OBi200 VoIP Telephone Adapter with 1-Phone Port & USB
With Support for Four (4) SIP and OBiTALK VoIP Services

With the OBi200, you are in control of your digital & analog communications life. Via the OBi200’s on-board telephone connection as well as via the Internet to other OBi endpoints via Obihai’s free OBiTALK network or up to four (4) available VoIP services, you have the power make and receive phone calls and faxes as well as bridge mobile, fixed line and Internet telephone services. The OBi200 supports the T.38 fax standard for reliable facsimile calls over the Internet.

The OBi200 USB port serves multiple purposes. Using the OBiWiFi Wireless Adapter, the OBi200 can be placed anywhere within range of an 802.11b/g/n access point. Or, the USB port can be connected to a storage device to enable local and remote access of stored files over the Internet by authorized users.

The OBi200 is a dedicated device, built with a high-performance system-on-a-chip platform to ensure high quality voice conversations. The OBi200 has high availability and reliability because it is always-on to make or receive a call.

With the OBi200, a computer is not required and a computer does not need to be on to talk to people. To get started, all you need is a phone, power and a connection to the Internet.

The OBi200 is Complemented by Other OBi Products & Services
OBiTALK: A web portal for device management and service configuration. OBiTALK also allows its members to add people and associated OBi endpoints to “circles of trust” such that additional functionality can be shared amongst authorized users. The OBITALK portal is also where members can download the OBiON applications for smart phones and Internet connected devices like the iPhone, iPad, iPod touch & Android.

OBiON iPhone, iPad, iPod touch & Android Devices: An application for iPhone, iPad, iPod touch and Android devices which makes possible placing and receiving calls to/from other OBi endpoints.

OBiON PC: A middleware application for a PC that facilitates placing and receiving calls to/from other OBi endpoints.

Key Features of the OBi200 VoIP Telephone Adapter:
SIP Service Provider Support for Up to Four (4) SIP Accounts
Any Available Service Can be Accessed from the Phone Port
Aggregation / Bridging of Four (4) SIP and One (1) OBiTALK Service
Automatic Attendant for Simplified Call Routing (AA)
Call Back Service – Automatic Call Back to Connect User to the AA to Make a New Call or Ring the Attached Phone
OBiTALK Web Portal Integration

- Configuration and Management of OBi Endpoints
- Download OBi Client Applications for Smart Phones, Internet Devices & PCs
- Creating & Joining Circles of Trust So You Can Share Your OBi
- Setting Up Your OBi Endpoint Speed Dial Directory

Configurable to Work with Any SIP Compliant Internet Telephone Service

Analog Phone Impedance Agnostic

Robust Telephony Features:

- Caller ID – Name & Number
- Call Waiting
- Message Waiting Indication - Visual and Tone Based
- Speed Dialing of 99 OBi Endpoints or Numbers
- Three Way Conference Calling with Local Mixing
- Hook Flash Event Signaling
- Call Forward - Unconditional
- Call Forward on Busy
- Call Forward on No Answer
- Call Transfer
- Anonymous Call
- Block Anonymous Call
- Do Not Disturb
- Call Return
- Repeat Dialing
- Caller ID Pass-Thru

Powerful Call Routing & Voice Service Features:

- SIP Support for Voice and Fax Over IP (T.38 and G.711 pass-thru) from Internet Telephony Service Providers
- OBiTALK Managed VoIP Network for OBi Endpoint Devices & Applications
- High Quality Voice Encoding Using G.711, G.726, G.729, iLBC Algorithms
- Recursive Digit Maps & Associated Call Routing (Outbound, Inbound)

<table>
<thead>
<tr>
<th>General</th>
<th>Obihai</th>
</tr>
</thead>
<tbody>
<tr>
<td>Brand</td>
<td>Obihai</td>
</tr>
<tr>
<td>Manufacturer</td>
<td>Obihai</td>
</tr>
<tr>
<td>Hardware Designer</td>
<td>Obihai Technology, Inc.</td>
</tr>
<tr>
<td>Model Name</td>
<td>OBI200</td>
</tr>
<tr>
<td>Release Date</td>
<td>May 2013</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Microprocessor</th>
<th>Width of Machine Word</th>
<th>32 bit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Instruction Set</td>
<td>ARM</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>FXS SLIC (Subscriber Line Integrated Circuit): Phone Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ringer Specifications</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Maximum Ring Load</td>
</tr>
<tr>
<td>FXS (PHONE Port)</td>
</tr>
<tr>
<td>Configuration Settings</td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>
**FXS (PHONE Port) Configuration Settings cont.**

- 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

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**Management – Configuration**

<table>
<thead>
<tr>
<th>Local Access Interface</th>
<th>IVR, Web Page – Password Protected (Admin &amp; User Level Log-in)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote Access Interface</td>
<td>Syslog (Multi-Level Granularity), Invokable via SIP Notify, Web, Provisioning</td>
</tr>
<tr>
<td>Device Web Page Standard</td>
<td>HTTP v1.1, XML v1.0</td>
</tr>
<tr>
<td>Remote Provisioning</td>
<td>XML via TFTP or HTTP, (TR069 / TR104 Parameter Naming Syntax)</td>
</tr>
<tr>
<td>Secure Remote Provisioning</td>
<td>SSL via HTTPS, Encrypted XML via HTTP or TFTP – Dedicated User Name &amp; Password</td>
</tr>
<tr>
<td>Secure Remote Firmware Update</td>
<td>Encrypted Binary File via TFTP or HTTP + Dedicated User Name &amp; Password</td>
</tr>
<tr>
<td>Customization</td>
<td>OBi-ZT: Obihai Zero-Touch Automatic Customization &amp; Configuration **</td>
</tr>
<tr>
<td>Call History (CDRs)</td>
<td>Call Detail Records on OBi Web Page, Export to XML</td>
</tr>
<tr>
<td>LED Indications</td>
<td>Power, Device Status, Upgrade Progress Status, Ethernet Activity, PHONE Status</td>
</tr>
</tbody>
</table>

**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

**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

**Security**

<table>
<thead>
<tr>
<th>Local Access Interface</th>
<th>IVR Password</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote Access Interface</td>
<td>User Name &amp; Password Access via HTTP, TFTP – HTTPS</td>
</tr>
<tr>
<td>Device Web Page Standard</td>
<td>HTTP v1.1, XMLv1.0</td>
</tr>
<tr>
<td>Secure Remote Provisioning</td>
<td>TFTP, HTTP, HTTPS</td>
</tr>
</tbody>
</table>
## Network – Application Details

### Data Networking
- MAC Address (IEEE 802.3)
- UDP (RFC 768)
- TCP (RFC 793)
- IP version 4 (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

### VoIP
- 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
- 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
- Media Loopback Support

### Telephony
- 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)
- Direct / Single-Stage Dialing (Route to Voice Gateway)
<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Fax Pass Through (G.711)</td>
<td>T.38 Fax Relay for Real-Time Fax over IP</td>
</tr>
<tr>
<td>Modem Pass Through (G.711)</td>
<td>In-Band DTMF (G.711)</td>
</tr>
<tr>
<td>Out of Voice Band DTMF (RFC 2833)</td>
<td>Out of Voice Band DTMF (INFO Method)</td>
</tr>
<tr>
<td>Call Progress Tone Generation</td>
<td>Tone Profile per SIP SP and OBiTALK service</td>
</tr>
<tr>
<td>Ring Profile per SIP SP and OBiTALK service</td>
<td>Star Code Profile per SIP SP and OBiTALK service</td>
</tr>
<tr>
<td>Full Duplex Audio</td>
<td>G.165, 168 Echo Cancelation</td>
</tr>
<tr>
<td>VAD – Voice Activity Detection</td>
<td>Silence Suppression</td>
</tr>
<tr>
<td>Comfort Noise Generation</td>
<td>Three Way Conference Calling with Local Mixing</td>
</tr>
<tr>
<td>Hook Flash Event Signaling</td>
<td>Flash Hook Timer</td>
</tr>
<tr>
<td>Caller ID – Name &amp; Number per Bellcore, ETSI and DTMF</td>
<td>MWI – Message Waiting Indicator</td>
</tr>
<tr>
<td>Visual Message Waiting Indication (VMWI)</td>
<td>Daylight Savings Time Support – North &amp; South Hemispheres</td>
</tr>
<tr>
<td>Caller ID Enable /Disable</td>
<td>Caller ID Enable /Disable</td>
</tr>
<tr>
<td>Caller ID Number</td>
<td>Caller ID Number</td>
</tr>
<tr>
<td>Caller ID Name (Alphanumeric)</td>
<td>Caller ID Name (Alphanumeric)</td>
</tr>
<tr>
<td>Caller ID Spoofing</td>
<td>Call Waiting</td>
</tr>
<tr>
<td>Maximum Session Control</td>
<td>Call Forward - Unconditional</td>
</tr>
<tr>
<td>Call Forward on Busy</td>
<td>Call Forward on No Answer (Ring Count Configurable)</td>
</tr>
<tr>
<td>Call Transfer Enable / Disable</td>
<td>Anonymous Call Block</td>
</tr>
<tr>
<td>Anonymous Call</td>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>Call Return</td>
<td>Repeat Dialing</td>
</tr>
<tr>
<td>Configurable Call Progress Tone</td>
<td>Call Progress Tone Profiles (2)</td>
</tr>
<tr>
<td>Dial Tone</td>
<td>Busy Tone</td>
</tr>
<tr>
<td>Ringback Tone</td>
<td>Reorder Tone</td>
</tr>
<tr>
<td>Confirmation Tone</td>
<td>Confirmation Tone</td>
</tr>
<tr>
<td>Holding Tone</td>
<td>Second Dial Tone</td>
</tr>
<tr>
<td>Stutter Tone</td>
<td>Howling Tone</td>
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<tr>
<td>Prompt Tone</td>
<td>Call Forwarded Tone</td>
</tr>
<tr>
<td>Conference Tone</td>
<td>SIT Tones (1-4)</td>
</tr>
<tr>
<td>Ring Patterns (10) - Configurable</td>
<td>Ringing &amp; Call Waiting Tone Configuration</td>
</tr>
<tr>
<td>Call Waiting Tone Patterns (10) - Configurable</td>
<td>Call Waiting Tone Pattern Profiles (2)</td>
</tr>
<tr>
<td>Star Code Configuration</td>
<td>Configurable Start Codes</td>
</tr>
<tr>
<td>Star Code Profiles (2)</td>
<td>Redial</td>
</tr>
<tr>
<td>Call Return</td>
<td>Activate Block Caller ID</td>
</tr>
</tbody>
</table>
| Star Code Configuration cont. | 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 |

| Interfaces & Indicator Lights | Internet (WAN) 1 x 10/100BaseT Ethernet Port (802.3)  
Phone (FXS) 1 x RJ-11 FXS Analog Phone Port  
USB USB 2.0  
Reset Button Yes – Located on Bottom of Case  
LEDs 3 – Power/Status, Ethernet Activity (WAN), Phone  
LED Indications Power On, Status, Upgrade in Progress Status, Packet RX/TX, Phone Port Status |

| Certifications | FCC Part 15 Yes – Class B  
A-Tick Yes  
CE Yes  
ICES-003 Yes  
RoHS Yes  
WEEE Yes  
UL/cUL Yes – Power Adapter |

| Environmental | Operating Temperature 0º to 45º C (32º to 113º F)  
Storage Temperature -25º to 85º C (-13º to 185º F)  
Operating Humidity 10% to 90% Non-condensing  
Non-operating Humidity 10% to 90% Non-condensing |

| Physical Attributes | Dimensions: 6.9 cm x 6.9 cm x 3.0 cm  
(width x depth x height) 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)  
Mounting Desktop Mountable |

| Power Supply | Type Universal Switching with Fixed US, EU, UK Style Plug Prongs (Model Dependent)  
Input Power AC Input: 100 to 240 Volts 0.3A 50-60Hz (26-34 VA)  
Output Power DC: +12V 1.0 Amp Max |
### Carton Specifications

<table>
<thead>
<tr>
<th>Units Per Carton</th>
<th>20 Units</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carton Dimensions</td>
<td>43.2 x 25.4 x 21.6 centimetres – 17 x 10 x 8.5 inches</td>
</tr>
<tr>
<td>Carton Weight</td>
<td>6.4 Kilograms / 14 pounds</td>
</tr>
<tr>
<td>Cartons Per Std. 20 / 40 ft Container</td>
<td>1,170 / 2,430 Cartons – Non-palletized</td>
</tr>
</tbody>
</table>

### Miscellaneous

| Requirements | Active Internet Connection  
<table>
<thead>
<tr>
<th></th>
<th></th>
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</thead>
<tbody>
<tr>
<td></td>
<td>Analog Touch Tone Phone</td>
</tr>
</tbody>
</table>
|              | 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 |
| Documentation | Quick Start / Installation Guide  
|              | User / Administrative Guide  
|              | Implementation Guide for Service Providers ** |
| Package Contents | OBi200 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 Hardware (Limited) |
| Engineering & Design Location | California, USA |
| HST Code      | 8517.62.00 |
| Data Sheet State | All content subject to change.  
|              | This data sheet is not a warranty. |
| Data Sheet Version | 131217.200.2 |

** For Service Providers Only