STONEHENGE IP250 /IP255

Field Proven VoIP Phone

Widely used voip telephone in Market

Stable and Deferentiable solution for provider or dealer

IP250 / 255

SIP

As VoIP phone for business based on standard SIP, it meets every function of various IP PBX and soft-switches embodying linked function enough. It proved their stability and VoIP technique by installing them into domestic and overseas sites. Based on the proved technique, it increase the convenience and productivity.

IP 250 / IP 255 제품 특징

SIPv2(RFC3261) of VoIP Phone

As Global standard SIP internet phone, It allows easy installing and accessing to anywhere connected with internet, SIP service server and installed soft switch.

Graphic LCD for user

Graphic LCD in English helps user to be accessible to various function easily.

RTP and Transport Layer Security

To avoid call hacking, it supports a high level signal assuring the excellent security.

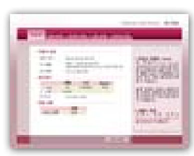
Excellent sound quality by various Codec

Adapting various Codec including G.711a Law/μLaw, G.279.A/B, guarantees high quality sound as much as conference phone.

IPv4/IPv6 dual stack

Supporting IPv4 and IPv6 together

Effective remote management



For effective management, it supports Auto Provision function which can easily upgrade software and install phone.

IP 250 IP Phone

FXO & PoE Optional

PoE

PoE(Power of Ethernet): Self power adapter over Ethernet.

FXO

FXO(Foreign Exchange Office): Additional PSTN port

IP 255 IP Phone

PoE Optional

PoE

PoE(Power of Ethernet): Self power adapter over Ethernet.

STONEHENGE IP250 /IP255 Specification

Call Function

- Call Forwarding
- Call Hold
- Call screen/ Do Not Disturb
- Call Pickup
- Call Transfer
- Call Waiting
- 3 Party Conference call
- SIMPLE based Instant Messaging and Presence
- Auto Provisioning (HTTPS, HTTP, TFTP)

SIP Protocal

- Proxy Registration and Failover
- Outbound Proxy
- Multi-user Registration
- Registration Timer
- SIP Transport – UDP, TCP, TLS
- Secured media negotiation(SRTP)
- Realm-based authentication (Digest authentication)
- Session Timers
- DNS query (A record, SRV, NAPTR)
- Codec Negotiation
- DTMF relay RTP payload(RFC 2833) or SIP info
- Hook flash signaling
- Visual Message Waiting Indicator

DTMF/Ring Signal

- DTMF (Dual-Tone Multi-Frequency)
- Multiple Ring Tones
- My bell: 10 Ring
- Call Progress Tone Generation
  (Dial tone, Busy, Audible Ring back)
- DTMF generation
  (RFC 2833 In-Audio or Out-of-band SIP info)

Voice & Codec

- CNG (Comfort Noise Generation)
- Echo Cancellation: G.168 compliance
- Codec Auto Negotiation
- Codecs
  - Narrow band -G.711μLaw/aLaw with PLC, G.722.1, G.723.1/A, G.726, G.728, G.729A/AB/E , Broadvoice®16
  - Wide band – G.722.1, Broadvoice®32
- Echo Suppression (G.164)
- Enhanced Packet Loss Concealment
- Silence Suppression (G.164)
- VAD (Voice Active Detector)
- Adaptive Jitter Buffer
- SIPFrag (RFC 3420)
- Dynamic Payload Support
- Adjustable Audio Frames per Packet
- Flexible Dial Plan Support with Inter-Digit Timers

IP Network

- IPv4 (RFC 791), IPv6(RFC1883) dual stack (optional), TCP, UDP, HTTP, ARP, ICMP
- RTP/RTCP, Secure RTP
- DNS: A record (RFC 1706), SRV record (RFC 2783)
- NAT/PAT
- VLAN: IEEE 802.1q
- QoS: IEEE 802.1p, DiffServ(RFC 2475)
- Network Address Assignment: Static IP/DHCP, PPPoE

Managemen

- Password Protection for Admin mode and User mode
- Management Protocol: SNMPv2 (RFC 2782)
- Auto Provisioning: DHCP TFTP, Static TFTP, HTTP
- Remote Software Upgrade: http,tftp
- Remote Configuration

Ethernet

- Dual switched 10/100 Based-T Through RJ-45 Interfaces
- 10/100BASE-T: 2 Ports Providing auto-MDI/MDIX, Enabling the Use of Straight or Crossover Cable in Either Port
- 1 RJ-45 Port to Connect LAN or Wan, this Port Supports Optional PoE (IEEE802.3af or In line power)
- 1 RJ-45 port to Connect PC

Key & Button

- Dial Keys: 12 keys (ITU E.161)
- 9 Function Keys: Redial, Transfer, 3 Way call, Call Pick up, Hold, Mute, Headset, DND, Call Forwarding
- Speaker Phone Button for Handfree conferencing
- Additional 3 Service Buttons: Do Not Disturb function, Call Forwarding, Message
- 6 Hotkeys
- 2 Volume Control Keys; can be used as navigation (up & down) Keys on the Menu
- Phonebook: Record 100 Entries
- Call history Logs: Record 60 Entries for In/ Outbound or Missed Calls

Display

- Black and White LCD: 2x16 Character LCD or 128x64 Graphic LCD with Back Light
  Multi-Language Support: English + one
- Ringing and Message Indicator: Green LED for status indication

Power

- Power Adaptor: Output DC5V/1A, rated Input AC100~240V. 50/60Hz
- Power Consumption: 5Watts(max)
- PoE (Power over Ethernet) *Optional

Technical Spec

- Measurements: 82x109x125mm (WxHxL)
- Weight: 800g
- Color: beige, black
- Operating Temp: 0~45°C
- Storage Temp: -20~60°C
- Humidity: 10~85% (Non-Condensing)
- Storage Temp: -20~60°C
- Phone Stand
- Handset: RJ-7 Standard Connector

Optional

- IP250: PoE or FXO
- IP255: PoE(Power over Ethernet)

RFCs supported

RFC2327	Session Description Protocol(SDP)
RFC2976	The SIP INFO Method
RFC3261	SIP: Session Initiation Protocol
RFC3262	Reliability of Provisional Responses in SIP
RFC3263	SIP : Locating SIP Servers
RFC3264	An Offer/Answer Model with SDP
RFC3265	SIP – Specific Event Notification
RFC3420	Internet Media Type message/sipfrag

RFC3428	SIP Extension for Instance Messaging
RFC3515	The SIP Refer Method
RFC3725	Best Current Practices for Third Party Call Control in SIP
RFC3842	A Message Summary and Message Waiting Indication Event Package for SIP
RFC3892	The SIP Referred-By Mechanism
RFC3903	SIP Extension for Event State Publication
RFC4568	SDP Security Descriptions for Media Streams

IETF drafts supported

Draft-ietf-sipping-cc-transfer-01	SIP call control – Transfer
Draft-ietf-sip-replaces-02	The SIP Replaces Header
Draft-ietf-sip-session-timer-08	The SIP Session Timer

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