snom 360

The Next Generation of VoIP Phones

- **Tiltable graphical display (128 x 64 pixels)**
- **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**
- **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**
- **CTI via snom Flexor Manager**

The **snom 360** is 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 your optimum reading angle.

Additional sophisticated call control features, full call detail, configuration options can be accessed via web browser. 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 snom 360

GENERAL FEATURES

- **Dimensions:** approx. 25 x 20 x 13 cm
- **Weight:** approx. 960 g
- **Safety:** IEC 60950-1:2001, CB Test Certificate: DE 2-008417
- **Certifications:** FCC Class B, CE Mark
- **Power consumption:** 1.7 – 2.7 watts
- **Colors:** snomBLACK and snowWHITE

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:** RJ11 connector
- **Headset:** RJ11 connector
- **Expansion Module:** Proprietary snom connector

USER INTERFACE

- 128 x 64 pixels, tiltable graphical display, 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
- URL Dialing support
- Ring tone selection, import of individual ring tones
- 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
- VLAN (802.1 pq)

CODECS, AUDIO

- G.711 A-law, µ-law
- G.729A, G.723.1, G.726
- GSM 6.10 (full rate)
- G.722
- 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 (NAT traversal)
- ENUM (RFC3261), NAPTR (RFC2915), rport (RFC3581), REFER (RFC3515)
- Bridged line appearance (BLA)
- Auto provisioning with PnP
- Presence/buddylist feature
- Busy lamp field support (BLF)
- Presence publishing

INSTALLATION

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

snom technology AG

Charlottenstraße 68–71
10117 Berlin
tel/enum: +49 30 39833-0
fax: +49 30 39833-111
sip: info@snom.com
mail: info@snom.com

Version 1.4 / en / May 2009