type (Tx/Rx)
- RTP Packetization - ms (Tx/Rx)
- RTP Packet Count (Tx/Rx)
- RTP Byte Count (Tx/Rx)
- Peer Clock Differential Rate - PPM
- Packets In Jitter Buffer
- 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
- DTMF Digits Received
- Jitter Buffer Underruns
- Jitter Buffer Overruns
- Sequence Number Discontinuities
- Skew Compensation – ms

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 & Call Waiting Tone Configuration

- Ring Patterns (10) - Configurable
- Call Waiting Tone Patterns (10) - Configurable
- Call Waiting Tone Pattern Profiles (2)

Star Code Configuration

- Configurable Start 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)

FXS (PHONE Port) Configuration Settings:

- Recursive Digit Map & 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 – 3dBm
- 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

Polycom OBi312 ships with:

- OBi312 Voice Service Bridge and Telephone Adapter
- Power Adapter
- 1 x RJ-45 Ethernet Cable (80 inches / 203 centimeters)
- Quick Start / Installation Guide
- 1 x RJ-11 Cable

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, Telco Cables, and Packaging)

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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