FREQUENTLY ASKED QUESTION

Configuring snom phones for Mass Deployment V3.4

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snom The SIP Experts

As stated in the FAQ „How can I update a snom phone?", it is possible to configure your snom phone with config files, which consist of the following settings.

This release v3.4 is describing firmware settings beginning with version 3.4 ! In case you are using an older firmware version please refer to v2.0 of this document.

Phone settings

Settings are non-volatile name/value pairs that are stored in the flash of the phone. They include registration information, dialled numbers, network settings and other information that should be available after rebooting. Settings can be made read only on a per setting basis. This is useful in environments where an operator sets up the phones and wants to avoid users changing settings that affect the stability of the phone. In this way, expensive trouble ticket searching can be reduced and the total cost of ownership of the phone be minimized.

Settings Files

Setting files are ASCII-based files containing lines (terminated with newline or carriage return/newline pairs). Comments start with a # or a < character. The < and > characters allow easy integration of html tags. Names may consist of the characters a-z, A-Z, 0-9 and _.

Appending a name with an "!" character means that this setting becomes writable by the user. Appending the name with an "&" states that the setting is read only (this is the default). If a setting is marked writable, the value in the settings file is only written to the phone if that setting has not yet been set up at the phone. In order to overwrite such settings, do not append the setting name with "!".

A sample file looks like this:

```
<html>
<pre>
#This is a settings file
#
phone_name: testphone
dns_domain: intern.snom.de
dns_server1!: 192.168.0.9
</pre>
<html>
```

Downloading procedure

Settings are downloaded from web servers. HTTP is a very powerful way to get configuration information from anywhere in the world. The location of a file is described as a URL, i.e. it begins with http://.

This is how the Settings are set up:

1. The settings that were stored in the flash memory are read.
2. Network and DNS is set up. This allows the phone to get settings from internet URLs.
3. It then checks the setting "setting_server" and loads the settings stored in the URL (e.g. http://www.snomag.de/snom200/snom200.htm) provided. For a small description of

the file format, see below. The setting_server URL can also consist of a {mac} statement like http://www.snomag.de/snom200/snom200.php?mac={mac}. The MAC address is a unique identification ID of Ethernet devices. snom devices have the form 000413xxxxxx, where xxxxxx is a hexadecimal number identifying the snom device. The MAC address is automatically replaced from the phone with its own mac address before using the URL. So phone specific settings can be provided for each phone via this mechanism. The "setting_server" setting can be set manually via webinterface or automatically via DHCP (options 66 and 67) and through snom4S SIP proxy.

4. The de facto settings are added. These settings may differ from what has been set up because DHCP may have changed them. The settings include:
    - the IP address of the device,
    - the net mask,
    - the IP gateway,
    - the hostname,
    - the DNS domain, first and second DNS server,
    - the UTC offset in seconds,
    - the time server,
    - DHCP on or off

Each setting is described by its ‚Name:' which should be added to the config file followed by a colon and its specific value. Name is followed by ‚Valid values:' where the valid values are described. Each possible value is bracketed in <>.The braces must be left out if you are using the values stored in a config file. ‚Default:' shows the value which is set by default after you have removed all of the setting with a manual update or a reset values! Lastly, ‚Description' gives brief details of each setting. See also the snom4S SIP proxy manual for PNP configuration via the proxy. And see the FAQ "How to update a snom phone" for further details regarding the update process.

Redirection

redirect_event

Valid values: <all>, <busy>, <none>, <time> Default: none Description: Event that causes redirection. <all> redirects always, <none> never, <busy> when the phone is in use and <time> after a timeout.

redirect_time

Valid values: e.g. <15>

Default: blank

Description: Number of seconds after which the phone will redirect the incoming call.

redirect_number

Valid values: e.g. <tb,sf> or <sip:tb@snom.de;q=0.8,sf@snom.de;q=0.9>.

Default: blank

Description: The redirection target. If the proxy executes the redirection, this may be a comma-separated list of alternate destinations (including probabilities).

redirect_busy_number

Valid values: e.g. <tb,sf> or <sip:tb@snom.de;q=0.8,sf@snom.de;q=0.9>.

Default: blank

Description: The redirection target when the redirect_event is set to busy.

Basic

language

Valid values: <English>, <English(UK)>, <Deutsch>, <Francais>, <Italiano>, <Espanol>, <Cestina>, <Nederlands>, <Polski>, <Portugues>, <Slovencina>, <Suomi>, <Svenska>, <Dansk>, <Japanese>, <Chinese>

Default: English

Description: Selects the language.

web_language

Valid values: <English>, <Deutsch>, <Nederlands>, <Suomi>, <Dansk>

Default: English

Description: Selects the language for the webinterface.

display_method

Valid values: <full_contact>, <display_name>, <user_name>

Default: full_contact

Description: Selects the display style of SIP URLs.

call_completion

Valid values: <true>, <false>

Default: false

Description: With <true> the phone offers after a "busy" the option to wait for the other party to become free again and inform the user to make a new call attempt.

auto_dial

Valid values: <off>, <2> , <5>, <10>, <15>

Default: off

Description: Phone begins after 2, 5, 10 or 15 seconds to dial the number already typed in.

block_url_dialing

Valid values: <true>, <false>

Default: false

Description: With <true> the user is not anymore able to change the inputtype for editing chars, it will remain numbers only.

challenge_response

Valid values: <true>, <false>

Default: true

Description: <true> turns on displaying challenge request on the display to the user.

broadsoft_call_control

Valid values: <true>, <false>

Default: false

Description: <true> turns on broadsoft call control feature support.

cmc_feature

Valid values: <true>, <false>

Default: false

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

call_waiting

Valid values: <true>, <false>

Default: false

Description: If call waiting indication (CWI) is on (<true>), during a call parallel incoming calls are signaled with a knocking sound and the caller hears a free signal. In case of <false> only one incoming call in parallel can be handled and the caller hears a busy signal.

cw_dialtone

Valid values: <true>, <false>

Default: false

Description: In case of holding a party you will hear a free tone <true> or nothing <false>

conf_hangup

Valid values: <true>, <false>

Default: false

Description: Someone maybe wants the hookswitch to not cancel a call, then it should be set to <false>.

transfer_on_hangup

Valid values: <true>, <false>

Default: true

Description: to switch off "transfer on hook" functionality, set it to <false>.

contrast

Valid values: Between <0> and <15>

Default: 8

Description: Determines the display contrast, but should not be used, because each phone reacts differently to it dependend by example from the temperature etc. Its better to set it manually.

use_backlight

Valid values: <true>, <false>

Default: true

Description: In case your phone has a backlit display (snom105/220) the light can be switched off completely with <false>.

image_src

Valid values: <true>, <false>

Default: false

Description: The pictures for the internal web interface are taken from the phone (<false>) or loaded via the Internet (<true>). Thus if you have a slow connection to the remote phone, <true> will significantly speed up the displaying of the phone's webpages.

guess_number

Valid values: <true>, <false>

Default: true

Description: Number guessing or auto completion while typing in numbers to dial.

deny_all_feature

Valid values: <true>, <false>

Default: true

Description: If set to <true>, the "Deny All" option is offered on incoming calls. Otherwise it is disabled.

admin_mode

Valid values: <true>, <false>

Default: true

Description: Administrator mode (all settings are accessible) or user mode (only a few basic settings are accessible).

admin_mode_password

Valid values: Numbers and character strings of unspecified length, e.g. <1234>, <0fg5gju>, <nhcndeve>

Default: 0000

Description: The password for entering the Administrator mode on the snom phone.

callpickup_dialoginfo

Valid values: <true>, <false>

Default: false

Description: If your phone should be able to get informed of calls to be picked up via the dialoginfo SIP message and to pickup calls, set this setting to <true>.

SIP Line Settings

The SIP stack on the phone supports seven lines, which are identified with the numbers 1 through 7. Line-specific settings are appended with the line number.

user_realname[1-7]

Valid values: e.g. <Jim Testa>

Default: blank

Description: This is the real name of the user that is displayed for line x.

user_name[1-7]

Valid values: e.g. for <sip:
abc@company.de>, the user_name[1-7] would be abc.

Default: blank

Description: The account name for line x. This is a mandatory setting to set up a SIP line!

user_host[1-7]

Valid values: e.g. for <sip: abc@company.de> the user_host would be <company.de>.

Default: blank

Description: The registrar for line x. This is also a mandatory setting to set up an SIP line!

user_pass[1-7], user_hash[1-7]

Valid values:

e.g. <company, 456876, answerIs42, a6 4bd7c29de0f23ba64bd7c29de0f23b0>

Default: blank

Description: A setting for registrar authentication. The username and the password can usually be set up on the proxy and registrar, user_hash is a MD5 encrypted password, which can be used instead of user_pass.

user_q[1-7]

Valid values: Values between <0.0> and <1.0>

Default: 1.0

Description: The probability of the registration for line x. This probability is used by some proxies to call the registered phones one by one (sequential and parallel forking proxy).

user_dp_str[1-7]

Valid values: dialplan regular expression, see snom4S proxy manual

Default: blank

Description: dialplan per line

user_transport[1-7]

Valid values: <auto>, <udp>, <tcp>, <tls> Default: auto Description:The IP protocol which should be used for this line.

user_expiry[1-7]

Valid values: <60>, <600>, <3600>, <7200>, <28800>, <86400>

Default: 86400 (one day)

Description: The proposed expiry time of the registration in seconds for line x.

user_mailbox[1-7]

Valid values: e.g. <abc> or <sip: abc@mailbox.bla.com>.

Default: Blank

Description: The SIP url of the mailbox associated with line x.

user_ice[1-7]

Valid values: <true>, <false>

Default: false

Description: Offer ICE <true> or not <false> associated with line x.

user_ringer[1-7]

Valid values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>

Default: Ringer1

Description: The ring tone to be used if a call comes to that specific line. See "ring_source" also.

user_moh[1-7]

Valid values: e.g. <sip:mh@snom.de>, <mh>, <192.168.0.40>

Default: blank

Description: SIP Address of music on hold server of that specific line. The music is automatically played if this line is on hold.

user_stream[1-7]

Valid values: e.g. <http:// www.radioxyz.de>, <http: //192.168.0.40> Default: blank

Description: HTTP URL of a music streaming server of that specific line. The music is automatically played if this line is on hold.

user_srtp[1-7]

Valid values: <true>, <false>

Default: false

Description: Use SRTP for audio connections <true> or don't <false>. This option is valid for snom190 only.

active_line

Valid values: [1-7]

Default: 1

Description: Number of the active SIP line. This is the line that is used as the originator of an outgoing call.

SIP Stack Settings

sip_postfix

Valid values: e.g. <proxy.company.de>

Default: blank

Description: Register postfix, which is automatically chosen as registrar if the logon wizard is used to logon an SIP user and the registrar has not been set up through the logon wizard.

user_outbound[1-7]

Valid values: e.g. <proxy.company.de>, <192.168.0.24>

Default: blank

Description: Address or path of the outbound proxy per SIP line that is used for calling.

route_proxy

Valid values: <true>, <false>

Default: false

Description: Treat as initial route (<true>), <false> for address only

nat_detection

Valid values: <auto>, <upnp>, <stunclient>, <static>, <off>
Default: auto
Description: NAT detection according to your network.

stun_server

Valid values: e.g. <217.115.141.99: 5062>
Default: blank
Description: IP address of a STUN server (hostname:port) if you have one.

stun_binding_interval

Valid values: Integer values e.g. 5, 10.
Default: blank
Description: STUN binding interval in seconds.

rtp_port_start

Valid values: Valid port numbers.
Default: blank
Description: First dynamic RTP port

rtp_port_end

Valid values: Valid port numbers.
Default: blank
Description: Last dynamic RTP port

network_id_name

Valid values: Valid name.
Default: blank
Description: Network identity (hostname)

network_id_port

Valid values: Valid port number.
Default: blank
Description: Network identity (port)

use_nw_port

Valid values: <true>, <false>
Default: false
Description: Use local SIP port (<false>) or network port (<true>)

sip_retry_t1, sip_retry_t2

Valid values: Integer values, e.g. <500>.
Default: 500, 4000
Description: The time for resending SIP messages in milliseconds. They should be set to 500 and 4000 respectively.

session_timer

Valid values: Integer values, e.g. <2400>, <3200>.
Default: 3600
Description: Default time for session timer in seconds. 0 disables the session timer, 3600 is a reasonable value.

dirty_host_ttl

Valid values: Integer values, e.g. <60>, <120>.
Default: 0
Description: Time in seconds until the phone waits before it repeats a non answered request again.

max_forwards

Valid values: Integer values, e.g. <40>, <60>.
Default: 70
Description: The maximum number of hops allowed for an SIP request/ response.

enum_suffix

Valid values: e.g. <e164.arpa>
Default: e164.arpa
Description: Route domain for ENUM,

see FAQ “ENUM on snom phones”

tcp_threshold

Valid values: <auto>, <tcp>, <udp>

Default: auto

Description: Set transport layer for SIP messages either to TCP, UDP or automatic.

trace

Valid values: <true>, <false>

Default: true

Description: Enable tracing output of the current protocol stack.

logon_wizard

Valid values: <true>, <false>

Default: false

Description: Logon wizard and mobility features available in snom key menu.

user_phone

Valid values: <true>, <false>

Default: true

Description: Use user=phone in SIP signalling.

use_mapped

Valid values: <true>, <false>

Default: true

Description: Use mapped address.

publish_presence

Valid values: <true>, <false>

Default: false

Description: Publish presence information on <true>.

refer_policy

Valid values: <auto>, <refer>, <bye>

Default: Auto

Description: Choose between REFER and BYE/Also method for initiating a transfer. Auto makes an automatic selection depending on the capabilities indicated by the other party.

refer_brackets

Valid values: <true>, <false>

Default: false

Description: In case you want to have your REFER framed with brackets set it to <true>.

require_prack

Valid values: <true>, <false>

Default: true

Description: In case PRACK is required, set to <true>, else <false>.

symmetrical_rtp

Valid values: <true>, <false>

Default: true

Description: Switch off symmetrical RTP support with <false>, else <true>.

offer_gruu

Valid values: <true>, <false>

Default: true

Description: This setting is used to toggle the support for GRUU (Globally Routable User agent URL’s) in SIP.

offer_mpo

Valid values: <true>, <false>

Default: false

Description: Using this setting the user can turn the Media Path Optimization on or off.

challenge_dialog

Valid values: <true>, <false>

Default: false

Description: Turning this setting on <true> enables a challenge response for dialog messages.

challenge_reboot

Valid values: <true>, <false>

Default: false

Description: This setting enables/disables a challenge response for remote reboot request.

challenge_checksync

Valid values: <true>, <false>

Default: false

Description: Turning this setting on <true> enables a challenge response for Check-Sync.

H.323 Settings

user_h323_e164_number1

Valid values: e.g. <3884576>

Default: Blank

Description: E164 number assigned to the phone.

user_h323_id1

Valid values: e.g. <kitchen>

Default: blank

Description: H.323 ID assigned to the phone.

user_h323_url_id1

Valid values: e.g. tx@company.de

Default: blank

Description: H.323 URL ID assigned to the phone.

user_gkgw_mode1

Valid values: <auto>, <gatekeeper>, <gateway>, <none>

Default: auto

Description: Gatekeeper or gateway mode. Can be <auto> (automatic discovery of Gatekeeper) or <gatekeeper> for explicitly using the gatekeeper mode, <gateway> for gateway mode and <none> for autonomous operation.

user_h323_gateway1

Valid values: e.g. <192.168.0.9>, <gtk.company.de>

Default: blank

Description: IP address of the gatekeeper or the gateway, depending on gkgw_mode (see above). This field will be automatically set to the discovered gatekeeper's IP Address if <auto> mode is selected in "gkgw_mode".

user_h323_ttl1

Valid values: e.g. <600>

Default: 600

Description: Time-to-live for Gatekeeper registration. Very low values result in more frequent refreshes of registration but also cause additional network traffic. Very high values result in large intervals of disconnection from the Gatekeeper before getting back online if the gatekeeper reboots without sending any un-registration request to the phone.

user_early_start1

Valid values: <true>, <false>

Default: true

Description: The early start option allows the H.245 channel to be set up earlier, and hence speeds up the call setup. When early start is turned on, the H.245 channel address is also supplied in the Setup message. This speeds up the call because the H.245 channel negotiations can proceed in parallel to H.225.

user_fast_start1

Valid values: <true>, <false>

Default: false

Description: Fast start sends encoded OpenLogicalChannel messages within the Setup messages and hence eliminates the need for a separate H.245 channel for Codec capability negotiation. This will only work if the other party is also supporting fast start. This feature in H.323 is optional, meaning it can be ignored by an endpoint not supporting this feature, hence making the snom phone revert to the normal H.245 signalling on a separate channel. Note that if Fast Start is being used, Out of band DTMF might not work because the DTMF info is relayed on the H.245 channel.

user_h323_h245_tunneling1

Valid values: <true>, <false>

Default: false

Description: H.245 Tunnelling option sends tunnelled H.245 messages within an H.225 pdu. This also accelerates connection setup if both parties are supporting this feature.

trace

Valid values: <true>, <false>

Default: true

Description: Enable tracing output of the current protocol stack.

user_h323_h4501

Valid values: <true>, <false>

Default: false

Description: If flag is set to <true>, H.450 supplementary services are used, otherwise (<false>) it is facility-based.

Gateway prefixes (H.323 only)

user_external_line_prefix1

Valid values: e.g. <0>, <111>

Default: blank

Description: Prefix used for reaching outside numbers via an H323 gateway. This prefix will be replaced by the gateway_prefix in a dialled number for calling outside numbers (see gateway_prefix).

user_gateway_prefix1

Valid values: e.g. <*5>

Default: blank

Description: Prefix with which an ISDN gateway is registered at the gatekeeper. The phone will replace this prefix with the external_line_prefix in a dialled number if it is starting with the external_line_prefix (see external_line_prefix).

Hence the user doesn't have to dial numbers like 5*1234567 but can instead dial 01234567 with the same effect.

PBX key codes (H.323 only)

key_seq_hold_on

Valid values: For example <*1>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and the "R" (hold on) key is pressed on the phone.

key_seq_hold_off

Valid values: For example <*2>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and the "R" (hold off) key is pressed on the phone.

key_seq_transfer_blind

Valid values: e.g. <*60#>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and the "Xfer" key is pressed on the phone for a blind transfer.

key_seq_transfer_consultation

Valid values: e.g. <*6>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and the "Xfer" key is pressed on the phone for a consultation transfer. The phone needs to have 2 active calls in order to use this feature.

key_seq_divert

Valid values: For example <*3>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and redirection is active with a redirection number.

key_seq_conf_on

Valid values: For example <*3>

Default: blank

Description: Key code that is sent by the phone to the PBX if it is set and the "Cnf.On" key is pressed on the phone.

key_seq_conf_off

Valid values: e.g. <*30>

Default: blank

Description: Key code that is sent by the phone to the PBX if its set and the "Cnf.Off" key is pressed on the phone.

key_seq_pickup

Valid values: e.g. <#11>

Default: blank

Description: Key code that is sent by the phone to the PBX to pick up an alerting call. This is activated by pressing the 'pickup' key when the key_seq_pickup is set.

key_seq_end

Valid values: e.g. <#>

Default: blank

Description: The end symbol that is automatically appended to the above key codes before they are sent to the PBX. The end symbol marks the end of a key code sequence.

key_single_call_mode

Valid values: <true>, <false>

Default: true

Description: Set to <true> the phone simulates a simple analog phone which is able to handle one call in parallel only.

Codec and DTMF settings

codec[1-5]_name[1-7]

Valid values: 0 (ulaw), 8 (alaw), 3 (gsm), 2 (G.726-32), 18 (G.729) (also 9 (G.722), 4 (G.723) for snom190 only)

Default: 0, 8, 3, 2, 18

Description: Name of the codecs for line1-7 to be used in preferred order 1-5.

codec_size[1-7]

Valid values: <80>, <160>, <240>, <320>

Default: 160

Description: Packet size in bytes. Affects only ulaw and alaw codecs (other codecs have fixed packet size). 80 (10 ms), 160 (20 ms), 240 (30 ms) and 320 (40 ms) bytes are available per line.

dtmf_type_outofband

Valid values: <true>, <false>

Default: true

Description: Options for Out of band DTMF. <true> forces the use of Out of band DTMF, <false> turns it off. Its used for H.323 only !

dtmf_type_inband

Valid values: <true>, <false>

Default: on

Description: Options for In band DTMF. <true> forces the use of In band DTMF, <false> turns it off. Its used for H.323 only !

dtmf_payload_type

Valid values: Integer values, e.g. <100>, <150>.

Default: 101

Description: Payload type for Out of band DTMF.

Base Network Settings

ip_adr

Valid values: e.g. <192.168.0.50>

Default: Blank

Description: The IP address of the device. This parameter is mandatory in order to enable the ethernet connection.

netmask

Valid values: e.g. <255.255.255.0>

Default: Blank

Description: The netmask for the device. This parameter is mandatory in order to enable the ethernet connection.

phone_name

Valid values: e.g. <phone1>

Default: Blank

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

gateway

Valid values: e.g. <192.168.0.1>, <10.0.0.1>

Default: Blank

Description: The IP address of the default IP gateway (not the VoIP gateway!!!). It is the address to which the packets get routed if the wanted packet address is not in the current subnet. Setting up this parameter is mandatory in order to reach an external network.

dhcp

Valid values: <true>, <false>

Default: true

Description: Use DHCP (<true>) or do not use DHCP (<false>).

timezone

Valid values: <USA-10>, <USA-9>, <CAN-8>, <MEX-8>, <USA-8>, <CAN-7>, <MEX-7>, <USA-7>, <CAN-6>, <CHL-6>, <MEX-6>, <USA-6>, <BHS-5>, <CAN-5>, <CUB-5>, <USA-5>, <CAN-4>, <CHL-4>, <PRY-4>, <BMU-4>, <FLK-4>, <CAN-3.5>, <GRL-3>, <PRT-1>, <FRO-0>, <IRL-0>, <PRT-0>, <ESP-0>, <GBR-0>, <ALB+1>, <AUT+1>, <BEL+1>, <CAI+1>, <CHA+1>, <HRV+1>, <CZE+1>, <DNK+1>, <FRA+1>, <GER+1>, <HUN+1>, <ITA+1>, <LUX+1>, <MAK+1>, <NLD+1>, <NAM+1>, <NOR+1>, <POL+1>, <SVK+1>, <ESP+1>, <SWE+1>, <CHE+1>, <GIB+1>, <YUG+1>, <BLR+2>, <BGR+2>, <CYP+2>, <EGY+2>, <EST+2>, <FIN+2>, <GAZ+2>, <GRC+2>, <ISR+2>, <JOR+2>, <LVA+2>, <LBN+2>, <MDA+2>, <RUS+2>, <ROU+2>, <SYR+2>, <TUR+2>, <UKR+2>, <IRQ+3>, <RUS+3>, <IRN+3.5>, <ARM+4>, <AZE+4>, <GEO+4>, <KAZ+4>, <RUS+4>, <KAZ+5>, <KGZ+5>, <PAK+5>, <RUS+5>, <IND+5.5>, <KAZ+6>, <RUS+6>, <CHN+7>, <RUS+7>, <SGP+7>, <KOR+8>, <JPN+9>, <AUS+9.5>, <AUS+10>, <RUS+10>, <AUS+10.5>, <NZL+12>, <RUS+12>, <NZL+12.75>, <TON+13>

Default: Blank

Description: Timezone string, if set, it sets utc_offset and dst automatically !

utc_offset

Valid values: e.g. in Germany it is <+3600>.

Default: Blank

Description: Signed UTC offset in seconds. See timezone also.

dst

Format: offset mm.ww.dd hh:mm:ss mm.ww.dd hh:mm:ss

Valid values: e.g. for Germany <3600

03.05.07 02:00:00 10.05.07 03:00:00>

Default: Blank

Description: Daylight saving time that observes yearly change and leap years. See timezone also.

ntp_server

Valid values: e.g. <192.53.103.103>

Default: 192.53.103.103

Description: Address of the NTP time server.

Advanced Network Settings

dns_domain

Valid values: e.g. <company.com>

Default: Blank

Description: The DNS domain. This parameter is mandatory in order to enable DNS searching.

dns_server1, dns_server2

Valid values: e.g. <194.25.2.129>

Default: Blank

Description: Server that may be used for DNS searches. Setting up one of these parameters is mandatory in order to enable DNS searching.

http_user, http_pass

Valid values: character strings, e.g. <john>, <jh24>

Default: Blank

Description: Username and password for accessing the embedded webserver on the phone.

http_proxy

Valid values: IP address or url

Default: Blank

Description: Sets the http proxy for outgoing http requests.

http_port

Valid values: e.g. <80>, <8080>

Default: 80

Description: Sets the port for http requests.

https_port

Valid values: e.g. <80>, <8080>

Default: 443

Description: Sets the port for https requests.

webserver_type

Valid values: <http>, <https>, <http_https>, <off>

Default: http_https

Description: Sets the connection type of http requests which are allowed.

webserver_cert

Valid values: base 64 encoded certificate along with the private key

Default: empty

Description: Certificate send from the phone to the browser in case of https.

register_http_contact

Valid values: <true>, <false>

Default: false

Description: Should the phone during a registration request send its own URL via which its web interface can be accessed to the registrar <true> or don't <false>.

http_scheme

Valid values: <true>, <false>

Default: false

Description: Type of authentication scheme used for web interface: Basic <false>, Digest <true>

register_http_contact

Valid values: <true>, <false>

Default: false

Description:

lcserver1

Valid values: e.g. <192.168.0.26>

Default: Blank (to use local LCServer on device)

Description: IP addresses of an alternative remote LCServer. You will not usually need to change this setting, its for snom internal use only.

vlan

Valid values: integers separated by a space, e.g. <128 5>

Default: Blank

Description: VLAN ID (0..4095) and Priority (0..7) separated by a space

Update

update_policy

Valid values: <auto_update>, <ask_for_update>, <never_update_firm>, <never_update_boot>, <settings_only>, <never_update>

Default: settings_only

Description: Determines the update behaviour of the phone. <never_update> does not connect a server regarding settings or updates at all, <settings_only> loads settings only from settings server. <auto_update> does not ask again, if you are really sure to update.

firmware_interval

Valid values: amount in minutes

Default: 1440 (24*60)

Description: The phone polls by default each 24 hours for a new firmware. If you want to change that, it has to be set to the amount of minutes after a new poll request should be performed.

setting_server

Valid values: e.g. <http://www.company.com/settings/snom200.htm>, <130.149.12.54>

Default: blank

Description: URL of the settings file.

firmware_status

Valid values: e.g. <http://www.company.com/settings/snom200-firmware.htm>

Default: Blank

Description: URL of the config file that consists of the bootloader and firmware setting, see below.

bootloader, firmware

Valid values: e.g. <http://www.company.com/files/snom200ABC.bin>, <130.149.12.54>

Default: Blank

Description: URL of the image file for bootloader or firmware respectively.

subscribe_config

Valid values: <true>, <false>

Default: false

Description: In case you want to subscribe to the settings send by the proxy use <true> otherwise (<false>) the settings were retrieved via setting files.

Miscellaneous

preselection_nr

Valid values: Integer values & '+' e.g. <001>, <+491234567>

Default: blank

Description: If a number is entered in this option, the phone dials this pre-selected number automatically every time the phone is taken off the hook. This is particularly useful for using calling/prepaid cards etc.

mute

Valid values: <true>, <false>

Default: false

Description: Mute <true> the

microphone.

disable_speaker

Valid values: <true>, <false>

Default: false

Description: Disable <true> casing speaker.

headset_device

Valid values: <none>, <external>, <headset_rj>

Default: none

Description: Select wether you want to use a headset with pc or rj connector or not.

mwi_notification

Valid values: <silent>, <beep>, <reminder>

Default: silent

Description: Do you want to get reminded that you have messages waiting or don't you ?

release_sound

Valid values: <true>, <false>

Default: true

Description: Usually the phone plays a short sound after the other party ended the call. You can switch that off with <false>.

Other Settings

DND_mode

Valid values: <true>, <false>

Default: False

Description: <true> means that the phone is in "do not disturb" (DND) mode, <false> is normal behavior.

edit_alpha_mode

Valid values: <123>, <abc>, <ABC>, <kana>

Default: <123>

Description: Preferred or default editing mode for typing in phone numbers/ addresses unless the user changes it by dialing a different type of phone number.

tone_scheme

Valid values: <AUS>, <CHN>, <GER>, <IND>, <JPN>, <MEX>, <GBR>, <USA>

Default: blank

Description: Country specific scheme of phone audio signals, like free tone, busy tone etc.

user_ringer[1-7]

Valid values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>

Default: Ringer1

Description: Default selection of the ring tone style per SIP line that signals incoming calls. See "ring_source" also.

family_ring_sound, friends_ring_sound, colleagues_ring_sound

Valid values: <Ringer1>, <Ringer2>, <Ringer3>, <Ringer4>, <Ringer5>, <Ringer6>, <Ringer7>

Default: Ringer1

Description: Phonebook contact type specific ringers. Selection of the ring tone style that signals incoming calls dependend on the contact type of the caller in the local phonebook. See also "ring_source".

ring_source

Valid values: <source>, <destination>

Default: source

Description: Defines if the local phone (destination) or the remote phone (source) determines the ring tone style on an incoming call.

ringing_time

Valid values: Between <0> and <86400>

Default: 60

Description: Time in seconds how long an incoming call should ring before the phone denies him.

vol_speaker

Valid values: Between <0> and <15>

Default: 8

Description: Selection of the casing speaker volume.

vol_headset

Valid values: Between <0> and <15>

Default: 8

Description: Selection of the headset speaker volume.

vol_handset

Valid values: Between <0> and <15>

Default: 8

Description: Selection of the handset speaker volume.

auto_connect

Valid values: <true>, <false>

Default: False

Description: If it is <true>, the phone will automatically answer incoming calls.

auto_connect_type

Valid values: <auto_connect_type_handsfree>, <auto_connect_type_handset>

Default: auto_connect_type_handsfree

Description: If the above setting auto_connect is <true>, auto_connect_type determines whether the auto-answered incoming call switches the phone to handsfree mode or normal handset mode.

log_level

Valid values: Between <0> and <9>

Default: 5

Description: Log level of the maintenance web page, 9 is the most verbose mode.

Key Mapping

fkey[0-4] ([0-64] in case of snom220 with up to 3 extension boards)

Valid values: <line>, <dest>, <icom>, <orbit>, <recorder>, <dtmf>, e.g. "dest kd@company.org"

Default: line

Description: Changes the meaning of the programmable keys. See the user/admin manual for more info. <dtmf> is for H323 only.

break_key

Valid values: <true>, <false>

Default: true

Description: Change the behaviour of the key below the casing speaker key to either break <true> or transfer <false> functionality, valid for snom200 only.

no_dnd

Valid values: <true>, <false>

Default: false

Description: Switch off the DND functionality at all with <true>.

Privacy

privacy_in

Valid values: <true>, <false>

Default: false

Description: Reject <true> or accept <false> anonymous incoming calls.

privacy_out

Valid values: <true>, <false>

Default: false

Description: Show <false> or hide <true> your own phone number on outgoing calls.

presence_timeout

Valid values: Between <0> and above

Default: 15

Description: 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. In other words, presence is disabled for all practical purposes.

Action URLs

action_dnd_on_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here DND has been switched on), a web GET to the specified URL is performed.

action_dnd_off_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here DND has been switched off), a web GET to the specified URL is performed.

action_redirection_on_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here redirection always has been activated), a web GET to the specified URL is performed.

action_redirection_off_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here redirection always has been deactivated), a web GET to the specified URL is performed.

action_incoming_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here an incoming call is ringing), a web GET to the specified URL is performed.

action_outgoing_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here an outgoing call has been started to dial out), a web GET to the specified URL is performed.

action_setup_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here the end of the setup function has been reached and the phone is setup), a web GET to the specified URL is performed.

action_offhook_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here the handset was lifted from the hook switch), a web GET to the specified URL is performed.

action_onhook_url

Valid values: HTTP URL

Default: empty

Description: In case the specific action has taken place (here the handset was put on the hook switch), a web GET to the specified URL is performed.

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