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OpenVox UC200 Series IPPBX

Overview

OpenVox UC200 Series IPPBX is a new generation Unified Communication terminal equipment designed for SME and SOHO users. It is compact and lightweight, and provides FXS/FXO interface. It integrates functions such as IP telephony, voice and recording. It is compatible with multiple service platforms and terminals and can seamlessly connect to VoIP networks, traditional telephone networks (PSTN) and provides diverse converged communications solutions. UC200 IPPBX also features fast installation, easy deployment, and high reliability, which brings a new experience of mobile office and communication to the enterprise.

Highlight features

- Convergence of telephone calls, recordings, messages
- Corporate headquarters and branch voice networking
- Compatible with IMS and Asterisk service platforms
- HD Voice & Video Call Support
- SIP Forking: Multiple extension registration with the same sip account
- Send/Receive Calls on Web (WebPhone integration)
- Hotel interface (minibar, alarm, room cleaning, extension privilege setting)
- Fxs Interface High Voltage Spotlight
- Open Application Programming Interface (MQTT API)
- High Avaibility (Hot Standby)
- Flexible Call control: Call failure rerouting, routing based on user privilege level, routing based on time period, routing based on caller and callee number.

Features

Physical Specification

- FXS: Up to 8, flexible selection
- FXO: Up to 8, flexible selection
- Storage: 1 * TF Slot and 1 * USB
- Network Interface: 3 *10/100 Base-T RJ45

FXS

- Interface Type: RJ11
- Caller ID Signaling: BELL,V23, V23_JP, DTMF
- Hang Up Detection: Off-hook, On-hook, Busy Tone
- Polarity Reverse
- Hooking Detection
- Fxs Interface High Voltage Spotlight

FXO

- Interface Type: RJ11
- Caller ID Detection: FSK, DTMF
- Reversed-Polarity Detection
- Delayed Response Off-hook
- Busy Tone Detection
- No Current Hang-up Detection

Voice Feature

- VoIP Protocols: SIP over UDP/TCP/TLS, SDP, RTP/SRTP
- Supported Codecs: G.711a/μ law, G.729A, GSM, G.726, G.722, iLBC, OPUS, VP8, H264
- VPN: N2N and OpenVPN
- Silence Suppression
- Comfort Noise Generator (CNG)
- Voice Activity Detection (VAD)
- Echo canceller(G.168), Maximum 128ms
- Adaptive Dynamic Buffering
- Adjustable Gain Control/Automatic Gain Control
- AutoCLIP Routing
- Auto Announcement with outgoing call
- MQTT API
- OPUS/VP8 HD Voice/Video Call
- TOS/QOS support for SIP and RTP
- Call Proceeding Tone: Dial Tone, Ring-back Tone, Busy Tone
- Support NAT Traversal
- Supports HD broadcasting (automatic broadcasting, timed broadcasting)
- DTMF Mode: RFC2833/Signal/Inband
- Customized Signal Tones
- Intra-group Pickup
- Hotline
- Do Not Disturb (DND)
- Tripartite Meeting

Call Features

- Ring Group/Routes Group
- Calling/Called Number Transform
- Call Duration Limitation/Call Failure Rerouting
- Caller ID Number Acquisition/DID Acquisition
- Remote Party ID/Remote Management
- P-Asserted-Identity/P-Preferred-Identity
- Routing based on user privilege level/time condition/ caller id number
- Time Condition
- Based on Destination Routing/Source Routing
- Dial Plan
- Failover Routing
- FXO Impedance Matching
- Customizable Multi-language IVR
- Auto Attendant Function
- Local CDR Storage
- SIP forking (multiple SIP device registration with same sip account) / Customized SIP Fields
- WebPhone (WebRTC)

Management & Maintenance

- Simple and convenient configuration via Web GUI
- CLI Management Config
- Support configuration files backup and upload
- Support Chinese and English page
- Firmware Update by HTTP
- Modify Password via Web & Telnet
- CDR Query & Export
- Syslog Query & Export
- Ping and Tracer Test
- Traffic Statistics: TCP, UDP, RTP
- Network Capture/Network Quality Test
- Automatic Time synchronization

Additional Service

- T38 fax
- Call Forwarding (Unconditional/No Reply/Busy/Not Reachable)
- Call Waiting/Holding/Transfer/Queue/Spy
- Permission Control/Broadcast control
- Secretary Extension
- High Availability (Hot Standby)
- Phone Book/Announcement/Morning Call (Wake up)
- Hotel interface (minibar, alarm, room cleaning, extension privilege setting)
- PBX License Control

Specifications

UC200 Series IPPBX				
Product Model	UC200-2S2O	UC200-4O	UC200-4S4O	UC200-8O
FXS	2	0	4	0
FXO	2	4	4	8
Max SIP Register	200			
Max SIP Trunks	30			
Max Concurrent Calls	24			
Weight	0.55kg			
Power Supply	12V DC，≥2A			
Maximum power consumption	20W			
Dimension (W/D/H)	188X128X25 mm			
WAN (WAN Port)	10/100 Base-T RJ45			
LAN (LAN1 Port)	10/100 Base-T RJ45			
LAN (LAN2 Port)	10/100 Base-T RJ45			
Built-in Storage	128M			
Micro SD Card Slot	1			
USB PORT	1			
Operation Temperature	0°C ~ 50°C			
Operation Humidity	10% ~ 90% Non-condensing			
Storage Temperature	-20°C ~ 80°C			

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