Interop T440 / T640 - VTX Telecom (FW Version 1.1.0)

VTX Switzerland

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	n/a
A-number forwarding	Yes

Gigaset T440/T640 PRO settings.
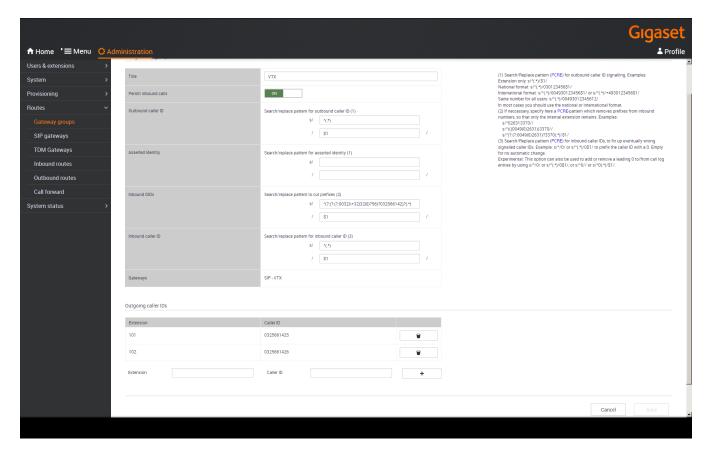
For entering a new SIP trunk into the PBX, you need some steps:

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

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.

Outbound caller ID

As the number is fixed with this account, every user who is allowed to use this line will present the same number to the callee.

Asserted Identity

These fields are used specially for external forwardings. Some providers need special settings in order to present the correct number at the receiving party.

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 first part of the extension, according to your line-settings.

Example: ^(?:(?:(?:0041|\+41|32|0)?56)?032566142)?(.*)

This will be forwarded to the Inbound Routes.

Inbound caller ID

For incoming calls, no additional **0** will be added.

Gateways

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

SIP gateway

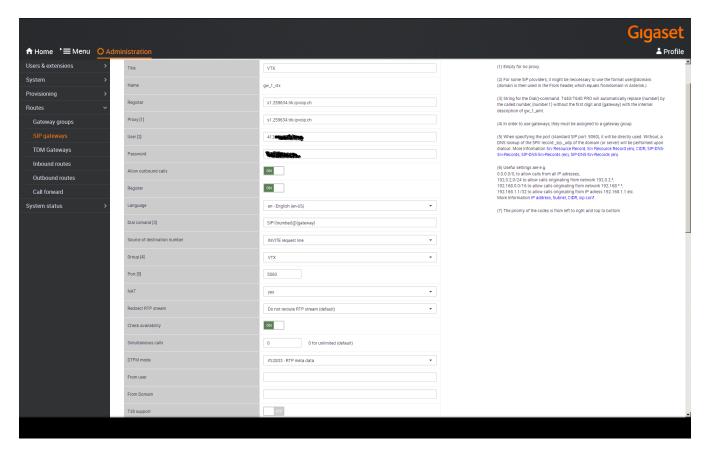In the Gigaset PBX go to *"Administration" - "Routes" - "SIP gateways"* enter a name for the new gateway and click on: **Create new gateway**

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 data you received from the provider.

Example of VTX gateway settings 1/2.

Title	VTX
Name	gw_1_vtx
Registar	s1.259634.trk.ipvoip.ch
Proxy [1]	s1.259634.trk.ipvoip.ch
User [2]	413█████████
Password	███████████
Allow outbound calls	ON
Register	ON
Language	en - English (en-US)
Dial comand [3]	SIP/{number}@{gateway}
Source of destination number	INVITE request line
Group [4]	VTX
Port [5]	5060
NAT	yes
Redirect RTP stream	Do not reroute RTP stream (default)
Check availability	ON
Simultaneous calls	0 0 for unlimited (default)
DTFM mode	rfc2833 - RTP meta data
From user	
From Domain	
T38 support	OFF

(1) Empty for no proxy.

(2) For some SIP providers, it might be neccessary 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 of 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) Useful settings are e.g.
0.0.0.0/0, to allow calls from all IP adresses,
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 adress 192.168.1.1 etc.
More Information IP address, Subnet, CIDR, sip.conf

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

Dial command

The dial command is the command which is used in the asterisk software. The term **{number:1}** means, that at the dialled number (e.g. 008987654321) the first digit is removed. If you don't use a line access code (in most cases '0' is used), you have to remove the '**:1**' ! ! ! (Example is without access code)

Group

Select here the previously created gateway group.
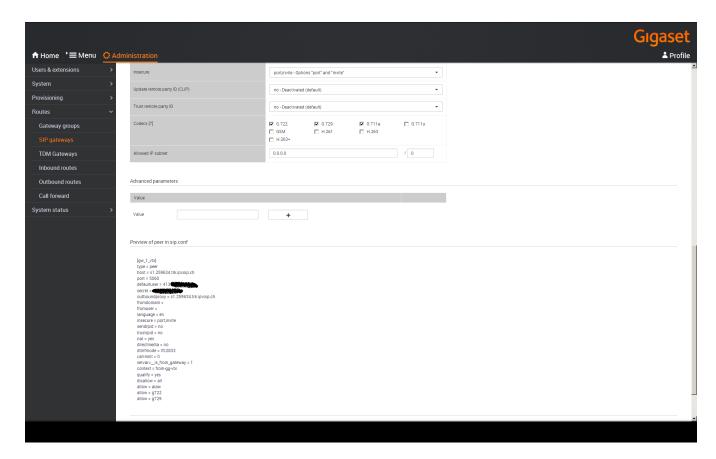
Update remote party ID (CLIP)

This setting is just optional, as the provider replaces outgoing CLIP which is not valid with the default number of the SIP account.

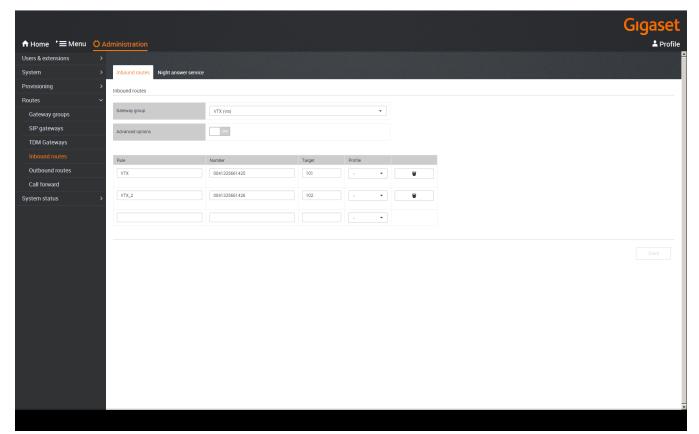For sending another CLIP, use the CLIP-No-Screening feature.

Advanced Parameter

Can be empty

Example of VTX gateway settings 2/2.

Inbound routes

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

Outbound routes

In the Gigaset PBX *"Administration" - "Routes" - "Outbound routes"* you don't have to activate the advanced options for the basic installation.