



Cloud managed & monitored
VoIP Gateways and Appliances

How to:

Connect a Flowroute SIP account with a beroNet
Gateway?



flowroute®

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I) Network configuration:

The gateway does not register with the SIP trunk of Flowroute. We need to specify the IP address of the company using the gateway in the Flowroute account.

I.1. Ports used by the gateway

In order to configure the link between the gateway and the Flowroute SIP trunk, we need to know the ports used by the gateway. Those ports can be modified and are found in the WEB GUI of the gateway in the “SIP General” page under “SIP +”:

Sip General	
Bind Port:	<input type="text" value="5060"/>
RTP Port Range:	<input type="text" value="8000-7000"/>
TOS RTP:	<input type="text" value="180"/>
TOS SIP:	<input type="text" value="180"/>
Reject calls under load:	<input type="checkbox"/>
RTP port selection:	<input type="text" value="standard"/> ▼
SIP transport:	<input type="checkbox"/> tls <input type="checkbox"/> tcp <input checked="" type="checkbox"/> udp

These are the default ports but they can be changed.

I.2. Inbound and outbound rules

One inbound rule and one outbound rule need to be created in the Flowroute account.

Navigate to “interconnection” and “inbound route”. There, enter the external IP of your company where the gateway is and the VoIP port used by the gateway. This port is 5060 by default but it can be modified via the web interface of the gateway under “SIP General”.

In my case:

Route	Type	Functions
<input type="text" value="Enter Route"/>	<input type="text" value="Host"/> ▼	<input type="button" value="[+] Add Route"/>
office2.beronet.com:50060	Host	<input type="button" value="Edit Route"/> <input type="button" value="Remove Route"/>

We then need to enter the allowed IP in the “outbound allowed IPS”. In my case I entered the external IP of my company:

<input type="text" value="Enter IP Address"/>	<button>+ Add IP Address</button>
IP	Functions
91.65.252.69	<button>Edit IP</button> <button>Remove IP</button>

Finally, it is necessary to open the “Bind Port” and the “RTP port range” in the router of the company where the gateway is installed in order to let the SIP flow and audio flow go through the firewall.

2) Configuration of the SIP trunk in the gateway

The information for the configuration of the flowroute sip trunk are to be found on your flowroute account under “interconnection – Status & Setup”. Three important fields are listed:

- SIP Registrar / Proxy
- Username & Auth Username
- Password

Those information are to be added in the SIP trunk configuration in the web interface of the gateway.

To configure a SIP account in a beroNet VoIP Gateway, navigate to “SIP” under “SIP+”. There, click on add and enter the information given by Flowroute. Here is an example:

- Name: give it the name you want
- Server Address: this is the SIP proxy given by Flowroute which is sip.flowroute.com
- User: enter the “Username & Auth Username” found in your flowroute account

- Password: enter the “Username & Auth Username” found in your flowroute account
- NAT options: you can use a STUN-Server
- Register: do not activate this option as the gateway does not register with the trunk
- Save
- Activate

SIP CONFIGURATION
✕

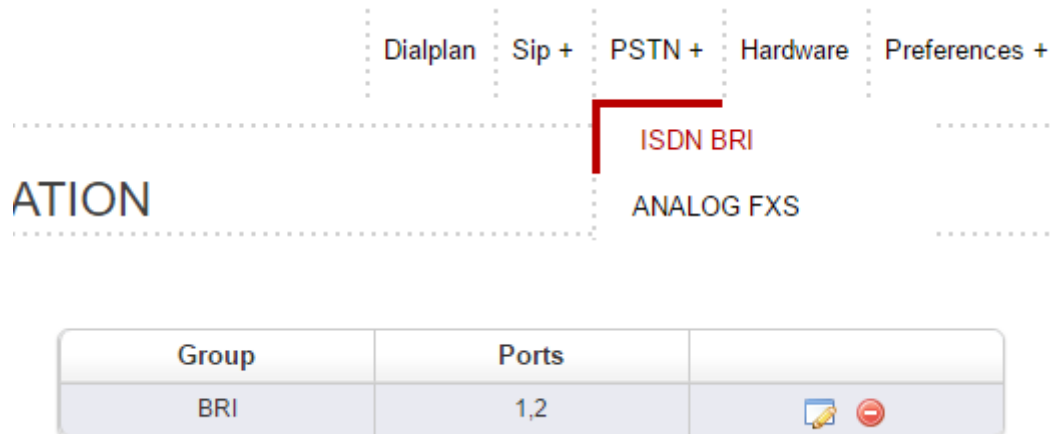
Name:	<input type="text" value="Flowroute"/>
Server Address:	<input type="text" value="sip.flowroute.com"/>
User:	<input type="text" value="30335754"/>
Authentication user:	<input type="text"/>
Displayname:	<input type="text"/>
Secret:	<input type="text" value="....."/>
Match type:	<input checked="" type="checkbox"/> IP Address <input type="checkbox"/> From User <input type="checkbox"/> To User <input type="checkbox"/> Contact User <input type="checkbox"/> Request-URI User <input type="checkbox"/> Manual <input type="text"/>
SIP transport:	<input type="text" value="udp"/> ▼
NAT options:	<input checked="" type="radio"/> No-NAT <input type="radio"/> STUN-Server <input type="radio"/> Extern-IP
Register:	<input type="checkbox"/>
more...	

As Flowroute and beroNet use SIP in the same way, there is not need to change the additional options that can be found under “more...”

The gateway is now connected with your Flowroute SIP trunk. In order to connect it to your ISDN lines or analog devices, please find more articles on our blog: beronet.com/blog

3) PSTN configuration

In order to configure analog, GSM or ISDN ports, navigate to the tab “PSTN+” in the web interface of the gateway.



If you are using an analog line (FXO module), a GSM (SIM card) or an ISDN line, then the CID sent will be recognized by the gateway and sent as such to SIP.

If you are using an FXS Gateway though (for an analog device) you need to set the CID in the CLIP / CNIP of the port or in the “new source” field of the dialplan. In this how to, we’ll show how to do this in the dialplan.

4) Dial plan configuration

The beroNet dial plan enables the gateway to connect two technologies together. There is always a source and a destination. In this example, we will show how to set rules for an FXS gateway but the same method can be applied for ISDN and GSM technologies as well.

a. Tech Prefix









Flowroute uses a “Tech Prefix” that is used if you choose to use IP-based authentication for outbound calls. Your Tech Prefix can be found under “Interconnection – Status & Setup” in your Flowroute account.

We need to take this into account while creating dial plan rules.

b. Creating a dial plan rule

Dial Plan rules usually work in pairs: from a technology to another and vice versa. Here is an example with an FXS gateway.

Navigate to “Dialplan” and two rules as such:

<input type="checkbox"/> Check all		Direction: all	Search:	Entries per page: 20					
	Direction	From ID	To ID	Destination	New destination	Source	New source	Position	
<input type="checkbox"/>	sip-analog	Flowroute	FXS1	(.*)	\1	(.*)	\1	1	   
<input type="checkbox"/>	analog-sip	FXS1	Flowroute	(.*)	30335754*\1	(.*)	15093022082	2	   

Here is a quick explanation of the different fields:

- **Direction:** here we set the technologies that will be connected by the gateway
- **From ID:** the SIP account or PSTN port that will be used a source technology
- **To ID:** the SIP account or PSTN port that will be used a destination technology
- **Destination:** the number we will call in order for the rule to apply. “(.*)” means anything. In this case, the rule will apply no matter the number we call
- **New destination:** the number that the gateway will actually send to the destination technology. “\1” takes what we had between the brackets in the “destination” field. In our case, it takes everything. With a Flowroute account, we need to add the “Tech prefix” in this field in order for the call to go thru. Just add the Tech prefix before the “\1” in order for the call to be sent to Flowroute in a proper way.
- **Source:** CID from which the call should be sent in order for the rule to apply. In this case, the rule will apply no matter which CID is calling
- **New Source:** the CID that the gateway will actually show to the destination technology. In a case of an analog telephone, we can add here the number it should generate. In my case, I added the number of my SIP trunk.