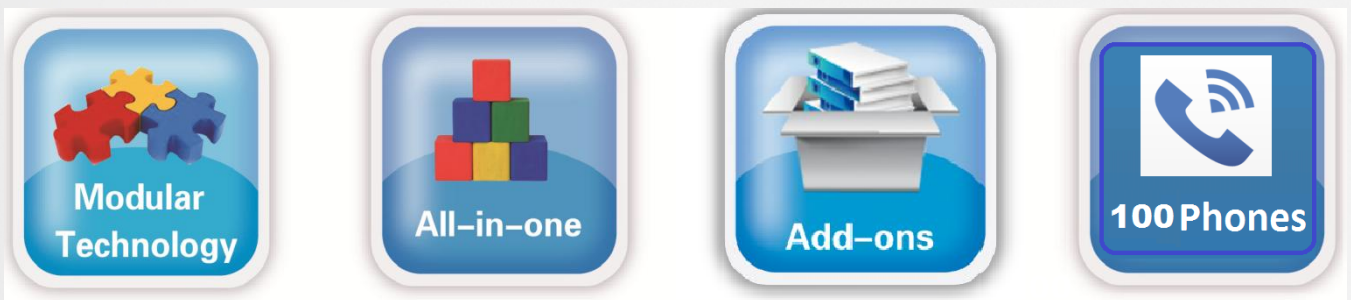


MyPBX U300



MyPBX U300

Embedded Hybrid IP-PBX for Your Business



MyPBX U300 boasts an embedded PRI (E1/T1/J1) port and 2 FXS ports in the one compact system, providing higher density trunking for offices using E1 PRI signaling. This system supports up to 300 users and 50 concurrent calls. Also it could be used as a gateway to legacy PBX systems in applications.

- **Easy to use**
Easy to deploy and manage via web-based configuration interface
- **Robust all-in-one features**
Deliver enterprise-class communication features and functionality to SMBs
- **Redundancy**
Support hot standby settings (automatic failover) in case of server failure
- **Match your IP phone**
Perfect interoperability with mainstream IP Phone
- **Speak your language**
Multi-language Web interface and voice prompts
- **No future licensing fees**
Scalable with plug-and play ease without licensing
- **Used as a gateway**
Could be used as a E1 gateway (E1 to VoIP or VoIP to E1)
- **Energy Saving**
Low power consumption for your green office



Basic Features

- Automated Attendant (IVR)
- Attended Transfer
- Blind Transfer
- Blacklist
- Call Back
- Call Detail Records (CDR)
- Call Forward
- Call Parking
- Call Pickup
- Call Routing
- Call Waiting
- Caller ID
- Conference
- Do Not Disturb (DND)
- Follow me
- Intercom/Zone Intercom
- Music on Hold
- Music on Transfer
- Queue
- Ring Group
- Skype Integration (Skype Connect)
- Speed Dial
- Voicemail (3000 minutes)
- Voicemail to email
- Voicemail Forwarding

Advanced Features

- Direct Inward System Access (DISA)
- Distinctive Ringtone
- Dial by Name
- LDAP Server
- One touch recording
- Phone Provisioning for Aastra, Cisco, Escene, Fanvil, Grandstream, Panasonic, Polycom, Snom, Yealink IP Phones
- QoS (voice quality)
- Redundancy (Hot Standby)
- Route by Caller ID
- Spy functions (Normal Spy, Whisper Spy, Barge Spy)
- Static Route

Add-on

- Call Recording*

Security

- Firewall
- SIP TLS transport
- SRTP (RTP encryption)

Faxes

- T30, T38 faxes
- Fax to email
- Incoming fax tone detection

Multiple Languages

- System voice prompt: American English, Australian English, British English, Chinese, Dutch, French, Canadian French, German, Greek, Hungarian, Italian, Polish, Portuguese, Brazilian Portuguese, Russian, Spanish, Latin American Spanish, Mexican Spanish, Turkish, Thai, Korean, Persian, Danish, Finnish, Norwegian, Swedish, Arabic
- Web GUI: English, Chinese Simplified, Chinese traditional, Portuguese, Spanish, Russian, Hebrew, Turkish, French, Italian, Polish, Romanian, Albanian, Thai, Korean, Persian, German, Dutch

Internet

- DHCP server
- DDNS
- Static IP
- DHCP client
- PPPoE
- VLAN: VLAN over LAN, VLAN over WAN
- VPN: OpenVPN, L2TP, IP Sec

Hardware Interface

- 1 LAN port (10/100Mbps)
- 1 WAN port (10/100Mbps)
- 1 E1/T1/J1 port (support PRI, MFC R2, SS7)

- 2 FXS ports
- 1 Audio input port
- 1 Audio output port
- 1 RS232 port
- 1 USB port (2.0)

System Capacity

- 300 IP phone users
- 50 concurrent calls
- 512 MB Onboard Flash
- 512 MB Onboard RAM
- Protocol: SIP (RFC3261), IAX2
- Transport: UDP, TCP, TLS, SRTP
- DTMF: RFC2833, SIP INFO, In-band
- Codec: G.711 (a-law, u-law), G.722, G.726, G.729 A, GSM, Speex, ADPCM, H261, H263, H263p, H264, MPEG4

Environment

- Size: 213x160x44 mm
- Weight: 1.2 kg
- Power Supply: AC 100~240V, 50~60Hz
- Operation Range: 0 to 50°C, 32° to 122° F
- Storage Range: -20 to 65°C, 4 to 149° F
- Humidity: 10-90% non-condensing

*Call recording files are stored in a USB device.