SUNCOMM

High-end Enterprise Business IP Phone

PSTN+VOIP

SC135-G

Call recording
Automatic deployment

Overview

SC135 intelligent telephone use CortexA7 1.2*quad-core design,support HD voice , The original ecological Android 7.1 system design supports application expansion based on the open Android platform. The device uses a 3.5 inch 320 ×480 IPS high-definition resolution display, supports rj45 wired network, and meets various network connections. Input method.

SC135 supports HD audio calls based on the SIP protocol, and has excellent hardware echo suppression and noise reduction effects;   supports the standard SIP2.0 protocol, and is compatible with mainstream soft switches, IMS and IPPBX systems on the market, as well as various SIP protocol mainstream audio terminal.

Optional PSTN phone function, and support TF card recording and answering functions, flexible hardware configuration, can be matched according to customer application scenarios.

HD voice

3.5-inch IPS screen

Dual gigabit Network port

Audio conference

Call Recording

POE power supply

6 SIP account

Product Highlight

Good user experience

The Simple and elegant outlook, It features a 3.5-inch 320×480 IPScolor screen, with a use-friendly UI, bringing a comfortable and efficient experience. Alsothe Snap-On bracket design on the back of the SC135-G fits perfectly with the desktop, making it easy to use and firm.

Powerful & good expansibility

Built-in 2.4G/5G WIFI，2*1000m network ports make the connection flexible and convenient .1* PSTN port, which can be connected to the telephone line & meet customer needs in different scenarios.Has a TF card, which can be expanded to store call records. It supports POE power supply, USB port and power line power supply, and multiple charging methods to meet various needs of customers.

HD AUDIO

Adopting industry-leading intelligent noise cancellation technology, it can provide high-definition, perfect sound quality, and provide users with a first-class communication experience.

Secure Deployment Configuration Protocol

Users do not need to manually configure and upgrade the phone, just need to power on and connect to the Internet, and the phone can be automatically deployed, saving a lot of time and cost.

Technical Parameters

AUDIO

- HD Voice Microphone/Speaker(Handset/Hands-free)
- Excellent Echo Cancellation Technology(ECT)
- Receiver with intelligent noise reduction ANC Technology
- Audio codec: G.722,GSM,PCMA,PCMU,G.729AB,iLBC,AMR, DTMF,ISACWB,ISACSWB,OPUS
- DTMF：In-band、SIP INFO
- Full duplex hands-free
- VAD，CNG，AEC，AGC，ANS

Phone Features

- Call hold, mute, DND, redial, One-touch speed dial
- Call forward/ call transfer/ blind transfer
- Redial, back and auto answer/Intercept answer/Intercept group
- Call log: all/out/received/missed/call forward
- 6-way conferences
- TF card automatic loop recording
- Direct IP call without SIP proxy
- Ringer selection
- Set date time manually or automatically
- Built-in dual-band WIFI：
  -Network Standards：IEE 802.11a/b/g/n
  -Band：2.4GHz/5.0GHz

Directory

- Phone book (1000 records)
- Backlist
- Phone book search/import/export
- Tree enterprise phone book/LDAP phone book

Others

- BLF
- Intercom, multicast
- Message Prompt
- Voice message, switch account
- Call Park, Call pickup
- Recording after calling
- Server-based recording function

Feature keys

- 34 touch keys
- 7 features keys: Hold/ meeting/ headphone/voice mail/call transfer/redial/Hands-free
- Menu key/ number keys
- Volume+/- key
- Speed dial key

PSTN Feature （optional)

- Support FSK/DTMF
- Answering: Automatic Answering/Workday Answering
- Calling Region Checking/auto dialing with "0"

Interface

- 2*RJ45 1000M ports
- 1* TF card port
- 1*USB 2.0 port
- 1*PSTN jack
- 1*Power adapter jack
- 2*RJ9 (4P4C) Handset/headphone jack

Management

- Online APP update, WEB terminal offline update
- · Support TFTP/PnP/DHCP/one key update mode and auto provisioning
- · TR069 remote provisioning
- · Reset to factory, reboot
- · Packet capture and export system log

Network and Security

- Call server redundancy supported
- NAT Traversal By STUN
- Proxy mode and peer-to-peer SIP link mode
- IP address allocation mode: static/dynamic
- Support HTTP web server
- Time and date synchronization using SNTP
- Support UDP/TCP/TLS
- QoS：DSCP
- Support STRP

Phone interface

- Incoming call /missed message led indicator
- Intuitive user interface
- support Chinese/English or other languages
- Incoming call show name and number

Other physical

- Color: Black
- Power adapter: 12V/1A DC input
- Relative humidity: 10%~90% No Condensing
- Operation temperature: -10~50 °C

Package Contents

- Package list:
  -S135-G IP phone /handset / bracket /power adapter/
  line cord/user manual
- N.W/CTN：9.1kg
- G.W/CTN：10.1kg
- GIFT BOX SIZE：265x239x63mm
- MASTER CARTON SIZE：555*335*258mm

Product pictures