snom
VoIP phones

snom 320

Innovative SIP based VoIP Phone

snom OCS edition

- Tiltable semi-graphical two-line display
- 47 keys, 13 LEDs
- 12 programmable function keys
- Speakerphone
- 2 x IEEE 802.3 10/100 Mbps switch
- Power over Ethernet
- Headset connection
- SIP RFC3261
- Security: SIPS/SRTP, TLS
- STUN, ENUM, NAT, ICE, IEEE 802.1X
- Codecs: G.711, G.729A, G.723.1, G.722, G.726, GSM 6.10 (full rate)
- National language support
- XML driven mini browser
- Very low energy consumption
- Expansion module available
- Unified Communications ready

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 a menu-driven user interface supports easy feature management.

12 programmable keys with LEDs support flexible identity access/key configuration. A 100 number call memory, a 100 entry onboard address book (to which data may easily be uploaded), custom call blocking, configurable/downloadable ringtones, auto-answer mode, DND and other sophisticated features ensure convenience and productivity. And the **snom 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 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), 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 ensuring high audio quality and low latency.

Technical Data snom 320

GENERAL FEATURES

- **Dimensions:** approx. 25 x 20 x 12 cm
- **Weight:** approx. 920 g
- **Certifications:** FCC Class B, CE Mark
- **Power consumption:** 2.1 – 2.4 watts

CONNECTORS

- **1 x LAN, 1 x PC:** RJ45 (Ethernet)
- **Power:** 5 V DC
- **Ethernet:** 2 x IEEE 802.3 10/100 Mbps switch
- **Power over Ethernet:** IEEE 802.3af, Class 1
- **Handset:** RJ-4P4C connector
- **Headset:** RJ-4P4C connector
- **Expansion Module:** Proprietary snom connector

USER INTERFACE

- 2 x 24 character, tiltable semi-graphical display with backlit
- 47 keys, 12 programmable function keys with LEDs (54 with the expansion module)
- Caller-ID
- Message waiting indication LED
- Address book (100 entries)
- Speed dialing
- Local dial plan
- Number guessing
- 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
- Handling of up to 12 calls simultaneously
- Menu-driven user interface
- Ring tone selection, import of individual ringtones
- URL Dialing support
- National language support for selected languages (NLS)
- Do not disturb
- Speakerphone (full duplex)
- Auto answer mode
- Keyboard lock

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 pickup (only via PBX)
- Call completion
- Client Matter Code (CMC)
- Call waiting/switching between calls
- Redialing
- RTP multicast paging
- Multiple audio device support

WEB SERVER

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

SECURITY, QUALITY OF SERVICE

- HTTPS (server/client)
- Transport Layer Security (TLS)
- SRTP (RFC3711), SIPS
- RTCP, S-RTCP
- VLAN (IEEE 802.1X)
- LLDP-MED

CODECS, AUDIO

- G.711 A-law, µ-law
- G.722, G.729A, G.723.1, G.726
- GSM 6.10 (full rate)
- Comfort noise, voice activity detection

SIP

- RFC3261 compliance
- UDP, TCP and TLS
- Digest/basic authentication
- PRACK (RFC3262)
- Error-information support
- 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)
- Dialog-state monitoring (RFC 4235)
- In-band DTMF/out-of-band DTMF/ SIP INFO DTMF
- STUN client, ICE (NAT traversal)
- ENUM (RFC3261), NAPTR (RFC2915), rport (RFC3581), REFER (RFC3515)
- Bridged line appearance (BLA)
- Auto provisioning with PnP
- Busy lamp field support (BLF)

INSTALLATION

- Automatic software update
- Automatic settings retrieval via HTTP/HTTPS/TFTP with authentication
- Installation via web interface
- Static IP provisioning, DHCP
- NTP