VoIP phones

VoIP phone snom 360

The Next Generation VoIP Phones

- tiltable graphical display (128x64 pixels)
- 47 keys, 13 LEDs
- 12 programmable function keys
- Speakerphone
- Dual Ethernet connection
- Power over Ethernet
- Headset connection
- Additional keypads with 42 programmable function keys
- SIP RFC3261
- Security: SIPS/SRTP, TLS
- STUN, ENUM, NAT, ICE
- Compression: G.723.1 and others
- National Language Support
- Asian Language Support

Usability

Security

Interoperability

The snom 360 was designed for maximum productivity and efficiency in the everyday business environment. Dedicated keys provide you with direct access to the functions for audio and call control, and context-sensitive menus offer you the additional functionality that you may need at any given moment. The graphical display can be tilted for optimum reading angle.

Additional sophisticated call control features, full call detail, configuration options can be accessed via browser from the attached PC.

Customized ring tones can easily be downloaded from the web server - including, of course, your favorite ring tone. Incoming calls can be marked with special ring tones to indicate the destination of the call.

12 programmable keys can be used to customize the functionalities according to your own specific needs. The LED associated to a function key shows you whether or not your colleague is currently on a call. And, of course, your colleagues can see whether your line is free or not.

The new mini browser, embedded in snom's top-of-the-line 360 executive SIP phone, lets users and developers create web-driven, screen-based telephone applications. Examples include custom contact-center apps, web-based phone directories, messaging, posting of news and other info on telephone screens, and more.

To spare you the annoyance of unwanted invasions of your speech data, the snom 360 supports the security standard SRTP - a current specification from the Internet Engineering Task Force (IETF) for protection against eavesdropping and the stealing of data.

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.

Technical Data

- **Dimensions:** approx. 25x 20 x 13.5 cm
- **Weight:** approx. 960 g
- **Safety:** IEC 60950-1:2001
  CB Test Certificate: DE 2-008417
- **Certifications:** FCC Class B, CE Mark Commercial

Connectors:

- **Network:** RJ45 (Ethernet)
- **PC:** RJ45 (Ethernet)
- **Power:** 5 V DC (stabilized)
- 2 port switch included (802.3 10/100 BT half duplex/full duplex with autosense)
- Power over LAN (IEEE 802.3af) on network port
- **Handset:** RJ11Connector
- **Headset:** RJ11 Connector
- **Extension Board:** Proprietary snom connector

User Interface

- Display: Graphical 128 x 64 pixels
- 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)
- Programmable function keys
- Menu-driven user interface
- Selectable ringing melodies
- National language support for selected languages (NLS)
- Asian language support (ALS)
- Downloadable ringing melodies
- URL Dialing support
- Do not disturb
- Speakerphone (Full Duplex)
- Auto answer mode
- UTF8-encoded Caller-ID
- Multi-Line registration
- 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)
- SIPS
- AES
- VLAN (802.1pq)

Codecs

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

SIP

- RFC3261 compliant
- UDP, TCP, TLS 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 client for configuration
- Automatic software update
- Completely automatic installation from web

For more information, contact your snom partner.

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