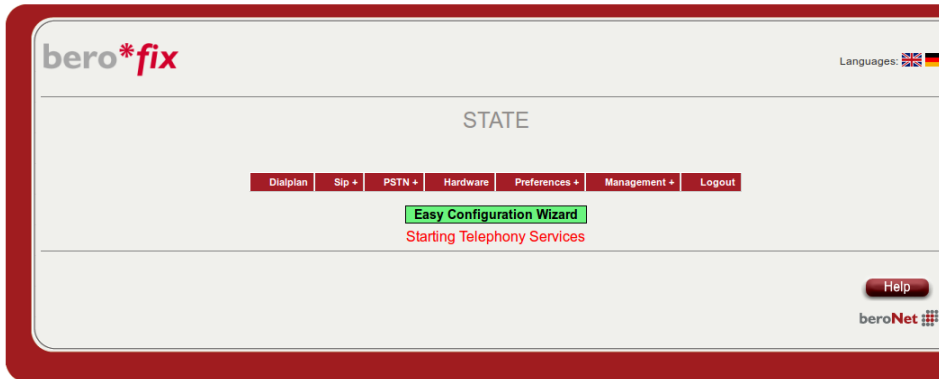


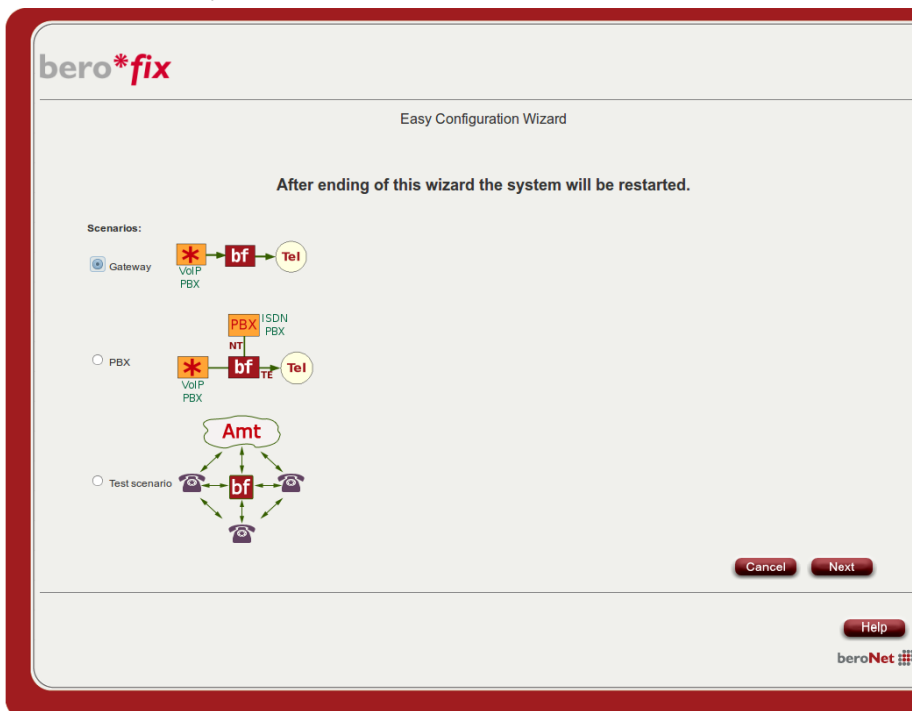
# Beronet berofix 400/1600/6400

This guide will show you how to connect the bero\*fix gateway to T440 /T640

1. Open the web configuration interface by pointing a browser to the IP Address of the Beronet gateway and login .(The default credentials for the Beronet web interface are Username: admin, Password: admin  
[blocked URL](#)
2. Click on the "Easy Configuration Wizard" Butto



3. Choose senario Gateway



4. Setup the following parameters:  
Name = PRI E1  
Server Address ( IP T440 or T640 / default 192.168.0.50) = 192.168.0.50



## Warning

Due to new IPBX firmware uses PJSIP, target port has to be defined after server address : 192.168.0.50:5070

### SIP CONFIGURATION



Name:

BeroT0

Server Address:

192.168.1.200:5070

User (Username to register T440 or T640 to Beronetgateway) = beronetpri  
Secret (Pwassword to register T440 or T640 to Beronetgateway) =beronetsecret

berofix

Easy Configuration Wizard

After ending of this wizard the system will be restarted.

Sip configuration

Name: PRI E1

Server Address: 192.168.0.50

User: beronetpri

Secret: beronetsecret

Cancel Next

Help

beronet

5. Select the PRI Port and enter Group Name= BeronetPort1

berofix

Easy Configuration Wizard

After ending of this wizard the system will be restarted.

PSTN-Network-Group (TE-Ports)

☒ PRI

Group Name: PRIport1

Ports:

U0(bf1E1)	U1()
Port 1 <input checked="" type="checkbox"/>	<input type="checkbox"/>

Cancel Next

Help

beronet

6. Finish the Beronetgateway configuration

**beronetfix**

Easy Configuration Wizard

After ending of this wizard the system will be restarted.

**SIP Entry:**  
Name: PRI E1  
Server Address: 192.168.0.50  
User: beronetpri  
Secret: beronetsecret  
Registrar: 0

**PSTN Entry:**  
Group Name: PRIport1  
Ports: 1

**Dialplan Entries:**  
From direction: sip  
To direction: isdn  
From ID: p-PRI E1  
To ID: p-PRIport1  
Destination: (\*)  
New destination: 1  
Source: (\*)  
New source: 1

From direction: isdn  
To direction: sip  
From ID: p-PRIport1  
To ID: p-PRI E1  
Destination: (\*)  
New destination: 1  
Source: (\*)  
New source: 1

Cancel Finish

Help

beronet

7. Open the web configuration interface by pointing a browser to the IP Address of the T440 or T640.  
(The default credentials for the Beronet web interface are **Username: admin**, **Password: 1234**)

**Gigaset**

Login

Username

PIN

Login

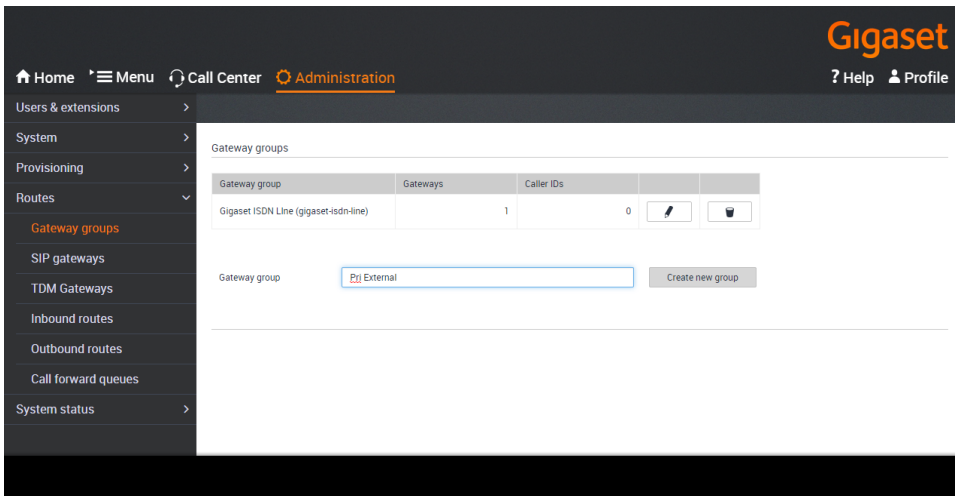
For entering a new SIP trunk to connect the Beronet gateway into the PBX, you need some steps:

- Adding a new Gateway group
- Adding a new SIP gateway
- Defining the inbound routes assignment of number to extension)
- Defining the outbound routes

Let's assume our provider allocate the following numbers: **0891234567[0-9]**  
and we choose following internal numberblock **67[0-9]**

8. Create a new 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 last 3 digits of the provider number blocks are matching the user extensions (e.g. 678), we just add the full extension to the trunk part 08912345. For external calls, we will present e.g. 08912345678 to the calling party.

### Inbound DIDs

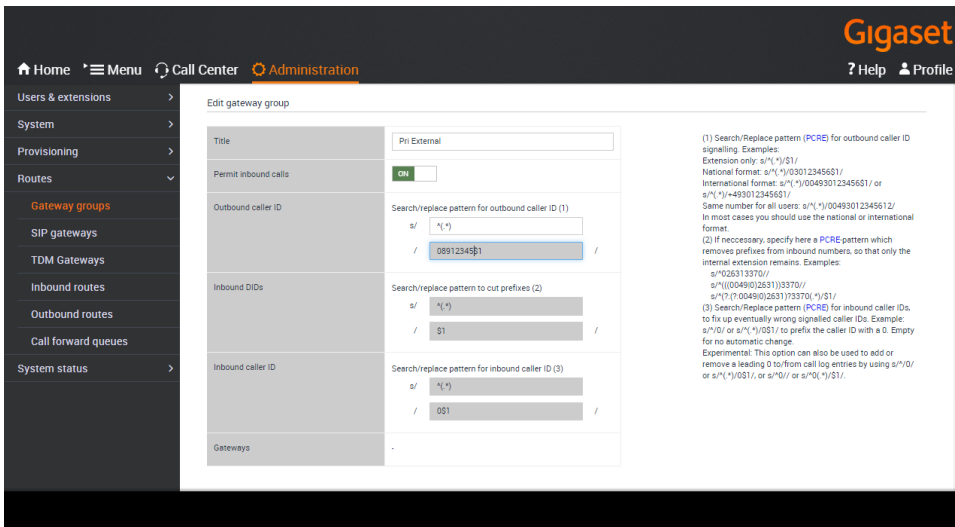
For incoming calls, we will route the complete number to the Inbound routes (e.g.08912345621).

### Inbound caller ID

For incoming calls, we will add an additional 0 in front of the external number of the caller, in order to use the callback-feature of the phone (e.g. 008912345678). During the call-setup the additional 0 will be automatically removed.

### Gateways

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



### 9. Create a new Sip gateway

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

### Registra

IP Address of Beronet gateway

### User

User (Step 4 Beronet gateway) = beronetpri

### Password

Password (Step 4 Beronet gateway = beronetsecret

## 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. 0089987654321) 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' !!!

## Group

Select here the previously created gateway group.

## Update remote party ID (CLIP)

To display the correct extension at external parties, you have to use the P-Asserted-Identity (PAI) setting. If this is not selected, only the head-number (e.g.0891234567-0) will be presented, no matter which extension is used to dial.

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Edit SIP gateway: PRI beronet

Title	PRI beronet	(1) Empty for no proxy
Name	gw_4_priberonet	(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 trdomain in Asterisk.)
Registrar	192.168.0.20	(3) String for the Dial() command. Gemeinschaft will automatically replace (number) by the called number, (number:1) without the first digit and (gateway) with the internal description of gw_1_amt.
Proxy [1]		(4) In order to use gateways, they must be assigned to a gateway group.
User [2]	beronetpri	(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 <a href="#">Srv Resource Record</a> , <a href="#">Srv Resource Record (en)</a> , <a href="#">CIDR</a> , <a href="#">SIP-DNS-Srv-Records</a> , <a href="#">SIP-DNS-Srv-Records (en)</a> , <a href="#">SIP-DNS-Srv-Records (en)</a>
Password	beronetsecret	(6) 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. <a href="#">More information</a> <a href="#">IP address</a> , <a href="#">Subnet</a> , <a href="#">CIDR</a> , <a href="#">sip.conf</a>
Allow outbound calls	<input checked="" type="checkbox"/>	(7) The priority of the codes is from left to right and top to bottom
Register	<input checked="" type="checkbox"/>	
Language	de - German (de-DE)	
Dial comand [3]	SIP/(number)@(gateway)	
Source of destination number	INVITE request line	
Group [4]	PRI external	
Port [5]	5060	

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Source of destination number	INVITE request line	192.168.1.1 etc. <a href="#">More information</a> <a href="#">IP address</a> , <a href="#">Subnet</a> , <a href="#">CIDR</a> , <a href="#">sip.conf</a>
Group [4]	Pri External	(7) The priority of the codes is from left to right and top to bottom
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)	
DTFM mode	rfc2833 - RTP meta data	
Insecure	portinvite - Options "port" and "invite"	
Update remote party ID (CLIP)	Use P-Asserted-Identity header	
Trust remote party ID	no - Deactivated (default)	
Codecs [7]	<input type="checkbox"/> G.722 <input type="checkbox"/> G.729 <input checked="" type="checkbox"/> G.711a <input type="checkbox"/> G.711u <input type="checkbox"/> GSM <input type="checkbox"/> H.261 <input type="checkbox"/> H.263	

## 10. Inbound routes

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

### 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 cut off the first part 08912345 and forward all additional numbers (e.g.678) to the target. When the PBX finds an according extension it will route the call to it. The shown example is the easiest way for blocks of numbers, but also other configurations or assignments are possible.

Rule	Active	Date	Weekdays	Time	Profile	Pattern	Target
Default	ON		M T W T F S S	00:00 to 24:00	-	08912345(*)	\$1
	ON		M T W T F S S	00:00 to 24:00	-		

## 11. 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.

### 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 External PRI.

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ON

M T W T F S S  
☑ ☑ ☑ ☑ ☑ ☑ ☑

00:00  
to  
24:00

\*0

[all]

Pri Ex

- +

ON

M T W T F S S  
☑ ☑ ☑ ☑ ☑ ☑ ☑

00:00  
to  
24:00

[all]

-

- +