

Interop T440/T640 2.0 Aarenet VoIP-Trunk

Aarenet VoIP-Trunk (T440/640 SW-version [2.1.0](#))

Feature	
Outgoing Calls	Yes
Incoming Calls	Yes
CLIP incoming	Yes
CLIP outgoing	Yes
Call Forwarding	Yes
Call Transfer	Yes
Call Waiting	Yes
DTMF	Yes
Anonymous Call	Yes
A-number forwarding	Yes
Fax	see chart below



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Gigaset T440/T640 PRO settings.

For entering a new SIP trunk into the PBX, you need some steps:

1. Adding a new Gateway group
2. Adding a new SIP gateway
3. Defining the inbound routes (assignment of number to extension)
4. Defining the outbound routes

Let's assume our SIP trunk contains the following numbers:

0441234567[0-9]

And we choose following internal numberblock:

67[0-9]

And we have no line-prefix.

Gateway Group

In the Gigaset PBX go to "Administration" - "Routes" - "Gateway groups" enter a name for the new group and click on: **Create new group**

In the Gateway group you have to define the Outbound Caller ID, the Inbound DIDs (how the number is forwarded to the Inbound routes) and the Inbound caller ID (number presentation of external caller).

In addition you can permit here inbound calls in general for this gateway group.

Edit gateway group

Title: Aarenet

Permit inbound calls: ☒

Outbound caller ID: Search/replace pattern for outbound caller ID (1)
s/ [^(*)] / 044123456\$1 /

Asserted Identity: Search/replace pattern for asserted identity (1)
s/ [^(*)] / 044123456\$1 /

Inbound DIDs: Search/replace pattern to cut prefixes (2)
s/ [^(*)?(?:0041|1+4141|00144|12345)?(.*\$) / \$1 /

Inbound caller ID: Search/replace pattern for inbound caller ID (3)
s/ [^(*)] / \$1 /

Gateways: *

Outgoing caller IDs:

(1) Search/Replace pattern (PCRE) for outbound caller ID signalling. Examples:
Extension only: s/^(*)/\$1/
National format: s/^(*)/004930123456\$1/ or s/^(*)/+4930123456\$1/
International format: s/^(*)/004930123456\$1/ or s/^(*)/+4930123456\$1/
Same number for all users: s/^(*)/00493012345612/
In most cases you should use the national or international format.
(2) If necessary, specify here a PCRE-pattern which removes prefixes from inbound numbers, so that only the internal extension remains. Examples:
s/^026313370//
s/^((00490)2631)3370//
s/^(?:00490)2631)73370(.*\$)/\$1/
(3) Search/Replace pattern (PCRE) for inbound caller IDs, to fix up eventually wrong signalled caller IDs. Example: s/^/0/ or s/^(.*)/0\$1/ to prefix the caller ID with a 0. Empty for no automatic change.
Experimental: This option can also be used to add or remove a leading 0 to/from call log entries by using s/^/0/ or s/^(.*)/0\$1/, or s/^0/ or s/^0(.*\$)/\$1/.

Cancel Save

Example of Aarenet VoIP trunk gateway group settings.

Outbound caller ID

As just the last digit of the trunk number block is changing, you can select just the last digit (in brackets) and put it into the variable \$1.

As the provider wants the signalling number in the format (country-code)(city-code)(number), you have to enter this number into the next line (here: 0441234567\$1, where the \$1 represents the changing part).

For external calls, we will present e.g. **04412345678** to the provider, and he will take care about the correct representation to the called party.

Asserted Identity

These fields are used specially for external forwardings. For Aarenet VoIP Trunk, you can use the same settings as for the Outbound caller ID.

Inbound DIDs

For incoming calls, you can use the regular expression, generated by the setup assistant. In your case you would have to adjust the city-code and the pilot-number, according to your line-settings.

```
^(?:?:?:?:0041|\+41|41|0)?44)?12345)?(.*$)
```

This expression cuts all possibly available country- and city-codes and the pilot-number from the incoming number and only the extension is remaining (here: **678**). This will be forwarded to the Inbound Routes.

Inbound caller ID

For incoming calls, we will forward the external number of the caller "as-is" to the phone (e. g. **044987654321**). This number can then be used from the call-lists for re-dialling.

Gateways

This field will be empty when you create this Gateway group. It will show later the assigned SIP gateways.

SIP gateway

In the Gigaset PBX go to "*Administration*" - "*Routes*" - "*SIP gateways*" enter a name for the new gateway and click on: **Create new gateway**

The SIP gateway contains all necessary data for the registration and dial command and how the number is delivered to/from the provider.

Just enter or choose the values according to your contract or VoIP trunk data you received from the provider.

Edit SIP gateway: Aarenet

Title	Aarenet
Name	gw_4aarenet
Registrar	[Flags]
Proxy [1]	[Flags]tr
User [2]	[Flags]
Password	*****
Allow outbound calls	<input checked="" type="checkbox"/>
Register	<input checked="" type="checkbox"/>
Language	de - German (de-DE)
Dial command [3]	PJSIP:(prefix)(number)g(gateway)
Transport name	default-udp
Source of destination number	INVITE request line
Group [4]	Aarenet
Port [5]	5060
NAT	yes
Redirect RTP stream	Do not reroute RTP stream (default)
Check availability	<input checked="" type="checkbox"/>
Simultaneous calls	0 0 for unlimited (default)

(1) Empty for no proxy.

(2) For some SIP providers, it might be necessary to use the format user@domain. (domain is then used in the From header, which equals fromdomain in Asterisk.)

(3) String for the Dial()-command. T440/T640 PRO will automatically replace (number) by the called number, (number:1) without the first digit and (gateway) with the internal description (e.g. gw_1_amt).

(4) In order to use gateways, they must be assigned to a gateway group.

(5) When specifying the port (standard SIP port: 5060), it will be directly used. Without, a DNS lookup of the SRV record _sip._udp of the domain (or server) will be performed upon dialout. More Information [Srv Resource Record](#), [Srv Resource Record \(en\)](#), [CIDR](#), [SIP-DNS-Srv-Records](#), [SIP-DNS-Srv-Records \(en\)](#), [SIP-DNS-Srv-Records \(en\)](#).

(6) The priority of the codes is from left to right and top to bottom

(7) Useful settings are e.g.
0.0.0.0/0, to allow calls from all IP addresses,
192.0.2.0/24 to allow calls originating from network 192.0.2.*,
192.168.0.0/16 to allow calls originating from network 192.168.*,
192.168.1.1/32 to allow calls originating from IP address 192.168.1.1 etc.
More information [IP address](#), [Subnet](#), [CIDR](#), [sip.conf](#)

Cancel Save

Example of Aarenet VoIP trunk gateway settings 1/2.

The screenshot shows the Gigaset administration interface. The sidebar on the left contains navigation links: HOME, MENU, ADMINISTRATION (highlighted), and PROFILE. Below these are categories: Users & extensions, System, Provisioning, Routes, Gateway groups, SIP gateways (highlighted), TDM Gateways, Inbound routes, Outbound routes, Call forwarding, and System status.

The main configuration area for a SIP gateway includes the following fields:

- DTMF mode:** rfc4733 - RTP meta data
- From user:** [empty field]
- From Domain:** [empty field]
- T38 support:** OFF
- Update P-Asserted-Identity (CLIP):** no - Deactivated (default)
- Update remote party ID (CLIP):** no - Deactivated (default)
- Trust remote party ID:** no - Deactivated (default)
- Codescs [s]:**
 - ☐ G.722
 - ☐ G.729
 - ☒ G.711a
 - ☐ G.711u
 - ☐ GSM
 - ☐ H.261
 - ☐ H.263
 - ☐ H.263+
- Allowed IP subnet [r]:** 0.0.0.0 / 0

Below the main fields is the **Advanced parameters** section, which includes a table for key-value pairs:

Value	
inband_progress=yes	<input type="checkbox"/>
Value	<input type="text"/> +

At the bottom is the **Preview of peer in sip.conf** section, showing the following configuration:

```
[gw_4.aarenet]
type=auth
auth_type=userpass
```

Buttons for **Cancel** and **Save** are located at the bottom right of the configuration window.

Example of Aarenet VoIP trunk gateway settings 2/2.

Registrar

Use here the **SIP-Server / Registrar** (received by Aarenet). In this case an IP-address is used.

Proxy

The tests were performed with the the same IP-address as used in the Registrar field, followed by ';lr' , which means "loose-routing".

User

Enter here the **Username** (Benutzername) you have received by Aarenet.

Password

Enter here the **Password** (Passwort) you have received by Aarenet.

Dial command

The dial command is the command which is used in the asterisk software. The term **{number}** means, that the number (e.g. **04455007000**) is dialled as entered. If you use a line access code (in most cases '0' is used), you have to add ':1' after "number".

Group

Select here the previously created gateway group.

Update remote party ID or P-Asserted-Identity (CLIP)

These settings can be left on the default setting (no - Deactivated)

From Domain

Enter here the **Domain/Realm** delivered by Aarenet. You can use the same value as used for the **Registrar**.

T38 support

In order to have a better fax-support for fax-machines, connected to the FXS ports please deactivate this option.

All tested scenarios were ok with this parameter deactivated.

Underneath the configuration parameter you will see a preview of the complete sip.conf file.

Inbound routes

In the Gigaset PBX go to "*Administration*" - "*Routes*" - "*Inbound routes*" select the correct Gateway group and press **Show**.

In addition it is advised to activate the **advanced options** by clicking it to '**ON**' and then pressing **Show**.

Inbound routes

Night answer service

Inbound routes

Gateway group

Aarenet (aarenet)

Advanced options

ON

Rule	Active	Date	Weekdays	Time	Profile	Pattern	Target	
General	ON	<div>to</div>	M T W T F S S <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>	<div>00:00</div> <div>to</div> <div>24:00</div>	-	^(*)	\$1	<div>-</div> <div>-</div> <div></div>
	ON	<div>to</div>	M T W T F S S <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>	<div>00:00</div> <div>to</div> <div>24:00</div>	-			<div></div> <div>+</div>

Example of Aarenet VoIP trunk Inbound routes.

Rule

Enter here a name for the according rule.

Date / Weekdays / Time

With these settings you can configure a time-controlled routing to different targets.

Pattern / Target

In the pattern you define which part of the incoming number is used to forward the call to the according extension.

In our example we receive already the correct extension from the gateway group. Therefore no further number-manipulation is necessary.

When the PBX finds an according extension it will route the call to it.

But you can add here exceptions from this rule, for example for internal fax users or waiting queues, etc.

Please have in mind the order of these rules, as the system is using **First Match!!!**

Outbound routes

In the Gigaset PBX go to "*Administration*" - "*Routes*" - "*Outbound routes*" activate the Advanced options by clicking it to '**ON**' and then pressing **Show**.

Advanced options
☐ ON

Rule	Active	Weekdays	Time	Pattern	Group	Gateway group	Add prefix	
General	<input checked="" type="checkbox"/> ON	M T W T F S S ☑ ☑ ☑ ☑ ☑ ☑ ☑	00:00 to 24:00	^0	[all]	Aarenet		<div>+</div> <div>-</div> <div>☒</div>
	<input checked="" type="checkbox"/> ON	M T W T F S S ☑ ☑ ☑ ☑ ☑ ☑ ☑	00:00 to 24:00		[all]	Sisters		<div>+</div> <div>-</div> <div>-</div>

Example of Aarenet VoIP trunk Outbound routes.

Rule

Enter here a name for the according rule.

Date / Weekdays / Time

With these settings you can configure a time-controlled routing.

Pattern

In the pattern you define how the outside line is seized. In our example all dialled numbers starting with '0' will use the Gateway group Aarenet VoIP trunk.

Remark: For calling anonymous to external parties, use the CLIR-setting in the HOME screen of the user!

Fax support

Following table shows you the current state (09.03.2017) of supported fax constellations. These results are without a guarantee. Due to different end-devices, configuration of PBX or other settings the fax transmission might fail. [More info about fax via VoIP networks can be found here.](#)

Receiving	Sending			
	Internal FXS	Internal T38-Fax	External SW-Fax	External machine-Fax
Internal FXS	---	---	OK* / NOK**	OK* / NT**
Internal T38-Fax	---	---	OK* / NOK**	OK* / NT**
External SW-Fax	OK* / NOK**	OK* / NOK**	---	---
External machine-Fax	OK* / NT**	OK* / NT**	---	---

Used devices or services:

Canon Fax-L100 (internal FXS), www.minifax.de (external SW-Fax), Triumph Adler DCC 2725 (external machine-Fax)

* = T38 option in SIP gateway deactivated

** = T38 option in SIP gateway activated