

VoIP Einstellungen

Nachfolgend wird erklärt, wie man seinen HAMNET SIP Client für den Zugang zur HAMNET VoIP Telefonie einrichtet. Es wird nach Herstellern unterschieden, und die gängigsten Modelle aufgeführt.

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SNOM 300

Das SNOM ist ein günstiges und gutes SIP Telefon, und kann gebraucht schon für kleines Geld erworben werden (bspw. bei eBay).

Bevor man jedoch die Konfiguration vornimmt, sollte das Telefon auf den letzten Stand der Firmware gebracht werden! Anleitungen dazu findet man direkt im [SNOM Wiki](#).

Identity 1

[Login](#) [SIP](#) [NAT](#) [RTP](#)**Login Information:**

Identity active: ☒ on ☐ off ?

Displayname: ?

Account: ?

Password: ?

Registrar: ?

Outbound Proxy: ?

Failover Identity: ?

Authentication Username: ?

Mailbox: ?

Ringtone: ?

Custom Melody URL: ?

Display text for idle screen: ?

Ring After Delay (sec): ?

Record Missed Calls: ☒ on ☐ off ?

Record Dialed Calls: ☒ on ☐ off ?

Record Received Calls: ☒ on ☐ off ?

Identity is hidden: ☐ on ☒ off ?

[Apply](#) [Re-Register](#) [Play Ringer](#)[Remove Identity](#) [Remove All Identities](#)[Login](#) [SIP](#) [NAT](#) [RTP](#)**RTP Identity Settings:**

Codec: ?

Packet Size: ?

Filtered codec list:

Full SDP Answer: ☒ on ☐ off ?

Symmetrical RTP: ☐ on ☒ off ?

RTP Encryption: ☐ on ☒ off ?

G.726 Byte Order: ☒ RFC3551 ☐ AAL2 ?

SRTP Auth-tag: ☐ AES-32 ☒ AES-80 ?

RTP/SAVP: ?

Media Transport Offer: ?

Media Transport Offer Setup: ?

Multicast relay address: ?

[Apply](#)

Linphone



Im Reiter "Codecs" sollte kontrolliert werden, dass GMS, PCMA (alaw), PCMU (ulaw) aktiviert ist.

CSipSimple

Auch ein Android Smartphone kann als SIP-Client genutzt werden.

Da vom Server kein Passwort benötigt wird kann die eigene Rufnummer eingetragen werden.

Datei: csipsimple.png
CSipSimple

Grandstream 2020

Auch das Hardwaretelefon Grandstream 2020 (oder auch 2000) kann ebenfalls für kleines Geld erworben werden (bspw. eBay, Willhaben)

Die Konfiguration über das Webinterface für z.B. Account1 sollte wie folgt aussehen:

Grandstream Device Configuration					
STATUS	BASIC SETTINGS		ADVANCED SETTINGS		EXT 1
ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4	ACCOUNT 5	ACCOUNT 6
Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes					
Account Name: <input type="text" value="Hamnet"/> (e.g., MyCompany)					
SIP Server: <input type="text" value="voip.oe2xtr.ampr.at"/> (e.g., sip.mycompany.com, or IP address)					
Outbound Proxy: <input type="text"/> (e.g., proxy.myprovider.com, or IP address)					
SIP User ID: <input type="text" value="<Ham Tel.Nr>"/> (the user part of an SIP address)					
Authenticate ID: <input type="text" value="<Ham Tel.Nr>"/> (can be same or different from SIP UserID)					
Authenticate Password: <input type="text"/> (not displayed for security protection)					
Name: <input type="text" value="CALL"/> (optional, e.g., John Doe)					
Use DNS SRV: <input checked="" type="radio"/> No <input type="radio"/> Yes					
User ID is phone number: <input type="radio"/> No <input checked="" type="radio"/> Yes					
SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes					
Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes					
Support SIP Instance ID <input checked="" type="radio"/> No <input type="radio"/> Yes					
Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)					
local SIP port: <input type="text" value="5060"/> (default 5060)					
SIP Registration Failure Retry Wait Time: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)					
SIP T1 Timeout: <input type="text" value="1 sec"/>					
SIP T2 Interval: <input type="text" value="4 sec"/>					
SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP					
Use RFC3581 Symmetric Routing: <input type="radio"/> No <input checked="" type="radio"/> Yes					
NAT Traversal (STUN): <input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> Yes					
SUBSCRIBE for MWI: <input checked="" type="radio"/> No <input type="radio"/> Yes					
SUBSCRIBE for Registration Event: <input checked="" type="radio"/> No <input type="radio"/> Yes					
PUBLISH for Presence: <input checked="" type="radio"/> No <input type="radio"/> Yes					
Proxy-Require: <input type="text"/>					
Voice Mail UserID: <input type="text"/> (UserID for voice mail system)					
Send DTMF: <input checked="" type="checkbox"/> in-audio <input type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO					
Early Dial: <input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)					
Dial Plan Prefix: <input type="text"/> (this prefix string is added to each dialed number)					
BLF Call-pickup Prefix: <input type="text" value="**"/> (this prefix is prepended when answering call with BLF key)					
Delayed Call Forward Wait Time: <input type="text" value="20"/> (Allowed range 1-120, in seconds.)					
Enable Call Features: <input type="radio"/> No <input checked="" type="radio"/> Yes (if yes, call features using star codes will be supported locally)					
Call Log: <input checked="" type="radio"/> Log All Calls					
<input type="radio"/> Log Incoming/Outgoing only (Missed calls NOT recorded)					
<input type="radio"/> Disable Call Log					

Session Expiration:	<input type="text" value="180"/> (in seconds. default 180 seconds)
Min-SE:	<input type="text" value="90"/> (in seconds. default and minimum 90 seconds)
Caller Request Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)
Callee Request Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)
Force Timer:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)
UAC Specify Refresher:	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)
UAS Specify Refresher:	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)
Force INVITE:	<input checked="" type="radio"/> No <input type="radio"/> Yes (Always refresh with INVITE instead of UPDATE)
Enable 100rel:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Account Ring Tone:	<input checked="" type="radio"/> system ring tone <input type="radio"/> custom ring tone 1 <input type="radio"/> custom ring tone 2 <input type="radio"/> custom ring tone 3
Ring Timeout:	<input type="text" value="60"/> (in seconds. Between 30-3600, default is 60)
Send Anonymous:	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)
Anonymous Method:	<input checked="" type="radio"/> Use From Header <input type="radio"/> Use Privacy Header
Anonymous Call Rejection:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Auto Answer:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Auto Answer by Call-Info:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Turn off speaker on remote disconnect:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Check SIP User ID for incoming INVITE:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Refer-To Use Target Contact:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Disable Multiple Media Attribute in SDP:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Preferred Vocoder: (in listed order)	<div>choice 1: <input type="text" value="GSM"/></div> <div>choice 2: <input type="text" value="GSM"/></div> <div>choice 3: <input type="text" value="GSM"/></div> <div>choice 4: <input type="text" value="GSM"/></div> <div>choice 5: <input type="text" value="GSM"/></div> <div>choice 6: <input type="text" value="GSM"/></div> <div>choice 7: <input type="text" value="GSM"/></div> <div>choice 8: <input type="text" value="GSM"/></div>
SRTP Mode:	<input checked="" type="radio"/> Disabled <input type="radio"/> Enabled but not forced <input type="radio"/> Enabled and forced <input type="radio"/> Optional
eventlist BLF URI:	<input type="text"/>
Special Feature:	<input type="text" value="Standard"/>
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>	

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Wichtig ist ganz unten bei den verwendeten Codec's alles auf **GSM** zu setzen da sonst ein falscher Codec verwendet wird und man am Telefon nichts hört (es läutet, aber man hört den Gesprächsparten nicht)