snom
VoIP phones

snom 320
VoIP Business Phone

Manual

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Version 1.00

www.snom.com

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Foreword

Congratulations on the purchase of your snom 320 Voice over IP telephone!

Telephony is part of our everyday life. Over a period of more than 100 years, a form of communication has evolved which we feel cannot be ignored. In spite of the new technology base of the snom 320, most of its look and feel will be very familiar to you and you should be able to use it intuitively.

On the other hand, the world of the Internet has opened a whole range of new possibilities. Many users are using web browsers and own one or more e-mail accounts. They will find it easy to manage the phone via its web interface or to make a call to “sip:john@domain.de”, for example.

We are confident that developments in the computer industry will follow those in the telecom world.

VoIP is not only about transporting speech over data networks. It is about interoperability and breaking up a vertical market, as well as streamlining business processes by seamlessly integrating the telephone into computer networks and applications. With its technical flexibility, our commitment to all open and relevant standards and our cooperation with other vendors in the VoIP industry, the snom 320 represents a safe investment for the future.

We would like to take this opportunity to wish you a great experience in the VoIP world.

snom technology AG

Note to the reader

This manual describes the snom 320 running in administrator mode.
The phone supports SIP protocol only!
The default password to reach administrator mode is “0000” (four zeros).
The current version of this manual can be obtained from:

http://www.snom.com

Table of Contents

Installation

Delivery Content

Please check whether the delivery contains the following parts:

- The base unit with display and keypad
- The handset
- The handset cable
- The power supply (optional)

Safety Information

Please read the following safety notices before installing or using your snom phone. They are crucial for the safe and reliable operation of the device.

Power supply

You have two options for providing the snom 320 with power:

- An external power supply (5 V)
- Power feeding over the network cable (IEEE 802.3af compatible)

If you want to use an external power supply, use the one that is included in the package. Other power supplies may cause damage to the phone, affect its behavior or induce noise.

Setting up the Phone

Your snom 320 is delivered with the footstand attached to the phone's bottom shell (shaded gray in Fig. 1). Place the snom 320 on an even, horizontal surface that gives the rubber pads a secure grip. Do not place it on carpets or other materials containing fibers that could block the air vents and cause overheating.

After connecting the phone (see chapter *Connecting the Phone*, below), clip the cords into the appropriate slots on the footstand or, in the case of the handset cord, on the bottom shell of the phone (Fig. 2).

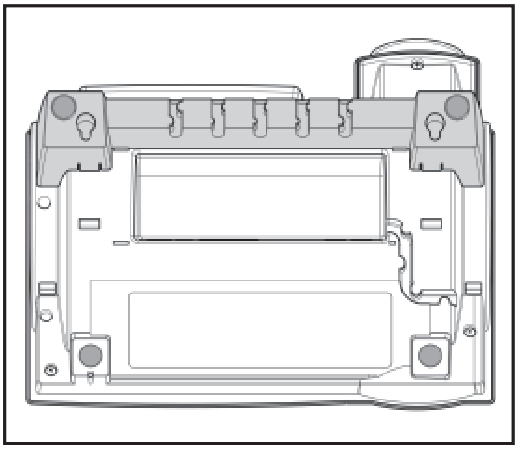


Fig. 1

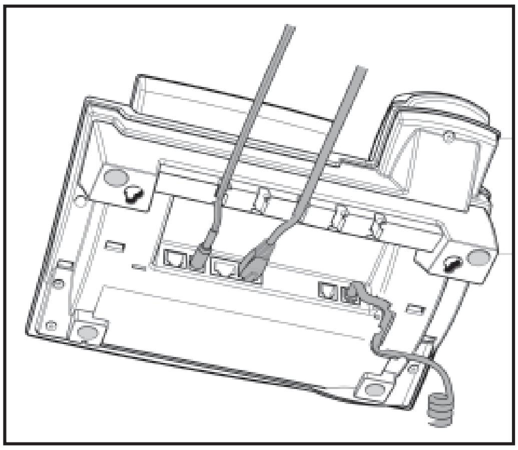


Fig. 2

Wall Mounting

It is also possible to mount the snom 320 on the wall. In order to do this, detach the footstand from the bottom shell and attach it in the appropriate position for wall mounting, as described in steps 1 through 6, below.

(1) Turn the phone upside down (Fig. 3). Please ensure that you do not damage the display and/or its hinge and that you do not drop the receiver. You may want to hold the phone in your lap or have a second person hold it.

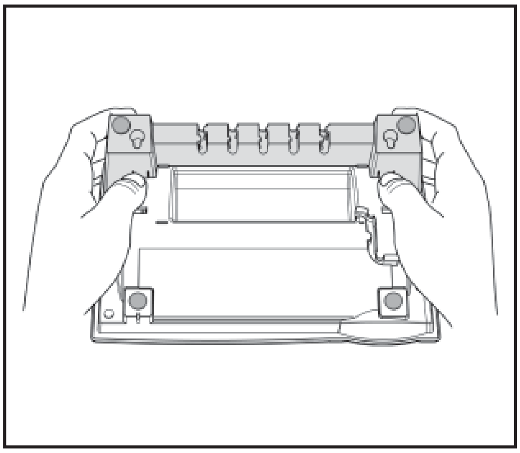


Fig. 3

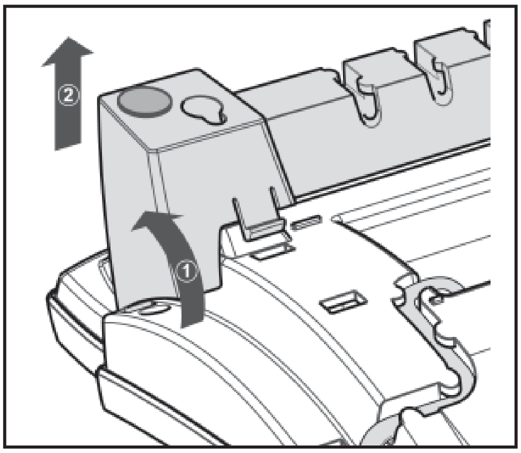


Fig. 4

(2) Lightly press the snap-fits of the footstand inwards (Fig. 3) and, at the same time, tilt the footstand backward and lift it from its anchorage in the mounting holes (Fig. 4).

(3) Rotate the footstand 180° around its vertical axis.

(4) Make sure that the handset cord has been clipped correctly into the groove on the bottom shell of the phone, as shown in Figs. 4 and 6. Failure to clip the cord into the groove will result in damage to the cord when the footstand is attached in the position for wall mounting.

(5) Insert the outside, rear snap-fits into the mounting holes on the bottom shell (Fig 5, arrow no. 1). Tilt the footstand forward and downward until the snap-fits snap into place in the mounting holes. **Please make sure that the four snap-fits have securely snapped into the holes before mounting the phone on the wall.**

1

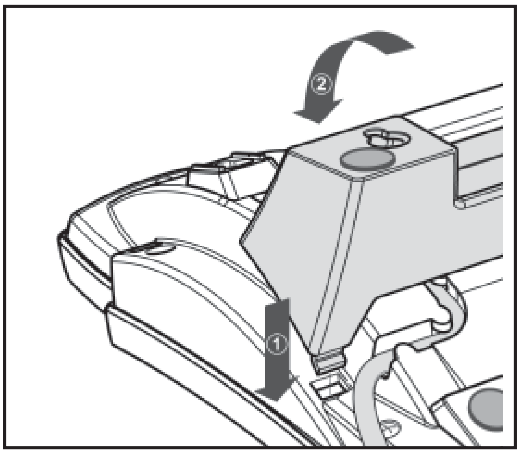


Fig. 5

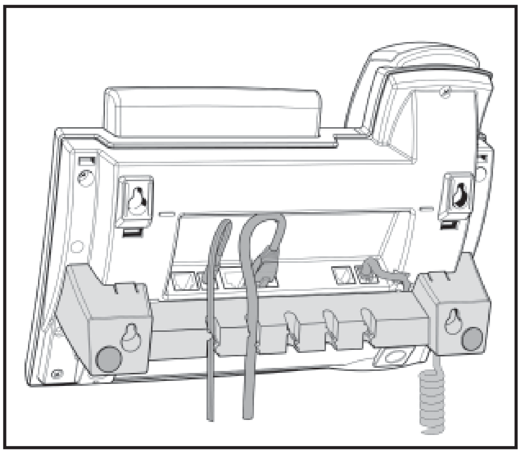


Fig. 6

(6) The slots on the footstand for the various cords can also be used when the phone is mounted on a wall. After connecting the phone (see chapter *Setting up the Phone*, above), form the cords into loops and gently press the long ends into the slots on the footstand (Fig. 6). If the loops are too big they will hamper the secure mounting of the phone on the wall; if they are too small they might bend or break the cords.

The delivery includes a template to be used for marking the position of the holes to be drilled in the wall.

(Use wall plugs, if necessary, and screws with half-round head profiles (diameters of screws dmax= 4.5 mm (0.1755”), diameters of heads dmax=8.5 mm (0.3315”).

The screws must protrude approx. 0.5 cm (0.195”) from the wall for easy wall mounting of the phone.

Adjusting the Handset Rest

The speaker of the handset has a small, rectangular indentation that fits over the rounded end of the plastic tab inserted into a slot on the top shell of the phone. When the phone is mounted on the wall, the handset must be placed on this rounded end to ensure that it will not fall off.

Slide the tab out of the slot, as shown in Fig. 7, rotate it 180°, and slide it back into the slot, as shown in Fig. 8.

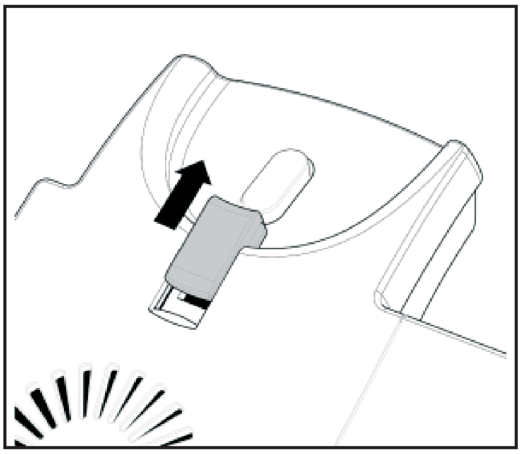


Fig. 7

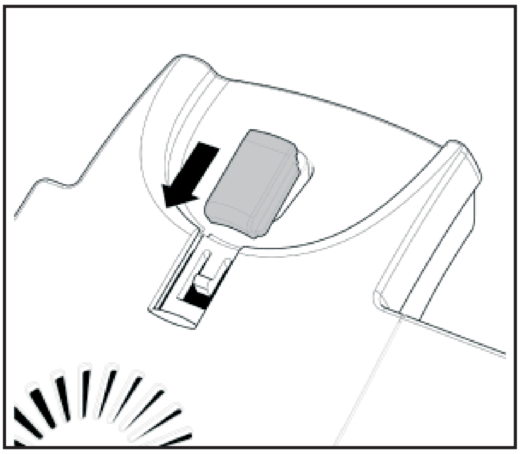


Fig. 8

Adjusting the Position of the Display

The position of the snom 320's display is adjustable up to an angle of 45° (Figs. 9 and 10).
Hold down the phone with one hand and adjust the display with your other hand. Do not use too much force and do not pull, wrench, or twist the display as this might damage or break the hinges.

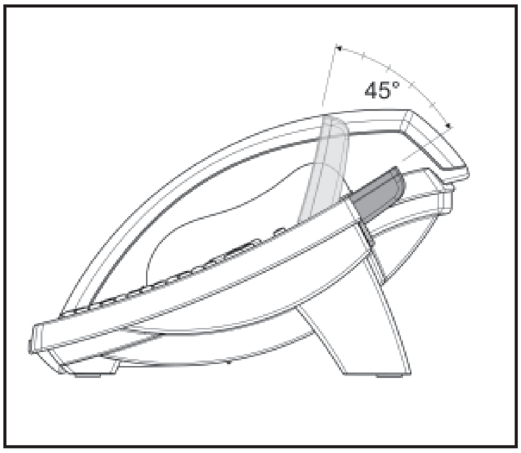


Fig. 9

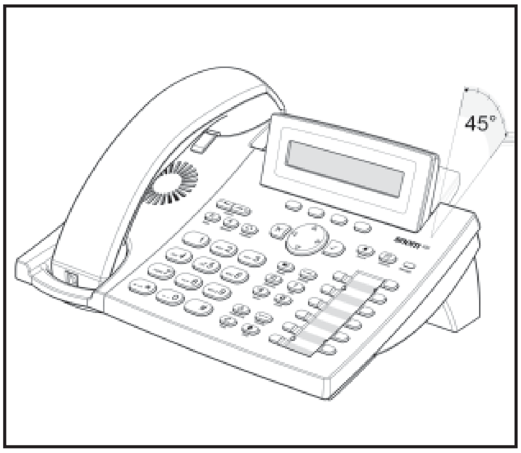


Fig. 10

Cleaning

To clean the snom 320, use an anti-static cloth. Please avoid cleaning liquids as they might damage the surface or the internal electronics of the phone.

Connecting the phone

First connect one end of the handset cable to the handset and then to the jack labeled “ ” on the left-hand side of the phone’s bottom.

Next plug the Ethernet (network) cable into the RJ45 connector labeled “NET”, and plug the other end into the network side to establish a data link. The second RJ45 connector, labeled “PC”, is for daisy-chaining further Ethernet devices without the need for a second Ethernet connection line.

If you are using an external power supply, please insert the plug of the power supply into the connector labeled “DC 5V” next to the data line and hook up the casing into the mains.

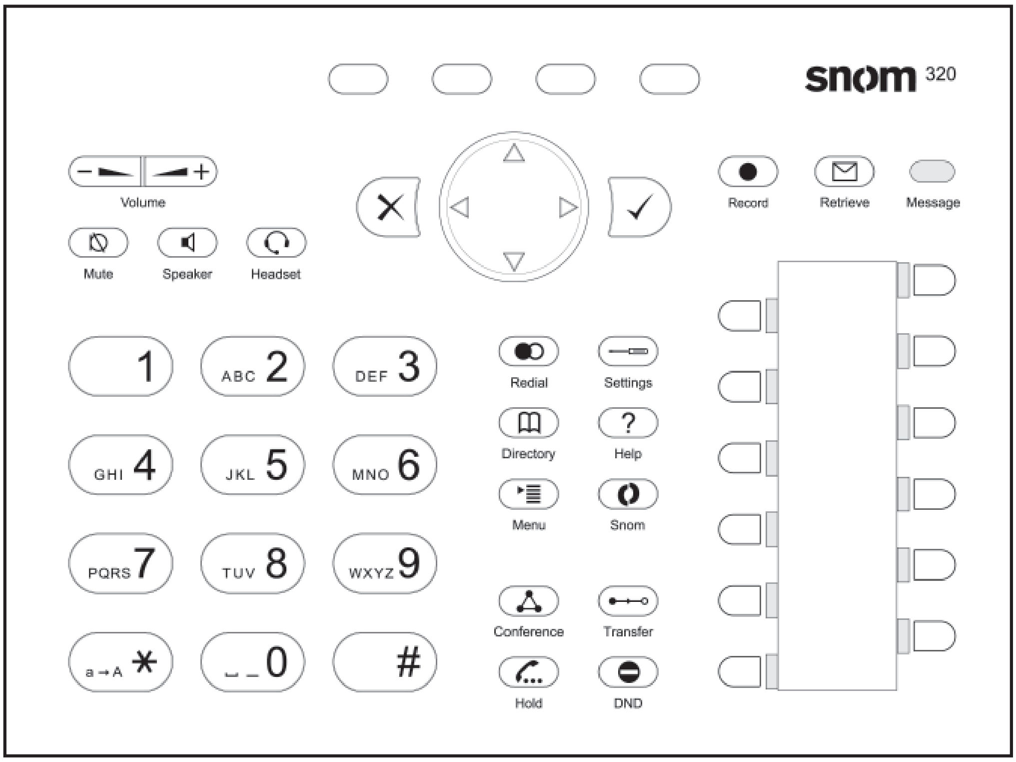
If you want to use a headset, it can be connected to the audio jack labeled “ ” on the bottom side of the snom 320.

Clip the cords into the appropriate slots on the footstand or, in the case of the handset cord, on the bottom shell of the phone (see Fig. 1 in the chapter *Setting up the phone*, above).

Keypad

The numeric keypad with the keys 0 to 9, *, and # is used to enter digits and letters. Depending on the operating mode, different actions can be performed (see the table below):

- Entry of digits only (e.g., when dialing a phone number)
- Typing in letters and digits by pressing the keys repeatedly (similar to a cellular phone), etc.



Use the MENU key to call up the menu. To cancel actions or input, use the CANCEL key. The ENTER key confirms actions, selections, and entries. For navigation, use the large round navigation key in the middle.

Depending on the operating mode, the keys can have context-specific meanings, which are described in the manual.

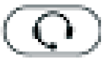
The four keys located below the display are context-sensitive function keys. Their current mapping is indicated by texts in the bottom line on the display.

The twelve keys on the right with LEDs / next to them are the programmable function keys onto which different functionalities can be mapped.

The keys of the numeric block in different operating modes:

Key	Digits	Lower case	Upper case
␣ 0	0	(SPC)_0	(SPC)_0
1	1	.@1,?!-/():;&%*#+<=>$[]	.@1,?!-/():;&%*#+<=>$[]
ABC 2	2	abc2	ABC2
DEF 3	3	def3	DEF3
GHI 4	4	ghi4	GHI4
JKL 5	5	jkl5	JKL5
MNO 6	6	mno6	MNO6
PQRS 7	7	pqrs7$	PQRS7$
TUV 8	8	tuv8	TUV8
WXYZ 9	9	wxyz9	WXYZ9
#	# or . after timeout, if not numeric	Number guessing	
a→A *	*	Toggles upper and lower case	

Additionally, the following keys are available:

Key	Description
	Mute microphone on/off
	Casing speaker on/off
	Headset on/off
	Adjust volume (lower/higher)
	Open phone book
	Help
	Menu
	Redial
	Settings menu
	XML Add-on (planned)
	Establish 3-party conference
	Do not disturb mode on/off
	Put call on hold/resume call
	Call transfer
	Record a call (with server support only!)
	Connect to Mailbox
	LED for MWI (Message waiting indication)

Initialization

Booting

The booting process comprises a series of different configuration steps that set up the phone for future use by any user.

Selecting the language

The default language setting is English. After startup, the phone offers you the option to select the language of your choice. To change to a different language, use the function keys beneath the right and left arrows or the navigation key / and press to confirm.

```
Select Language:
←        English        →
```

DHCP Configuration

If your network supports DHCP, press when this screen appears:

```
Are you using DHCP?
 *On                 Off
```

The phone has a built-in DHCP client. It will receive an IP address and other network-related settings (Netmask, IP gateway, DNS server) from the DHCP server.

If your network does not support DHCP, press when the above screen appears. The user will subsequently be asked for the following settings:

Setting the IP Address

The user can provide a static IP address for the phone if DHCP is not used. A valid IP address appropriate to the network in which the phone is being used can be provided.

```
IP Address:
192.168.139.212
```

The following edit functions for IP address editing are available via the navigation key:

- Clear IP address
- Move cursor to the right
- Backspace and delete sign to the left of the cursor
- Move cursor to the left
- Confirm input

Setting the Netmask

Similarly, the user will be asked to provide a Netmask. After entering the information, confirm with .

```
Netmask:
255.255.0  .0
```

Setting the IP Gateway

If a valid Netmask has been provided, the user will be asked for the IP address of the IP gateway. After entering the information, confirm with

```
IP Gateway:
192.168.0  .1
```

Setting the DNS Server

The last item for this series of network-related configurations is the IP address for the DNS server. After entering the information, confirm with .

```
DNS Server:
192.168.0  .9
```

Selecting the Tone Scheme

You will then be asked for the tone scheme to be used on the phone. Use the navigation key to move the cursor to the tone scheme you wish to use and confirm with .

```
Select Dialtone:
←        Australia       →
```

Selecting the Timezone

Select the timezone to be used on the phone by moving through the different timezone options available in the menu and pressing to confirm your selection.

```
Select Timezone:
←  -10: USA (Honolulu) →
```

If the settings explained above are set up properly, the phone will ask for the first account registration.

Logging on the first account

If no number is assigned to the phone yet, you will be prompted to type in your account name. Confirm your input with .

```
Account:                abc
█
```

This will be followed by the address of your registrar. The phone tries to guess the correct registrar, so the display could show something like this:

```
Registrar:              abc
intern.snom.de█
```

Confirm your input with ; you will then be asked if you want to use ENUM.

```
Use ENUM?
  On                   Off
```

Press if you wish to use ENUM or , if you do not. If you have chosen ENUM you will be asked for your country code (e.g., 1 for the US) and your area code (e.g., 802 for Vermont). ENUM means that a conventional E.164 number (normal phone number) is mapped to a SIP URI so that a pure IP call can be started instead of an IP/PSTN call. Regarding ENUM see also our white paper “ENUM on snom phones”!

```
Country Code:
█
```

```
Area Code:
█
```

After entering each code, confirm with .

The phone tries to register your given account name at the given registrar. The idle screen appears. Your registration has been successful if the identity you have just configured is shown in brackets in the top line. If, however, the letters „NR“ are shown instead, the registration has not been successful.

```
4/28 (447)        2 57PM
 DND PhoneBk Reg CallLog
```

When this state has been reached, the time and date are also shown on the display.

Logon

Mobility

In business office environments, desktops are sometimes shared by different employees at the same time or at different points in time (e.g., employees working different shifts, etc). The phones located on the desktops need to be able to cater to this.

To ease the effort of assigning the appropriate phone number to the phone for each individual employee, the snom 320 offers so-called mobility features, which are described in the following subchapters.

Logon wizard

If no number is assigned to the phone, you will be prompted to type in your account name. See chapter 1, subchapter *Logging on the first account,* for further information.

Registration Menu

```
5/19 (447)       11 12AM
 DND PhoneBk Reg CallLog
```

After pressing the function key “Reg” from the idle screen you can choose the registration line you wish to use.

```
1* 447@intern.snom.de  →
 Next              Edit
```

Press the function key “Next” to move through the up to twelve available registration lines.

Logon user

Suppose you want to log on a user 777 on the third registration line.

```
3:                          →
 Next                  Edit
```


Press “Edit” to start editing the registration line information. If you need to backspace during editing, use the nagigation key . First type in the user’s phone number:

```
Account:                 123
█
```

```
Account:                 123
777█
```

Confirm with and type in the registrar address:

```
Registrar:               123
█
```

```
Registrar:               123
intern.snom.de█
```

After you press , the entire registration line is shown again.

```
3: 777@intern.snom.de      →
 Next Activate          Edit
```

Logoff user

Choose the registration line where you want to logoff the user and press “Edit”.

```
3: 777@intern.snom.de   →
 Next  Activate      Edit
```

```
Account:              123
777█
```

Remove the current phone number with and confirm with . You will then see the registrar entry again. Confirm it with to leave it for later use.

Select Outgoing Line

Select the registration line you want to use as your outgoing identity for the next call by pressing “Next”.

```
1* 447@intern.snom.de   →
 Next                Edit
```

Then activate the selected registration line by pressing “Activate”.

```
2: 555@intern.snom.de   →
 Next  Activate      Edit
```

By choosing a different line, your originator phone number is changed to this line. This means that different people can start calls from the same phone with their own originator phone number, and the called phone will display the current outgoing line information. The current outgoing line is marked on the registration screen with an “*” directly in front of the registered SIP line.

```
2* 555@intern.snom.de  →
 Next              Edit
```

To change the outgoing identity from the idle screen, use the function key “Reg” to access the first registration, then move through the currently registered lines as described above.

Challenge/Authentication

In SIP, a user can set up authentication requirements for each registration on the phone. The password for each account name is set up via the web interface of the phone on the configuration pages for Lines 1-12, index card ‘Login Information’.

If the authentication password is not set, or is set wrongly, the account will not register on that domain and the phone will be challenged for that line. A challenge response will ask for the correct password. This may look like the following:

```
Password (intern.sno 123
█
```

The default mode for password input is integer. You can switch to alphanumeric input by pressing the asterisk key “*”. The password is replaced by a series of ‘*’s to keep it hidden from prying eyes.

```
Password (intern.sno 123
**█
```

Repeat this procedure for all accounts with authentication. The challenge responses are stored in the phone and will be used at re-registration or upon reboot. You can also edit this information via the web interface.

Basic Functions

Idle state

In idle state, the phone shows the date and time as well as the active outgoing identity (in brackets) in the top line of the display. The second line shows the functions currently mapped onto the function keys.

```
5/19 (447)      11 12AM
 DND PhoneBk Reg CallLog
```

Dialing

There are two methods to begin a call from the idle state of the snom 320. You can either dial with the handset resting in the cradle and then pick up the handset or press to use the speakerphone, or you can pick up the handset or press and then dial the number. In the latter case, press to indicate that the number is complete and dialing can commence.

The snom 320 shows the current communication status in the display.

```
448
                 Cancel
```

Input Modes

Depending on the context, various types of input are possible. The active input mode is indicated in the lower left corner of the display:

Key mapping for input:

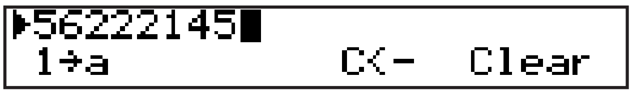
	Move cursor to the left
	Move cursor to the right
C<-	Delete the character to the left of the cursor
A→1	Change input mode to numbers
a→A	Change input mode to capital letters
1→a	Change input mode to small letters

3

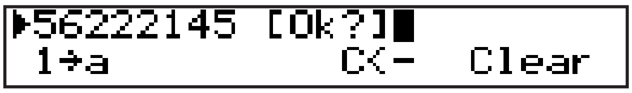
As mentioned in the table above, you may use "A->1", "a->A", and "1->a" to change the input mode. After using a specific input mode for dialing a number, the phone saves this mode as default for future use. If you discover during dialing that you want to use a different mode, press the left function key until the right mode is shown and continue dialing.

Dialing a phone number

Telephone numbers are dialed on the numeric keypad.



If the user has not pressed any keys for several seconds, the phone will remind the user to press in order to start the call.

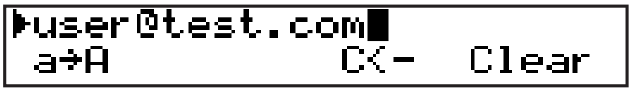


Key mapping:

	Move cursor to the left
	Move cursor to the right
C<-	Delete the character to the left of the cursor
A→1	Change input mode to numbers
a→A	Change input mode to capital letters
1→a	Change input mode to small letters
✓	Dial the number
✕	Abort the dialing

Dialing a SIP address

Enter a SIP address via the alphanumeric block. To find the ".", press the numeric key "1" in capital or small letter mode once, to find the "@" symbol, press it twice.



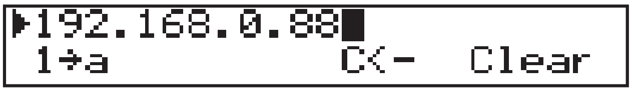
Key mapping:

See *Dialling a Phone Number*, above. ✓ dials the SIP address if entered correctly.

Dialing an IP address

Enter the IP address via the numeric block. Use the "*" key instead of the dot (".") or press the hash "#" key a little longer until the hash character changes to a dot!

If IP address dialing leads to an error message, make sure both of the phones involved are locally using port 5060 for SIP signaling (see option "network identity port")!



Key mapping:

See *Dialling a Phone Number*, above. dials the IP address if entered correctly.

TIP

Dialing an IP Address

When you are calling an IP address, the called phone or computer cannot determine which user you want to reach. It is assumed that "anonymous" is being called. This is the case with many phones and applications. Therefore, you should only use this method in exceptional cases, as the entry of IP addresses is rather cumbersome.

Number guessing

For your convenience, this functionality offers you the first number from dialed numbers, missed calls, received calls, or numbers in the address book that matches the beginning of the number you have begun to type in.

If the displayed number is not the one you want, you can either keep pressing the “#” key (in alphanumeric modes only) to get the next matching number, continue typing your desired number, or press to temporarily switch off number guessing for this session of editing a phone number.

Terminating a call

End calls by placing the handset on its cradle, pressing the hook switch, or pressing . The snom 320 will terminate the call and return to the idle state.

Incoming call

When your snom 320 is called, it rings and displays the following screen:

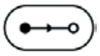
```
448
Deny All                  Deny
```

Accepting a call

Picking up the handset or pressing will accept the call. The snom 320 is now in a call.

```
448                       2:35
 Xfer   Mute            Cancel
```

Key mapping:

	Hold/Resume call
 / Xfer	Transfer call
	Handsfree mode on/off
	Mute/Unmute the microphone
	Change the volume

See below for further particulars.

3

Denying/blocking a call

When the snom 320 signals an incoming call, you have the option to deny the call with “Deny”.

If you press “Deny All”, the caller will be placed on the deny list and will always be denied automatically. If “Deny All” is not displayed, it is switched off in the settings. You can enable it via the web interface (“Deny All” feature).

Adjusting the volume

In idle state or while making a call, you can use the  keys to increase or lower the volume of the audio device (casing speaker, handset, headset) you are currently using.

```
Handset Volume:
    ██████████______
```

Key mapping:

	Reduce the volume in steps
	Increase the volume in steps
	Sets the volume and returns to idle state
	Returns to idle state without setting the volume.

Mute / Unmute

To mute the microphone during a call, press  or the function key “Mute”. Press  or the function key “Unmute” to enable the microphone again.

DND Mode on/off

The DND functionality is available in the snom menu. DND is short for “Do Not Disturb”. If this mode is activated with  , incoming calls will no longer come through to the phone and you will be completely undisturbed by the phone, except for VIP calls. If there is a mailbox set up for you, the call is redirected to it. Before this mode is activated, you will be asked if you really want to change to this mode.

```
Do not disturb?
  On                *Off
```

When DND is activated, “DND active!” is displayed in the top line of the idle screen.

```
5/19 DND                 2 52PM
 DND PhoneBk  Reg CallLog
```

Please keep in mind that the contact type “VIP” overrides the DND mode, i.e., a call from a number you have defined as contact type “VIP” will be put through to your phone even when DND is active.

Handset / Handsfree / Headset

The snom 320 supports calls made using the handset or the handsfree capability. During a call, it is possible to switch between the two modes by pressing the Speaker key . The small speaker symbol in the top line indicates speaker usage.

```
448                     0:03
 Xfer   Mute          Cancel
```

When a headset is connected and the audio output is set to headset (via or web interface), as indicated by the headset symbol in the top line, the snom 320 will treat the headset as the output device for calls.

```
448                     0:08
 Xfer   Mute          Cancel
```

When a connected headset is the default setting for speaking on the phone, which will, for example, normally be the case in a call center environment, the key is used to accept incoming calls.

Key mapping:

	Switch to handsfree mode and back
	Switch to headset mode and back
	Accept incoming calls in headset mode
	Return phone to idle state

Programmable Keys

The twelve keys P1 - P12 to the right of the number block are the programmable function keys, which have the following options:

Line
Destination
Intercom
Park Orbit
Voice Recorder
Shared Line
DTMF

Each of these keys has an LED indicating the current status of the respective programmable key.

Some features are common to all keys:

- Pressing a key when its LED is blinking and the phone is ringing accepts the call.
- Pressing a key when a call is in progress at that key, as indicated by the constantly lit LED, puts the call on hold.

- Pressing a key twice while editing a number for dialing at that key, with the receiver in the cradle, returns the phone to idle state.
- Pressing a key twice while editing a number for dialing at that key, with the receiver off the hook, erases the number.

Each of these different options will now be explained in detail.

Line

‚Line' can be used in one of the following ways:

a. To map a local SIP line: A user can assign the local lines to programmable keys by selecting this option and setting the URL of the local line as argument to that key setting. For example, if a phone has two registrations, 501@my.proxy.com and 502@my.proxy.com, the user has the option to map two programmable keys to each one of these lines by selecting ‚Line' and setting the respective SIP URL as argument. In this case, all the calls to a particular line will go to its matching programmable key, e.g., if 502@my.proxy.com is mapped at key P2, the LED on that key will start blinking when there is an incoming call on that line. Similarly, if the user presses P2 in the idle state with the receiver on hook, 502@my.proxy.com will become the active line for that call. This feature enables the customer to use his different SIP accounts as he would use different PSTN phone lines. It is also possible to assign different ring tones to each SIP line in order to make an acoustic differentiation. This can be done either on the proxy or on the phone's web interface page Configuration Line 1-12.

b. To map a SIP URL for call pickup: Selecting this option with a SIP URL as argument will subscribe to dialog state changes of the phone with that registration. The LED on that key will show the status of the registration as idle, talking, or ringing by varying frequencies of blinking. This allows the user to pick up a ringing call remotely by simply pressing a key. An example of its usage would be the mapping of the office reception phone line to the phone of a secretary. If, for some reason, a ringing phone is not answered at the reception, the secretary can see its ringing status by the blinking LED and pick up the call by pressing that programmable key. This way, no calls go unanswered.

c. Free Key: ‚Line‘ is also the default setting for the programmable keys. If no argument is set, the keys are treated as free. Outgoing and incoming calls not bound to any other key go to the first such key that is not already occupied.

Destination

The user can map a SIP URL to a particular programmable key by setting this option and providing the URL as argument. This option can be used in the following ways:

a. If the sip line 505@my.proxy.com is bound to key P3 with this option, all calls coming to the phone from this number will go to P3.

b. If P3 is pressed during the idle state, 505@my.proxy.com will be dialed, as it is set as destination for this key.

c. In the ringing state, if the call comes from any line other than 505@my.proxy.com, pressing P3 will transfer the incoming call to 505@my.proxy.com.

Intercom

This option is similar to ‚Destination‘, with the exception that pressing the key bound to ‚Intercom‘ enables the intercom mode, and the phone will be directly connected to the set snom phone if authentication is set up properly. This feature is useful in an office environment as a quick access key to connect to the operator or the secretary.

Park orbit

snom4s (snoms iPBX solution) provides its customers with the opportunity to set up parking orbits on the media server, where calls can be parked and picked up. The option ‚Park Orbit' enables the phone to provide this feature.

Suppose key P4 is bound to orbit1@my.proxy.com: The LED on this key now displays the status of calls, if any, that are parked on this orbit. A blinking LED indicates that a call is parked there; to pick up this call, press P4. Pressing P4 during a call will park it at orbit1@my.proxy.com until the same or another user picks it up later. The caller will hear the holding music. This feature is useful for call center environments and all places where there is a great inflow of calls and some kind of queuing is required to manage them.

Voice recorder

This option can be set up with a valid voice recording account. Suppose that vr@my.proxy.com offers voice recording and is bound to key P5 on the phone. Its usage is as follows:

During a call, the user is able to record his conversation with the other party by pressing P5. Pressing P5 again will end the recording process. The recorded media can later be listened to by accessing the recorder account vr@my.proxy.com.

This feature is also useful for recording short messages or memos to self. Press P5 in the idle state and record a message you wish to listen to later. The same applies to the recording of a debate or discussion, to keep audio minutes of a meeting, or to record a conference call hosted on the phone.

Shared Line

The snom 320 offers users the option of sharing a single registration among multiple phones. This can be the case for a single outgoing identity for a SOHO environment using an ISDN/PSTN connection to the outside world, or when sharing a single line amongst multiple users in a localized office environment. In each case, all the phones sharing the same line will register each line independently and then have the option to map that particular SIP line as a 'Shared Line' on their programmable keys. The status of the LED will reflect the status of the particular line and users are able to pick up calls on the same line that have been put on hold by their colleagues. Regular users of remote pickup will find this feature very helpful. (See also our white paper "Key System Setup"!)

DTMF

This option allows the specification of arbitrary key sequences, which will be sent via DTMF when this button is pressed.

Menus

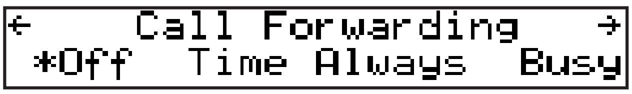
Press the menu key  to call up the menu. The below described submenus are available. Use the navigation key to scroll from one menu to the next; use the four function keys underneath the display to select an item in the bottom line of the display.

Key mapping:

	Move to the previous menu
	Move to the next menu
	Return phone to idle state

Call Forwarding

In this menu, an administrator can set up the options for call forwarding.

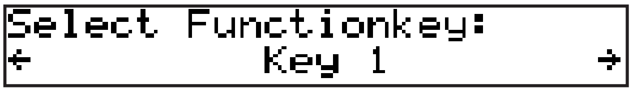


See *Call Diversion* in the chapter *Advanced Functions*, below, for a detailed description.

Phone Behavior

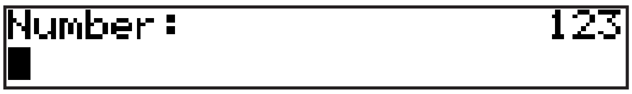
Press “CWI” to enter the submenu where you can switch Call Waiting Indication (CWI) on or off by pressing the corresponding function key.

Press “FKeys” to enter the selection menu for the mapping of the function keys.



Use the navigator key / to move to the desired function key and press .

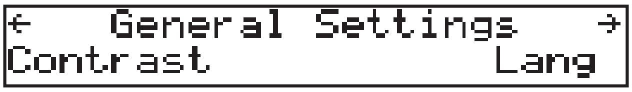
Use the navigator key / to move to the desired function you want to map onto the selected key and press .



Enter the number to which the key should be mapped.

General Settings

In this submenu, the display contrast can be adjusted and a display language selected.



Press the function key underneath “Contrast” to enter the submenu.

Contrast

The current contrast setting is displayed. Use the  keys to increase or decrease the contrast in steps.

Key mapping:

	Reduce the contrast in steps
	Increase the contrast in steps
	Confirm new contrast setting and return to idle state
	Return to idle state without setting the contrast

Lang

Select the language of your choice, as explained in the chapter *Initialization*, above.

Headset Device

```
←    Headset Device    →
*None            RJ Conn
```

Select whether you want to enable the headset or not.

Volume Settings

```
←    Volume Settings   →
 Hand  Head      Speaker
```

Here, the volume of the handset speaker, headset speaker, and casing speaker can be adjusted. Select one of them.

A tone is played at the currently set volume. Use the  keys to increase or decrease the volume in steps.

	Reduce the volume in steps
	Increase the volume in steps
	Set volume and return to idle state
	Return to idle state without setting the volume

Other Settings

```
←    Other Settings    →
TimeZone        DialTone
```

In the subsequent submenus you can set up the timezone and the tone schemes for the phone, as explained in the chapter *Initialization*, above.

Web Interface

```
←       Web Interface      →
Server          HTTP HTTPS
```

In the subsequent submenus you can set up whether the web interface should be accessible via HTTP only, HTTPS only, or via HTTP and HTTPS. The appropriate ports for both protocols can also be set here.

VLAN Settings

```
←       VLAN Settings      →
  ID  Priority     Reset
```

The ID (0..4095) (80.2.1q) and priority (0..7) (802.1p) values for VLAN can be set here. “Reset” removes both settings from the phone.

System Info

```
←        System Info       →
Network              Memory
```

Enter the respective submenus to view the Network status (number of packets sent/received) and the total and free memory in kB.

SW Update

When a new software version for your phone is available for download, an additional menu item “SW Update” can be found in the menu.

```
←         SW Update        →
          Ok
```

Pressing “Ok” or ✓ reboots your phone after a few seconds. During boot-up, the phone asks you whether or not it should proceed with the firmware update. If the phone doesn’t reboot by itself, please start the reboot manually.

Information Menu

Press ? to get into this menu.

```
      Information
IPAdr  MAC       Version
```

Here, you can look up the IP and MAC addresses and the software version of the phone.

IPAdr: Press the function key “IPAdr” to view the IP address currently assigned to the phone.

```
IP Address:
192.168.200.239
```

MAC: Press the function key “MAC” to view the MAC address of the phone.

```
MAC:
000413240005
```

Version: Press the function key “Version” to view the firmware version currently installed in the phone.

```
Version:
snom320-SIP 3.60i 6394
```

Maintenance Menu

Press to get into this menu where you can change the IP configuration of the phone and perform other maintenance like resets and reboots.

```
        Configuration
Reset Reboot  DHCP Reg
```

Reset: Press the function key "Reset" to clear all current settings of the phone and restore the default factory settings. This should only be used with the utmost care, which is why a second screen appears, requiring the entry of the administrator password. Press to confirm your password or to return to the menu.

If the correct administrator password has been entered, the phone will ask for a reboot. The default password is "0000".

```
Admin Mode Pwd         123
**█
```

Reboot: Press the function key "Reboot" to restart the phone without unplugging the power cable. This is called a "Soft Restart". This operation also requires confirmation.

```
Reboot?
                    Cancel
```

DHCP: Press the function key "DHCP" to reach this submenu.

```
Are you using DHCP?
 *On                   Off
```

If your network supports DHCP, press the function key "On". The phone has a built-in DHCP client. It will receive an IP address and other Network-related settings (Netmask, IP gateway, DNS server) from the DHCP server.

If your network does not support DHCP, press the function key "Off". You will then be asked for the basic network settings as explained in the chapter "*Initialization*", above.

Reg: See the subsection "*Registration Menu*" in the chapter "*Logon*", above, for a description of this option.

Advanced Functions

Hold and resume

When a call is in progress, pressing the hold key puts the call on hold, i.e., neither party can hear the other.

```
Calls on Hold: 1
444
```

Pressing again will resume the call. While a call is on hold, you can establish another call by dialing the desired number. When more than one call is on hold, the following window appears:

```
Calls on Hold: 2
474
```

Select the party you wish to talk to with or . Press to be connected.

Key mapping:

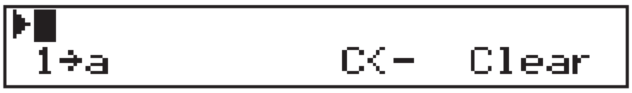
	Terminate the call being held
	Resume the call being held
*, #, digits	Can be used to initiate another call
	Handsfree mode on/off
	Headset on/off

4

Transfer

Direct Transfer

During a call, press the transfer key or Xfer to put the connected party on hold, then dial the number to which the call is to be directly transferred.



As soon as you press , the snom 320 will transfer the held party to that number.

Key mapping:

	Move the cursor to the left
	Move the cursor to the right
C<-	Delete the character to the left of the cursor
a→A	Change input mode to capital letters
1→a	Change input mode to small letters
A→1	Change input mode to numbers
	Transfer the call
	Abort the transfer

Consultation Transfer

During a call in progress, put the connected party on hold by pressing the hold key .

While the call is on hold, you can establish a second call by dialing the desired number and pressing . When the second call is established, you can consult the second party, e.g., to announce the call. You can connect the two parties by pressing of Xfer or place the handset on the hook. This behavior is available only when the option "Call join on Xfer" on the "advanced" web page is set to "on". When it is not, select the party you want the call to transfer to by pressing or , followed by the transfer keys or Xfer.

Conference

If the phone is connected with two calls, one on hold and one active, you can connect all three phones in a conference by pressing or "Cnf.On".

```
474                    0:19
 Xfer   Mute         Cnf.On
```

```
448                    0:48
 Xfer   Mute        Cnf.Off
```

By pressing again or by pressing "Cnf.Off", the conference is disconnected and the calls are all put on hold.

Call Diversion

All kinds of call diversions can be set, changed, and deactivated in the menu item "Call Forwarding". Press to access it. The currently active setting is preceded by an asterisk "*".

```
←     Call Forwarding     →
 *Off   Time Always   Busy
```

Diverting all calls

By setting the option "Menu / Call Forwarding" to "Always", every incoming call is diverted immediately to the number set in the following window "Redirect target" without the phone ringing.

To disable this feature, set "Menu / Call Forwarding" to "Off".

Divert when Busy

Set the option “Menu / Call Forwarding” to “Busy” if you want to divert all incoming calls to the number set in “Redirect target” when another call is already in progress.

Divert when not answered

Enter the number of seconds in “Menu / Call Forwarding / Time” after which every incoming unanswered call is diverted to the number set in “Redirect target”.

Note: To activate call divert immediately, see “Diverting all calls”, above.

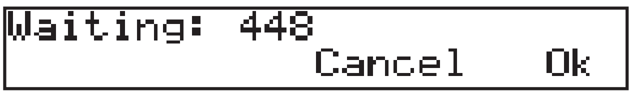
Call completion

The call completion functionality allows the user to establish calls successfully when the other party is busy or not answering. This feature can be enabled via the menu item “Phone Behavior / Call Completion”.

On busy (CCBS)

When the party you are trying to reach is busy, the following window will appear during the call attempt:

Activate call completion by pressing ; the phone then waits for the other party to return to idle.



If you wish to receive and make calls while you are waiting for call completion, press ✓ to return to the idle screen, which now shows the letters 'CC' to the left of the date in the upper left corner. Pressing ✕ cancels the call completion procedure.

```
5/20 CC                2:26PM
 DND PhoneBk  Reg CallLog
```

When the other party becomes idle, the following window appears:

```
Dial: 448?
                  Cancel    Ok
```


If you press "OK" or ✓ , the phone will dial the number again in order to establish a connection to the other party.

On no response (CCNR)

When the phone rings and nobody answers your call, activate call completion by pressing the 'CC' function key.

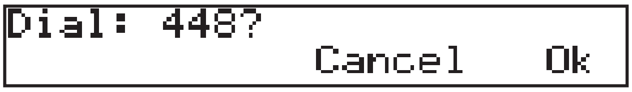
```
448
                     CC    Cancel
```

The phone will now wait for the dial destination to become active again.

```
Waiting: 448
                  Cancel    Ok
```

If you wish to receive and make calls while you are waiting for your call completion, press ✓ to return to the idle screen, which now shows the letters 'CC' to the left of the date in the upper left corner. Pressing cancels the call completion procedure.

When the other side is available again, the display will prompt you to redial the number.



Be aware that the activity detection only detects whether the phone you are calling is being used. It cannot detect whether the person you want to reach is in the room or not!

DTMF Tones

During a call, e.g., with a voicemail system, pressing the digits 0-9, *, or # will generate and send DTMF tones to the other party. The phone supports in-band and out-of-band DTMF functionality. It prefers out-of-band DTMF, but, if the other party does not support it, the phone falls back to in-band DTMF. This standard phone behavior cannot be changed.

Short Messages (SMS)

Incoming short messages are automatically indicated by a small letter symbol on the display (when the phone is in idle state), a blinking message LED, and, if "MWI Notification" is set to "beep/reminder", with a short beep (Menu item "Preferences / MWI Notification").

To display the message, press either or the SMS key. A message could look like this:

```
sipsak: usrloc test mess
age from SIPsak for user
```

Return to the idle screen with .

Message Waiting Indication (MWI)

If you did not answer a call and the mailbox recorded a message for you, the idle screen shows that a recorded message is waiting for you.

```
5/20 (447)         2 48PM
 DND  VMail Reg  CallLog
```


The text above the second function key from the left now says “VMail”. Press this function key or to connect to your mailbox and listen to the recorded message(s).

If you want to connect to your mailbox independently when there is no “new message” alert, press . This will only work, however, if you have entered the correct number of the mailbox on the Login index card of the web page for the corresponding line (Configuration Line 1-12)!

Conducting a Software Update

When a new software version for your phone is available for download, you will see the text “SW” to the right of the date in the top line of the screen.

```
5/20 SW            3:16PM
 DND PhoneBk Reg CallLog
```

In this case, an additional menu item “SW Update” can be found in the menu by pressing and using the navigation key to scroll to it.

Pressing “Ok” or reboots your phone after a few seconds. During boot-up, the phone asks you whether or not it should proceed with the firmware update. If the phone doesn’t reboot by itself, please start the reboot manually.

4

Call Register

Phone book

The snom 320 contains an internal phone book that can be accessed from the phone's idle state by pressing or the "PhoneBk" function key.

```
5/20 (447)           4 05PM
 DND PhoneBk Reg CallLog
```

You will then see the first entry:

```
Kate Wilson
Details       Edit Clear
```

Key mapping:

	Scroll to the next item
	Scroll to the previous item
Details	Toggles between displaying name and number
Edit	Enter edit mode for the selected entry
Clear	Delete current item
/ off hook	Dial the number of the selected entry
	Return phone to idle state

5

Adding an entry

In order to add a new entry, select the list item <New item> and press “Edit”.

```
<New Item>
             Edit
```

Enter the name and phone number and confirm each item by pressing .

```
Edit Name:          abc
Kate Wilson█
```

```
Edit Number:        123
448█
```

The title of the window indicates which input is expected. Pressing aborts the addition of a new entry. Up to 100 entries can be placed in the phone book.

Key mapping:

	Change the input mode
	Backspace
	Confirm the entry
	Cancel the entry

Editing an entry

After pressing the “Edit” key on the entry to be modified, the name and phone number will be brought up in sequence. Confirm each item by pressing .

The title of the window indicates which input is expected. Press to confirm the data; press to abort the editing.

Key mapping: See *Adding an entry, above.*

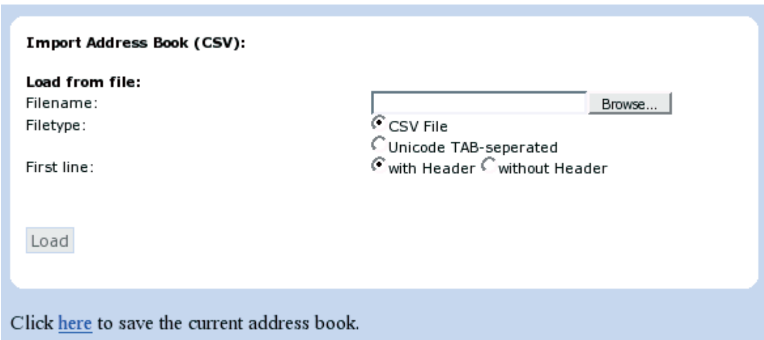
Please keep in mind that the contact type "VIP" overrides the DND mode, i.e. calls from someone on your "VIP" list will be put through to your phone even when DND is 'active'!

Delete whole address book

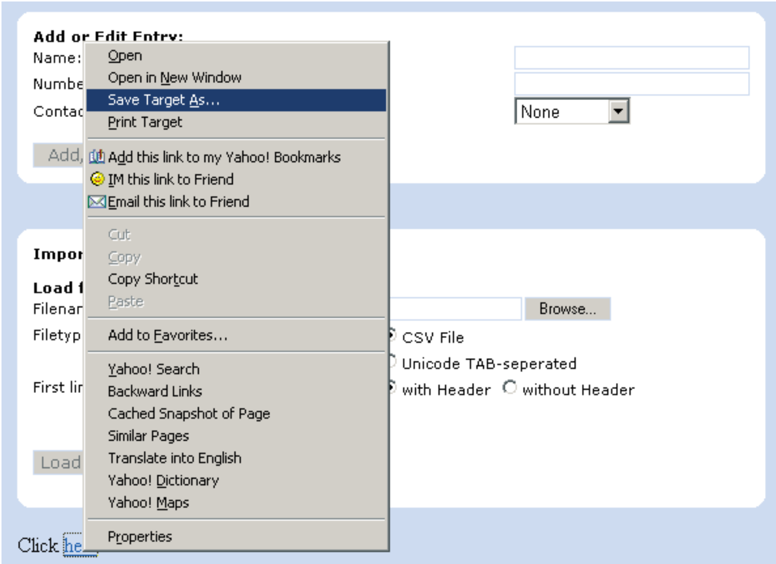
If you want to delete the entire address book, click onto the "Delete" button on the web interface page of the address book .

Export

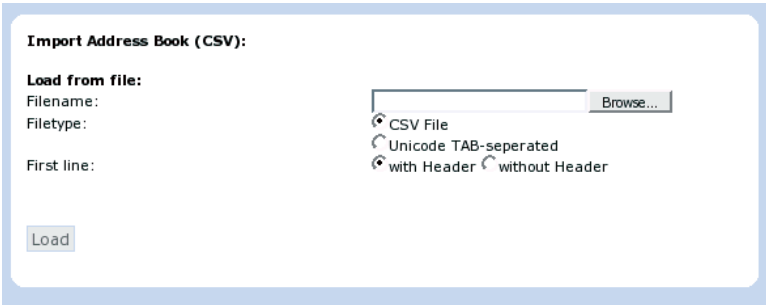
To save the content from a filled phonebook, right-click onto the the "Click here" link at the bottom of the "Addressbook" page on the phone's web interface.



With "Save Target As..." you can save the phone book content in a comma-separated file containing the current address book entries (CSV = Comma-Separated Values).



Import



To load an Address book from a file, click on the Browse button on the page Phone / Phonebook in the section entitled “Import Address Book (CSV)”. Select the pre-stored csv file from the storage medium and then press the ‘Load’ button.

5

It does not matter if some of the entries already exist in the phone book. They will not be duplicated if both name and number are exactly the same. If the number field is the same but the name is not, the file entries will replace the old ones (just like editing the old entry to save a different name). If the name is the same but the number is not, a new entry will be made.

Preview

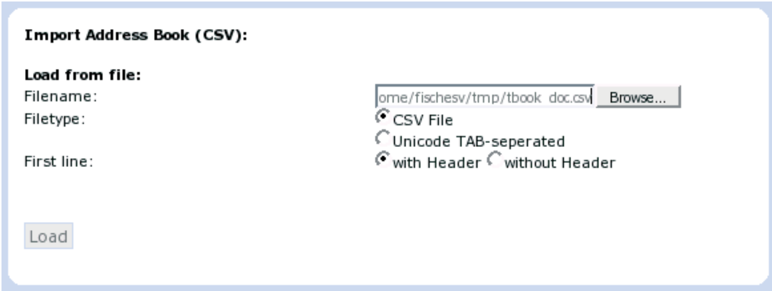
A preview feature is available while importing files for the phone book. One way to do this is to use a Comma-Separated Values (CSV) file (perhaps one you previously exported from a phone). A CSV file has entries separated by a comma (and without spaces between the comma and the preceding and following entry). It can look like this:

```
Name,ZIP,City,Street,Phone number,Type
Kate,123243,New York,21 Street,278139232,family
George,52765,Tokyo,Mainroad,13153243,friends
Harry,10364,Berlin,Pankestr,112984382,colleagues
```

The first line represents the header, which shouldn't be imported, which is why the option "without header" has to be selected in this case.

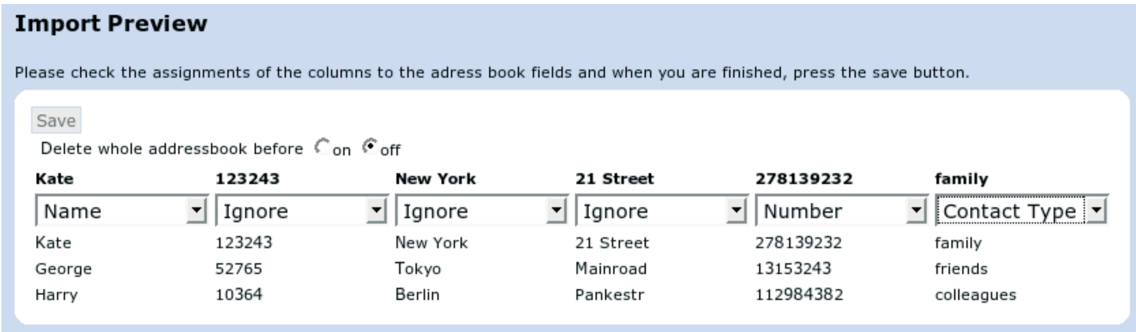
5

Selecting a CSV or Unicode TAB-separated file is possible through the web interface, as shown in the following:

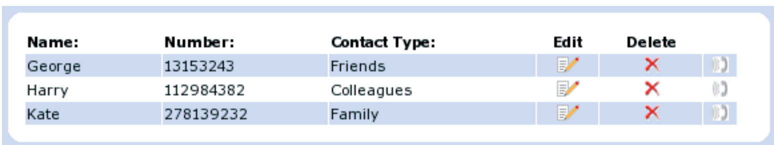


After choosing a file, a preview of the selected file contents will be shown.

Now assign the three possible data types (name, number, contact type) that can be imported to the columns in the preview representing those kinds of data.



Once satisfied with the preview, press “Save” to save the file contents to the phone book, which could look like this:



These new entries will become part of the existing address book. They will be stored on the flash and can be retrieved anytime until they are deleted or a factory reset is performed.

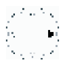
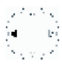
Call Lists

The snom 320 maintains lists of missed, received, and dialed calls that can be accessed by pressing the call list key "CallLog" from the idle state. Each list can contain up to 100 entries.

```
Missed: 1          4:46PM
 DND PhoneBk Reg CallLog
```

```
Select List:
Missed Dialed   Received
```

Key mapping:

	Scroll to the next call
	Scroll to the previous call
Details	Show details of this call
Clear	Delete the current entry
 / off hook	Dial the number of the current entry
	Return phone to idle state

For example, after choosing the missed calls, the most recent missed call is displayed first.

```
4:46PM▶ Kate Wilson
Details           Clear
```


5

Press “Details” to have call details about this missed call displayed:

```
To: sip:447@192.168.20 →
 Edit  Save       Clear
```

Scroll through the “Details” with or . The details shown are “To”, “From”, “Time”, and “Missed” (number of missed calls from the same phone number), or the duration of the call in the case of dialed and received calls.

```
From: Kate Wilson      →
 Edit  Save       Clear
```

```
Time:  4:46PM          →
 Edit  Save       Clear
```

```
Missed: 1              →
 Edit  Save       Clear
```


Key mapping:

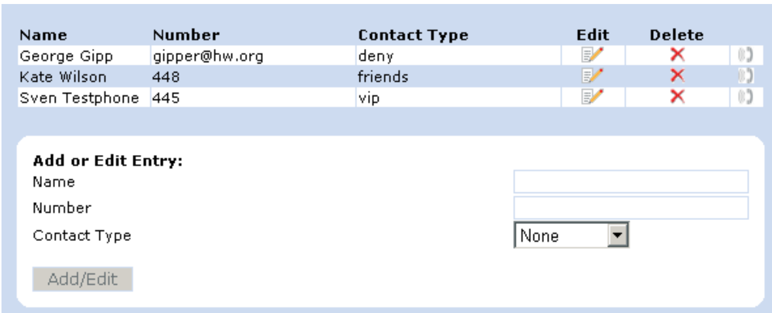
Edit	You can use the current entry for your next call and possibly edit the number before calling it
Save	Save the current calling party to the phone book
Clear	Delete the current entry
/ off hook	Dial the number of the current entry
	Return the phone to the idle state

Deny List

The snom 320 gives you the option of putting telephone numbers on a deny list to prevent incoming calls from these numbers.

Browsing

To look at the numbers currently on your deny list, go to the web interface Address Book page.



The numbers to be denied are listed as contact type "deny".

Adding a number

While you are being called, you can press the "Deny all" function key if this feature is activated on the webpage "Advanced Settings". This will not only deny the present call but will also add the number to the deny list. Pressing "Deny" on an incoming call will only deny that particular call and will not put the phone number on the Deny List.

In addition, any number can be added to the deny list by entering the phone number in the Address Book and selecting the contact type "Deny list".

Removing a number

There are two ways to remove a number from the deny list: (1) On the "Address Book" webpage, click onto the small red crossed icon at the end of the line to delete that particular entry. (2) Change the contact type on the "Address Book" webpage.

Speed Dial

The snom 320 supports speed dial of up to 33 numbers. These are mapped onto the numbers 0-30 and the * and # keys. Set up the speed dial numbers on the built-in web interface page of your phone (see next chapter *Settings,* subchapter *Settings via Web Browser*, *Setup Speed Dial*).

Dialing

Speed dialing is initiated by typing in a number from 0-30, *, or #, respectively, and confirming with ✓. In this way, 33 speed dial numbers can be called up without having to look at the display.

Settings

A long list of different settings is available, which can be used to control the behavior of the snom 320. These settings are explained in the white paper "Configuring snom phones for Mass Deployment", which you will find at http://www.snom.com. The white paper "How can I update a snom phone?" describes the different options available to update the phone, also via settings file.

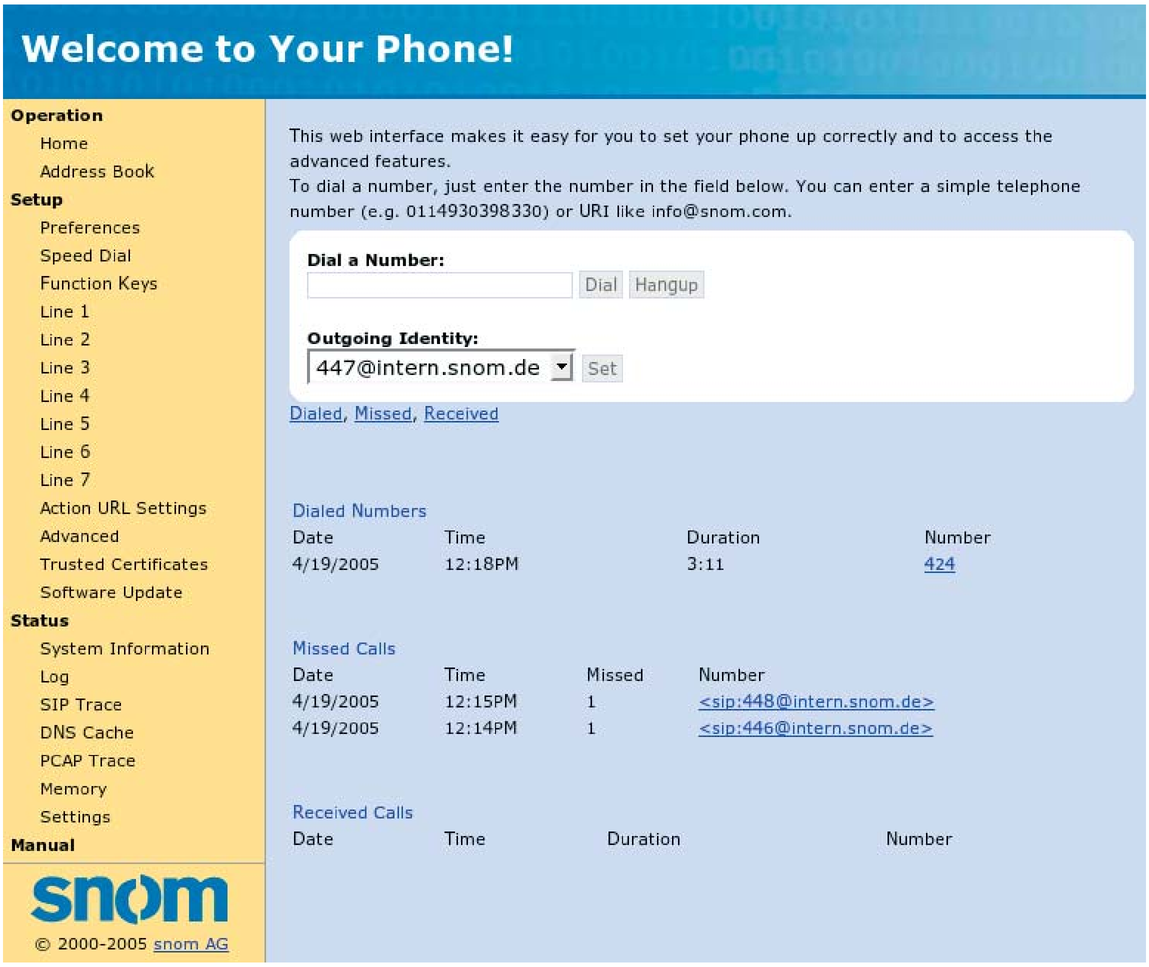
You can specify and change the settings via the phone or the phone's webpage.

Settings via Phone

For the settings that can be adjusted via the phone, see the section *Menu* in the chapter *Basic Functions*, above. Additional settings can be adjusted only via the phone's webpage which is usually quicker and more convenient to use anyway.

Settings via Web Browser

The snom 320 has an integrated webpage for this purpose. If the phone is connected to a network that provides DHCP, it can be accessed via the browser immediately after boot-up. If you do not want to use DHCP, you must specify IP address, netmask, gateway, DNS domain, and DNS server statically to ensure correct operation.



Start your web browser.

Enter the IP address of the phone as the URL (e.g. 192.168.0.100). If you do not know the IP address, you can look it up on the phone's display by pressing the ? key.

There is a selection menu on the left side of the web page.

Click on the desired submenu; the current settings of this submenu will be displayed in the larger blue field on the right.

You can now modify and store the values by using mouse and keyboard. To save the changes, click on the **SAVE** button. Do NOT press SAVE if you want to discard the changes.

Settings Options

The following options are available via web interface when the phone is operating in Administrator mode.

Setup Preferences

GENERAL INFORMATION

The webpage subsection may look like this:

General Information:	
Webinterface Language:	English
Language:	English
Number Display Style:	Name
Tone Scheme:	United States
MWI Notification:	Silent
MWI Dial Tone:	Stutter
Use Headset Device:	None
U.S. date format (mm/dd):	(•) on ( ) off
24 Hour clock:	( ) on (•) off
Use Flash Plugin	(•) on ( ) off

Webinterface Language: Your phone is able to show all display texts in a number of languages. Select the language of your choice which may be different from the one used on the phone.

Language: This is the language used on the display of your phone. Choose a language from the drop-down menu.

Number Display Style: Specifies how incoming and outgoing calls are displayed:

- Full Contact: The complete URL is shown
- Name: Only the name is displayed
- Number: Only the number is displayed
- Name+Number: Name and number are displayed

Tone Scheme : Select the dialtone you prefer for your phone. Also, DTMF echo will differ (on/off) on different schemes.

MWI Notification: Specify the type of MWI notification that will inform you when a new message arrives.

MWI Dial Tone: Set the dial tone to stutter mode in the case of an active MWI.

Use Headset Device: Select the headset device you would like to use. Select none if you don't want to use a headset.

U.S. Date Format: Here, you can select either U.S. (month/day) or European (day.month) format for displaying the time and date.

24 Hour clock: When you select "on", the timestamps will be formatted in 24-hour format, otherwise in 12-hour format.

Use Flash Plugin: If you want to have a live reaction on incoming or outgoing calls on the phone's "Home" page, switch this option to "on". Your web browser has to support the Macromedia flash movie format.

REDIRECTION

You can have all incoming calls diverted to a specific number.

"Event" drop-down menu options:

- Never: This deactivates all call redirections.
- Always: All calls are diverted to the number specified in "Number Redirect".
- When Busy: When a call is in progress, any other call made to that number is rejected and will receive a busy indication. The phone can be set in such a way that the second caller is diverted to another number set in "Number Busy".
- After Timeout: Specify the "Timeout" in seconds after which calls are to be diverted to the number specified in "Number Redirect.

The following fields for redirections are available:

Timeout: Specifies the timeout in seconds after which the call will be diverted.

Number redirect: Specifies the number to which calls will be diverted.

Number busy: Specifies the number to which calls will be diverted when the phone is busy.

RINGTONES

Ringtone defaults

Ringer Device for Headset: If you want to hear the ring tone via the headset only, choose “headset”; otherwise, “speaker”.

Default Ringer: Choose the default ringing melody for your snom phone through this setting.

Address Book Ring Tones

Specify the ringing melodies for different contact types of your personal phone book entries (e.g.., “friends”) by choosing a ringer from the respective pull-down menu.

Custom Melody URL: If you have chosen *Custom Melody URL* in one of the pull-down menus, you can specify an URL to your own ringing melody. The type of files that should be supplied to the phone are: “PCM 8 kHz 16 bit/sample (linear) mono WAV”.

AUTO ANSWER

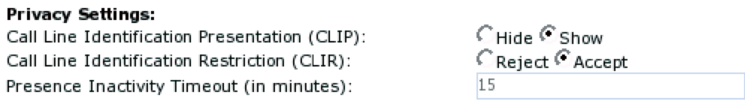
snom phones have auto-answering capabilities. Through these settings you control the behavior of auto answer.

Auto answer for the seven lines on your phone is turned on or off on the "Configuration Line 1-12" pages, tab "SIP". "Auto answer indication" and "type of answering" will only affect your phone when auto answer is on.

Auto Answer Indication: If you want to be informed with an audible indication when an incoming call is automatically answered by your phone, select "on".

Type of Answering: Select how you want to receive the incoming call, i.e., in handsfree mode, on the headset, or on the handset.

Privacy Settings:



Call Line Identification Presentation (CLIP): Show or hide your own phone number when making a call.

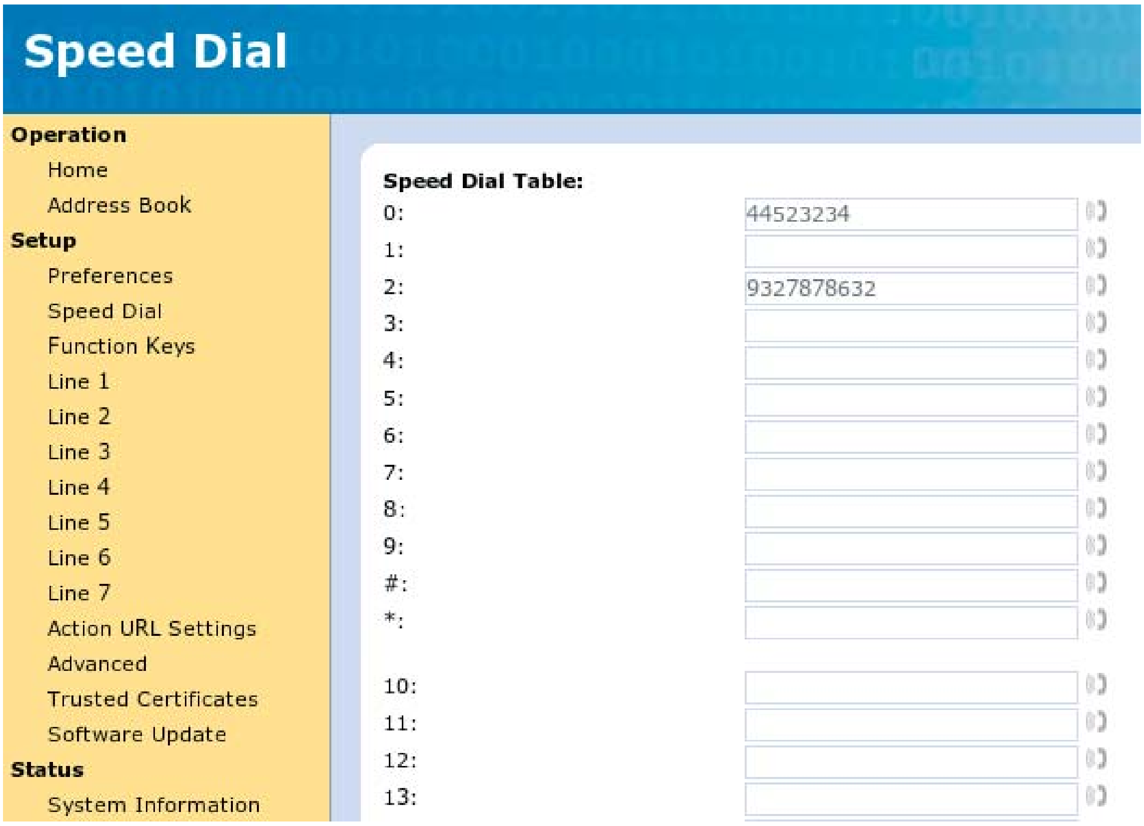
Call Line Identification Restriction (CLIR): Reject or accept anonymous incoming calls.

Presence Inactivity Timeout: This is the time after which, if there is no activity, presence is set to "closed". The default is 15 minutes. If it is set to 0, the presence stays closed and nothing is published at all. In other words, presence is disabled for all practical purposes.

Setup Speed Dial

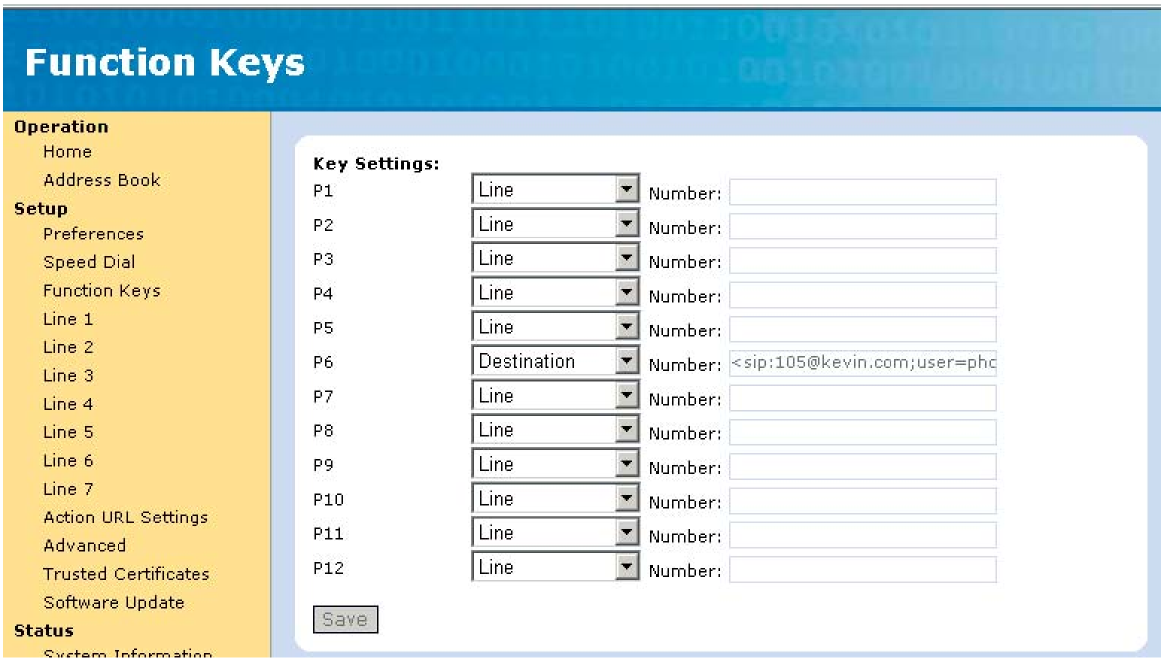
Speed Dial Table

Set up your speed dial numbers on this webpage.



The available speed dialing codes are the numbers 0 through 30 and the # and * keys. To dial one of them, press the respective key(s) and confirm with ✓.

Setup Function Keys



It is possible to program the functions “Line”, “Destination”, “Intercom”, “Park Orbit” and “Voice Recorder” onto the function keys P1 - P12 with LEDs. Each of these functions was explained in detail in the section “*Programmable Keys*”, above. The specific key utility will work only if you have also entered the number for each of the keys.

Setup Lines 1-12

On these pages you can set up the SIP lines you wish to use on your phone. It is possible to set up twelve lines. Each line setting page looks like the following illustration. Click on the tabs to access “Login”, “SIP”, “NAT” and “RTP”.

Login Information

Displayname: Set the name you would like to associate with each line, e.g. "John Smith". This information is also sent out to any party you are calling.

Account: This is the account with which you register to a registrar/proxy. It could be alphanumeric, e.g. "js", or based on digit like "445" in the screenshot above.

Password: This is the password to be used for challenge responses. In order to protect them from unauthorized use, passwords are not displayed in their true form, but as a series of asterisks (**..).

Registrar: Specify the IP or DNS address of the registrar/proxy where you want to register this account. After a successful registration, the registrar knows how to reach this specific line and can route requests (e.g., incoming calls) from other registered parties to this phone.

Authentication Username: Registrar environments may need different user names for registration and authentication.

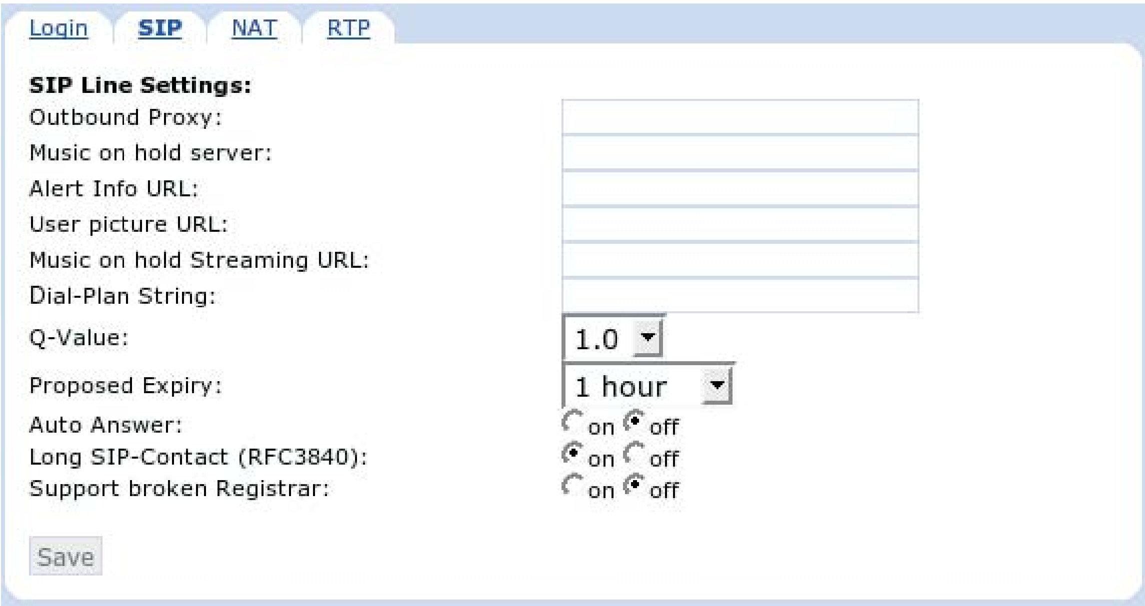
Mailbox: If you have set up a mailbox, specify the account name for that mailbox here to associate it with this particular SIP line. This is important for contacting your mailbox - by pressing the key - when there are no new messages indicated or when the MWI message does not include the proper mailbox SIP URI.

Ringtone: Select a ring tone from this pull-down menu that will alert you when a call comes in on this particular line.

Custom Melody URL: Specify an URL to your own ringing melody. The type of file that should be supplied to the phone is: „PCM 8 kHz 16 bit/sample (linear) mono WAV". This only has an effect when you have chosen "Custom Melody" from the "Ringtone" pull-down menu and when the incoming call matches this SIP line.

Display text for idle screen: If you enter a name of up to 8 characters in this field, then this name will be shown on the idle screen associated with this particular line instead of the name you have entered in the "Displayname" field, if any. This information is not sent out to anyone, but is merely shown on the phone's display for your information.

SIP Line Settings



Outbound Proxy: Specify the outbound proxy in this field (format: addr:port). This is to ensure that all SIP packets are going via this specified communication point.

Music on Hold Server: If you specify a SIP URI pointing to a media server account, the phone will, when a call is put on hold, invite this SIP URI to call the held phone to play music on hold.

Alert Info URL: This URL should point to a web server where audio alert messages are accessible.

User Picture URL: Specify an URL to a small JPEG picture. When the flash plugin feature (Preferences page) is enabled, this picture will be shown on the "Home" web page during a call.

Music on Hold Streaming URL: This turns MOH streaming on, no matter what appears on the RTPport. When this setting is available, the phone adds a supported header „x-moh-stream", which stops the other phone from inviting the MOH server (avoiding double MOH bandwidth usage).

The streaming format is always http (or https!). The phone expects WAV file format, either in ulaw-format (mono 8kHz) with 64 kbit/s or in linear format (mono 8kHz) with 128 kbit/s. Example files are available at

http://snom.com/download/moh0.wav
http://snom.com/download/stream-linear.wav
http://snom.com/download/stream-ulaw.wav

The files are automatically repeated (no caching). The phone expects a content-length header in http; if the file is „endless", the content-length header should be omitted. Currently, Chunked encoding is not supported. The phone buffers 4-5 seconds of media. Tests show that this is a reasonable value, but it might need adjustment in other network environments.

Dial Plan String: You can set up the dial plan for this line here. With a dial plan, you can match user input (digits via keyboard) to specific actions like dialing, using a distinct outgoing identity, etc. Please have a look at the white paper "*Dial plan on snom phone*" on our website.

Q-Value: You can set up the probability of a registration for each line through this setting (the default is 1.0). This means that different registrations with different Q-values will ring in serial order (serial forking) in contrast to different registrations with the same Q-values, which will ring in parallel (parallel forking).

Proposed Expiry: You can select the time when the registration on this line is to expire. Upon expiration of the registration, the phone will send a fresh re-registration request.

Auto Answer: Line-specific "Auto Answer" can be used to establish an intercom functionality. Set it to "on", map the number of the extension to be reached by intercom onto one of the programmable keys on the page "Function Keys", and set the key type to "destination".

Long SIP-Contact (RFC3840): When your SIP Registrar is not properly supporting long contacts specified in accordance with RFC3840, you may want to switch this behavior off.

Support broken Registrar: If your VoIP provider works only when you turn on „Support broken registrar" on the snom 320's web interface, this means your provider does not call your phone the way the phone requested to be called.

What happens is that incoming INVITEs from your VoIP provider do not contain the contact URI which was previously registered by your phone as its contact. Thus the phone cannot safely identify the target line of the incoming call. When you compare the URI in the first line of the incoming INVITE and the URI in the Contact of the REGISTER, which the phone sends to the registrar of your provider, they will presumably differ. This is what we mean by "broken registrar".

It is as though your provider has sent a letter to an apartment building with the city, the street address, and the house number on it, but without the recipient's name. When you turn on „Support broken registrar", the phone tries to find the right apartment by guessing, but this guessing will fail when there are two parties with the same name in the building.

NAT Line Settings

Offer ICE: Choose whether or not you want to use ICE (Interactive Connectivity Establishment). ICE optimizes the media path. This would be the case, for example, when two phones in the same network are calling each other via a long media path through other, external networks. With ICE, the short media path in the same network would be chosen, which will presumably have better quality than the long one.

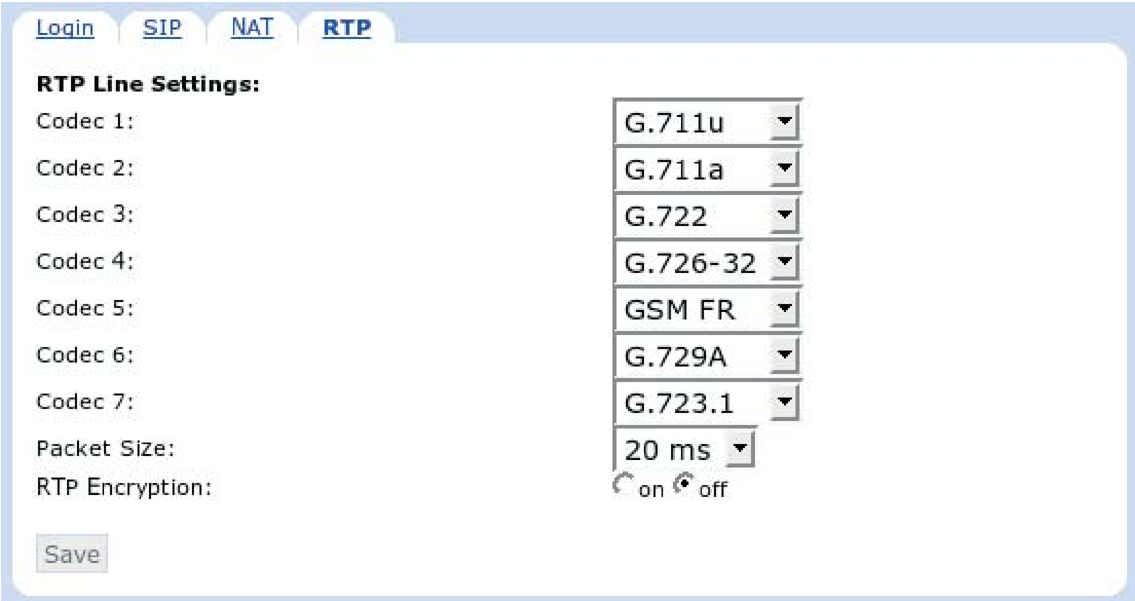
Sometimes this feature will stop you from being able to make calls. When this occurs, switch it off.

STUN Server (IP-addr:port): We reintroduced a STUN keep-alive mechanism for SIP, which can be turned on manually by specifying the address of the STUN server followed by the port number. However, we strongly discourage you from using it, because it will not work properly with symmetrical NAT (i.e., Linux-based router/firewall). The only general SIP NAT solution is a session border controller (SBC) on the service provider's side.

STUN Binding Interval (seconds): Similarly, set the STUN binding interval time in seconds through this setting.

Symmetrical RTP: If you want to use symmetrical RTP switch it "on" here.

RTP Line Settings



Codec 1-7: You can select the preferred order for the codec potentially to be used. The available options are G.711 ulaw, G.711 alaw, G.722, G.723.1, G.726-32, GSM and G.729a. Your phone will offer exactly this codec list to the called party and will, in return, receive a selection of the codec list the other party wants to use for this call.

Packet Size: Select the packet size in ms. G.723.1 needs 30 or 60 ms. All other codecs work with 20, 40 and 60 ms only.

RTP Encryption: The snom 320 supports RTP encryption via SRTP. If you want to encrypt your outgoing RTP (audio) stream, switch this option to "on". If, during a call, a small lock sign is shown on the display, this means that an SRTP encrypted call is currently taking place. Both parties have to enable the RTP Encryption option to establish an SRTP call.

RTP encryption has nothing to do with SSL/TLS. The keys are sent in the SDP part of SIP messages. Certificates are not used for this. (See also our white paper "Providing Security in VoIP Environments"!)

Action URLs

An “Action URL” triggers external URLs on phone-internal events. In the above example, for instance, on an incoming call, the URL of the field “incoming call” is triggered. “Setup finished” is triggered when the phone is booted up properly and ready for usage.

Parameters that will make this feature much more usable are not available yet.

Setup Advanced

Network

With these settings you can set up the basic network settings of your phone. The web interface for this looks like the following:

DHCP: Turn the use of DHCP on or off with this option. For further information, see our white paper regarding DHCP on our website.

IP address: You can change the IP address of the device through this setting. This parameter is mandatory in order to enable the Ethernet connection.

Netmask: Change the netmask for the device.

Phone name: Change the Hostname of the phone here. If this parameter is available, it is used for identifying the device in SIP signalling.

IP Gateway: This setting shows the IP address of the default IP gateway (NOT the VoIP gateway). It is the address to which the packets are routed when the desired packet address is not in the current subnet. Setting up this parameter is mandatory in order to reach an external network.

DNS

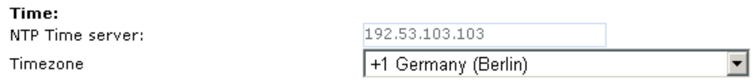
Domain: Specify the DNS domain for your phone here.

DNS Server 1: Specify the IP address of the DNS server for your network here. This parameter is extremely important for a properly functioning phone, so please make sure it is set up correctly.

For troubleshooting, check the DNS cache page so see whether the phone is able to resolve DNS addresses!

DNS Server 2: Specify the IP address of a backup DNS server for your network here.

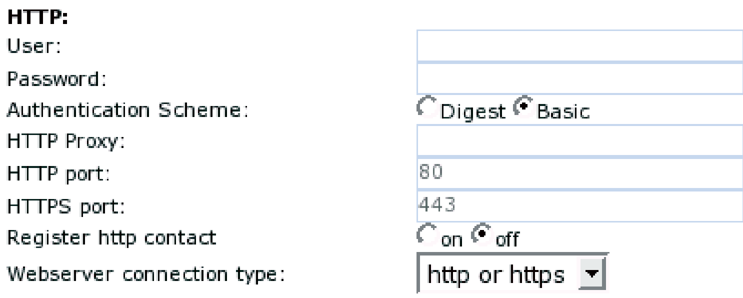
TIME SETTINGS



NTP Time server: Specify the url or IP address of the NTP server here.

Timezone: Select the time zone of your geographical location through this option.

HTTP



User: Here, you can select the HTTP username for your phone. This and the next option (Password) protect your web interface if you so desire.

Password: Set up the HTTP password for your phone here.

Authentication Scheme: Define whether “Basic” or “Digest Authentication Scheme” should be used. The latter is the more secure option.

HTTP Proxy: You can select the HTTP proxy address for your phone here. This is needed if you are also surfing the web via such a proxy.

HTTP port: Specify the HTTP port to be used by your phone through this setting. By default, it is port 80.

HTTPS port: Specify the HTTPS port to be used by your phone for HTTPS connections (default 443).

Register HTTP contact: Should the phone add the http URL of the phone as additional contact information? **WARNING:** Turning this setting on may cause a complete loss of VoIP ability if the proxy/registrar does not support it. We urge you strongly to leave it on “off” if you are not absolutely sure that it is supported by your proxy/registrar.

Webserver connection type: Set up the type of connection the phone’s webserver is willing to answer to:

- http
- https
- http & https
- off

Please be advised that **you will no longer be able to use the web interface of the phone when you select “off”**! If this has happened and you wish to change this setting, use the phone to do it. Press the menu key and select the submenus Network / WebserverType.

Phone Behavior

Phone Behavior:	
Call Completion:	● on ○ off
IDNA (RFC 3490) Support:	○ on ● off
Action on Ethernet cable replug:	Ignore
Auto Dial:	off
Number Guessing:	● on ○ off
Block URL Dialing:	○ on ● off
Deny All Feature:	● on ○ off
Challenge Response on Phone:	● on ○ off
Enable Intercom:	○ on ● off
CMC Feature:	○ on ● off
Dialog-Info Call Pickup:	○ on ● off
Call Waiting Indication:	on
Dialtone during Hold:	● on ○ off
Disconnect on Hook:	● on ○ off
Call join on Xfer (2 calls):	○ on ● off
Alert Info playback:	● on ○ off
AOC Amount Display:	off
AOC Pulse Currency:	$
AOC Cost/Pulse:	1

Call Completion: Turning this setting to "on" will prompt the user to activate call completion, if possible, while calling a number (see the CC soft key). When the called party becomes available again, your phone will be able to automatically redial the number. CCNR (on not reachable) as well as CCBS (on busy) is supported.

IDNA (RFC 3490) Support: Switch on support for Internationalizing Domain Names in Applications (IDNA). IDNA support is the ability to handle domain names including international special characters.

Action on Ethernet cable replug: Choose the action to be performed after the network connection is reestablished: „Do nothing“ or „reboot“.

Auto Dial: This setting is switched off by default. You can set a timeout after which a number is dialed automatically without pressing or taking the handset off the hook.

Number Guessing: Here, the number guessing functionality can be activated or deactivated. This is the automatic number completion which begins after you have entered the second digit of the number you are dialing.

Block URL Dialing: You can block the dialing of SIP URLs by turning this setting on. In this case only numeric numbers will be allowed as input.

Deny All: When this feature is set to "on", the "Deny Call" function key will be available on the display when calls are coming in, so that any undesirable incoming numbers may be put on the "Deny List".

Challenge Response on Phone: As explained in an earlier chapter, snom phones can handle challenge responses on the phone. Turning this setting off will disable this feature and you will only be able to handle authentication through the web interface of the phone.

Enable Intercom: Here, you can explicitly select whether or not you want to participate in the intercom functionality. If you enable it, the intercom functionality can be programmed onto the function keys set to type "Intercom" (see "Configuration Line 1-12).

Usually, intercom uses authentication, but if the line has registered at the registrar without authentication, intercom also works without authentication.

CMC (Client Matter Code): When this setting is turned on, the user is offered the function key "CMC" during a call; its use sends a code to the server using the INFO message in SIP. This code can later be used for billing or bookkeeping along with the call ID of that call.

Dialog-Info Call Pickup: snom phones subscribe to the status information of SIP URLs mapped as "Destination" on the programmable keys. By turning this setting on, the user will be offered a pickup window when a mapped destination gets a call and is in the ringing state. The display will change from the idle state to show the source and destination of that call and the user can pick up the call by pressing or the programmable key currently blinking.

```
From: 444
To: 474
```

The pickup offer is available as long as the destination is ringing; it will disappear when is pressed, when the call is connected, when the call is cancelled by the caller, or when a third party picks up the call.

When the feature is set to "off", the message will not be shown on the display; the only indication of the incoming call is the blinking LED of the programmable key. The pickup, however, will also work in this case.

Call Waiting Indication: The three available options are "on", "visual only", and "off". If you select "on", the second incoming call is displayed in the lower left corner of the display:

```
444                       0:22
474
```

You will also hear a short audible knocking signal behind your currently ongoing call, indicating that another call is coming in. When you select "visual only", the visual indication will be shown on the display and there will be no audio indication. Set on "off", Call Waiting Indication (CWI) is disabled, which means that only one call at a time can be handled to and from the phone.

Dialtone during Hold: Turning this setting to "on" will produce a dial tone when a call is being held and enables the user, for example, to dial a second number. A dial tone will not be played when this setting is turned to "off".

Disconnect on Hook: Sometimes it is useful to disable the disconnection of a call when the handset is placed on hook, e.g., during conference calls or in handsfree-mode, etc. This is achieved by turning this setting off.

Call join on Xfer: When this feature is turned to "on", you will connect an incoming call to, for example, a colleague you already have on hold by pressing . You will not be able to pick one of severals calls on hold to transfer the call to. If this is what you normally want to do, set this feature to "off" and use the navigation key to select the calls to be joined! Press and to join them.

Alert info playback: If you want your phone to replay audio system messages when they are provided, set this option to "on". Additionally, you will see a message on the display. When you set the option to "off", you will only see the message on the display.

AOC Amount Display: If your provider supports "Advice of Charge" (AOC) information (i.e., the costs for your call) during or at the end of the call, you can turn on this feature by selecting one of the following options: Select "Charged" to show the accumulated amount of the current call on the display; select "Balance" to show the amount remaining on your account.

AOC Pulse currency: Sets the currency symbol that will be shown next to the amount (e.g., $).

AOC Cost/Pulse: Specify how much money one pulse costs (e.g., 0.12 means 12 cents per pulse)

Keys

Here you can change the key behavior on the phone to suit your wishes. The webpage subsection may look like this:

Transfer on Onhook: If you do not want calls to be transferred when you place the receiver on the hook, you can switch it off here.

Block DND: If you don't want the users of the phone to have the option to turn on the "Do not disturb" (DND) mode, set "Block DND" to "on". This may be desirable in call center or switchboard environments.

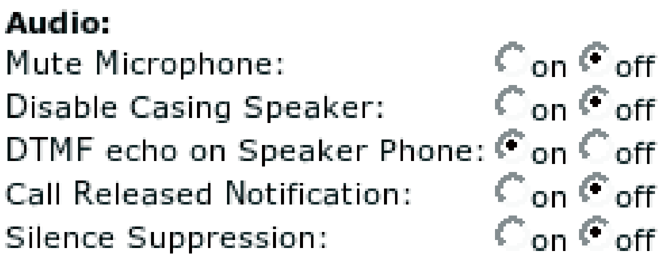
*Logon Wizard***:** The Logon Wizard assists you during the SIP line registration process. Turn this setting on if you want to use the Logon wizard, switch it off if you don't.

PRESELECTION

Prefix: Specify the number to be prefixed to each dialled number.

AUDIO

Here you can set up audio-related settings on your phone. These settings may look like this:



Mute Microphone: Setting this to on will mute the microphone of the phone. Tuning it off will enable the microphone again.

Disable Casing Speaker: Turn this setting on to disable your speaker.

DTMF echo on Speaker Phone: Switch DTMF echo on or off.

Call Released Notification: Turn this to "on" if you want to have an audible indication when a call is terminated. Turning it off will take you directly to the idle state when a call drops.

Silence Suppression: To save bandwidth in the case of silence, select "on" to stop silent RTP audio streams for as long as the silence lasts. When silence suppression is on, comfort noise (CNG/VAD) will be generated locally at the other end of the call so that the other party will not mistakenly believe that the call has been terminated.

Advanced Network

In this subsection you can change the advanced network settings of your phone:

Advanced Network:	
Dynamic RTP port start:	10000
Dynamic RTP port stop:	10009
Type of Service (TOS):	160
DTMF Payload Type:	101
Network identity (port):	
SIP T1 (ms):	500
SIP Session Timer (s):	3600
SIP Dirty Host TTL (s):	
SIP Max Forwards:	70
ENUM Suffix:	e164.arpa
Use user:phone:	(•) on ( ) off
Publish Presence:	( ) on (•) off
Refer-To Brackets:	( ) on (•) off
Require PRACK:	(•) on ( ) off
Offer GRUU:	(•) on ( ) off
Offer MPO:	( ) on (•) off
Filter Packets from Registrar:	( ) on (•) off
Authentication for SIP Reboot:	( ) on (•) off
Authentication for SIP Check-Sync:	( ) on (•) off
Session Refresher:	(•) Client ( ) Server

Dynamic RTP port start, Dynamic RTP port stop: If you want to set up the port range out of which the RTP ports will be dynamically taken, specify the start port and end port number, respectively, in these fields.

Type of Service (TOS): You can set up the TOS value in this field, for example, 160 in the above screenshot. This option enables the phone to support quality of service (QOS) in a network, if all of the involved network parts also support QOS.

DTMF Payload Type: Set up the payload type for Out-of-Band DTMF here The default setting is 101. This can be an arbitrary 8-bit value as long as the involved communication partners are both using the same value.

Network identity (port): Set a static local port number, which is used for the SIP protocol communication, in this field. Usually, the phone chooses a random one!

SIP T1 (ms): Set the retry timer in milliseconds after which an unanswered request is resent. If it is set to 500, the phone will resend the unanswered request after 500, 1000, 2000, 4000, 6000…31500 ms. If the request is still unanswered after this procedure, an error message will be shown on the display.

SIP Session Timer (s): Specify the session timer for SIP in seconds in this field. For instance, a Re-INVITE will be sent after its expiration.

Dirty Host TTL (s): Specify the “Time to Live” (TTL) for dirty hosts in seconds. This means that, when a phone was unable to reach a host, the phone will not try to reach this host again until the time specified in this field has elapsed.

SIP Max Forwards: If you set a maximum number of forwards in this field, each time a forward is sent the counter is reduced by one. When zero is reached, the forwarding will stop. This prevents the phone from running into a SIP message-forwarding loop.

ENUM Suffix: When using ENUM, you can specify a service suffix here, if desired. There is more than one service that supports ENUM lookups, and you can select here which one you want to use. Leave the default value e164.arpa if you don’t know.

Use user=phone: Turn this setting on if you want to use user=phone in SIP URIs. This is to distinguish phones from different non-phone devices like gateways, etc. (RFC2543 deprecated).

Publish Presence: When this feature is set to "on", the phone sends out PUBLISH messages showing the phone's status.

Refer-To Brackets: Switch additional brackets on or off in the Signaling for Refer-To. As some devices rely on this setting, we are kind enough to offer a solution!

Require PRACK: To force the use of PRACK, choose "on" here. "PRACK" messages are used to acknowledge the receipt of "180 Ringing" messages, which are usually not acknowledged. This helps, for example, to inform gateways whether the phone has actually begun to ring.

Offer GRUU: This setting is used to toggle the support for GRUU (Globally Routable User agent URLs) in SIP. When several phones have the same account, each one of them can be identified by the proxy through this GRUU ID, which is unique for each phone and stays the same even after reboot.

Offer MPO: Using this setting, the user can turn the Media Path Optimization support on or off. Turning it on makes sense only when you have MPO-supporting session border controller devices in your environment (e.g., Jasomi).

Filter packets from Registrar: If set to "on", all SIP packets not coming from the registrar/proxy will be ignored. For security reasons, "on" is the default setting. This may cause big problems in an environment where SIP packets from other sources also have to be accepted for proper functionality!

Authentication for SIP Reboot: This setting enables and disables challenge responses for remote reboot requests.

Authentication for SIP Check-Sync: Turning this setting on enables challenge responses for Check-Sync requests.

Use SIP Compact Headers: In order to use SIP compact header notations, you can switch this functionality on here.

UPDATE

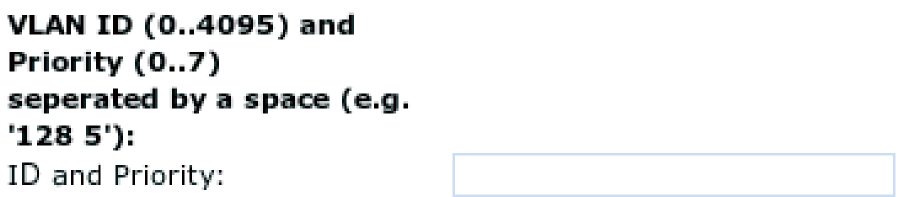
Update Policy: Select the update policy you wish to adopt for your phone.

"Update automatically" does not ask again whether you are really sure that you want to update. "Ask for update" asks whether you are really sure that you want to update firmware or bootloader. "Ask for updating firmware only" asks whether you are really sure you want to update the firmware. "Ask for updating bootloader only" asks whether you are really sure you want to update the bootloader. "Never Update, Load settings only": loads only settings from settings server.

Setting URL: Enter the URL of the settings server from where you would like to obtain the configuration file to configure your phone. Please read our white paper on "Mass deployment", which is available on our website.

Subscribe Config: The phone can subscribe to setting changes delivered via SIP when this option is switched to "on".

VLAN



ID and Priority: Enter your VLAN ID (802.1q) and Priority (802.1p) separated by a space (e.g. "128 5"). The first number specifies the VLAN number the phone should be joining and the second number gives the priority of this device.

Please be advised that the whole phone resides in the specified VLAN and that it can only be assigned to one VLAN at a time.

DEBUG

Debug:	
Syslog Server:	
LCServer:	

Syslog Server: Type in the host where a central Syslog Server is running to store the log messages coming from the phone.

LCServer: Type in the IP address of the remote LCServer if you want your phone to connect to it. Normally, you do not need to make an entry here. The LCServer is for snom's internal development only.

SNMP

Regarding SNMP on snom phones, see our white paper "SNMP"!

SNMP:	
Port:	161
Trusted Address:	192.168.0.0/16

Port: Type in the port to be used for SNMP communication.

Trusted Address : Specify the address range (in CIDR notation) solely from within which subnet SNMP requests will be accepted.

SECURITY

Security:	
Administrator Mode:	◉ on ○ off
Administrator Password:	****
Administrator Password (Confirmation):	****

Administrator Mode: This setting allows you to switch between the User and Administrator modes of the phone.

Administrator Password: When the phone is running in administrator mode, you can change the admin password through this setting. A password should be a numeric string of any length. The default password to reach the administrator mode is “0000”.

Administrator Password (Confirmation): You have to confirm the password to ensure no typing errors have been made.

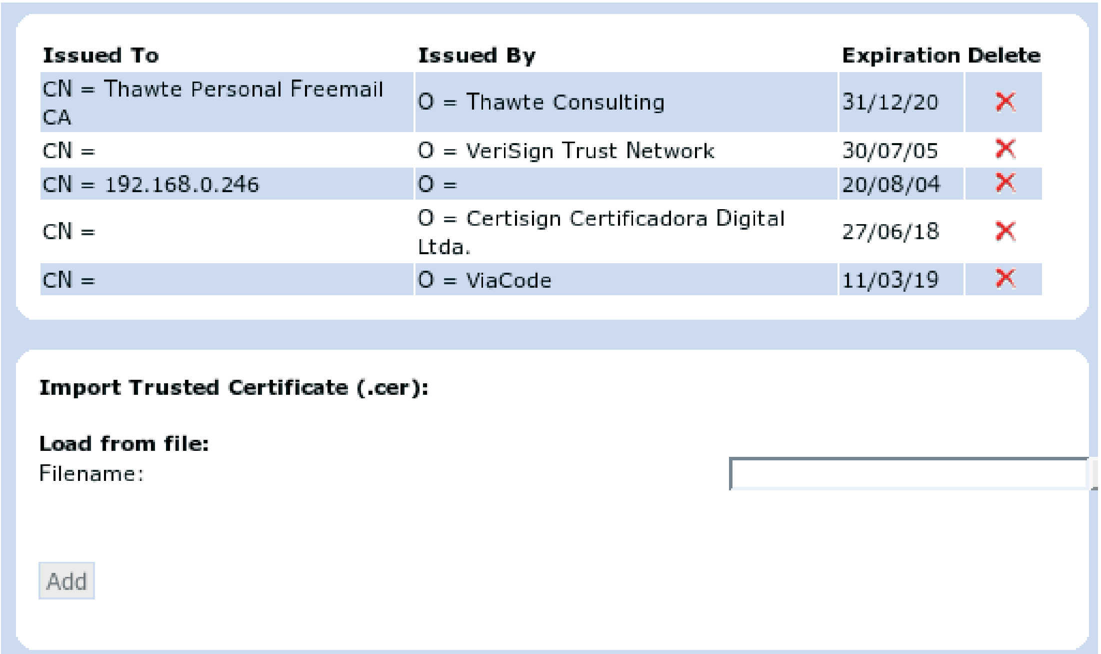
Certificates

Upload your own server certificate for TLS-secured communication. The default certificate is the same for every phone. A real signed certificate costs money, and we do not distribute free certificates. If you have a signed certificate you can upload it on his here.

For secure SIP communication (SIPS), the phone acts as a client. It is thus the server/proxy who sends the certificate and not the phone. This certificate contains the public key of the proxy server, which the phone uses for its web interface. The certificate loading option that you see here is therefore for the web server of the phone only.

Setup Trusted Certificates

On this page, certificates from trusted authorities can be imported to create an internal CA list. The page may look like this:



Issued To	Issued By	Expiration	Delete
CN = Thawte Personal Freemail CA	O = Thawte Consulting	31/12/20	✕
CN =	O = VeriSign Trust Network	30/07/05	✕
CN = 192.168.0.246	O =	20/08/04	✕
CN =	O = Certisign Certificadora Digital Ltda.	27/06/18	✕
CN =	O = ViaCode	11/03/19	✕

Import Trusted Certificate (.cer):

Load from file:
Filename:

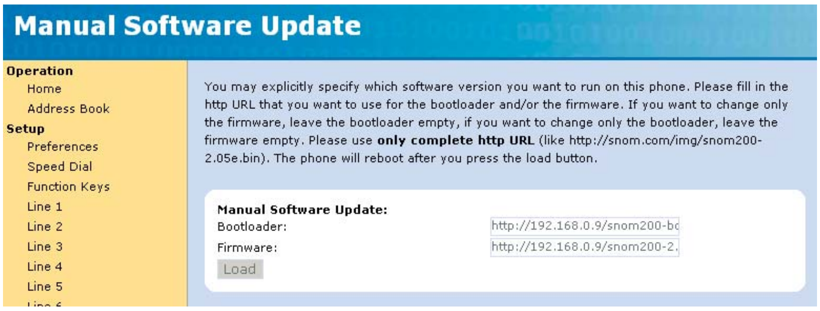
Add

The phone looks up on this table whether an incoming TLS request should be accepted because it is verified or has been signed by a CA. In the case of SIPS, the phone acts as a client. Hence, the certificate of the phone is not used in this case. The server sends a certificate that the client can accept or reject. The criterion for this is the CA list.

Setup Software Update

MANUAL SOFTWARE UPDATE

Here you can select the binary files you want to run on the phone. This section may look like this:



Bootloader: Specify the URL for the bootloader file (-b) you would like to place on the phone through this setting. This field is currently inactive, the bootloader will not change when you make an entry here!

Firmware: Type in the firmware (-l, -r, -j) URL in this field.

Press the "Load" button if you made changes to these settings.

The phone will now attempt to load the binary files from the newly provided addresses. It will show "SW" on the display and then start rebooting after a couple of seconds. If it shows "SW" but does not start to reboot, please start the reboot process yourself.

Naming conventions of the different images:

Bootloader „-b" is the bootloader and does not do a great deal. We do not expect any significant changes here.

Linux „-l" consists of the linux kernel. This will only rarely change.

Ramdisk „-r" is the ramdisk, which consists of the fundamental files that have to be present to get the linux system running. And with 3.14, we have added a small application that is able to update all partitions mentioned here via TFTP. This will only rarely change.

Application Filesystem (JFFS2) „-j"; here we have the application performing as "the phone". This is what will usually be updated.

Status

System Information

System Information:	
Phone Type:	snom320-SIP
MAC-Address:	000413240005
IP-Address:	192.168.200.239
Version-Code:	snom320-SIP 3.60i
Bootloader:	
Firmware:	http://192.168.0.9/snom320-3.60i-SIP-j.bin
SIP Line Status:	
Line 1 Status:	447@intern.snom.de: Ok
Line 2 Status:	555@intern.snom.de: Ok
Line 3 Status:	777@intern.snom.de:
Line 4 Status:	
Line 5 Status:	
Line 6 Status:	
Line 7 Status:	

Here, you find information regarding phone type, MAC Address, IP Address, firmware version code (this is extremely important!), boot loader, and firmware URLs.

The SIP Line Status displays the current status of each SIP line as long as it is configured completely.

Log

Depending on a selectable log level, the log messages are shown here. Don’t forget to attach this log to your support request!

SIP Trace

A quite helpful feature to display SIP signaling. For a support request, clear this page, perform your not-working scenario, reload the page, and attach it to your support request.

DNS Cache

An option to look into the current DNS cache. It is highly recommended to add this page to a potential support request!

PCAP Trace

Create packet traces from current network traffic right on your phone! This is a really powerful tool to find out what is going on in the network around your phone.

When you press the “start” button on this page, the tracing will start and record every incoming or outgoing packet addressing your phone. Press the “stop” button to stop the tracing. Click onto the “here” link and save the file, which can be easily analyzed with ethereal.

Please be aware that the ring buffer used has a limited size, which is filled up rapidly, especially when tracing audio streams, and will overwrite the first packets so that they disappear. Usually, they are the interesting ones, so please try to record scenarios that are as short as possible!

Memory

Look at the current memory usage of your phone.

Settings

Displays all settings of your phone including their current values. This is a good starting point to create your own setting files, but not all settings need to be specified in your setting file, so please don’t save the whole page to your setting file, but only the number of settings that you really want to specify. Please see the white paper regarding “Mass Deployment” on our website. Each individual setting is described there. Read the part about the flags “! and &”, too!!!

This page is also important for us to consult in the event of a support request, so please attach it to your request, as well.

Appendix

Standard Conformance

Name and address of manufacturer

snom technology AG

Gradestr. 46

12347 Berlin

The snom technology AG assures that the product

Type: VoIP Phone
Model: snom 320

conforms with the following standards

Product Standard EN 55 024
FCC Part 15 2004-04-23 Emission

Place, Date	Dr. Christian Stredicke
Berlin, 21 June 2005	Vorstand (MD)

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Reader‘s Feedback

snom technology AG welcomes your evaluation of this manual and any suggestions you may have. These help us to improve the quality and usefulness of our documentation.

Please send your comments and suggestions to:

snom technology AG

Attention: Marketing Department

FAX: +49 (30) 39833 111

Manual Name: User Manual, snom 320

Version: V1.00 Date: 21 June, 2005

	Excellent	Good	Fair	Poor
How would you rate the manual overall?	☐	☐	☐	☐
Are the installation instructions effective?	☐	☐	☐	☐
Are the Configuration Instructions effective?	☐	☐	☐	☐
Is the manual properly organized?	☐	☐	☐	☐
Are the illustrations usefull and easy to understand?	☐	☐	☐	☐
Are the suggested and default values useful?	☐	☐	☐	☐
Is the index useful?	☐	☐	☐	☐

Did you find any errors in the manual? (Please reference page and paragraph)_____

____

How might we improve this manual?____

____

____

____ Name________

______________________________Title_____________________________________

Company Name____________________________Telephone (   )_________________

Thank you for taking time to fill out this form.

snom VoIP Phone snom 320
User Manual
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