UC350 Series IP PBX

Overview

UC350 Series IP PBX is a new generation IP PBX for large capacity unified communication solutions. Based on the powerful hardware platform, UC350 Series IP PBX includes UC350 and UC350 Pro. UC350 and UC350 Pro support 1,000/5,000 extensions and 120/500 concurrent calls which are integrated voice, video, paging, fax, conference, recording, and other useful functions. It also provides four slots that can install E1/T1 boards, FXS, and FXO boards by hot-plug mode, so that it can be flexibly configured and combined according to the actual use scenarios. It is suitable for helping to build the telephony system of large and medium-sized enterprises and can meet the branch office needs of large group enterprises and government agencies, helping enterprises and industry customers to establish a convenient and efficient IP telephone system.

In addition, the UC350 Series IP PBX provides an intuitive and easy-to-use graphical interface to facilitate user management and maintenance. With the open API, users can easily integrate the phone system with third-party systems of CRM and hotel system platforms. It provides a convenient and intelligent one-stop telephony solution for enterprises.

UC350

UC350 Pro

Key Features

- Supports Voicemail/ Voice recording
- Supports E1/T1, FXO, and FXS ports with flexible and alternative capability
- FXU User board supports power failure lifeline
- Distributed multi-core CPU, greatly improves the call processing capacity
- Flexible dial rules based on time, number or source IP etc.
- Supports Multi-level IVR, helps to build personalized voice navigation for enterprise
- User-friendly web interface, classification of web user's access permission

Specification	UC350	UC350 Pro
SIP Users	1,000	5,000
Concurrent Calls	120	500
MCU board Slots	1	1
Gigabit Ethernet ports	2 （By default, GE1 is the management port）	4 （By default, GE3 is the management port）
USB2.0	1	2
USB3.0	✕	1
Console port	1*RJ45	1*USB-B
User board Slots	4	4
FXS Board (8 FXS Ports)	2*RJ45	2*RJ45
FXO Board (8 FXO Ports)	2*RJ45	2*RJ45
FXU Board (4 FXS and 4 FXO Ports)	2*RJ45	2*RJ45
DTU Board (4 E1/T1 Ports)	4*RJ45	4*RJ45
1+1 Power Supply (100-240 VAC, 50/60 Hz)	✓	✓
Dimensions (W/D/H)	437*345*49 mm	437*345*49 mm
Power Consumption	50W	55W
Weight	5.7kg	5.6kg
Operating Temperature	0 ℃ ~ 45 ℃	0 ℃ ~ 45 ℃
Storage Temperature	-20 ℃~80 ℃	-20 ℃ ~80 ℃
Humidity	10%-90% Non-Condensing	10%-90% Non-Condensing

FXS

- Connector: RJ45
- 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: RJ45
- Caller ID: FSK, DTMF
- Polarity Reversal
- Answer Delay
- Detection of Busy Tone
- Detection of No Current
- Auto Match of FXO Impedance

PSTN

- 4* E1/T1 Ports at max
- Interface: RJ48 (120 Ohm)
- ISDN PRI:
   23B+D(T1), 30B+D(E1),
   NT or TE ITU-T Q.921, ITU-T Q.931, Q.Sig
- Signal 7/SS7:
   ITU-T, ANSI, ITU-CHINA,
   MTP1/MTP2/MTP3, TUP/ISUP
- R2 MFC
- E1 Frame Format: DF, CRC-4, CRC_ITU
- T1 Frame Format:
   2-Frame Multi-frame (F12, D3/4),
   Extened Super-frame (F24, ESF)
- Line Code:
   E1: HDB3, T1: B8ZS
- Clock: Local/Remote Clock Source

Maintenance

- Web GUI Configuration
- Configuration Restore/Backup
- Multiple Languages Supported
- HTTP Firmware Upgrade
- Syslog, Ping/Nslookup/Traceroute
- Traffic Statistics: TCP, UDP, RTP
- Network Capture, Web Multi-User
- NTP, FTP server

Maintenance

- Classification of Web Users' Rights
- HTTP&HTTPS/NATS API
- Schedule Task, Event Report
- Remote management via cloud services
- Firewall, Hosts

Voice Capabilities

- VoIP Protocols: SIP over UDP/TCP/TLS, SIP-T, SDP, RTP/SRTP
- Silence Suppression
- Dynamic Jitter Buffer
- Adjustable Gain Control
- Automatic Gain Control(AGC)
- Comfort Noise Generator(CNG)
- Voice Activity Detection(VAD)
- NAT: STUN/DDNS/Rport/Static IP
- DTMF: RFC2833/Signal/Inband
- FAX: T.38 and Pass-through
- Echo Cancellation: G.168 with up to 128ms
- Call Progress Tones: Dial Tone, Ring Back Tone, Busy Tone
- Audio Codecs: G.711, G.723.1, G.729, G.722, G.726, OPUS
- Video Codecs: VP8, H.261, H.263, H.264

PBX Services

- 3-Way Conference
- Voicemail to Email
- CDRs, Multi-level IVR
- Auto-attendant Function
- Voicemaill, Voice Recording
- Event Report, Email Client
- Zero configuration of the phone
- Dial Rules, Failover Routing
- Routing Groups, Ring Group, Call Queue
- Paging/Intercom, Hotline, Do-not-disturb
- Routing Based on Time Period
- Routing Based on Source Trunks
- Routing Based on Caller/Called Prefixes
- Caller/Called Number Manipulation
- Call Forward (Unconditional/No Answer/Busy)
- Call Waiting/Call Holding/Call Transfer

DINSTAR

The Gateway to VoIP World