# VoIP phone snom 320

## Innovative SIP based VoIP Phones

- two-line semi-graphical display
- 47 keys, 13 LEDs
- 12 programmable function keys
- Speakerphone
- Dual Ethernet connection
- Power over Ethernet
- Headset connection
- SIP RFC3261
- Security: SIPS/SRTP, TLS
- STUN, ENUM, NAT, ICE
- Compression: G.723.1 and others
- National Language Support

Usability | Security | Interoperability

Ideal for the office and everyone who makes a lot of calls, the snom 320 is an affordable, yet powerful SIP business telephone with built-in, full duplex speakerphone and three-party conference bridging.

A 2 x 24 semi-graphical LCD display and menu-driven user interface support easy feature management.

12 programmable keys with LEDs support flexible trunk access/key configuration. A 100 number call memory, 100 entry onboard address book (to which data may easily be uploaded), custom call blocking, configurable/down-loadable ring-tones, auto-answer mode, DND and other sophisticated features ensure convenience and productivity. And the 320's built-in web server supports even simpler end-user configuration, screen dialing, and access to call history.

The snom 320 is remote-manageable and firm-ware-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), ENUM (for dialed-number resolution) and other state-of-the-art features enables flexible deployment behind local proxies, IP PBXs or hosted VoIP services.

The snom 320 supports the security standard SRTP - a current specification from the Internet Engineering Task Force (IETF) for protection against eavesdropping - and TLS for protection against sniffing of signaling and authentication data.

By limiting the need for external conference bridges/ media server capacity or use of conference services for routine multiparty calls, the snom 320's built-in three-party conference bridge helps limit total cost of ownership, while also insuring high audio quality and low latency.

# Technical Data

- **Dimensions:** approx. 25 x 20 x 12 cm
- **Weight:** approx. 920 g
- **Certifications:** FCC Class B, CE Mark Commercial

### Connectors:

- **Network:** RJ45 (Ethernet)
- **PC:** RJ45 (Ethernet)
- **Power:** 5 V DC (stabilized)
- 2 port switch
- Power over LAN (IEEE 802.3af) on network port
- **Handset:** RJ11Connector
- **Headset:** RJ11 connector

### User Interface

- Display: 2 x 24 character display
- 47 keys, 13 LEDs
- Last calls (100 entries)
- Address book (100 entries)
- Address book import/export
- Number guessing, speed dialing
- Missed calls, dialed calls
- Call waiting indication
- Clock, daylight saving, call-timer
- Call blocking (Deny List)
- 12 programmable function keys
- Menu-driven user interface
- Selectable ringing melodies
- National language support for selected languages (NLS)
- Downloadable ringing melodies
- URL dialing support
- Do not disturb
- Speakerphone (full duplex)
- Auto answer mode
- UTF8-encoded caller-ID
- Multi-line registration (12)
- Handling of 12 simultaneous calls
- Keyboard lock

### Call Features

- Hold
- Blind transfer, attended transfer
- Music on hold support (PBX)
- Divert
- Call intrusion
- Conferencing (3-way conference bridge on phone)
- Call park (PBX)
- Call pick-up
- Call completion
- CMC (Client Matter Code)

### Web Server

- Embedded web server
- Easy configuration of the phone, remote configuration
- Dial from web interface
- Password protection
- Diagnostics (tracing, logging)

### Security

- HTTPS (server/client)
- Transport Layer Security (TLS)
- SRTP (RFC3711)
- SIPS
- VLAN (802.1pq)

### Codecs

- G.711 aLaw, µLaw
- G.729A, G.726, G.723.1, GSM 6.10 (full rate)
- G.722 (16 kHz)

### SIP

- RFC3261 compliant
- UDP, TCP support
- Digest authentication
- Loose routing and strict routing support
- Error information support
- Reliability of provisional responses (RFC3262)
- DNS SRV (RFC3263), redundant server support
- Offer/answer (RFC3264)
- Message waiting indication reception (RFC3842), subscription for MWI events (RFC3265)
- Dialog-state
- In-band DTMF/Out-of-band DTMF (RFC2833)
- STUN client (NAT traversal)
- ENUM (RFC3261)
- NAPTR (RFC2915)
- rport (RFC3581)
- REFER (RFC3515)
- Many other SIP features

### Installation

- Static IP provisioning, DHCP
- HTTP/HTTPS client for configuration
- Automatic software update
- Completely automatic installation from web

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