UC200 Pro IP PBX for SMEs

Overview

UC200 Pro is a highly efficient IP phone system solution specifically designed for small and medium-sized enterprises. UC200 Pro provides 2 FXS and 2 FXO interfaces and allows users to expand with 2S/2O/1S1O modules, supporting up to 3 additional modules to flexibly meet various communication needs. It supports both desktop and rack-mounted options to meet deployment needs across different scenarios. Desktop devices are compact and flexible, ideal for quick deployment , making them perfect for homes and small offices. Rack-mounted devices offer high scalability and centralized management, suited for medium to large enterprises and data centers, supporting complex IT infrastructures.

UC200 Pro enables remote office through SIP voice terminals and can seamlessly integrate with other IPPBX or traditional PBX systems, catering to the diverse communication requirements of enterprises. It employs advanced encryption and security strategies to fully ensure communication security.

UC200 Pro can help small and medium-sized call centers or enterprise branches to enhance communication efficiency and reduce communication costs, which can provides a stable, secure, and efficient communication environment for businesses.

UC200 Pro (Desktop)

UC200 Pro (Rack-mounted)

Key Features

- Supports up to 500 SIP users and 50 concurrent calls
- Supports 2 FXS and 2 FXO interfaces
- Expandable with 2S/2O/1S1O modules, supporting up to 3 modules
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR(Interactive Voice Response)
- Support voicemail/ Voice recording
- User-friendly web interface, classification of web user’s rights

SIP Extensions	60	200	500
Concurrent Calls	15	30	50

UC200 Pro provides the above 3 different types of licenses to grow with your business.

Specification	UC200 Pro (Desktop)	UC200 Pro (Rack-mounted)
SIP Users	60/200/500	60/200/500
Concurrent Calls	15/30/50	15/30/50
Recording Calls	7/15/25	7/15/25
Gigabit Ethernet ports	2 (By default, GE1 is the management port)	2 (By default, GE1 is the management port)
USB2.0	1	1
Console port	1*USB-B	1*USB-B
TF Slots	1	1
FXS Ports	2	2
FXO Ports	2	2
Expandable Modules (2S/2O/1S1O)	3	3
Power Supply	Input: 100-240VAC, 50/60Hz Output: DC12V, 2A	Input/Output: 100-240VAC, 50/60Hz
Dimensions（W/D/H)	260×154×42mm	440×160×44 mm
Power Consumption	10W	10W
Weight	1.1kg	2.0kg
Operating Temperature	0 ℃ ~ 45 ℃	0 ℃ ~ 45 ℃
Storage Temperature	-20 ℃~80 ℃	-20 ℃ ~80 ℃
Humidity	10%-90% Non-Condensing	10%-90% Non-Condensing

FXS

- Connector: RJ11
- Hook Flash
- Polarity Reversal
- Answer and Disconnect Signaling: Answer, Disconnect, Busy Tone
- Caller ID: Bellcore Type 1&2, ETSI, BT, NTT and DTMF

FXO

- Connector: RJ11
- Caller ID: FSK, DTMF
- Answer Delay
- Polarity Reversal
- Detection of Busy Tone
- Detection of No Current
- Auto Match of FXO Impedance

Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS,RTP/SRTP/ZRTP
- Silence Suppression
- VPN Server/Client
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control (AGC)
- FAX: T.38, VBD and Pass-through
- NAT: STUN/DDNS
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- DTMF: RFC2833/Signal/Inband
- Codecs: G.711a/μ law,G.723.1, G.729A/B, G.726, OPUS, G.722
- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- Echo Cancellation: G.168 with up to 128ms

PBX Services

- CDRs
- Video call
- Voicemail, Voicemail to Email, Email Client
- Paging/Intercom
- Dial Rules
- Caller ID display
- Blacklist/Whitelist
- Event Report
- Manager/Secretary Function
- Feature Code
- Business Hours & Holidays
- Call Restriction(Local/National/International)
- Hotline, VIP Extension
- Follow me (one number service)
- IVR (Multi-level/Multi-lingual/Time-based)
- 3-Way Conference, Conference call
- Failover Routing
- One SIP account with multi device registrations
- BLF, PIN list
- SMS Route
- Personal/Enterprise Phonebook, LDAP
- Auto-attendant Function
- Auto CLIP
- WebRTC
- Support Dinlink (App)
- Up to 25 concurrent calls in full recording mode
- Caller/Called Number Manipulation
- Routing Based on Time Period, Caller/Called Prefixes, Source Trunks
- Local Recording (Support USB Storage)
- Attendant Console
- PMS (Property Management System)

Call Feature

- Call Waiting
- Call Holding
- Call Forward (Unconditional/ No Answer/Busy)
- Call Transfer(Attended/Blind)
- DID/DOD
- DNIS (Dialed Number Identification Service)
- Call Pickup
- Call Parking
- Alarm call
- Speed Dial
- Emergency call
- Call Return
- DISA (Direct Inward System Access)
- Do-not-disturb (DND)
- Music on Hold
- Coloring Ring Back Tone(CRBT)
- Distinctive Ringtone, Custom Prompts
- Call Routing Groups, Ring Group
- Call Queue
- Auto-recording
- Call Monitoring (Listen/Whisper/Barge-in)

Network

- IP Firewall, Fail2ban
- Dynamic/Static NAT
- Dual-machine Hot Standby
- Anti-brute Force Attack Mechanism

Maintenance

- NTP
- Syslog
- Ping and Tracert
- Auto Provision
- Network Capture
- Web-based Management Portal
- HTTP&HTTPS/NATS API
- Configuration Restore/Backup
- Multiple Languages Supported
- HTTP/HTTPS/TFTP/FTP Firmware Upgrade
- Traffic Statistics: TCP,UDP,RTP
- Classification of Web Users' Rights
- Personal Portal Management
- PNP Auto Provision
- Network Management System

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DINSTAR

About Us

Founded in 2011 in Shenzhen, DINSTAR is a leading global provider of IP Unified Communication products including VoIP Gateways, IP PBXs, IP Phones and SBCs, we have been delivering more agile, efficient and affordable communication solutions and unparalleled communication experiences to our customers with our reliable, innovative and future-proof products for years. Through our value-added distributors and resellers worldwide, now DINSTAR serves telecom operators, service providers, system integrators, enterprises, SMBs and OEM partners in over 100 countries.