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# 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

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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 auf den Screenshots aussehen. 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)

## Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	EXT 1	EXT 2
ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4	ACCOUNT 5
				ACCOUNT 6

**Account Active:** ☐ No ☒ Yes  
**Account Name:**  (e.g., MyCompany)  
**SIP Server:**  (e.g., sip.mycompany.com, or IP address)  
**Outbound Proxy:**  (e.g., proxy.myprovider.com, or IP address)  
**SIP User ID:**  (the user part of an SIP address)  
**Authenticate ID:**  (can be same or different from SIP UserID)  
**Authenticate Password:**  (not displayed for security protection)  
**Name:**  (optional, e.g., John Doe)  
**Use DNS SRV:** ☒ No ☐ Yes  
**User ID is phone number:** ☐ No ☒ Yes  
**SIP Registration:** ☐ No ☒ Yes  
**Unregister On Reboot:** ☒ No ☐ Yes  
**Support SIP Instance ID** ☒ No ☐ Yes  
**Register Expiration:**  (in minutes. default 1 hour, max 45 days)  
**local SIP port:**  (default 5060)  
**SIP Registration Failure Retry Wait Time:**  (in seconds. Between 1-3600, default is 20)  
**SIP T1 Timeout:**   
**SIP T2 Interval:**   
**SIP Transport:** ☒ UDP ☐ TCP  
**Use RFC3581 Symmetric Routing:** ☐ No ☒ Yes  
**NAT Traversal (STUN):** ☒ No ☐ No, but send keep-alive ☐ Yes  
**SUBSCRIBE for MWI:** ☒ No ☐ Yes  
**SUBSCRIBE for Registration Event:** ☒ No ☐ Yes  
**PUBLISH for Presence:** ☒ No ☐ Yes  
**Proxy-Require:**   
**Voice Mail UserID:**  (UserID for voice mail system)  
**Send DTMF:** ☒ in-audio ☐ via RTP (RFC2833) ☐ via SIP INFO  
**Early Dial:** ☒ No ☐ Yes (use "Yes" only if proxy supports 484 response)  
**Dial Plan Prefix:**  (this prefix string is added to each dialed number)  
**BLF Call-pickup Prefix:**  (this prefix is prepended when answering call with BLF key)  
**Delayed Call Forward Wait Time:**  (Allowed range 1-120, in seconds.)  
**Enable Call Features:** ☐ No ☒ Yes (if yes, call features using star codes will be supported locally)  
**Call Log:** ☒ Log All Calls  
☐ Log Incoming/Outgoing only (Missed calls NOT recorded)  
☐ 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)	choice 1: <input type="text" value="GSM"/> choice 2: <input type="text" value="GSM"/> choice 3: <input type="text" value="GSM"/> choice 4: <input type="text" value="GSM"/>	choice 5: <input type="text" value="GSM"/> choice 6: <input type="text" value="GSM"/> choice 7: <input type="text" value="GSM"/> choice 8: <input type="text" value="GSM"/>
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"/>	

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