ESCENE

Call Center IP Phone

CC800V2

ESCENE CC800V2 is one of the SayHi series IP Phone in the Call Center. It has has the unique style, good utility, clear voice etc feature. Cooperating with communication platform to finish strong phone functions, such as: call transfer, hotline function (immediately/delay), a key cancellation and registration, a key automatic response, etc.

Key Features

- One key enable or disable accounts register function.
- One key enable or disable auto-answer function.
- One key change the ringing type.
- Multi-language, e.g. Chinese, English, Russian, French etc .
- Two SIP accounts and support three-way conference, SMS.
- 2xLAN, PoE, RJ9Headset.
- 5 programmable keys.
- USB port for external unit charging.
- XML/LDA, BLF/BLA
- Auto-provision, HTTP/TFTP/FTP.
- Light of status.

CC800V2 Technical Datasheet

Phone Features

2 SIP accounts, Hotline
Call hold, Call waiting, Call forward, Call return
Call transfer (blind/busy/ask)
Caller ID display, Redial, Mute, DND
Auto-answer, 3-way conferencing
Speed dial, SMS, Voicemail
Message Waiting Indication LED
Tone scheme, Volume control
Direct IP call without SIP proxy
Ring tone selection/import/delete
Black list, Hand-free indicator.
Call history: dialed/received/missed (50 -entries);
Multi-Language
Soft keys programmable, Supports PC control

Advanced Features

XML Phone Book search and input & output
Enterprise phone book(800 entries)
LDAP phone book
Functions customizable keys

IP PBX System Integration

Busy lamp field (BLF), BLF list
Bridged line appearance (BLA)
DND & Forward synchronization
Intercom, Paging, Music on hold
Call pickup, Dial Plan, Call recording
Anonymous call, Anonymous call rejection
Network conference, Distinctive ringtone

Codes and Voice Features

Wideband Codec: G.722
Narrowband codec: G.711μ/A, G.723.1
G.726, G.729AB,iLBC
VAD, CNG, AEC, AGC.Full-duplex,

Security

VLAN QoS (802.1pq)
Phone lock for personal privacy protection
Admin/User 2-level configuration mode , LLDP, CDP
L2TP VPN tunneling protocol – added

Network Features

SIP v1 (RFC2543), v2 (RFC3261)
SIP connection: Proxy mode & Peering Points mode;
DNS SRV (RFC3263);
Redundant server support
NAT Traversal: STUN mode
DTMF: In-Band, RFC2833, SIP Info, Auto
HTTP/HTTPS Web Management
IP Assignment: Static/DHCP/PPPoE
Bridge/Router mode for PC port
TFTP/DHCP/PPPoE client
DNS client, NAT/DHCP server
Layer 3 ToS, DSCP

Management

Auto provision via FTP/TFTP/HTTP/HTTPS for mass
Configuration: browser/phone/auto-provision
Trace package and system log export

Physical Features

128x64 graphic LCD
25 keys including 2 programmable keys
6 LEDs: 1* headset light, 2* accounts lights,
3*function lights (Answer Key\Ringing\Mute)
1* RJ-9 handset port
1*RJ-9 headset port
2*3.5mm PC headset port
2xRJ45 10/100M Ethernet ports
Power adapter: AC 100~240V input and DC 5V/1A output
Power over Ethernet ,IEEE 802.3af,class 0
Power consumption: 2.5-3.5W
Net weight: 0.342KG
Dimension: 162x105x62MM
Operating humidity: 10~95%
Storage temperature: up to 60°C

Package Features

Qty/CTN: 20 PCS
N.W/CTN: 6.84KG
G.W/CTN: 10.08KG
Carton Meas: 285*460*300MM

Certifications

Compatible with the platform

ESCENE COMMUNICATION CO.,LTD

Add: Floor 4 building C YueAn Industrial Park No.51 Huangcun Road Tianhe District,Guangzhou,China.

Tel: (86)20-82320720 Fax: (86)20-82326757