

VoIP Phone SI 120

SIP based VoIP Phone



- Three-line display (3 x 15 characters)
- 28 keys, 1 LED
- 2 x IEEE 802.3
- 10/100 Mbps switch
- 2 identities
- SIP 2.0 (RFC3261)
- IAX 2
- STUN, NAT, ICE
- Codecs:
G.711, G.729A, G.723.1, G.722

→ Usability

The IP telephone with the latest VoIP technology for the discerning caller! Ideal for ITSPs, offices, and private users.

As the basic model of the Snom India business telephone family, the SIP telephone SI 120 fulfils the requirements of VoIP telephony and additionally offers numerous functions that are indispensable for everyday working life. Due to its excellent cost-performance ratio, it is ideal both for ITSP packages/products and for small and medium sized enterprises, home offices, and private users.

→ Security

The SI 120 sets new standards when it comes to user-friendliness: A three-line LCD display shows call information, and the menu-driven user interface provides simple feature management.

The SI 120 is remote manageable and firmware-upgradable, uniquely easy to install, and largely self configuring. Broad codec support and full compatibility with current SIP recommendations ensure interoperability; support for STUN (NAT traversal), and other state-of-the-art features enables flexible deployment behind local proxies, IP PBXs or hosted VoIP services.

→ Interoperability

With SIP (Session Initiation Protocol) you are ensuring your own personal independence. Most vendors are touting SIP to be the communication protocol of the future. SIP components can be combined into a complete system without you being tied to a single provider.

By limiting the need for external conference bridges/media server capacity or use of conference services for routine multiparty calls, the SI 120'S built-in three-party conference bridge helps limit total cost of ownership, while also ensuring high audio quality and low latency.

Technical Data SI 120

General Features

- **Dimensions:**
Approx. 220 x 148 x 75 mm
- **Weight:** approx. 940 g
- **Safety:** IEC 60950-1:2001,
CB Test Certificate: DE 02011039
- **Certifications:** FCC Class B,
CE Mark
- **Power consumption:** 2.5 – 2.8 watts

Connectors

- **1 x LAN, 1 x PC:** RJ45 (Ethernet)
- **Power:** 5 V DC
- **Ethernet:** 2 x IEEE 802.3
10/100 Mbps switch
- **Handset:** RJ11 connector

User Interface

- 3 x 15 character display, backlit
- 28 keys, 15 function keys
- Caller-ID
- Message waiting indication
- Address book (500 entries)
- Address book import/export
- Speed dialing
- Local dial plan
- Lists of missed, received and dialed
calls (100 entries each)
- Call waiting indication
- Clock, daylight saving, call-timer
- Call blocking (deny list)
- Blocking of anonymous calls
- Menu-driven user interface
- Ring tone selection
- Speakerphone (full duplex)

SIP

- RFC3261 compliance
- Digest/basic authentication
- PRACK (RFC3262)
- Reliability of provisional responses
(RFC3262)
- Early media support
- DNS SRV (RFC3263), redundant server
Support
- Offer/answer (RFC3264)
- Message Waiting Indication
(RFC3842), subscription for
MWI events (RFC3265)
- In-band DTMF/out-of-band DTMF/
SIP INFO DTMF
- STUN client (NAT traversal)
- Auto provisioning

Call Features

- Hold
- Blind transfer, attended transfer
- Music on hold support (only via PBX)
- Divert
- Conferencing (3-way conference
bridge on phone)
- Call park, call pick-up (only via PBX)
- Call completion
- Call waiting/switching between
calls
- Redialing
- Support voice record in this phone, 240
seconds max or 3 record max, user-
defined record prompt with 1 minute max

Web Server

- Embedded web server HTTP/HTTPS
- Easy configuration of the phone,
Remote configuration

- Password protection
- Diagnostics (tracing, logging,
syslog)

Network Features

- WAN/LAN: Support bridge and
router model
- Support PPPoE for xDSL
- Support basic NAT and NAPT
- Support DHCP client on WAN
- Support DHCP server on LAN
- Support main DNS and secondary
DNS server.
- Support VLAN (optional: voice
vlan/data vlan)
- Support DMZ
- Support DNS Relay, SNTP Client,
Firewall
- Support VPN (L2TP/UDP
TUNNEL)
- Support NTP
- QoS with DiffServ

Codecs, Audio

- G.711 A-law
- G.711 μ -law
- G.729A
- G.722
- G.723.1

Maintenance

- Automatic software update
- Automatic settings retrieval
via HTTP/HTTPS/TFTP
- Web, Telnet, Keypad Management

For More Information, Please Contact