ZAGMA5
PHONE SOLUTION NORTH AMERICA

snom 370

Innovative SIP based VoIP Phone

snom OCS edition

- **Tiltable high-definition graphical display (240 x 128 pixels)**
- **Call indication LED**
- **47 keys, 13 LEDs**
- **12 programmable function keys**
- **Speakerphone**
- **2 x IEEE 802.3 10/100 Mbps switch**
- **Power over Ethernet**
- **Multiple audio device support**
- **SIP RFC3261**
- **Security: SIPS/SRTP, TLS, IEEE 802.1X**
- **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**
- **OpenVPN inside**

The **snom 370** is the VoIP phone for the business user who needs immediate access to all of their important information.

With the large graphical, high-definition display, the **snom 370** offers an improved and extended presentation of call lists, address books and caller information. Caller information can be customized easily through XML to depict the information the user wants displayed.

Through the mini browser, users can have direct access to their own applications via the display screen. Users can not only customize the design of the display but, also view news tickers and other information as well as access central or public phone directories. The large LED (red) announces incoming calls.

More flexibility and productivity for your everyday work life. The improved interface guarantees ease of use and comfort. Context-sensitive menus offer additional options according to individual requirements. The **snom 370** supports several audio devices simultaneously, for example, it is possible to use the handset or headset and the loudspeaker concurrently.

More performance through expanded memory. The **snom 370** offers more additional customer-oriented functionalities and applications. The expanded memory capacity enables the depiction of graphics and high resolution pictures to show the status of contacts (e.g., busy, on-line, off-line) similar to Instant Messenger. It is also possible to play music and media files and to integrate applications like VPN for increased security.

Afraid of unwanted listeners, data theft or spam calls? To avoid problems with unwanted violations of your audio data, the **snom 370** supports the security standards TLS (Transport Layer Security), SRTP, and SIPS which are necessary to protect against electronic eavesdropping and data theft.

Technical Data snom 370

GENERAL FEATURES

- **Dimensions:** approx. 25 x 20 x 13.5 cm
- **Weight:** approx. 1000 g
- **Safety:** IEC 60950-1:2001, CB Test Certificate: DE 2-008417
- **Certifications:** FCC Class B, CE Mark
- **Power consumption:** 2.8 – 2.9 watts
- Ring tone selection, import of individual ring tones
- National language support for selected languages (NLS)
- Do not disturb
- Speakerphone
- Auto answer mode
- Keyboard lock

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

- 240 x 128 pixels, tiltable grayscale graphical display, backlit
- 47 keys, 12 programmable function keys with LEDs (54 with the expansion module)
- Call indication LED
- Caller-ID (with picture display)
- Message waiting indication LED
- Address book (250 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

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.729A, G.723.1, G.726, G.722
- 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
- Presence/buddylist feature
- Busy lamp field support (BLF)
- Presence publishing

INSTALLATION

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

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