beroNet

Cloud managed & monitored
VoIP Gateways and Appliances

How to:

Become a beroNet certified partner

Author: Jean-Eudes AMBROISE – Sales manager at beroNet GmbH

beroNet Partner

Index

About the beroNet technical training:

The beroNet technical training aims at teaching beroNet partner candidates how to install a well-working VoIP system using beroNet appliances, gateways and the beroNet cloud.

This technical training requires the devices available in the beroNet Starter Pack. These are:

- One BNTA20-2S02FXS-L
- One BF4002S02FXSbox
- One beroNet Cloud account

Very important information: a m0n0wall (firewall / router) is pre-installed and running in a virtual machine on the appliance for the purpose of this training.

DO NOT PLUG THE LAN PORT TO YOUR NETWORK. ONLY PLUG THE WAN PORT.

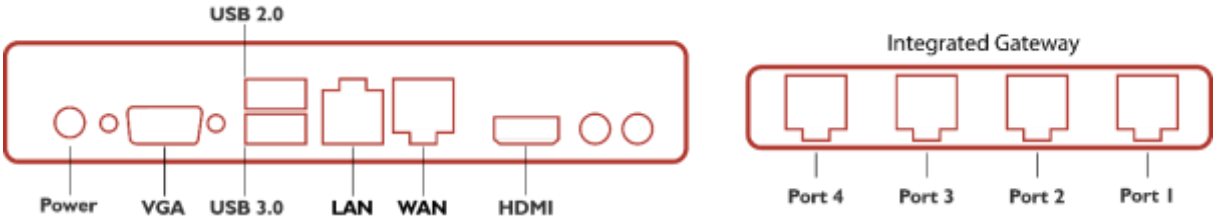


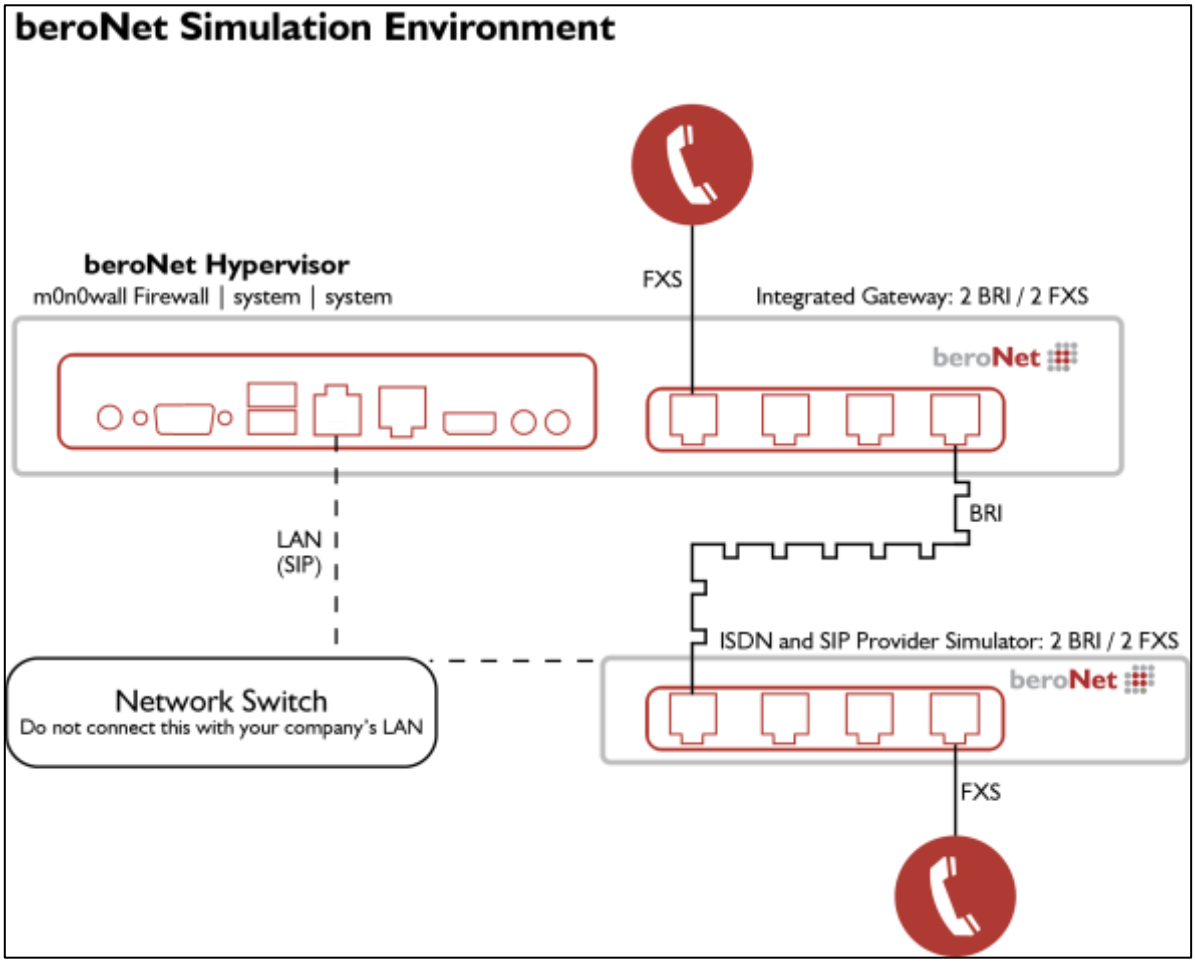
Diagram: beroNet Telephony Appliance 2.0

Presentation of the technical training

During this training you will install a complete VoIP system in a simulated environment; similar to what you will work with while installing VoIP equipment at a customer location.

- The beroNet gateway (BF4002S02FXSbox) will simulate an ISDN and SIP provider. An analog telephone will be connected to one of the FXS ports (port 3 or 4) to send and receive calls.
- The beroNet appliance with an integrated beroNet gateway will be used to simulate the internal network of a company. Preinstalled on the appliance is the beroNet Hypervisor enabling the device to run different virtual machines. The device will be used as:
    - A router / firewall. A m0n0wall virtual machine is pre-installed. With this you will be able to distinguish between the inside (LAN) and the outside (WAN) of the company. The IP address of the m0n0wall is 10.0.0.1.
    - A Gateway to connect the company to ISDN and SIP. An analog phone will be connected to the integrated gateway. Calls will be done through the integrated gateway to the simulated provider via SIP and ISDN.
- Both gateways and the hypervisor will be connected with the beroNet Cloud.

Here is a representation of the scenario:



Information about the settings:

- The external gateway is the ISDN and VoIP provider
- The m0n0wall is the router. It gives IP addresses to the IPBX simulator (internal gateway) and to the hypervisor (data center simulator)
- The IP range of the LAN is 10.0.0.10 to 10.0.0.99.
- The IP address of the m0n0wall is 10.0.0.1 and the one of the hypervisor is 10.0.0.4
- Both gateways are set with DHCP

At the completion of this training you will be able to make calls from within the appliance to the outside via SIP and ISDN. Here is how to proceed:

- Follow the steps outlined in this to create your virtual scenario.
- Send each trace taken during this process to training@beronet.com.
- Both gateways and the hypervisor will then be registered in the beroNet cloud.

Connecting the Appliance

During the training you will be using a closed network.

1. Connect the WAN port (the one on the right) to your local network. The m0n0wall will get a WAN IP address from your DHCP.
2. Connect the LAN port (the one on the left) to your computer.
3. Use the supplied power source to connect the appliance to electricity.

Test 1: Call via SIP

In this scenario you will connect the integrated gateway to the external one via SIP. To do this, configure the router pre-installed in the beroNet Hypervisor on the appliance.

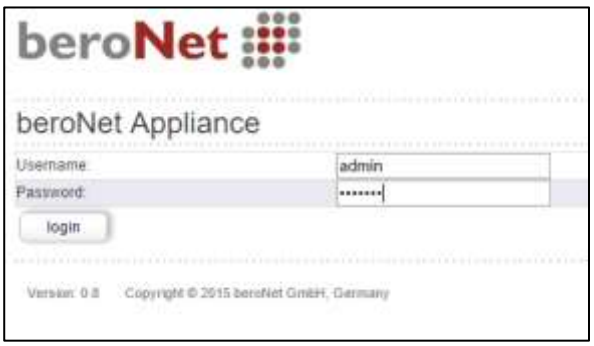
Configure the m0n0wall

Accessing the web interface of the hypervisor

M0n0wall is an open source firewall / router that runs on a VM of the hypervisor on the beroNet appliance. It has been pre-configured and a DHCP server is running on it. After you connect your computer to the LAN port (the one on the left) it will get an IP address with the 10.0.0.x subnet and will enable you to access the WEB GUI of the hypervisor (10.0.0.4).

Did you know: beroNet has developed a tool called bfdetect to identify and manage the IP-Addresses of beroNet technology in your LAN. This can be downloaded from: http://www.beronet.com/bfdetect

1. Enter the hypervisor's IP address in a browser (we recommend Mozilla Firefox or Microsoft Edge).
2. The credentials are "admin / beronet"



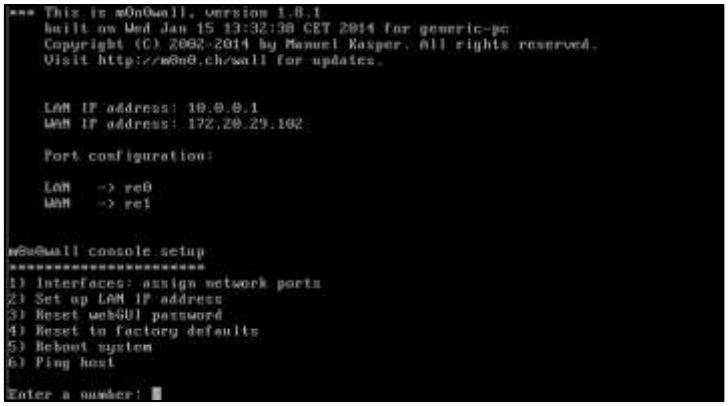
3. The dashboard of the hypervisor will load. Here you will see that the m0n0wall VM is already running.

More Info: More information about how the beroNet Hypervisor works is available at: http://www.beronet.com/products/telephony-appliance/ or http://wiki.beronet.com/index.php/BeroNet_Telephony_Appliance-v2

Accessing the web interface of the router

In order to access the m0n0wall web interface, we need its IP address.

1. Click on the "WEB-VNC (5901)" link. A new tab will open. Click on "connect" to access the configuration of the firewall. Here you can discover the LAN and WAN IP addresses of your m0n0wall. The login credentials are: admin / beronet



Additional Information: When accessing a VM via the WEB-VNC, we recommend using Mozilla Firefox or Microsoft Edge.

2. You can then access its web interface with the credentials "admin / beronet":
    a. With the LAN IP address when you are connected to the LAN of the appliance
    b. With the WAN IP address if your computer is not connected to the local network of the m0n0wall. To access it via the WAN, a port configuration has been configured, simply enter the port like this: "WANIP:2081". Ex.: 172.20.29.102:2081

Information regarding the pre-configuration of the m0n0wall :

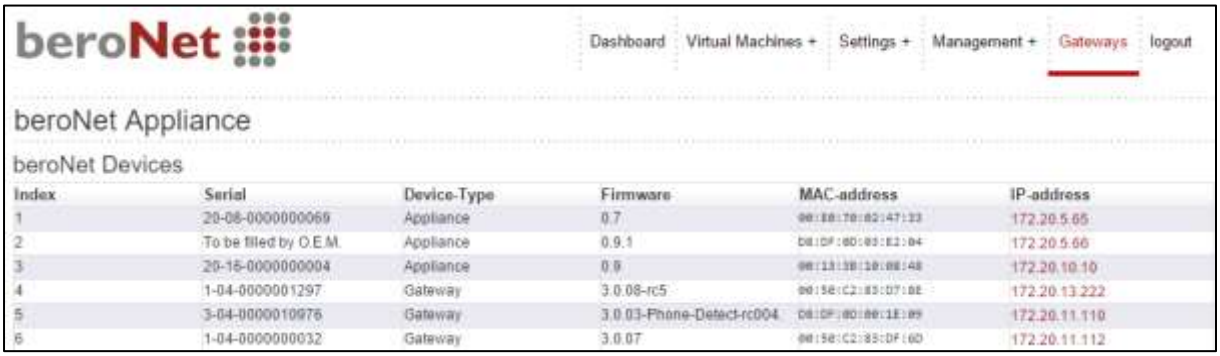
- DHCP is activated on LAN port. The DHCP range is set between 10.0.0.10 and 10.0.0.99.
- The LAN IP address of the LAN port is 10.0.0.1
- NAT rules have been configured for the hypervisor (port 2084) and the m0n0wall (port 2081).

3. Access the web interface of the m0n0wall using its LAN IP address.

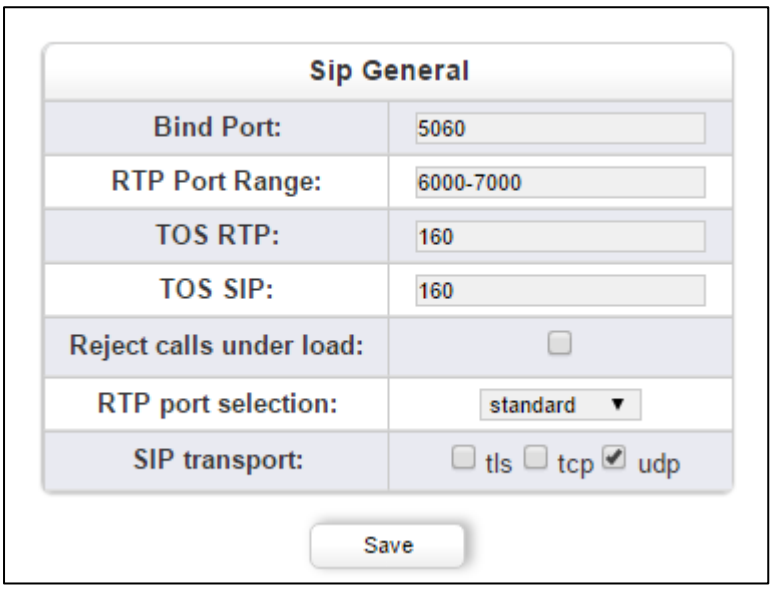
Port configuration of the m0n0wall

To allow incoming SIP calls from the provider we need to first configure our firewall ports. If this step is not done correctly the m0n0wall will block all calls coming from the provider.

1. Use a different browser window or tab to access the web interface of your gateway; you can discover it by using bfdetect or by clicking on "Gateways" on the web interface of the hypervisor. The default login credentials for a beroNet Gateway are: admin / admin

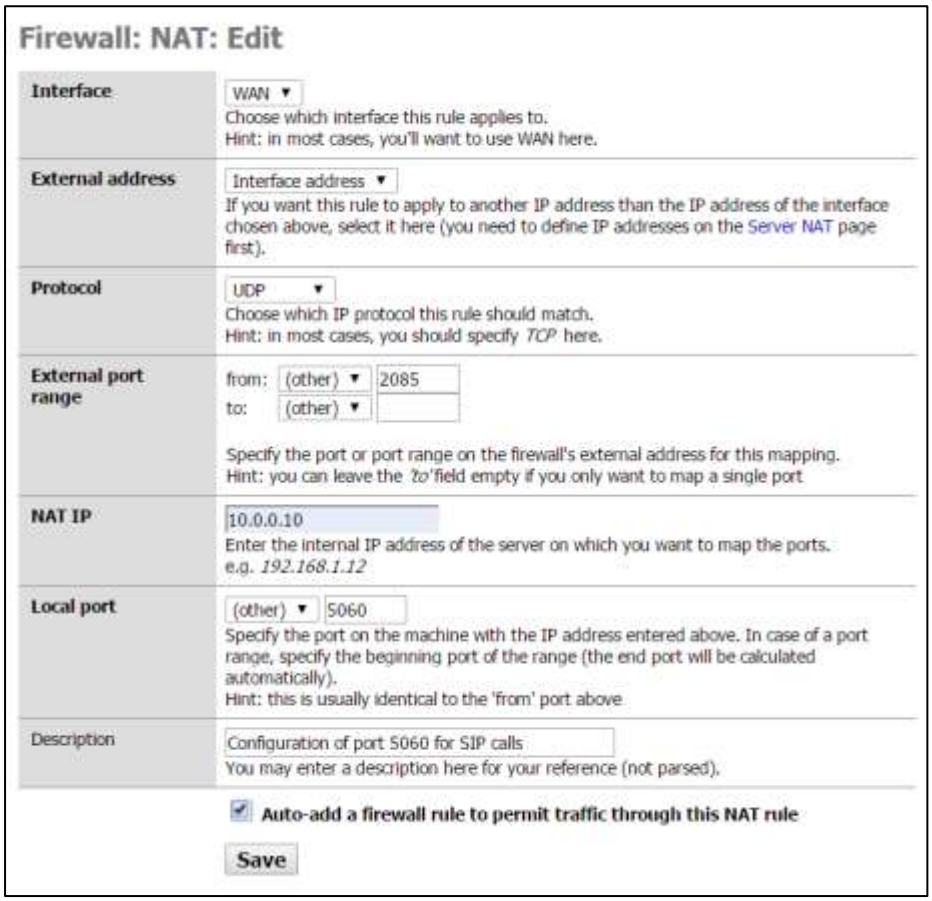


2. Look at the "SIP general" menu under "Sip+" in the WEB interface of the gateway, we see which ports need to be open in the firewall.

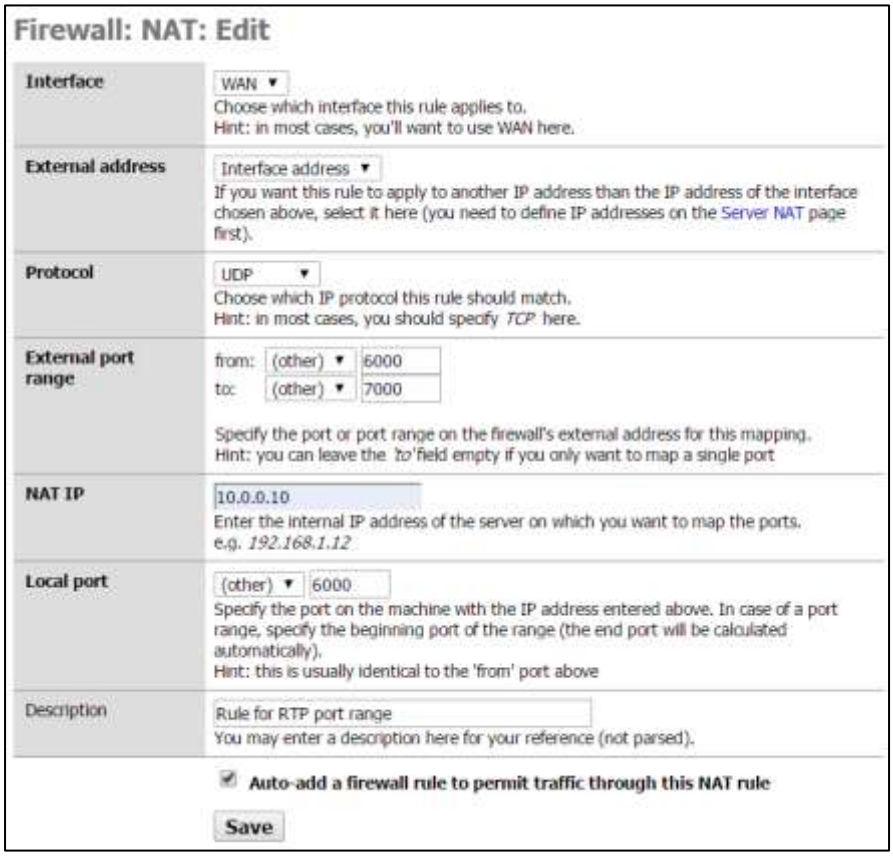


3. Return to the browser window or tab with the m0n0wall.

4. Go to “NAT” under “firewall” in the m0n0wall web interface.
5. Add a rule by clicking on the “+” icon and enter the following configuration:
    a. Interface: WAN
    b. External address: interface address
    c. Protocol: UDP
    d. External Port range: choose the port of your choice (I use 2085 in my case as it is more secure than using the port 5060 which is the standard VoIP port)
    e. NAT IP: enter the local IP address of the internal gateway (10.0.0.10 in my case)
    f. Local port: 5060 as it is the port set in the general SIP settings inside of the gateway
    g. Select “**Auto-add a firewall rule to permit traffic through this NAT rule”** in order to add a firewall rule that fits
    h. Save and activate the settings



6. A second NAT rule needs to be added in order to allow the audio in both ways. Here are the settings to choose in this second rule:
    a. Interface: WAN
    b. External address: interface address
    c. Protocol: UDP
    d. External Port range: set from 6000 to 7000
    e. NAT IP: enter the local IP address of the internal gateway (10.0.0.10 in my case)
    f. Local port: 6000
    g. Select "**Auto-add a firewall rule to permit traffic through this NAT rule"** in order to add a firewall rule that fits
    h. Save and activate the settings

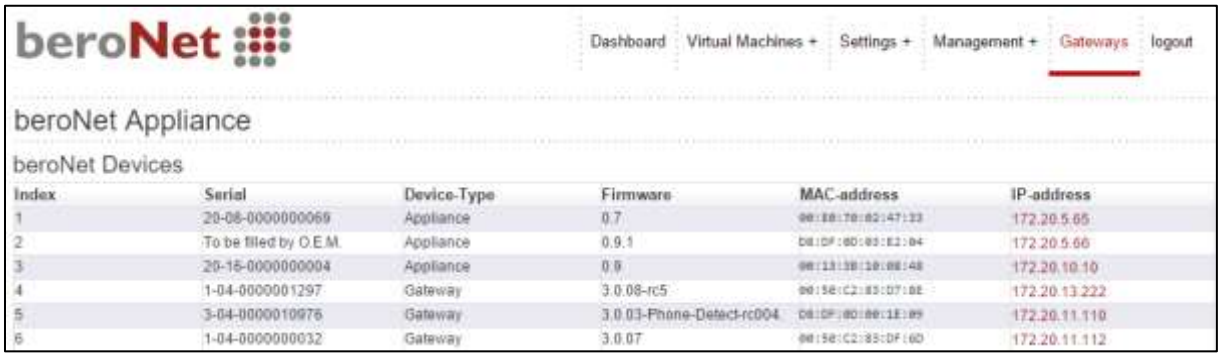


Configure the Gateways

The aim of this test is to call from one gateway to the other via SIP. The external gateway plays the part of a SIP provider. For this, add a SIP trunk in each gateway. You need to register the internal device on the external one and vice versa. In the "server address" field, simply enter the IP address of the other gateway.

Configure the integrated gateway

1. Access the web interface of your gateway; you can discover it by using bfdetect or by clicking on "Gateways" in the web interface of the hypervisor.

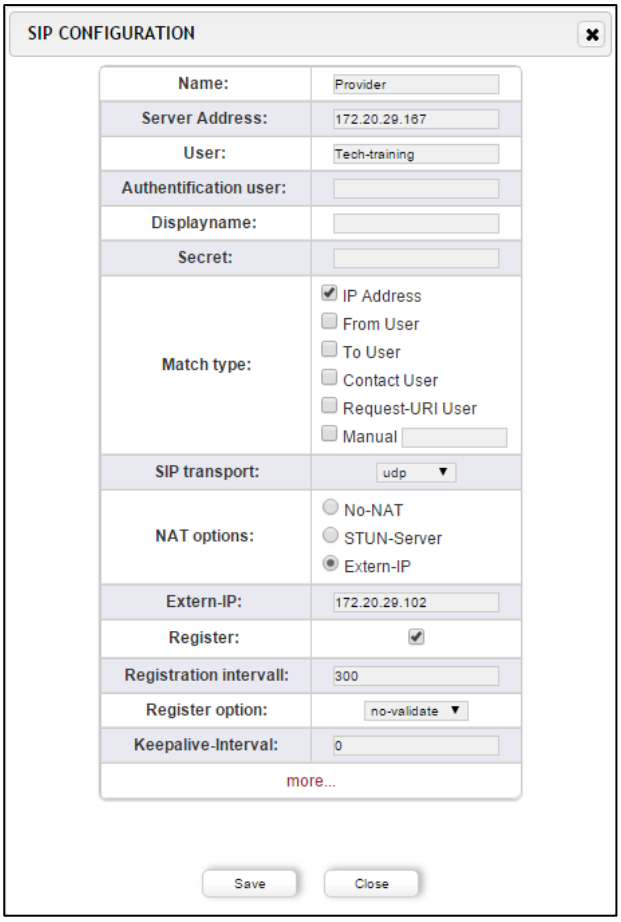


Index	Serial	Device-Type	Firmware	MAC-address	IP-address
1	20-08-0000000069	Appliance	0.7	00:50:70:02:47:33	172.20.5.65
2	To be filled by O.E.M.	Appliance	0.9.1	D8:DF:0D:03:E2:04	172.20.5.66
3	20-16-0000000004	Appliance	0.9	00:13:3B:10:08:48	172.20.10.10
4	1-04-0000001297	Gateway	3.0.08-rc5	00:50:C2:83:07:0E	172.20.13.222
5	3-04-0000010976	Gateway	3.0.03-Phone-Detect-rc004	D8:DF:0D:00:1E:09	172.20.11.110
6	1-04-0000000032	Gateway	3.0.07	00:50:C2:83:DF:0D	172.20.11.112

2. Click the link in the hypervisor or enter the IP address in your browser and login to the gateway Web interface using the default credentials: admin / admin

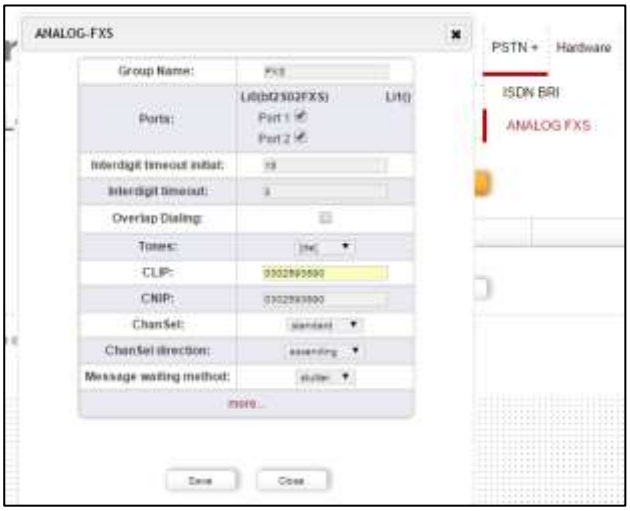
Configuration of the SIP account

1. Go to "SIP" under "SIP+". There, you can add a SIP account. Enter the following information:
    a. Name: choose a name of your choice
    b. Server Address: enter the IP address of the external gateway
    c. User: choose a user of your choice, this will be the same in the other gateway
    d. Secret: enter the password of your choice
    e. NAT-options: choose the extern IP option – the WAN of the m0n0wall (172.20.29.102 in my case)
    f. Register: select this option
2. Save



Configure the analog ports

1. Navigate to "Analog FXS" under "PSTN+".
2. Add a group and enter the following information:
    a. The name: enter a name of your choice
    b. Ports: choose one or both of them
    c. Tones: choose the tone of your country
    d. CLIP and CNIP: enter the telephone number the FXS port will simulate
3. Save
4. Activate



Configure of the dialplan

Two rules need to be configured in the dialplan:

1. Add a rule that routes all calls from the analog ports to SIP
2. Add a second rule that routes all calls from SIP to analog
   Here is an example of the two rules:

3. Activate the settings you have configured.

Configure the External Gateway

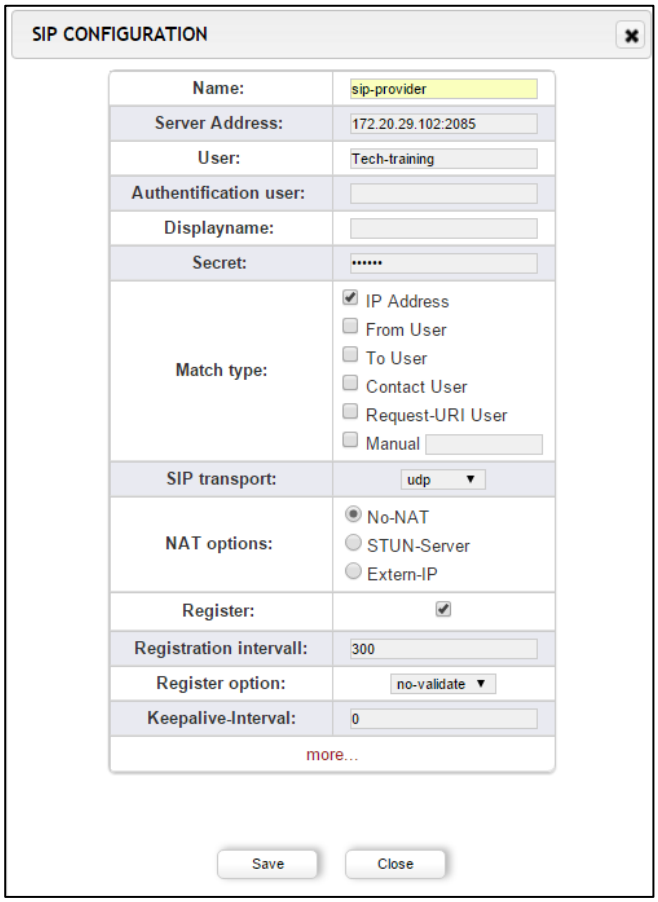
Accessing the gateway's web interface

Because the external gateway is not connected to the LAN of the m0n0wall you need to use the following steps to discover the IP address of the gateway.

1. Connect your computer to the LAN of your company
2. Use bfdetect to discover the IP address of the external gateway
3. Reconnect your computer to the LAN of the m0n0wall
4. Use your browser to navigate to the IP Address of the gateway and use the following credentials to login to the gateway's web interface: admin / admin

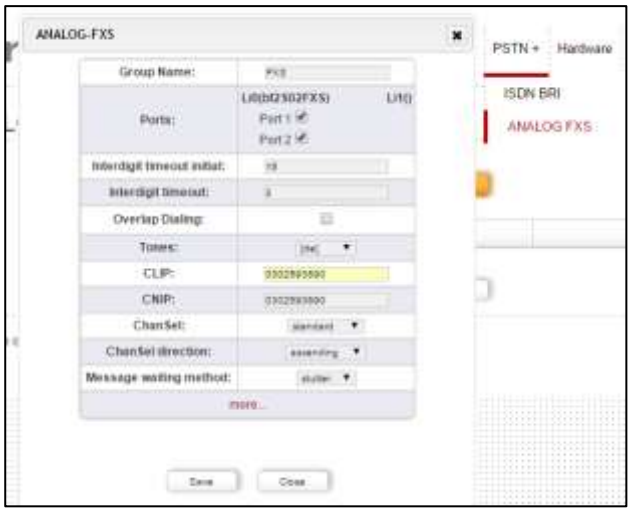
Configure the SIP account

1. Go to "SIP" under "SIP+". There, you can add a SIP account. Enter the following information:
    a. Name:
    b. Server Address: enter the WAN IP address of the m0n0wall and the port set for the gateway – 2085. In my case it is 172.20.29.102:2085
    c. User: choose a user of your choice, this will be the same in the other gateway
    d. Secret: enter the password of your choice
    e. NAT Options: leave the default "No-NAT" option
    f. Register: select this option
2. Save
3. Activate



Configure the analog ports

1. Navigate to "Analog FXS" under "PSTN+".
2. Add a group and enter the following information:
    a. The name: enter a name of your choice
    b. Ports: choose one or both of them
    c. Tones: choose the tone of your country
    d. CLIP and CNIP: enter the telephone number the FXS port will simulate
3. Save
4. Activate



Configure the dialplan

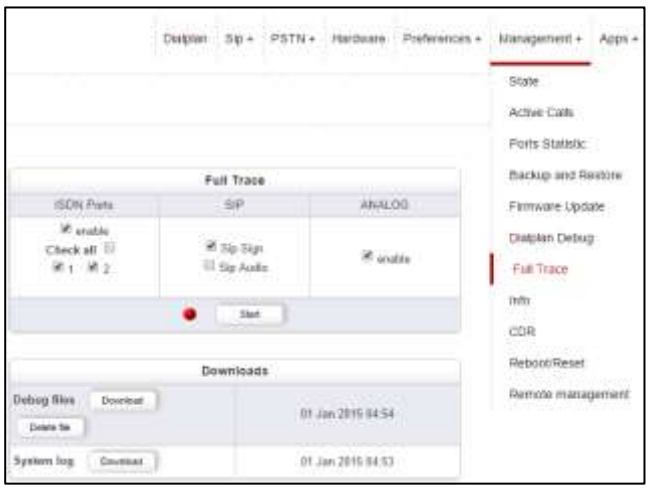
Two rules need to be configured in the dialplan:

1. Add a rule that routes all calls from the analog ports to SIP
2. Add a second rule that routes all calls from SIP to analog
   Here is an example of the two rules:

3. Activate the settings you have configured.

Test Run: Calls from Analog to SIP

Now that you have both devices configured, you can make a call from analog to SIP.



1. Navigate to "Fulltrace" under "Management+". Start the trace.
2. Make a call from the analog phone connected to the appliance to the phone connected to the external gateway using the number you entered into the "CLIP" field of that device.
3. Make a call from the analog phone connected to the external gateway to the phone connected to the appliance using the number you entered into the "CLIP" field of that device.
4. Stop the "Fulltrace" and if the calls were successful download the file. The file extension should be ".tar.gz". If the calls were not successful, review your settings and try again.

Test 2: calls via ISDN

In this section you will learn how to create ISDN rules and how the beroNet dialplan works.

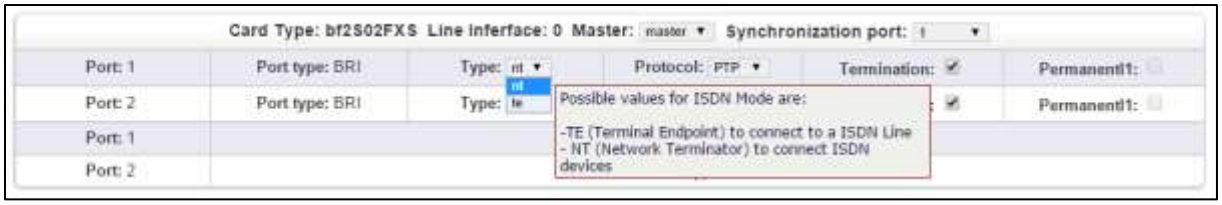
Configure the External Gateway

The external gateway will simulate the ISDN provider. Access its WEB interface again and configure the ISDN ports as follow:

Configure the ISDN Connection

First of all, as this gateway plays the ISDN provider, we need to set the right hardware settings for its ports.

1. Go to the "hardware" tab and set the "NT" mode (the appliance integrated gateway will have the "TE" mode).



2. Then, go to "ISDN BRI" under "PSTN+" and configure the ports as follow:
    a. Give a name to the group
    b. Choose the ports to add in the group
    c. Set the tone of your country
    d. Set the country code (49 in Germany)
    e. Set the city code (30 for Berlin)
    f. Save
3. Activate

Configure the Dialplan

Two rules need to be configured in the dialplan:

1. Calls coming from ISDN should be routed to the analog ports.
2. Calls coming from the analog ports should be routed to ISDN..

3. Activate the settings you have changed.

Configure the Integrated Gateway

We will configure this gateway so that calls to Germany (intended to represent local calls) will be sent via ISDN, and the rest via SIP.

Configure the ISDN Ports

1. Setup the same port configuration you used for the external gateway. Make sure that the hardware mode of the ports is set to "TE".
2. Use the orange ISDN cables delivered in the training kit to connect both gateways. **DO NOT USE A NORMAL ETHERNET CABLE!!!!**

Configure the Dialplan

Information about the beroNet dialplan

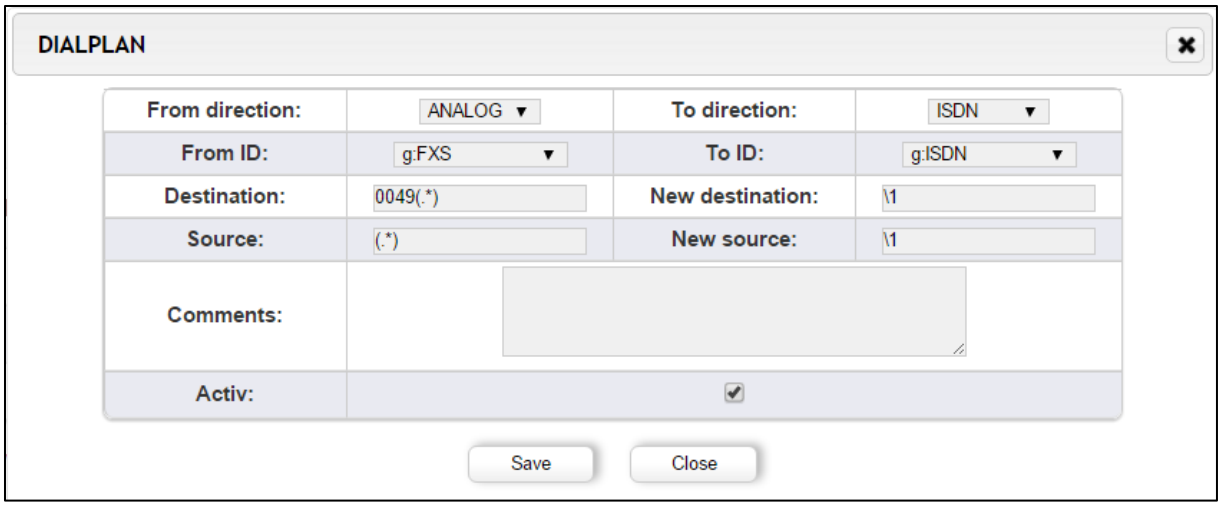
We need to add two additional rules and differentiate them. Here is the most important information to know about the dialplan:

- It uses regular expressions (more information here: http://www.zytrax.com/tech/web/regex.htm)
- It works from top to bottom. This means that general rules should be placed under precise ones. When a call comes in, the dialplan will go thru each rule, starting from the top. As a call comes through the device, the first rule that matches the parameters of this call (technology where it comes from, DID and CID) will be applied.

Precise dialplan rules

Currently the two existing rules send all calls coming from SIP to the analog ports and vice versa. Now we want to make sure that calls going to Germany are sent via ISDN. Because this new rule is more specific, it should be placed at the top of the dialplan.

1. Click on add and choose the following settings:
    a. From: Analog
    b. To: ISDN
    c. Destination: enter 0049(.*) meaning that this rule will apply for all calls dialing a number starting with 0049
    d. New destination: \1 meaning that the information within the brackets of the "destination" field will be kept
    e. Source and new source can be left empty



2. Click save. Now when you call a number starting with 0049, the call will be routed via ISDN.
3. Navigate to dialplan debut under "management+ to check whether the rule is working. Start a debug.
4. Attempt to make a call. You should see something like this:

Issue:

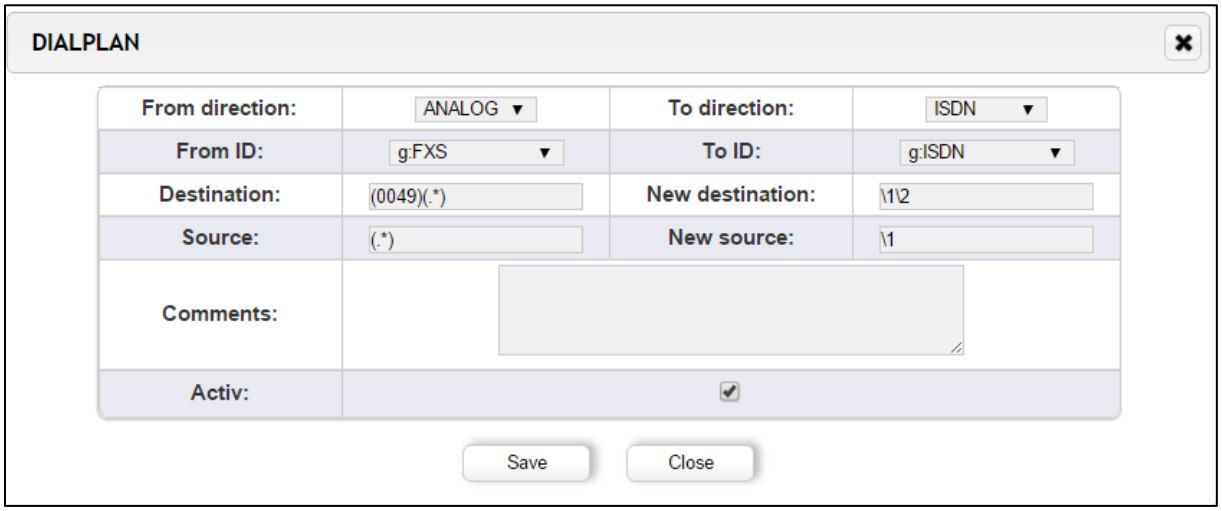
If you take a closer look at the rule, you will notice that the number I call is not properly sent to the ISDN line. The dialplan actually cuts the prefix "0049". According to the above screen shot, I called the number: 0049123456789 but the ISDN line only receives "dad=123456789". In order to keep the prefix, we need to slightly change the rule.

Changes in the rule:

According to the rule we created, the dialplan makes sure that all numbers starting with "0049" are sent to ISDN. The "\1" of the "new destination" field takes what we have between brackets. We notice that the prefix stands before the brackets, it is therefore cut.

1. Add another set of brackets and and "\2" in the "new destination" field.

Here is a screenshot:



2. Click Save. Now, when you call Germany, the number will be kept in the correct format.

The same rules apply for the fields "Source" and "New source". In order to have more information about the dialplan, please view our comprehensive article in the beroNet blog.

Test Run: Calls from Analog to ISDN

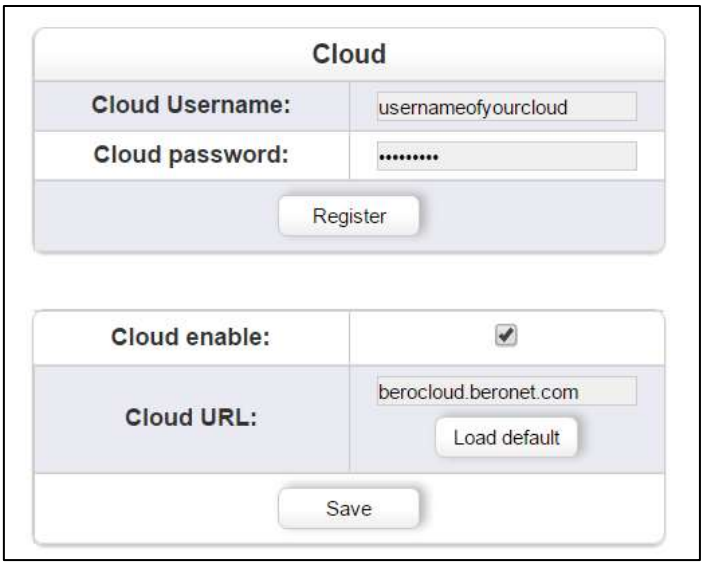
1. Make a trace of a call from analog to ISDN in each way.
2. Download the file (Debug files). This is one of the file that you need to send to training@beronet.com at the end of the training.

Connect the Devices to the beroNet Cloud

beroNet's Cloud service makes it possible for you to manage and monitor your devices from any browser, anywhere in the word.

Connect the Gateways to the Cloud

1. Navigate to "Remote management" under "management+".
2. Activate the option by ticking "Cloud enable".
3. Enter your Cloud Username and password and click on register.



Connect the beroNet Hypervisor to the Cloud

1. Go to "Cloud settings" under "Settings" in the WEB GUI of the hypervisor.
2. Enter your Cloud username and password and click on register

Summary of the training

During this training, we have configured the following:

- An internal VoIP system based on the beroNet appliance in which we configured a router and a beroNet gateway simulating a PBX.
- Calls can be sent to the outside via SIP or ISDN
- An external gateway plays the role of a SIP and ISDN provider in order to simulate the calls

In order to finish the training and become a beroNet partner, different traces need to be sent to training@beronet.com:

- The trace of two SIP calls: from and to the external gateway
- The trace of two ISDN calls: from and to the external gateway