

STONEHENGE IP270

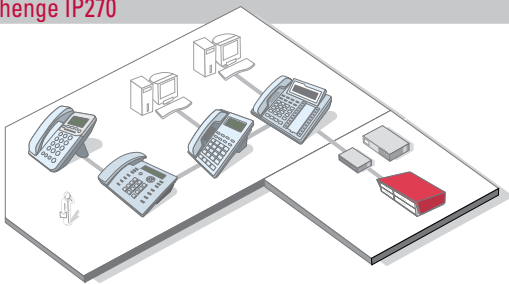
SIP based VoIP Superset

High Class IP phone based on standard SIP

Office VoIP solution for many functions and simple user interface



Stonehenge IP270



This IP270 is one of high class VoIP telephone based on Standard SIP, for convenient use, it support various function key as well as supplementary function. Graphic LCD and Bicolor LED help to facilitate the function. The proved the VoIP technique and stability of Moimstone from the market are the good solution.

20 Memory dials and menu button



20 speed dials are available for user, and 4 direction keys and frequently used function key are available also.

Graphic Icon on Menu



IP 270 shows environment for use as Graphic Icon. Those icons facilitate the understanding for VoIP function.

PoE(Power over Ethernet)



IP270 earns power from Network Ethernet line. So it doesn't need additional power adapter.

FXO for PSTN(Optional)



By building additional FXO(Foreign Exchange office) port optionally, it offers the change to PSTN from VoIP.

PSTN backup Battery for power outage

Backup Battery is built in the phone in case the abrupt power outage while using PSTN Line. With Backup Battery, LCD, LED and all functions will be operational when power is out.

Bicolor LED for BLF

BLF(Busy Lamp Filed) enables user to check if other lines previously installed are busy.

STONEHENGE IP270 Specification

Call Function

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

SIP Protocol

- 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@32
 - 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,ftp
- 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 Defaulted PoE (IEEE802.3af or In line power)
- 1 RJ-45 port to connect PC

FXO

- 1 RJ-11 FXO port for PSTN Use

Key & Button

- Dial Keys: 12 keys (ITU E.161)
- 11 Function Keys: Transfer, 3 Way call, Pick up, FXO, Hold, Redial(MUTE), Intercom, Menu, Headset, DND(ENT), Call Forwarding, Esc, Message(DEL)
- Speaker phone button for hands free conferencing
- Additional 20 Keys for Hot key
- 4 navigation keys: Up&Down keys can be used for volume control
- Phonebook: Record 100 entries
- Call history Logs: Record 60 entries for In/ Outbound or Missed Calls

Display

- Black and White (Gray) LCD: 255x64 Graphic LCD with back light
- Multi-Language Support: English + one
- 20 LED: 4 special key + 16 user setting key - 00 Bicolor LED for line status indication

Power

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

Technical Spec

- Measurements: 252x205x112.5mm (WxHxL)
- Weight: 1.2Kg
- Color: beige, black
- Operating Temp: 0-45°C
- Storage Temp: -20-60°C
- Humidity: 10-85% (Non Condensing)
- Phone Stand
- Handset: RJ-7 standard connector

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

IETF drafts supported

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

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