Aarenet System und IP-PBX Settings für den Anschluss von Innovaphone via SBC

Revision History

Autor	Datum	Info	Version
F.Remmers	22.09.2017	Initial Version	1.0

Alle Angaben ohne Gewähr und rein informativ. Die Einstellungen können je nach Aarenet System abweichen.

Inhalt

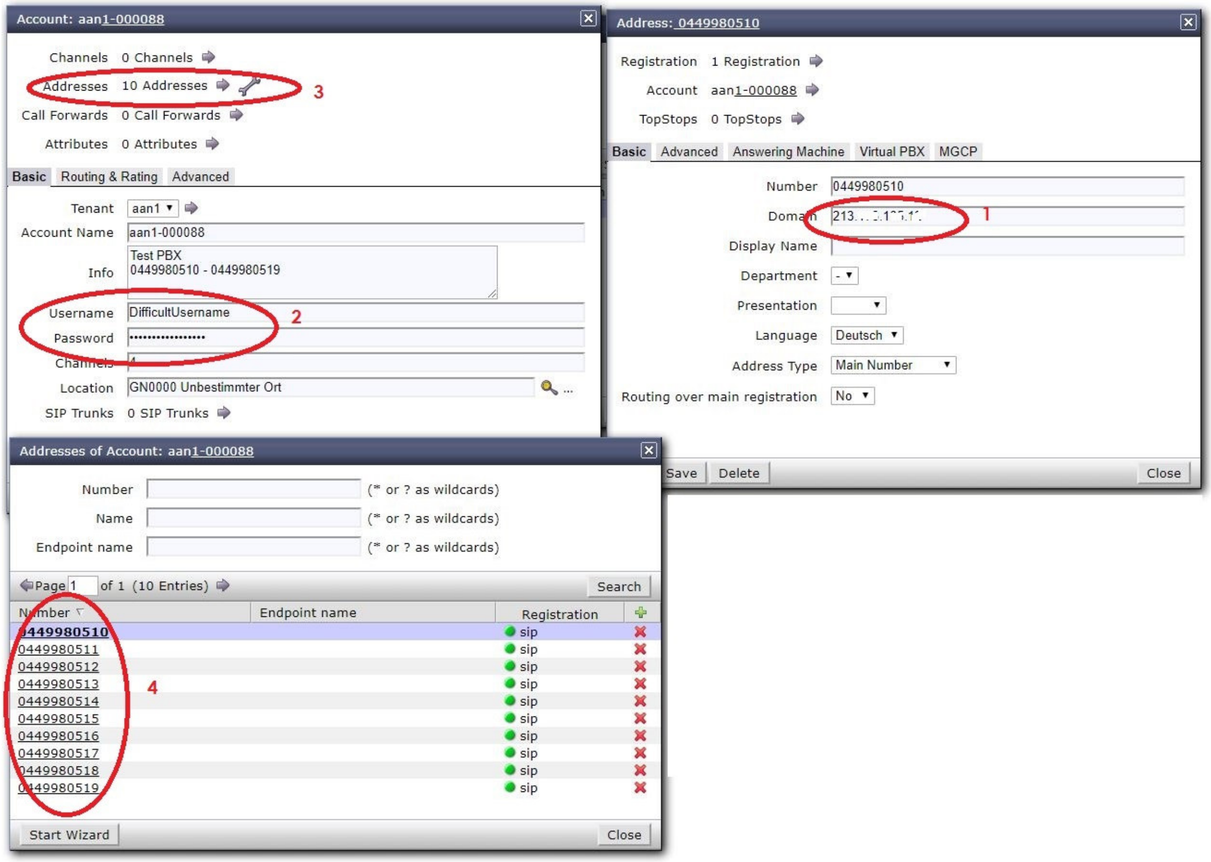
Basiseinstellungen – von Aarenet empfohlen

Die Einstellungen sind Endgeräte spezifisch und können (teils) auf IPPBX-en eingestellt werden. Bei abweichenden Werten können Verbindungsprobleme auftreten. Alle Angaben ohne Gewähr und rein informativ. Die Einstellungen können je nach Aarenet Systeme abweichen.

Empfohlene System / Endgeräte Einstellungen:

- Voice Codec G711A
- FAX Codec: G711A
- Hold Methode: direction attribute „sendonly“
- DTMF Methode: RFC 2833
- Session Timer: 1800s
- SIP Registration Expiry: 300s

Aarenet SIP Trunk Config Center Einstellungen



1. IP Adresse für SIP Anmeldung
2. Username / Passwort Info
3. Info 10 DDI Nummern
4. Liste mit den einzelnen DDI

Innovaphone setup Testresultate

Testresultate Swissnet: http://wiki.innovaphone.com/index.php?title=Howto:CH_-_swissnet_telecommunication_ag_-_swissnet_NxT_pro_SIP-Provider_%282017%29

Testresultate Quickline: http://wiki.innovaphone.com/index.php?title=Howto:CH_-_QUICKLINE_-_Office_Voice_SIP-Provider_%282017%29

Testresultate Sunrise:
https://wiki.innovaphone.com/index.php?title=Howto:Sunrise_Business_voice_-_SIP_Trunk_-_SIP_Provider_Compatibility_Test

ALG Funktion

Wichtig: ALG Funktion im NAT Router oder in der PBX ausschalten.

Testresults SIP Provider: Innovaphone V11 build 11.3127 on SBC

Source:
https://wiki.innovaphone.com/index.php?title=Howto:Sunrise_Business_voice_-_SIP_Trunk_-_SIP_Provider_Compatibility_Test

All information only informative. Technical support is only granted if tests have been performed and concluded together with Innovaphone and Aarenet.

Supported Features:

- Direct Dial In
- DTMF
- CGPN can be suppressed
- CLIP No Screening
- Hold/Retrieve
- Blind Transfer

Supported Codecs:

- G.711a, G.711u and in special cases G.729 only

This scenario describes a setup where the PBX and phones are in a private network. The SIP Trunk does not connect directly to the internet, but to the Gateway ("SBC").

The SIP trunk must be configured with "Media Relay", "exclusive G.711a" and "NO T.38".

Bold lines in the test results indicate a KO-criteria.

Basic Call

Tested feature	Result
call using g711a	**OK**
call using g711u	**OK**
call using g723	NA
call using g729	**OK** (*1)
call using g722	NA
Overlapped sending	NA
early media channel outgoing	**OK**
Fax using T.38	NA

T.38 Transcoding by the provider	NA
Fax using G.711	OK (*2)
Reverse Media Negotiation	NOK
CGPN can be suppressed	OK
CLIP no screening	OK
Long time call possible(>30 min)	**OK**
External Transfer	**OK**
NAT Detection	NA
Redundancy	OK (*3)
SIP over TCP	NA
Voice Quality OK?	**OK**

Direct Dial In

Tested feature	Result
Inbound(Provider -> Innovaphone)	**OK**
Outbound(Innovaphone -> Provider)	**OK**
Loop In call(Innovaphone -> Provider -> Innovaphone)	**OK**

DTMF

Tested feature	Result
DTMF tones sent correctly via RTP-events(RFC 2833)	**OK**
DTMF tones sent correctly via SIP-Info	OK
DTMF tones received correctly via RTP-events(RFC 2833)	**OK**

Hold/Retrieve

Tested feature	Result
Call can be put on hold	**OK**
Held end hears music on hold / announcement from PBX	OK

Transfer with consultation

Tested feature	Result
Call can be transferred	**OK**
Held end hears music on hold	OK

The following tests are made to test if call transfer is working.

Tested feature	Voice Ok?	MoH Ok?
inno1 calls inno2. inno2 transfers to PSTN-phone.	OK	OK
inno1 calls PSTN-phone. inno1 transfers to inno2.	OK	OK
inno1 calls PSTN-phone. PSTN-phone transfers to inno2.	OK	OK
PSTN-phone calls inno1. inno1 transfers to inno2.	OK	OK
PSTN-phone calls inno1. PSTN-phone transfers to inno2.	OK	OK
PSTN-phone calls inno1. inno1 transfers to other PSTN-phone-2.	OK	OK

Transfer with consultation (alerting only)

Tested feature	Result
Call can be transferred	**OK**
Held end hears music on hold or dialling tone	OK
Call returns to transferring device if the third Endpoint is not available	**OK**

The following tests are made to test if call transfer is working.

Tested feature	Voice Ok?	MoH Ok?
inno1 calls inno2. inno2 transfers to PSTN-phone.	OK	OK
inno1 calls PSTN-phone. inno1 transfers to inno2.	OK	OK
inno1 calls PSTN-phone. PSTN-phone transfers to inno2.	OK	OK
PSTN-phone calls inno1. inno1 transfers to inno2.	OK	OK
PSTN-phone calls inno1. PSTN-phone transfers to inno2.	OK	OK
PSTN-phone calls inno1. inno1 transfers to other PSTN-phone-2.	OK	OK

Blind Transfer

Tested feature	Result
Call can be transferred	OK
Held end hears dialling tone	OK

The following tests are made to test if call transfer is working.

Tested feature	Voice Ok?
inno1 calls inno2. inno2 transfers to PSTN-phone.	OK
inno1 calls PSTN-phone. inno1 transfers to inno2.	OK
inno1 calls PSTN-phone. PSTN-phone transfers to inno2.	OK
PSTN-phone calls inno1. inno1 transfers to inno2.	OK
PSTN-phone calls inno1. PSTN-phone transfers to inno2.	OK
PSTN-phone calls inno1. inno1 transfers to other PSTN-phone-2.	OK

CFU / CFB Transfer

Tested feature	Result
Call can be forward	**OK**
Held end hears dialling tone	**OK**

CFNR / Blind Transfer (alerting only)

Tested feature	Result
Call can be transferred or forward	**OK**
Held end hears dialling tone	**OK**

The following tests are made to test if call transfer is working.

Tested feature	Voice Ok?
inno1 calls inno2. inno2 transfers to PSTN-phone.	OK
inno1 calls PSTN-phone. PSTN-phone transfers to inno2.	OK
PSTN-phone calls inno1. inno1 transfers to inno2.	OK
PSTN-phone calls inno1. inno1 transfers to other PSTN-phone-2.	OK

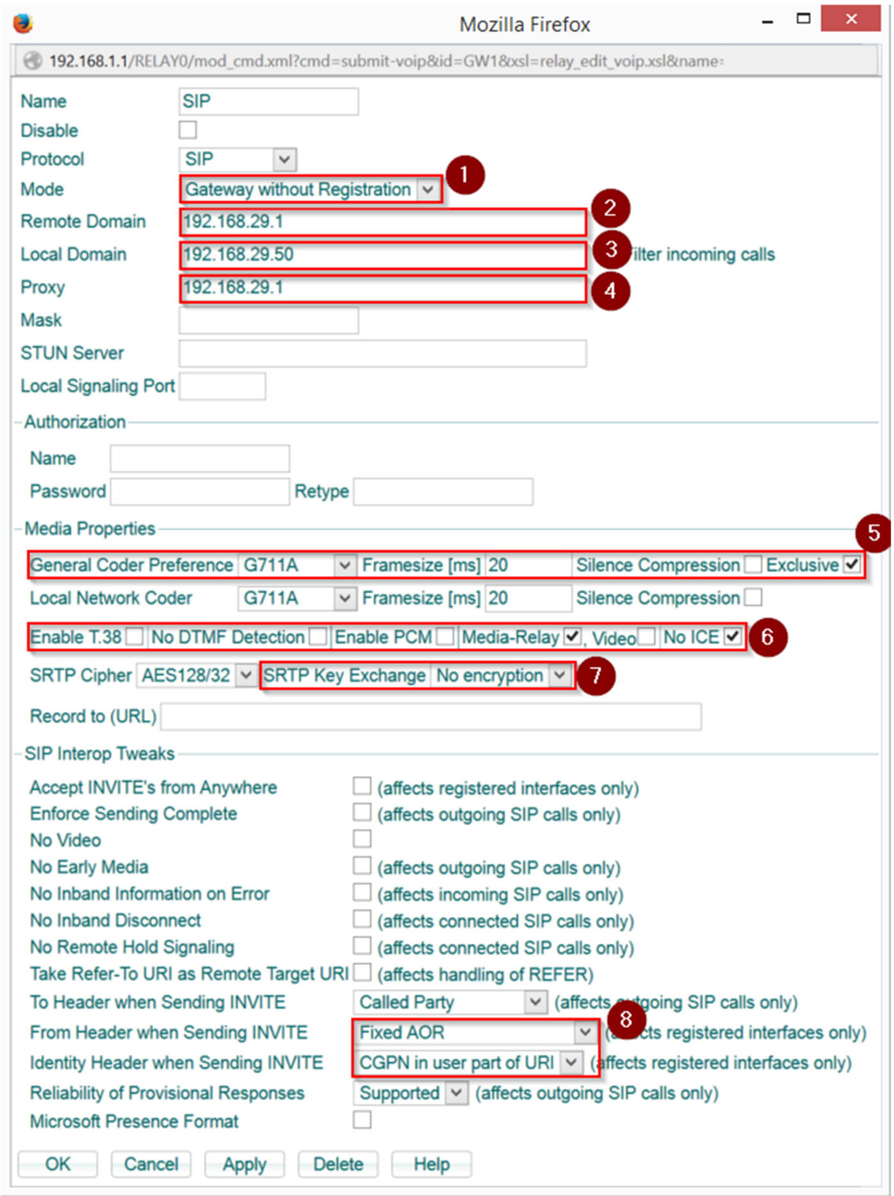
Broadcast Group & Waiting Queue

Tested feature	Result
Caller can make a call to a Broadcast Group	**OK**
Caller can make a call to a Waiting Queue	**OK**
Announcement if nobody picks up the call	**OK**

SIP - Trunk

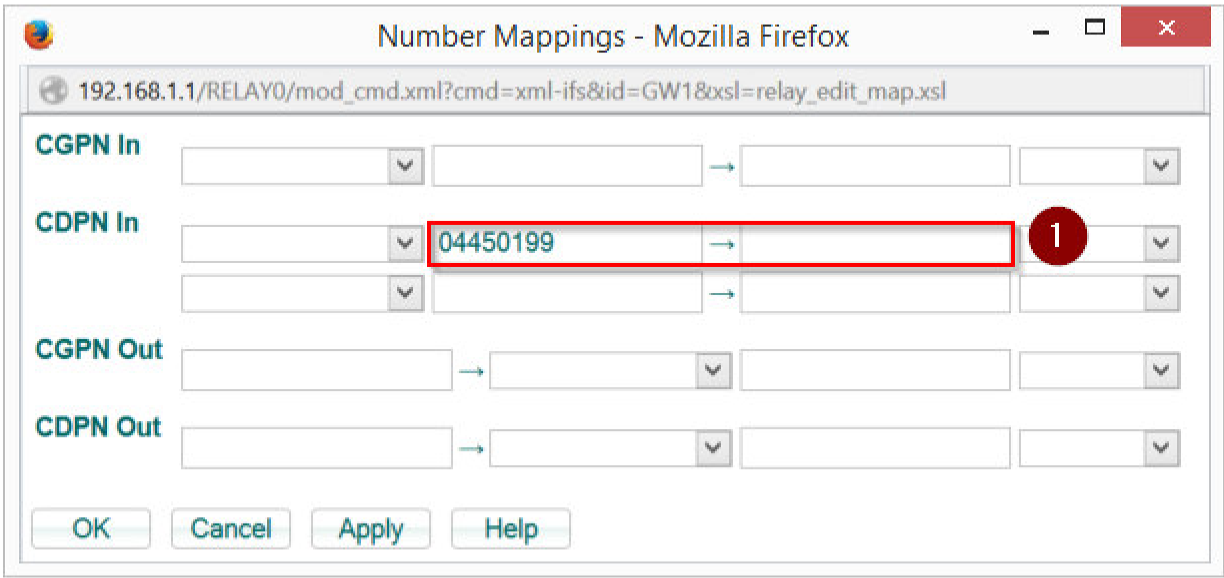
Following configuration needs to be done in the SIP Gateway Interface settings:

- 1: Use "Gateway without Registration" as "Mode"
- 2: Enter the IP address of SBC for "Remote Domain"
- 3: Enter the IP address of the innovaphone Gateway for "Local Domain"
- 4: Enter the IP address of SBC for "Proxy"
- 5: Select G.711a as coder 20ms as Framesize and Exclusive
- 6: Disable "T.38", Enable Media-Relay" and No ICE"
- 7: Disable encryption
- 6: Use "Fixed AOR" in "From Header" and "CGPN in user part of URI" in "Identity Header"



Number Mapping

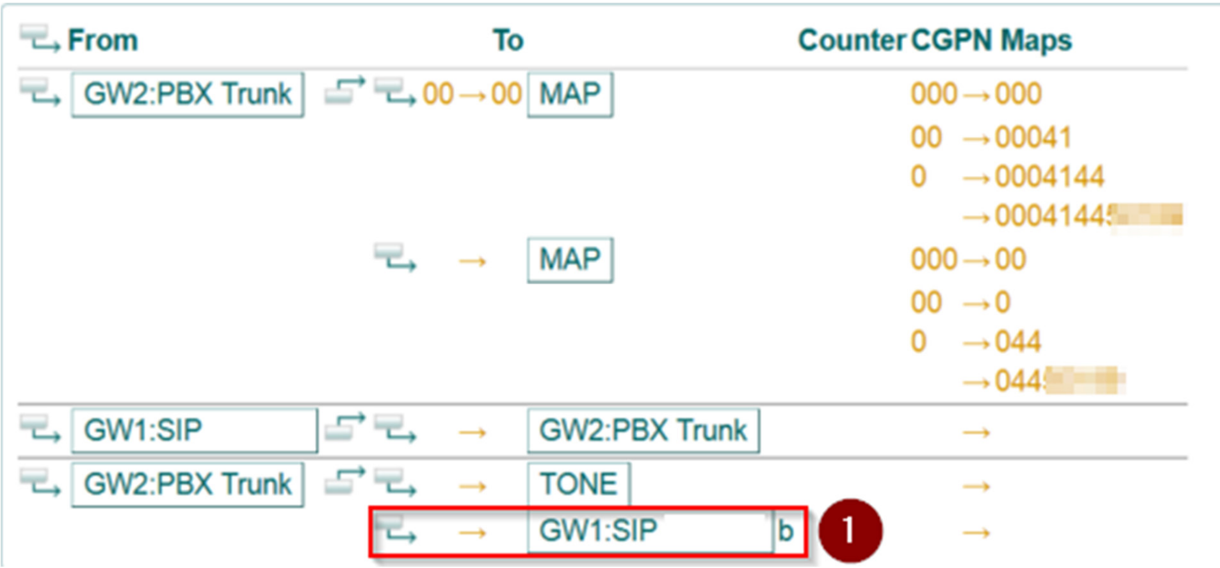
If the provider is using the full national number for incoming calls, you need to cut off the digits before an extension. In example, 04450199xx will be translated to xx.



Route Settings

A default Route Setup is used:

- 1: Force Enblock Setting is required for outgoing calls.



Fax

It is advised that if you want to use our Faxserver, you need to set T.38 to G.711 translation (Audio Fax support, available from V11). This works in a lot of cases where you have no problems with QoS.

Example to set T.38 to G.711 translation (Audio Fax support) on the Faxserver interface: