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.

Derzeit gibt es folgende Asterisk-SIP Server:
voip.oe2xzr.ampr.at
voip.oe6xrr.ampr.at
voip.oe7xwi.ampr.at
voip.oe9xfr.ampr.at

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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:	OE2WAO
Account:	912163
Password:	••••••••
Registrar:	voip.oe2xzr.ampr.org
Outbound Proxy:	
Failover Identity:	Identity 2
Authentication Username:	912163
Mailbox:	912163
Ringtone:	Ringer 9
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

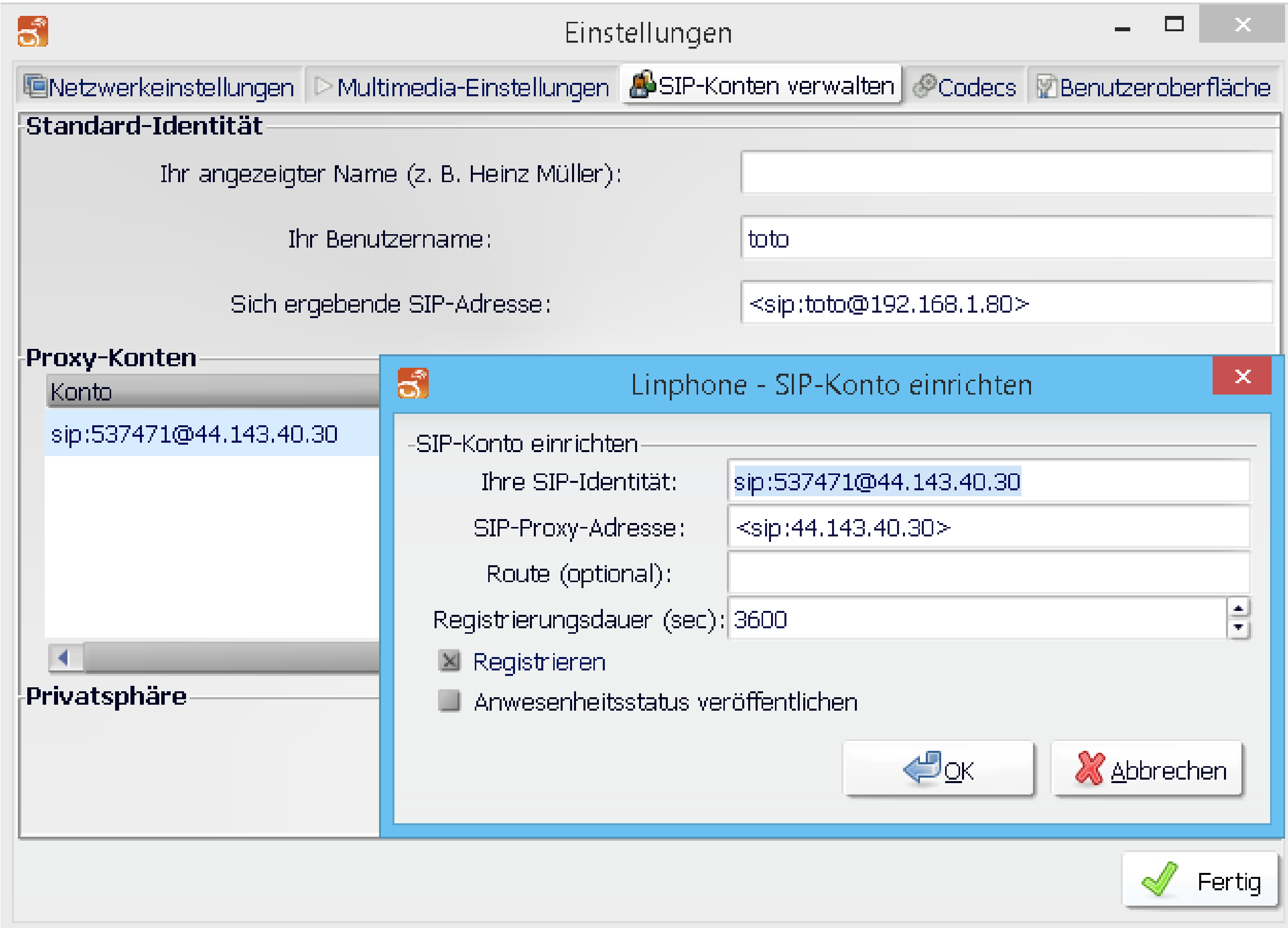
Login SIP NAT RTP

RTP Identity Settings:

Codec:	gsm,telephone-event
Packet Size:	20 ms
Filtered codec list:	gsm, telephone-event
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:	mandatory
Media Transport Offer:	UDP
Media Transport Offer Setup:	active
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.

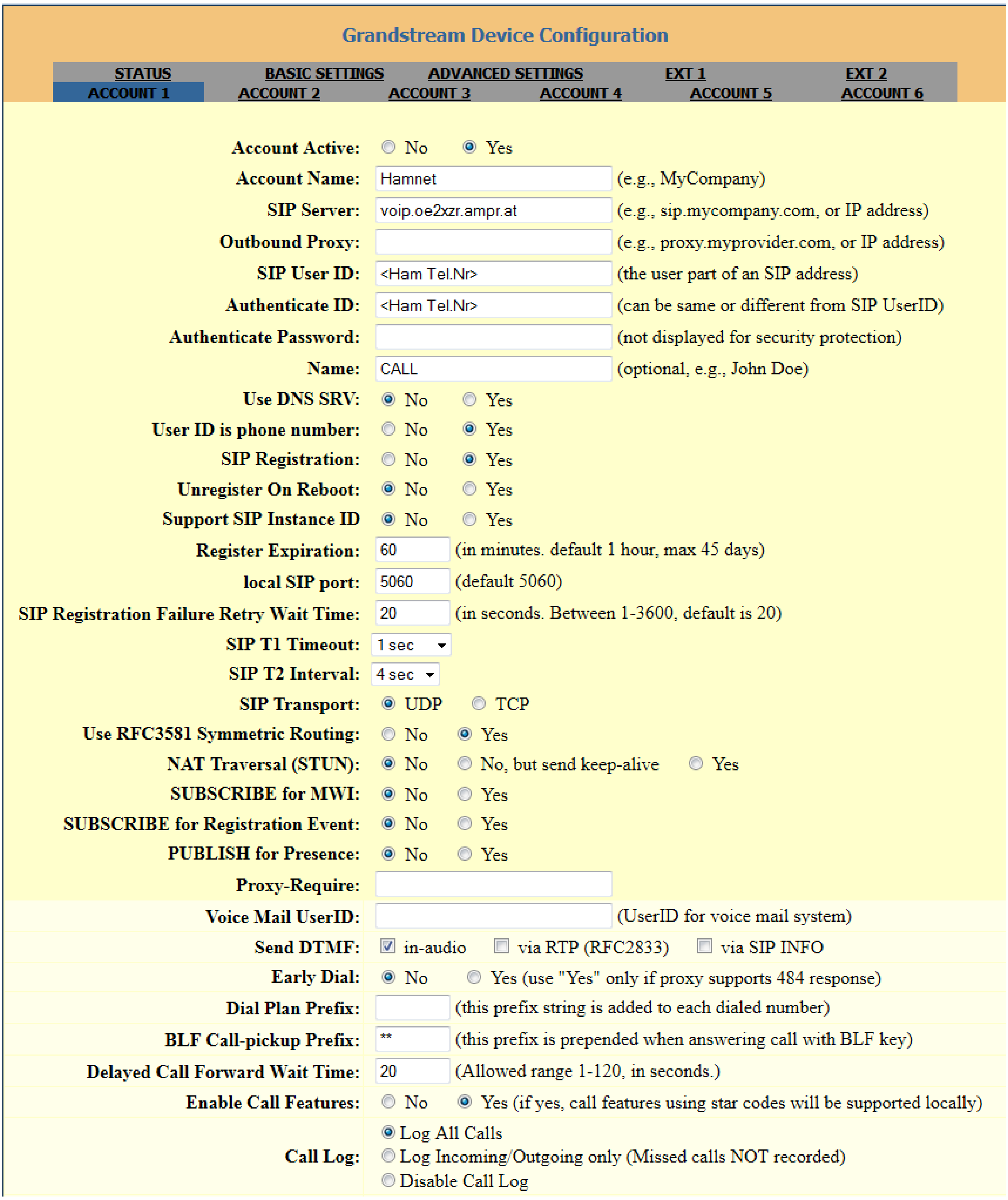
Da vom Server kein Passwort benötigt wird kann die eigene Rufnummer eingetragen werden.

Datei:csipsimple.png

CSipSimple

Grandstream 2020

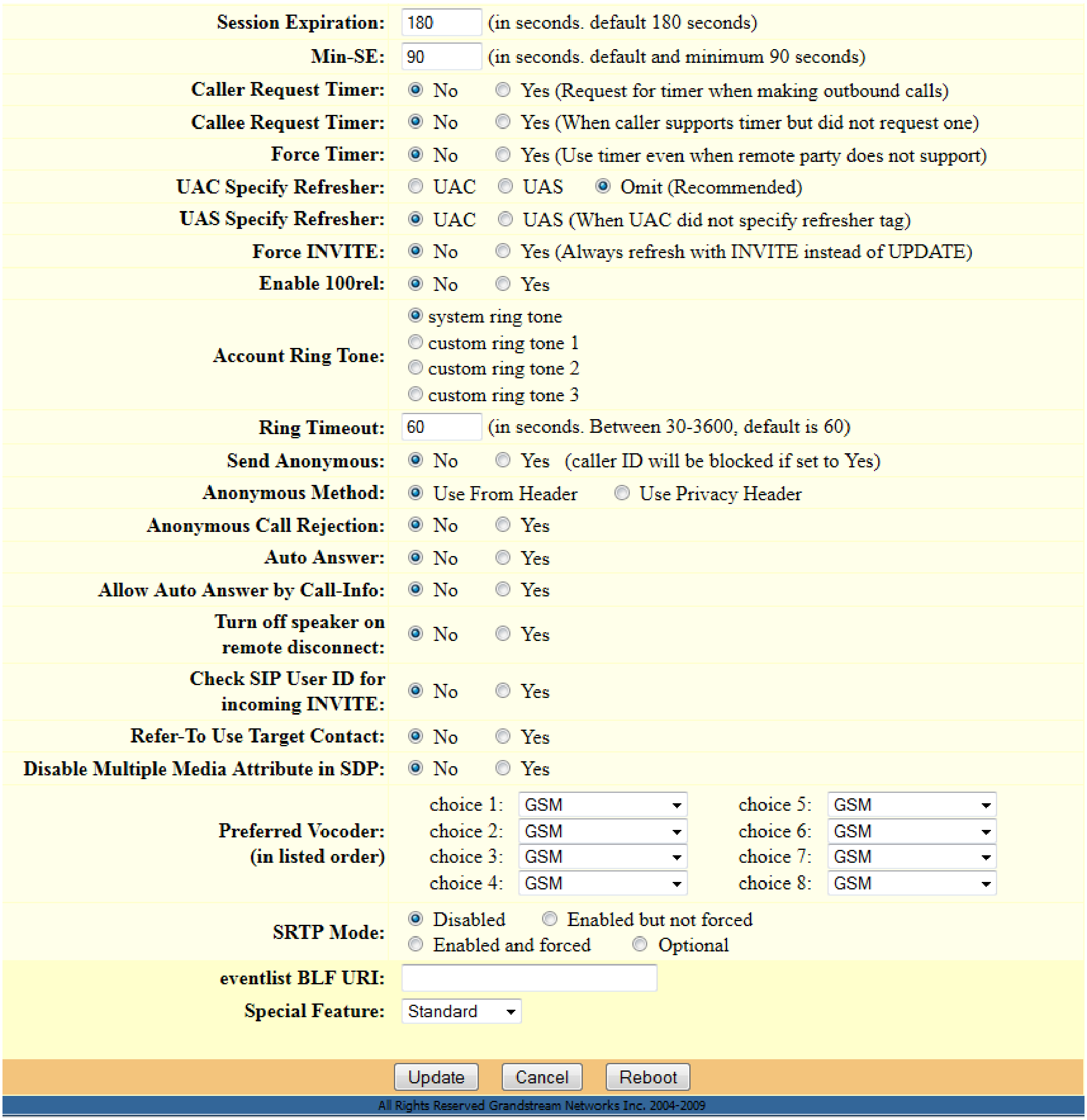
Auch das Hardwaretelefon Grandstream 2020 (oder auch 2000) kann ebenfalls für kleines Geld erworben werden (bspw. eBay, Willhaben)
Die Konfiguration über das Webinterface für z.B. Account1 sollte wie auf den Screenshots aussehen. Wichtig ist ganz unten bei den verwendeten Codec's alles auf **GSM** zu 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 (selected: Yes)
Account Name:	Hamnet (e.g., MyCompany)
SIP Server:	voip.oe2xzr.ampr.at (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	(e.g., proxy.myprovider.com, or IP address)
SIP User ID:	<Ham Tel.Nr> (the user part of an SIP address)
Authenticate ID:	<Ham Tel.Nr> (can be same or different from SIP UserID)
Authenticate Password:	(not displayed for security protection)
Name:	CALL (optional, e.g., John Doe)
Use DNS SRV:	No / Yes (selected: No)
User ID is phone number:	No / Yes (selected: Yes)
SIP Registration:	No / Yes (selected: Yes)
Unregister On Reboot:	No / Yes (selected: No)
Support SIP Instance ID	No / Yes (selected: No)
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
local SIP port:	5060 (default 5060)
SIP Registration Failure Retry Wait Time:	20 (in seconds. Between 1-3600, default is 20)
SIP T1 Timeout:	1 sec
SIP T2 Interval:	4 sec
SIP Transport:	UDP / TCP (selected: UDP)
Use RFC3581 Symmetric Routing:	No / Yes (selected: Yes)
NAT Traversal (STUN):	No / No, but send keep-alive / Yes (selected: No)
SUBSCRIBE for MWI:	No / Yes (selected: No)
SUBSCRIBE for Registration Event:	No / Yes (selected: No)
PUBLISH for Presence:	No / Yes (selected: No)
Proxy-Require:	
Voice Mail UserID:	(UserID for voice mail system)
Send DTMF:	in-audio (checked) / via RTP (RFC2833) / via SIP INFO
Early Dial:	No / Yes (use "Yes" only if proxy supports 484 response) (selected: No)
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:	20 (Allowed range 1-120, in seconds.)
Enable Call Features:	No / Yes (if yes, call features using star codes will be supported locally) (selected: Yes)
Call Log:	Log All Calls (selected) / Log Incoming/Outgoing only (Missed calls NOT recorded) / Disable Call Log


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