

# AREDN: Snom Setup

## Direct Call / Peer to Peer SIP Calls

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## Devices

This guide was written based on a Snom D120 with firmware version 10.1.54.24.

It was confirmed working for the following other Snom phone / firmware combinations:

-

## Setup

**General advice:** Apply/save the changes after each step. This is not mentioned separately.

## Identity

There are two potential ways to set this up:

### 1. Run the phone without a registration

- Downside: The phone keeps displaying a welcome message instead of a nicer status display once the phone is registered. It will still place and receive phone calls though.

### 2. Set up an actual SIP server or SIP Proxy

- Note: A fake server like [SipServer](#) does **not** suffice.
- Downside: This is a more complex setup and depends on the SIP server.

**Rationale:** When the Snom phone places a call, as long as a registration is active, it will send the SIP INVITE call message to the SIP Server even if the target is an identity not registered at the same SIP server.

The rest of this section is accordingly split up to describe the two approaches. Pick one and follow only that guidance. We suggest using *Approach 1* for now.

## Approach #1: Login

In "Identity 1", make at least the following changes:

- Identity active: off

**Note:** The phone will keep displaying the "Welcome! Press a key to log on." message. If you know how to turn this off, let us know. The phone can still be used to dial from the directory and can be called. However, the display is somewhat useless. Do **not** log on on the phone itself as that will activate the identity and thus outgoing calls won't work anymore.

**Configuration Identity 1** snom

[Logout](#)

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**Login** Features SIP NAT RTP Audio

**Login Information**

Identity active ☐ on ☒ off ?

Displayname 870831 (HB9GVM) ?

Account 870831 ?

Password \*\*\*\*\* ?

Registrar localnode.local.mesh ?

Outbound Proxy ?

Failover Identity None ?

Hidden Identity ☐ on ☒ off ?

Authentication Username ?

Mailbox ?

Mailbox Dial-in (if different from Mailbox) ?

Conference Server ?

Display Text for Idle Screen ?

Display Text for Call Forwarding Target ?

[Apply](#) [Re-Register](#)

[Remove Identity](#) [Remove All Identities](#)

## Approach #2: Login

Note: This assumes that you have already set up a SIP Server or Proxy on your AREDN Node. Note that this has not been tested fully (testing only with [SipServer](#) which permitted incoming calls to work but outgoing calls did not function).

In "Identity 1", make at least the following changes:

- Identity active: on
- Account: <your phone number>
- Registrar: localnode.local.mesh

# Configuration Identity 1

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Login

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**Login Information**

Identity active ☒ on ☐ off ?

Displayname 870831 (HB9GVM) ?

Account 870831 ?

Password \*\*\*\*\* ?

Registrar localnode.local.mesh ?

Outbound Proxy ? ?

Fallover Identity None v ?

Hidden Identity ☐ on ☒ off ?

Authentication Username ? ?

Mailbox ? ?

Mailbox Dial-in (if different from Mailbox) ? ?

Conference Server ? ?

Display Text for Idle Screen ? ?

Display Text for Call Forwarding Target ? ?

[Apply](#)
[Re-Register](#)

[Remove Identity](#)
[Remove All Identities](#)

## SIP

In the "SIP" tab, make at least the following changes:

- Support Broken Registrar: on  
Otherwise calls directed at "<number>@<number>.local.mesh" will not be accepted because the phone is registered at "<number>@localnode.local.mesh".

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SIP Identity Settings

Voice Quality Report Collector

Music on Hold Server

Send Hold as Inactive

Alert Info URL

User Picture URL

Dial Plan String

Count all Groups in Dial Plan

ENUM Support

Country Code

Area code

Proxy Require

Additional Supported Headers

Contact Source Priority (SIP)

Q-Value

Proposed Expiry

Auto Answer

Long SIP-Contact (RFC3840)

Support Broken Registrar

Shared Line

Publish Presence on Bootstrap

DTMF via SIP INFO

Extension Monitoring & Call Pickup List URI

Contact List

?

?

on off ?

?

?

?

on off ?

on off ?

?

?

?

?

PAI RPID FROM

1.0 ?

3600

on off ?

on off ?

on off ?

Off ?

?

on off ?

## Advanced

### SIP/RTP

In the "SIP/RTP" tab, make at least the following changes:

- Network Identity (Port): 5060  
[https://service.snom.com/display/wiki/network\\_id\\_port](https://service.snom.com/display/wiki/network_id_port)
- Listen on SIP TCP Port: on  
[https://service.snom.com/display/wiki/tcp\\_listen](https://service.snom.com/display/wiki/tcp_listen)

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SIP/RTP
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**SIP**

Network Identity (Port) 5060 ?

SIP T1 (ms) 500 ?

Timer Support (RFC4028) ☒ on ☐ off ?

SIP Session Timer (s) 3600 ?

SIP Dirty Host TTL (s) ? ?

SIP Max Forwards 70 ?

ENUM Suffix e164.arpa ?

Retry Interval after Failed Registration (s) 300 ?

Use user=phone ☒ on ☐ off ?

Require PRACK ☒ on ☐ off ?

Send PRACK ☒ on ☐ off ?

Offer GRUU ☒ on ☐ off ?

Offer MPO ☐ on ☒ off ?

Use Outbound ☐ on ☒ off ?

Use SIP Compact Headers ☐ on ☒ off ?

Listen on SIP TCP Port ☒ on ☐ off ?

Register HTTP Contact ☐ on ☒ off ?

Disable Blind Transfer (REFER) ☐ on ☒ off ?

Disable Deflection (Code 302) ☐ on ☒ off ?

Show History-Info ☒ on ☐ off ?

## QoS/Security

In the "QoS/Security" tab, make at least the following changes:

- Filter Packets from Registrar: off  
[https://service.snom.com/display/wiki/filter\\_registrar](https://service.snom.com/display/wiki/filter_registrar)

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Quality of Service

RTP Type of Service (TOS/Diffserv)160?

SIP Type of Service (TOS/Diffserv)160?

VLAN

VLAN Id (1-4094)?

VLAN Priority (0-7)?

Un-/Tag VLAN Traffic on Specific Switch Portson off?

PC Port

VLAN Id (1-4094)?

VLAN Priority (0-7)?

IEEE 802.1X Authentication:

Off?

User?

Password\*\*\*\*\*?

Security

Ignore Security Adviceson off?

Use Hidden Tagson off?

Restrict URI Querieson off?

Allow CSTA Controlon off?

Empty Client Certon off?

Filter Packets from Registraron off?

Authentication for SIP Rebooton off?

Authentication for SIP Check-Syncon off?

Administrator Modeon off?

## Phonebook / Directory

**Note:** This assumes that you have set up the phonebook as described in <https://github.com/dhamstack/AREDNstack/blob/main/Documentation/Installation%20of%20phonebook%20replication%20on%20hap%20router.pdf> and that you have enabled "snom" as an output in settings.txt.

Under "Function Keys", set the "Directory" button to type "Action URL" and set the URL to "[http://localnode.local.mesh/phonebook\\_snom\\_direct.xml](http://localnode.local.mesh/phonebook_snom_direct.xml)". This way, whenever the directory button is pressed, it will load the directory from the given URL and display it.

Function Keys

snom

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Key Settings

On this page you can specify the settings for programmable keys on your phone. Use Context to specify the identity context for that key e.g. this identity will be used to subscribe for a particular extension. Type will select the actual functionality of a particular key. In the last argument field Number, the actual telephone number, sip url, dtmf sequence, action url or key type can be stored. Please refer to your phone manual for more details.

Key Assignment

SmartLabel

SmartLabel Keys

Context	Type	Number	Full Label	XML Label
Active	Line			P1
Active	Key Event	Silent Mode		P2

Context-Sensitive Keys

Type	Number	Label
Key Event	Settings	F1
Key Event	Call History	F2
Call Forward		F3
Key Event	Help	F4

Navigation Keys

Type	
Settings	Up
Next Identity	Down
Redial	OK
Cancel	Cancel

Dedicated Keys

Type	Number	
Key Event	Voicemail	Voicemail
Key Event	DND	DND
Action URL	http://localhost.local	Directory
Transfer		Transfer
Key Event	Hold	Hold

## References

- Snom on peer to peer calling  
<https://service.snom.com/display/wiki/Can+I+establish+pure+IP%2C+Peer-to-peer+calls+with+snom+Phones>
- AREDN CH Documentation  
<https://github.com/dhamstack/AREDNstack/tree/main/Documentation>