

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 | 3 |
| Aarenet SIP Trunk Config Center Einstellungen | 3 |
| Innovaphone setup Testresultate..... | 4 |
| Testresults SIP Provider: Innovaphone V11 build 11.3127 on SBC..... | 5 |
| Basic Call | 5 |
| Direct Dial In | 7 |
| DTMF | 7 |
| Hold/Retrieve | 7 |
| Transfer with consultation | 7 |
| Transfer with consultation (alerting only)..... | 8 |
| Blind Transfer..... | 9 |
| CFU / CFB Transfer | 9 |
| CFNR / Blind Transfer (alerting only) | 9 |
| Broadcast Group & Waiting Queue | 10 |
| SIP - Trunk | 11 |
| Number Mapping..... | 12 |
| Route Settings | 12 |
| Fax | 13 |

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

The image displays three screenshots from the Aarenet SIP Trunk Config Center interface:

- Top Left:** Account settings for 'aan1-000088'. The 'Addresses' link is circled in red and labeled '3'. The 'Username' field is circled in red and labeled '2'.
- Top Right:** Advanced settings for address '0449980510'. The 'Domain' field is circled in red and labeled '1'.
- Bottom:** 'Addresses of Account' window showing a list of DDI numbers. The first entry, '0449980510', is circled in red and labeled '4'.

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 |

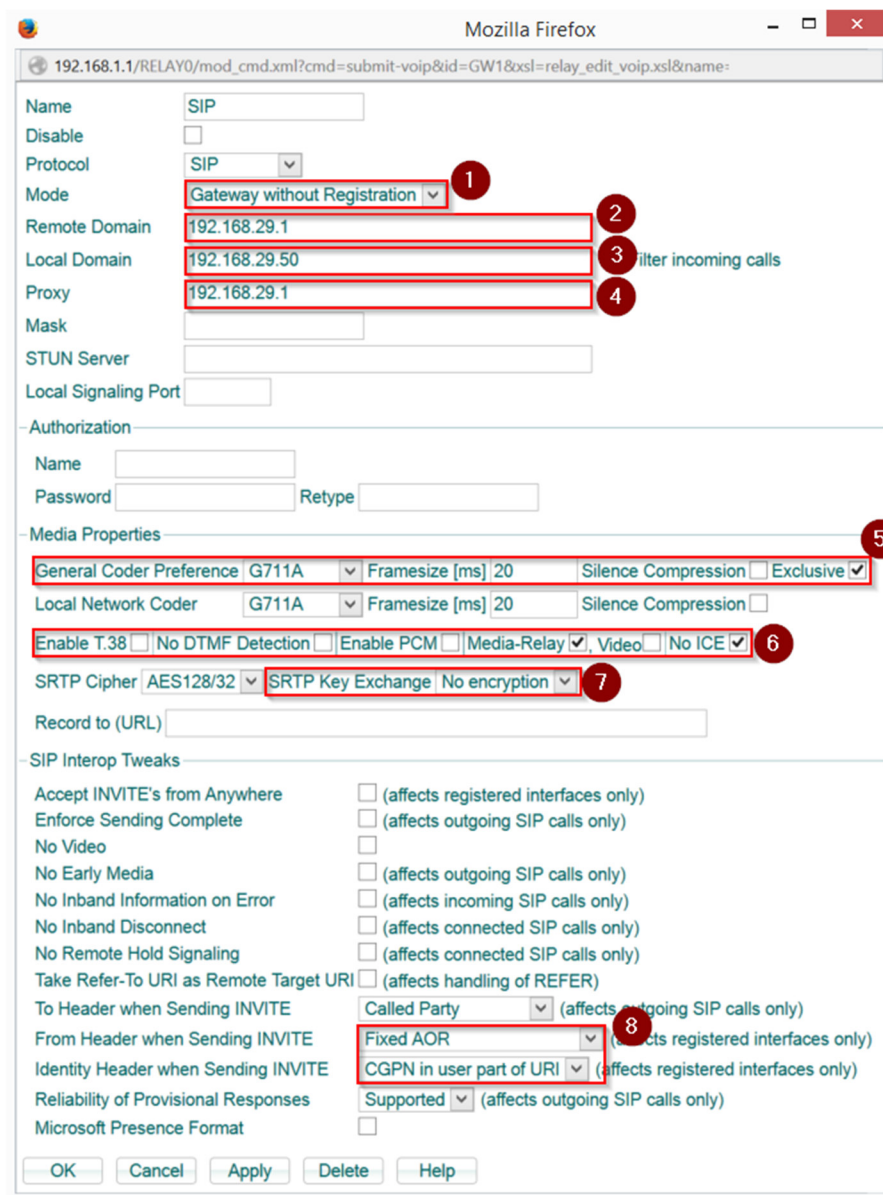
Broadcast Group & Waiting Queue

| 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
- 8: Use "Fixed AOR" in "From Header" and "CGPN in user part of URI" in "Identity Header"



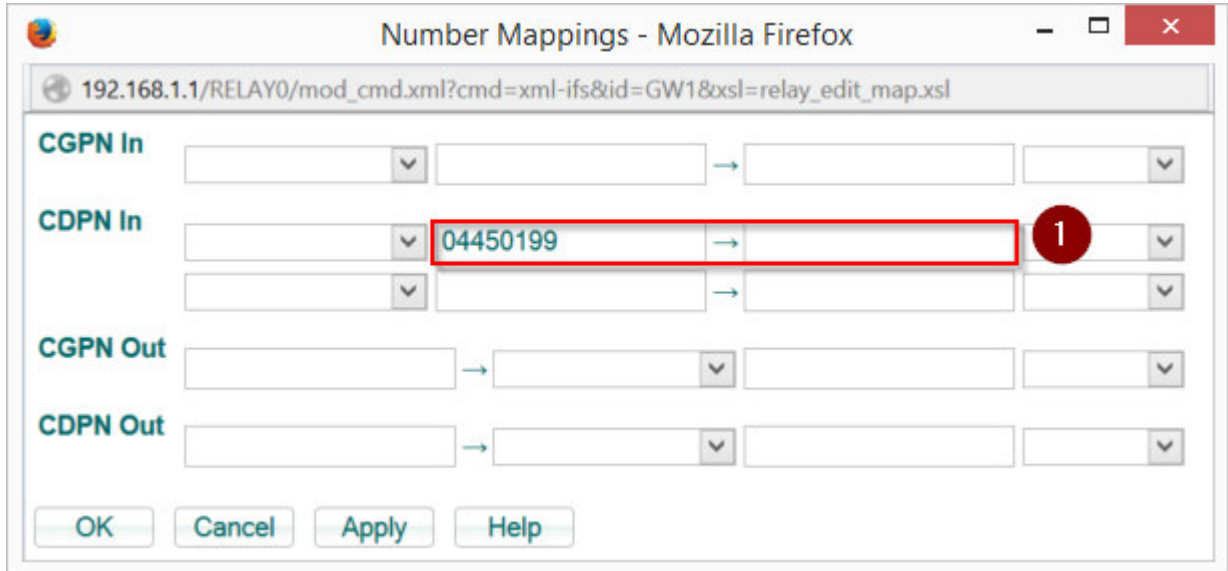
The screenshot shows the SIP Gateway Interface settings in Mozilla Firefox. The form is titled "192.168.1.1/RELAY0/mod_cmd.xml?cmd=submit-voip&id=GW1&oxsl=relay_edit_voip.xml&name:". The form contains the following fields and settings:

- Name: SIP
- Disable:
- Protocol: SIP
- Mode: Gateway without Registration (1)
- Remote Domain: 192.168.29.1 (2)
- Local Domain: 192.168.29.50 (3) filter incoming calls
- Proxy: 192.168.29.1 (4)
- Mask:
- STUN Server:
- Local Signaling Port:
- Authorization:
 - Name:
 - Password: Retype:
- Media Properties (5):
 - General Coder Preference: G711A
 - Framesize [ms]: 20
 - Silence Compression: Exclusive
 - Local Network Coder: G711A
 - Framesize [ms]: 20
 - Silence Compression:
 - Enable T.38: No
 - No DTMF Detection: No
 - Enable PCM: No
 - Media-Relay: Yes
 - Video: No
 - No ICE: Yes (6)
 - SRTP Cipher: AES128/32
 - SRTP Key Exchange: No encryption (7)
 - Record to (URL):
- SIP Interop Tweaks:
 - Accept INVITE's from Anywhere: (affects registered interfaces only)
 - Enforce Sending Complete: (affects outgoing SIP calls only)
 - No Video:
 - No Early Media: (affects outgoing SIP calls only)
 - No Inband Information on Error: (affects incoming SIP calls only)
 - No Inband Disconnect: (affects connected SIP calls only)
 - No Remote Hold Signaling: (affects connected SIP calls only)
 - Take Refer-To URI as Remote Target URI: (affects handling of REFER)
 - To Header when Sending INVITE: Called Party (affects outgoing SIP calls only)
 - From Header when Sending INVITE: Fixed AOR (8) (affects registered interfaces only)
 - Identity Header when Sending INVITE: CGPN in user part of URI (8) (affects registered interfaces only)
 - Reliability of Provisional Responses: Supported (affects outgoing SIP calls only)
 - Microsoft Presence Format:

Buttons: OK, Cancel, Apply, Delete, Help

Number Mapping

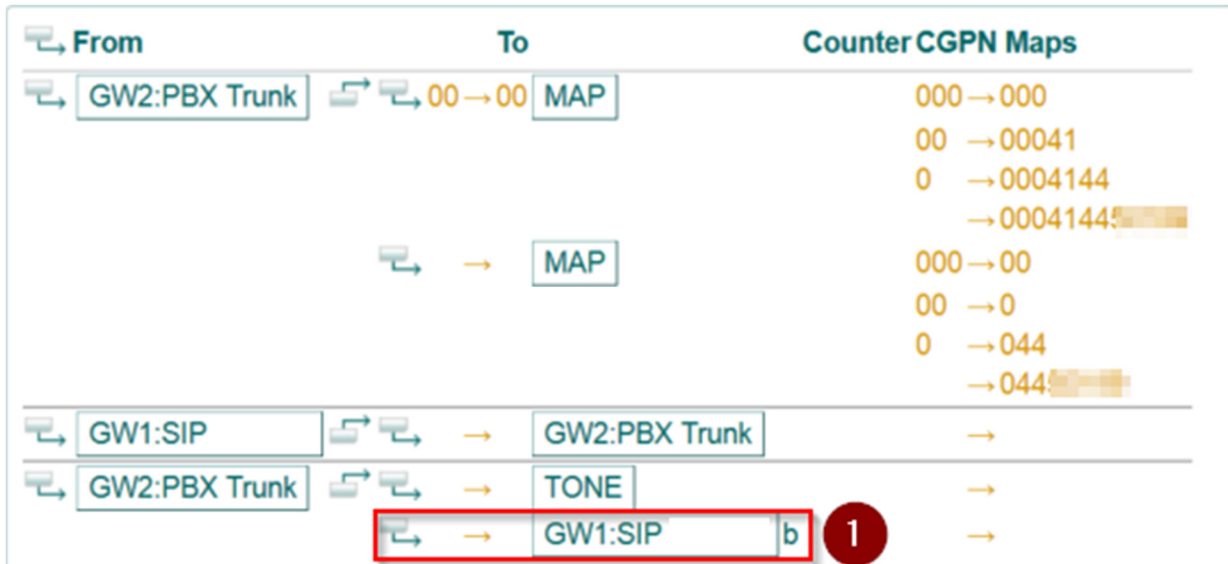
If the provider is using the full national number for incoming calls, you need to cut off the digits before an extension. In example, 04450199xx will be translated to xx.



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:

The screenshot shows the configuration interface for IP6010: innovaphone IP6010. The 'Gateway' tab is selected. A table lists various interfaces, with 'FAX Faxserver88' highlighted. A configuration window for 'FAX Faxserver88' is open, showing the following settings:

| Interface | CGPN-In | CDPN-In | CGPN-Out | CDPN-Out | State | Alias | Registration |
|-----------------|---------|---------|----------|----------|-------|-------|--------------|
| PRI1 | + | | | | | | |
| PRI2 | + | | | | | | |
| PRI3 | + | | | | | | |
| PRI4 | + | | | | | | |
| TEL | + | | | | | | |
| TEST | | | | | | | |
| TONE | | | | | | | |
| HTTP | | | | | | | |
| ECHO | | | | | | | |
| FAX Faxserver88 | + | | | | | | |
| CONF | + | | | | | | |

The configuration window for 'FAX Faxserver88' includes the following fields and options:

- Name: Faxserver88
- Disable:
- Interface Maps: Manual
- Internal Registration:
 - Protocol: H.323
 - STUN Server: [empty]
 - Gatekeeper Address: 127.0.0.1 (primary)
 - Gatekeeper Address: [empty] (secondary)
 - Gatekeeper ID: [empty]
 - Name: Faxserver88
 - Number: [empty]
 - Password: [masked] Retype: [masked]
- Media Properties:
 - General Coder Preference: G729A
 - Framesize [ms]: 30
 - Silence Compression: Exclusive
 - Local Network Coder: G711A
 - Framesize [ms]: 30
 - Silence Compression:
 - Enable T.38: Audio FAX support: No DTMF Detection: MOH Mode:
 - SRTP Cipher: AES128/32
 - SRTP Key Exchange: No encryption
 - Record to (URL): [empty]