STONEHENGE IP930 /IP950

Conference Butler

The best conference solution for conference room



IP 950 / 930

Echo Cancellation

Through canceling Static noise and Echo noise, it upgraded the bi-directional sound quality which is most important function to Conference phone.

Self Installing AI Microphone

Regardless of the size of the room, Artificial Intelligence Microphone is automatically installed to the best environment of microphone itself.

DSP(Digital Signal Processor)

For the perfect voice quality, it adapted DSP. It guarantees successfully multilateral conference with clear and natural sounds without delaying.

360 omni directional sound recognizing

From anywhere in the room Microphone recognizes the voice.

Recording function

While doing conference, user can simply record their conference through the recording terminal.

LCD Display

LCD display shows the telephone number and the status of telephone for convenient use.

IP 950

IP Conference Phone

POE (Power over Ethernet)

IP950 earns power from Network Ethernet line. AC/DC adapter is not necessary.

FXO Port for PSTN Line

IP950 is transferable to PSTN line. User can experience Excellent and flexible scalability from IP950.

930

PSTN Conference

PSTN Conference

IP930 is PSTN conference phone. Connecting to normal circuit, all function for conference phone is available.

Speed Dial

Speed Dials to save the conference time.

STONEHENGE IP930/IP950 Specification

Call Function

- Call Hold
- Call Mute
- Call Waiting
- 3 Party Conference call
- 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

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

Audio

- Microphones: 300 to 3500Hz
- Microphone pickup range: up to 3m
- Microphone with intelligent microphone mixing
- Extension microphones 300 to 3500Hz(Optional)
- Dynamic noise reduction

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
- 1 RJ-45 port to connect LAN or Wan, this port supports
Defaulted PoE (IEEE802.3af or In line power)
- 1 RJ-45 port to connect PC

FXO

- 1 RJ-11 FXO port for PSTN back up (Q1,2008)

Key & Button

- Dial Keys: 12 keys (ITU E.161)
- 4 Function Keys: On-hook/Off-hook, Hold, Mute
- 2 navigation keys: Up&Down keys can be used
for volume control

Display

- Character LCD: 12 character
- Language Support: English

Power

- PoE (Power over Ethernet) defaulted
- Power Consumption: 5Watts(max)

Technical Spec

- Measurements: 310x310x60mm(WxHxL)
- Weight: 760g
- Color: Dark gray
- Operating Temp: 0~45°C
- Storage Temp: -20~60°C
- Humidity: 10~85% (Non Condensing)

930(PSTN) Specipication

PSTN Conference Phone

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- Speed Dial button
- LCD Display
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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 Wait-ing 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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