

BerofixDocuV2

From berofix - professional SIP gateway

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BeroNet VoIP gateway family

The beroNet VoIP Gateway Family consists of the following products:

* beroNet VoIP card
 * modular beroNet Gateway
 * Small business Line
 * 16 FXS Gateway
 This Documentation applies to all above mentioned devices.

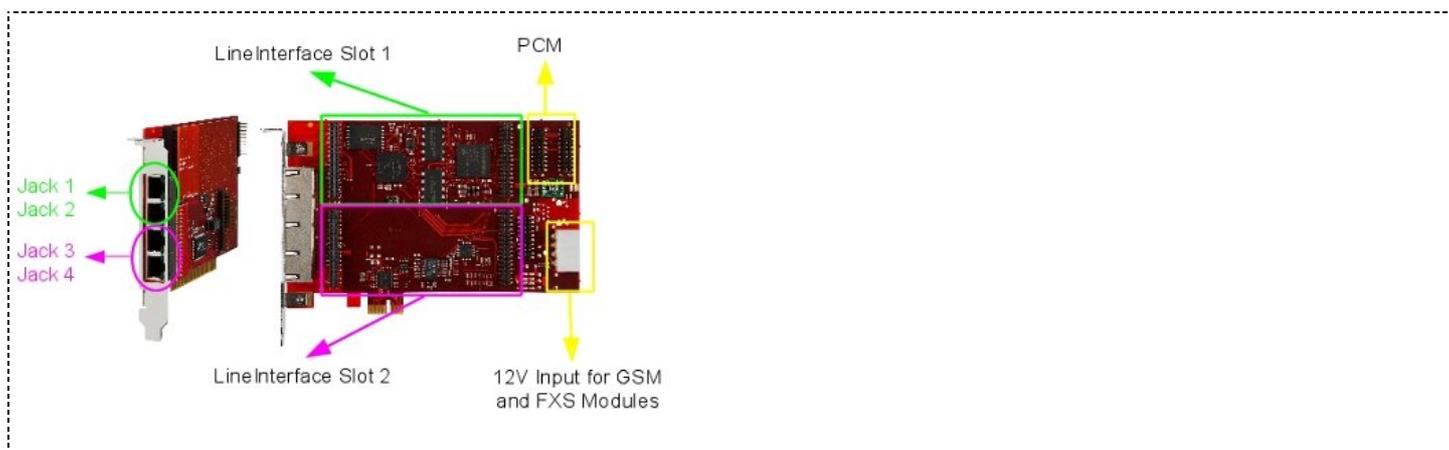
The beroFix product family are a powerful and flexible hardware solution to connect ISDN (BRI, PRI), analog (FXO/FXS) as well as GSM lines to any SIP based VoIP system. The beroFix product family consist of a baseboard and the so called modules. The beroFix baseboards are available as PCI- /PCIexpress-cards as well as an external Box variant. The baseboard is modular, so you can plug in and mix different Modules as analog (FXO/FXS), digital ISDN (BRI/PRI) as well as GSM modules all on the same baseboard. A PCM bus interconnection between beroFix baseboards enables you to bridge hardware to build a mixed and high-density system which transparently transmits Voice, Video, Data and Fax transmissions.



BeroFix PCI-/PCIe-cards are OS independent and can be used under Linux, Unix, Windows as well as MAC OS. This new advancement in technology incorporates a proprietary BeroNet OS on the baseboards which is automatically detected by your system as a 'Network card'. All necessary drivers are automatically loaded by the host machine's OS.

beroNet VoIP gateway baseboard overview

Independent of the type (PCI-/PCIe-card or external Box) the beroFix baseboard has 2 Lineinterface sockets ("Lineinterface Socket 0" / "Lineinterface Socket 1"). While "Lineinterface Socket 0" is in charge of Jack 1 & 2 "Lineinterface Socket 1" is in charge of jacks 3 & 4. The following picture should show you the scheme of the beroFix baseboards.



Depending on the Modules you are using, different slots with different Pin Out's are used which will be described in the following Chapters regarding the modules.

Bf4S0 4 port ISDN / BRI module

The bf4S0 module has 4 ISDN/BRI ports. The bf4S0 module can be configured individually for each port to NT (Network Termination) or TE (Terminal Equipment) mode. In either modes the module supports Point-To-Multi-Point (PMP) or Point-To-Point (PTP). The TE/NT mode as well as the line-termination of 100 ohms are selectable via software (Jumperfree).

Depending on which lineinterface socket you plug the bf4S0 module, the pinout is different. To best fit your needs the use of the bfBridge could be useful instead the bfTAdapters are necessary. The following Examples gives you an overview about it:

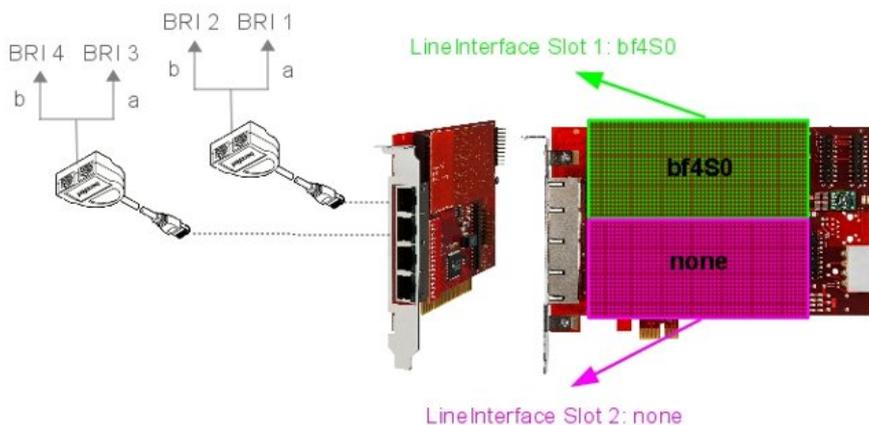
PinOut port 1 and 2 of bf4S0 module through Jack 1

Port 1:
 PIN 3 Tx+
 PIN 4 Rx+
 PIN 5 Rx-
 PIN 6 Tx-

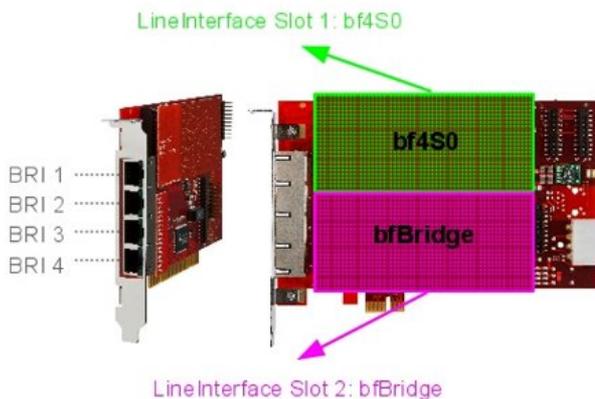
Port 2:
 PIN 1 Tx+
 PIN 2 Rx+
 PIN 7 Rx-
 PIN 8 Tx-

Port 3 and 4 are similar to the above mentioned scheme only they use Jack 2.

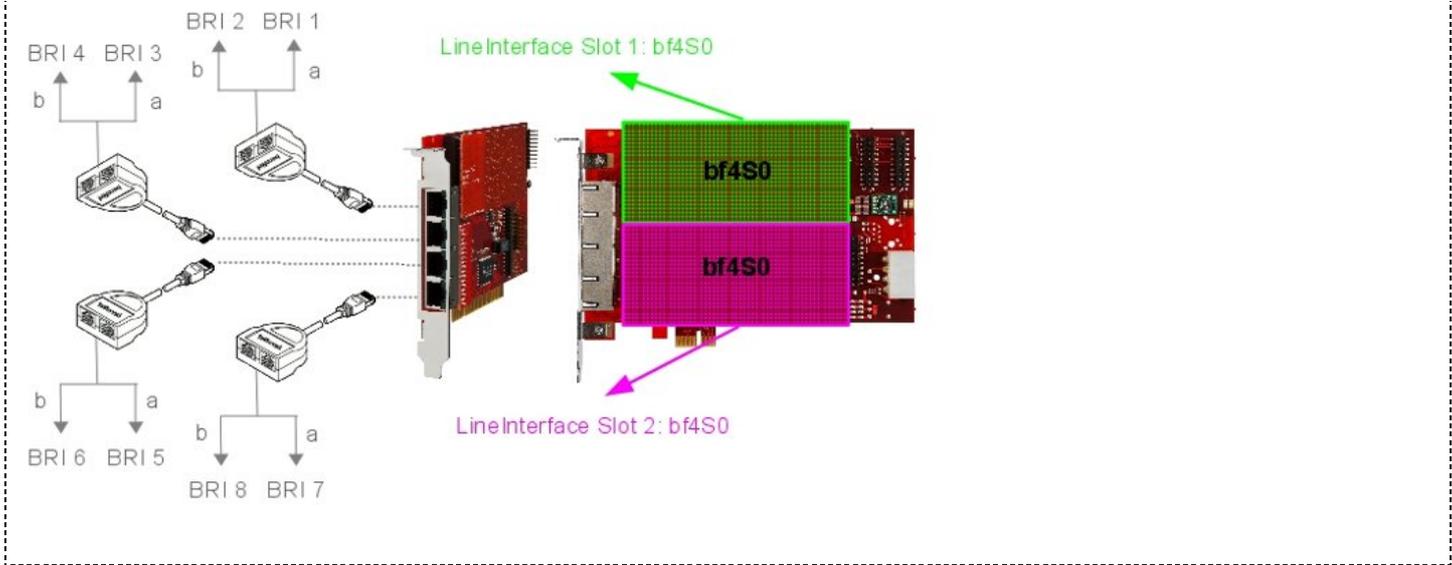
Example1:single bf4S0
 Lineinterfacesocket 0: bf4S0
 Lineinterfacesocket 1: none



Example2:bf4S0 with a bfBridge
 Lineinterfacesocket 0: bf4S0
 Lineinterfacesocket 1: bfbridge



Example3: 2x bf4S0
 Lineinterfacesocket 0: bf4S0
 Lineinterfacesocket 1: bf4S0



BfXE1 1-2 port ISDN PRI module

The bfXE1 PRI modules are available as 1 port (bf1E1) or as 2 port (bf2E1) variants. The bfXE1 module can be configured individually for each port to NT (Network Termination) or TE (Terminal Equipment) mode. If you want to use NT-mode you may need a cross cable which is optionally available (the bnE1Cross). Line termination (120 ohms) is selectable for each port by DIP switches on the module(2 per port).

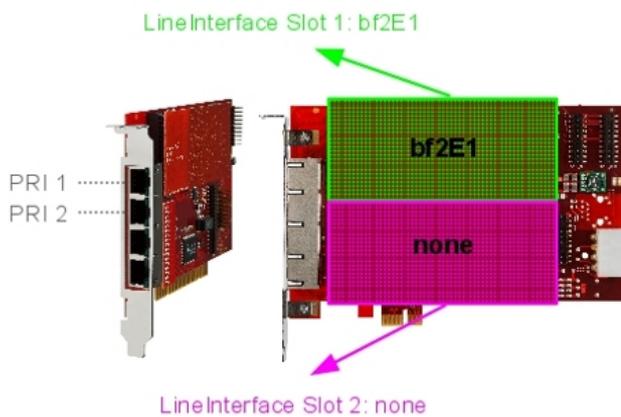
PinOut of bfXE1 module

- PIN 1: RX-
- PIN 2: RX+
- PIN 4: TX-
- PIN 5: TX+

The following examples will show you the different combinations of the module:

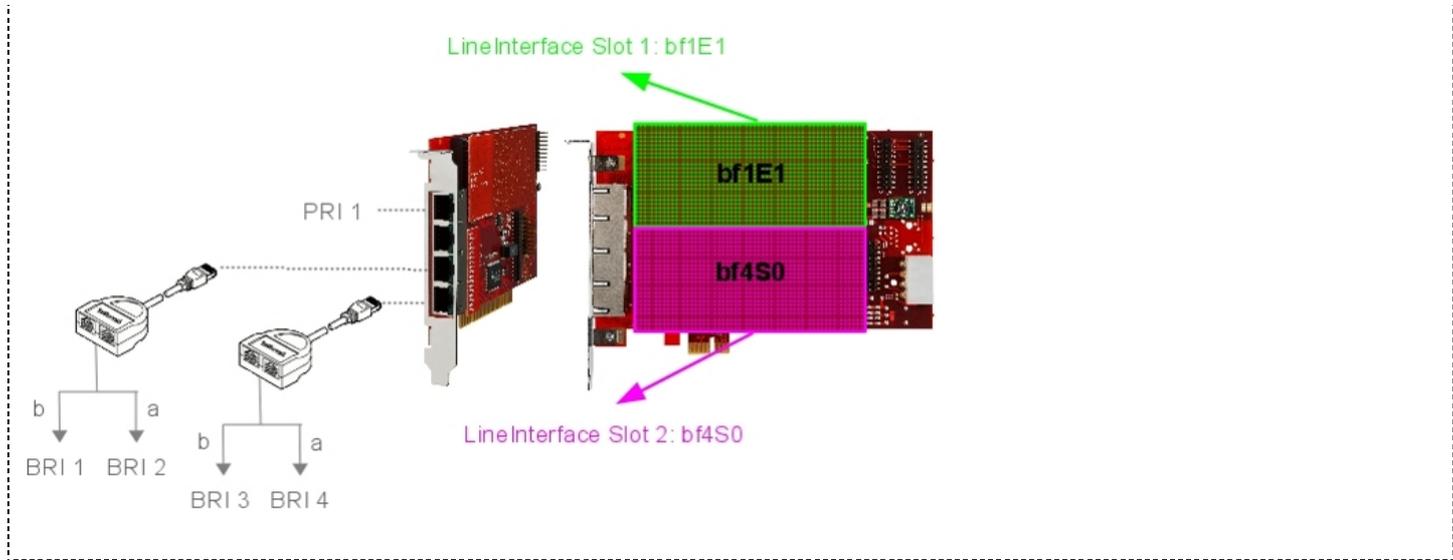
Example1: 1x bf2E1

```
Lineinterfacesocket 0: bf2E1
Lineinterfacesocket 1: none
```



Example2: 1x bf1E1 and 1x bf4S0

```
Lineinterfacesocket 0: bf1E1
Lineinterfacesocket 1: bf4S0
```



Bf4FXS 4 port analog module

The BF4FXS module is a 4 port FXS module for the beroFix baseboards and boxes. The BF4FXS module is used by small and medium enterprises which need internal analog FXS ports to connect analog phones, faxes and modems. Our 4 port FXS module is fully compatible and combinable with all other beroFix modules, cards and boxes.

The pinout of the bf4FXS module is similar to that of the bf4S0 module. Depending on the module configuration the use of the bfBridge or the bfTAdapters could be necessary. The following Examples give you an overview about the Pinout of this module:

```

PinOut port 1 and 2 of bf4FXS module through Jack 1
Port 1:
PIN 4
PIN 5

Port 2:
PIN 2
PIN 7
    
```

Port 3 and 4 are similar to the above mentioned scheme but use Jack 2.

Attention: When using this module on a beroFix PCI-/PCIe-card, the card has to be powered additionally, by using the 12V Molex Connector (See berofix product overview)

Bf8FXS 8 port analog module

The BF8FXS module is a 8 port FXS module for the beroFix baseboards and boxes. The BF8FXS module is used by small and medium enterprises which need internal analog FXS ports to connect analog phones, faxes and modems. Our 4 port FXS module is fully compatible and combinable with all other beroFix modules, cards and boxes.

The pinout of the bf8FXS module is similar to that of the bf4S0 module. Depending on the module configuration the use of the bfBridge or the bfTAdapters could be necessary. The following Examples give you an overview about the Pinout of this module:

```

PinOut port 1 - 4 of bf8FXS module through Jack 1
Port 1:
PIN 4
PIN 5

Port 2:
PIN 3
PIN 6

Port 3:
PIN 2
PIN 7

Port 4:
    
```

```
PIN 1
PIN 8
```

Port 5-8 are similar to the above mentioned scheme but use Jack 2.

Attention: When using this module on a beroFix PCI-/PCIe-card, the card has to be powered additionally, by using the 12V Molex Connector (See berofix product overview)

Bf4FXO 4 port analog module

The bf4FXO module is a 4 port FXO module for the berofix Mainboards and Boxes. The bf4FXO provides 4 FXO (Foreign Exchange Office) ports for the beroFix. It can be used to connect a Softpbx to the PSTN as well as a traditional PSTN based PBX. It is fully compatible and combinable with all beroFix modules, cards and boxes.

The pinout of the bf4FXO module is similar to that of the bf4S0 module. Depending on the module configuration the use of the bfBridge or the bfTAdapters could be necessary. The following Examples gives you an overview about the Pinout of this module:

```
PinOut port 1 and 2 of bf4FXO module through Jack 1
```

```
Port 1:
```

```
PIN 4
PIN 5
```

```
Port 2:
```

```
PIN 2
PIN 7
```

Port 3 and 4 are similar to the above mentioned scheme but they use Jack 2.

Bf2S02FXS 2 ISDN BRI Ports / 2 Analog FXS Ports Hybrid module

The BF2S02FXS Hybrid Module is a 2 Port BRI and 2 Port FXS Module for the berofix Baseboards and Boxes. Each BRI port of the BF2S02FXS can be configured as NT (Network Termination) or TE (Terminal Equipment). In both modes, each BRI port can operate in Point-to-Multi-Point (PMP) or in Point-to-Point (PTP) mode. The switching of the TE / NT mode and the deactivation of the termination resistors can easily be set in the software (Jumper Free) which makes the troublesome repositioning of jumpers unnecessary. Additionally to the above mentioned 2 BRI Ports the BF2S02FXS module provides 2 Ports FXS (Foreign Exchange Station) for the berofix. FXS interfaces are used to connect devices like analog phones or faxmachines to your berofix.

Attention: When using this module on a beroFix PCI/PCIe card, the card has to be powered additionally, by using the 12V Molex Connector (See berofix product overview)

Depending on the module configuration the use of the bfBridge or the bfTAdapters could be necessary. The following Examples gives you an overview about the Pinout of this module:

```
PinOut BRI Port 1 and FXS Port 1 of bf2S02FXS module through Jack 1
```

```
BRI Port 1:
```

```
PIN 3
PIN 4
PIN 5
PIN 6
```

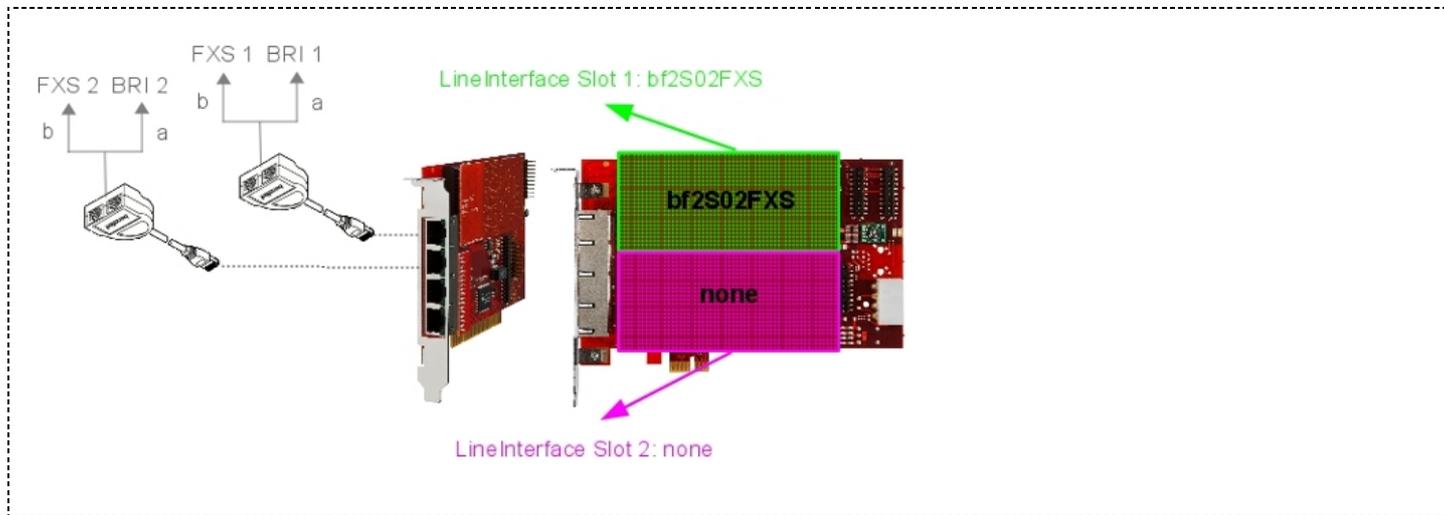
```
FXS Port 1:
```

```
PIN 2
PIN 7
```

BRI Port 2 and FXS Port 2 are similar to the above mentioned schema but uses Jack 2.

```
Example1:single bf2S02FXS
```

```
Lineinterfacesocket 0: bf4S0
Lineinterfacesocket 1: none
```



Example2:bf2S02FXS with a bfBridge

Lineinterfacesocket 0: bf480

Lineinterfacesocket 1: bfbridge



Bf2GSM 2 port GSM module

The bf2GSM module is a 2 port (QuadBand(850/900/1800/1900 Mhz) GSM module for the berofix baseboards and boxes. The BF2GSM is used by small enterprises which need up to 2 concurrent GSM calls. It can also be used for mobile convergence, LCR and SMS. Our GSM module is fully compatible and combinable with all other berofix modules and Cards.

Attention: When using this module on a beroFix PCI-/PCIe-card, the card has to be powered additionally, by using the 12V Molex Connector (See berofix product overview)

Regarding Voice Calls, the GSM Module behaves like all other modules. For SMS it has multiple interfaces:

1. Web-interface
2. Bero-API
3. Email-interface

The web-interface is explained below in this document: [GSM Options (http://wiki.beronet.com/index.php/BerofixDocuV2#GSM_options)]. The Bero-API makes it possible to Send and Receive SMS by external Applications via HTTP Requests, for more details look at: [SMS API (http://wiki.beronet.com/index.php/Howto_use_berofix_API#GSM_SMS)]

The email-interface is realized as a beroFix App called "smgw". It can be installed via the Firmware Installer and has it's own configuration web-interface. Details on this coming soon..

Upgrading beroFix box/card to be GSM Ready box/card

The GSM module can only be used with one of 3 different GSM upgrade kits:

1. BFEXTGSMBOX
2. BFEXTGSMCARD
3. BFGSMBracket

To upgrade a beroFix box with such a kit see the upgrade howto:

<http://www.beronet.com/downloads/berofix/docs/Howto-GSM-Upgrade-box.pdf>

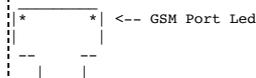
To upgrade a beroFix card with such a kit see the upgrade howto:

<http://www.beronet.com/downloads/berofix/docs/Howto-GSM-Upgrade-card.pdf>

NOTE: The GSM Module behaves like a BFBridge, which means that in combination with a 4 Port Module (BF4S0,BF4FX0,BF4FXS,BF2S02FXS) it should always be put in slot 2. This way you don't need to use the BFTadapter, but you can directly use all 4 RJ45 Ports.

LED Status

The bF2GSM Module uses always the right hand LED of the RJ45 Jack.



If the BF2GSM module is on the second slot(LI1) and a 4 port module is on the first slot(LI0) (eg. 4 BRI) then the RJ45 Jack 3 and 4 look like:



The GSM port LED states are:

OFF - Not initialized
 Blinking - initialized, trying to connect to network
 ON - Network in Reach (not necessarily registered to network)

Network setup

The next chapters will describe the basic steps that are necessary to bring your new berofix Box or card into your network, so that you can access the web-interface for further configurations.

BeroFix box network setup

By default all berofix devices use the same following network settings:

- **IP Address 10.0.0.2**
- **Subnet mask 255.0.0.0**

You can access the berofix Box WebInterface by typing <http://10.0.0.2> in your favourite Webbrowser. To be able to do this, the machine on which you want to use your favourite webbrowser has to be in the same network segment. Please check your Operating System manual for information on how to change the IP address / Subnet mask of your Network devices. You can log into the web-interface using the following default login credentials:

- **username:admin**
- **password:admin**

BeroFix PCI/PCIe cards network setup

The berofix PCI/PCIe cards use Realtek Network Chip-sets to communicate with the Host PC. This means that the berofix PCI/PCIe cards is recognized as a network card from your Operating System. You don't have to compile or load any additional driver except the network card driver for berofix PCI/PCIe cards, which should normally be done automatically by your Operating System. The berofix PCI cards use a RTL8139 Chipset and the berofix PCIe cards use a RTL8111 Chip-set. If you plug the berofix PCI/PCIe card into your computer it will have a new Network device. You can identify this new Network device by its MAC address, because all berofix devices use MAC addresses beginning with **D8:DF:0D** or **00:50:C2**.

By default the berofix PCI/PCIe cards have the IP-Address 10.0.0.2 with the Subnet mask 255.0.0.0. To be able to reach the berofix PCI/PCIe card you have to give your new Network device an IP-address in the same Network segment, e.g. IP: 10.0.0.1 with the same Subnet mask (255.0.0.0). Please check in your Operating System manual for information on how to change the IP address / Subnet mask of a device. After you have done these steps, you should be able to access the web-

interface of the berofix PCI/PCIe card by typing `http://10.0.0.2` in your preferred browser. You can log into the web-interface by using the default login credentials `username:admin` and `password:admin`.

Keep in mind that in this scenario the berofix card is only accessible from your machine, but not over the network. For several reasons, especially for SIP re-invites, it might make sense to advance your network settings so that the berofix PCI/PCIe card is reachable from your whole network. There are 2 common ways how to do this.

- Ethernet Bridge (recommended way)
- IP-Forwarding with a Static Route

These two ways will be described in more detail in the following chapters. We recommend to use the bridge setup, since this is the easiest to setup and allows the most flexibility.

BeroFix PCI/PCIe Network Setup using IP-Forwarding

Next we will show you how to configure IP-Forwarding on Linux and Windows based Machines, so that your berofix Card can be accessed via Network.

Linux IP-Forwarding

Before we start explaining how to setup IP-Forwarding, we need to give the berofix card an IP address. We will use 'ifconfig' to check out network settings.

berofix: `#ifconfig -a`

```
eth0  Link encap:Ethernet  HWaddr 00:0C:6E:D3:86:C6
      UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
      RX packets:797562 errors:0 dropped:0 overruns:0 frame:0
      TX packets:35206 errors:0 dropped:0 overruns:0 carrier:20
      collisions:7447 txqueuelen:1000
      RX bytes:141351732 (134.8 MiB)  TX bytes:19431622 (18.5 MiB)
      Interrupt:11 Base address:0xc000

eth1  Link encap:Ethernet  HWaddr 00:50:C2:83:D0:01
      UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
      RX packets:35124 errors:0 dropped:0 overruns:0 frame:0
      TX packets:276362 errors:0 dropped:0 overruns:0 carrier:0
      collisions:0 txqueuelen:1000
      RX bytes:20636164 (19.6 MiB)  TX bytes:28931756 (27.5 MiB)
      Interrupt:11 Base address:0xa000

lo    Link encap:Local Loopback
      inet addr:127.0.0.1  Mask:255.0.0.0
      UP LOOPBACK RUNNING  MTU:16436  Metric:1
      RX packets:9 errors:0 dropped:0 overruns:0 frame:0
      TX packets:9 errors:0 dropped:0 overruns:0 carrier:0
      collisions:0 txqueuelen:0
      RX bytes:720 (720.0 b)  TX bytes:720 (720.0 b)
```

In this example eth0 is the NIC in our computer and eth1 is the berofix NIC. The default internal IP-address of the berofix Card is 10.0.0.2. So need to give eth1 an IP address like 10.0.0.1, to access the berofix device. Take a look at the small scenario to make this more clear:

media:card-scenario.png

We can change the IP-Address of eth1 (berofix) by using ifconfig as described in the following example: `berofix:#ifconfig eth1 10.0.0.1 up`

To check this is working, try to ping 10.0.0.2

```
berofix:#ping 10.0.0.2
PING 10.0.0.2 (10.0.0.2) 56(84) bytes of data:
64 Bytes from 10.0.0.2: icmp_seq=1 ttl=63 time=1.12 ms
```

By default any modern Linux distribution will have IP Forwarding disabled. This is normally a good idea, as most people will not need IP Forwarding, but in our case we need it to support SIP reinvites. We can query the sysctl kernel value 'net.ipv4.ip_forward' to see if IP Forwarding is enabled or not. See the next examples:

```
sysctl net.ipv4.ip_forward
net.ipv4.ip_forward = 0
```

or just check out the value yourself in the /proc system

```
cat /proc/sys/net/ipv4/ip_forward
```

As we can see in the examples above IP Forwarding is disabled and represented by the value 0 for 'net.ipv4.ip_forward'. To Enable IP Forwarding you can use sysctl again, like described in the next example:

```
sysctl -w net.ipv4.ip_forward=1
or
echo 1 > /proc/sys/net/ipv4/ip_forward
```

Note that if you use sysctl as described in the above mentioned examples, IP-Forwarding will only be activated temporarily, that means this change will not be saved persistent. To make this setting persistent you should edit the configuration file `/etc/sysctl.conf` and add the following line:

```
net.ipv4.ip_forward = 1
```

If you already have an entry 'net.ipv4.ip_forward' with the value "0" you can change it to "1". To activate this changes it's necessary to run the command:

```
sysctl -p /etc/sysctl.conf
```

On Red Hat based systems this is also done by restarting the network service:

```
service network restart
```

On Debian/Ubuntu systems this can also be achieved by restarting the 'procps' service:

```
/etc/init.d/procps.sh restart
```

After activating IP-Forwarding the berofix device will still not be accessible from other devices in your LAN. To make it accessible you have to add a route either by adding a static route in your router or just change the standard gateway of the device from which you want to reach berofix:

```
route add 10.0.0.2 <ip-eth0-of-berofix-host>
```

or change standard gateway

```
route add default gw <ip-eth0-of-berofix-host>
```

If everything was successfully configured, you should be able to reach the berofix device from your network

Windows IP-Forwarding

Before explaining how to setup IP-Forwarding we need to assign an IP Address to the berofix PCI/PCIe card. Windows based systems can use ipconfig, from the command line interface, to check the network settings:

```
C:\Documents and Settings\Administrator>ipconfig -all

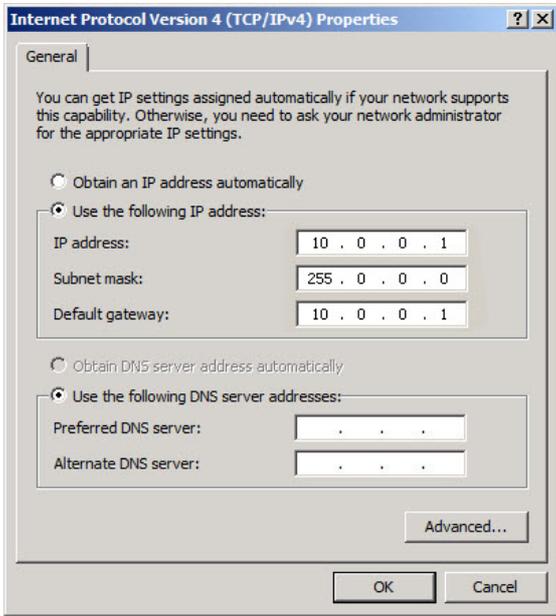
Windows-IP Configuration

   Hostname . . . . . : test-li3hfoglbs
   Primary Dns Suffix . . . . . :
   Node Type . . . . . : Mixed
   IP Routing Enabled . . . . . : No
   WINS Proxy Enabled . . . . . : No
   DNS Suffix Search List . . . . . : beronetoffice.local

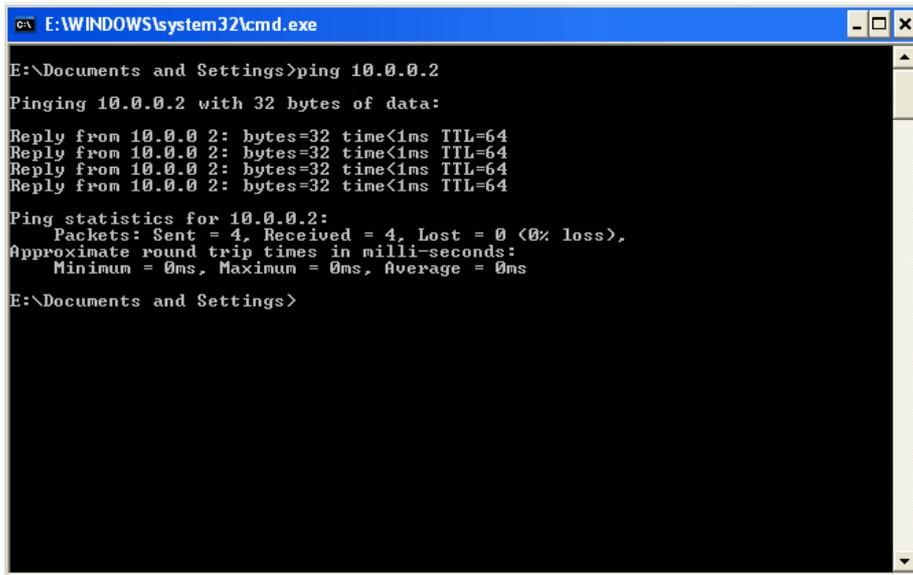
Ethernet adapter Local Area Network Connection 2: this is our berofix PCI card as you can see looking at it's MAC-Address
   Connection-specific Dns Suffix . . :
   Description . . . . . : Realtek RTL8139-Familie-PCI-Fast Ethernet-NIC #2
   Physical Address . . . . . : 00-50-C2-83-D0-00
   DHCP Enabled . . . . . : No
   IP Address . . . . . : 10.0.0.1
   Subnet Mask . . . . . : 255.0.0.0
   Default Gateway . . . . . :

Ethernet adapter Local Area Network Connection: this is the standard NIC. It has the same RTL Chip but a different MAC-Address
   Connection-specific Dns Suffix . . : beronetoffice.local
   Description . . . . . : Realtek RTL8139-Familie-PCI-Fast Ethernet-NIC
   Physical Address . . . . . : 00-0B-6A-EE-E7-4F
   DHCP Enabled . . . . . : Yes
   Autoconfiguration Enabled . . . . . : Yes
   IP Address . . . . . : 172.20.5.75
   Subnet Mask . . . . . : 255.255.0.0
   Default Gateway . . . . . : 172.20.0.1
   DHCP Server . . . . . : 172.20.0.1
   DNS Server . . . . . : 172.20.0.11
                           172.20.0.1
   Primary WINS Server . . . . . : 172.20.0.11
   Lease Obtained . . . . . : Monday, 15. June 2009 11:58:46
   Lease Expires . . . . . : Monday, 15. June 2009 23:58:46
```

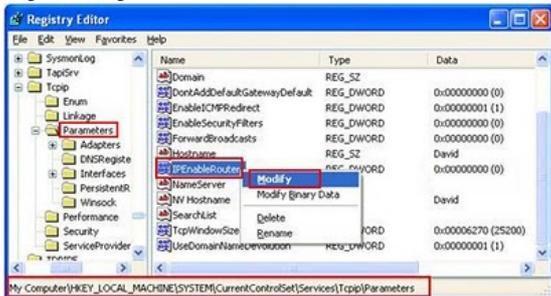
To assign an IP-address to the berofix device go to "Control Panel" and open "Network connections". Double click on "Local Area Network Connection 2" (Realtek RTL8139-Familie-PCI-Fast Ethernet-NIC #2), which is our berofix PCI card. Go to "Preferences" and choose "Internetprotocol (TCP/IP)" to get to the IP-Configuration dialog:



As seen in the picture above we have assigned the IP Address, the Subnet mask as well as the default gateway. After the network connection has been restarted, you can open the command line interface and use the "ping" command to verify that the IP Connection is working:



As you can see the berofix device is now reachable from this particular machine. To enable Windows IP-Forwarding you can use "regedit" in the command prompt to get the RegEdit32-Tool:



To enable IP-Forwarding please go to "**HKEY_LOCAL_MACHINE\System\CurrentControlSet\Services\Tcpip\ Parameters**" and change the entry "IPEnableRouter" (Type REG_DWORD) and change it to "1".

After activating IP-Forwarding the berofix device will still not be accessible from other devices in your LAN. To make it accessible you have to add a corresponding route. You can add a static route by entering the following in the command prompt:

```
route add 10.0.0.2 <ip-eth0-oE-berofix-host>
```

If you want to make this settings persistent please give the "-p" parameter to the route tool. If everything was successfully configured, you should be able to reach the berofix device from your network.

Berofix PCI/PCIe Network Bridge Setup

In the following Chapters we will describe how you can configure a Network Bridge under Linux and Windows based systems.

Linux Network Bridge

You can setup the Network Bridge under Linux with the "brctl" utility. This utility is provided by the "bridge-utils" package on debian based systems (like Ubuntu). As mentioned above the berofix PCI/PCIe cards use MAC Addresses beginning with **D8:DF:0D** or **00:50:C2**, so you can again use the "ifconfig -a" command to identify berofix PCI/PCIe cards:

```
beropbx:#ifconfig -a

eth0  Link encap:Ethernet  HWaddr 00:0C:6E:D3:86:C6  this is standard NIC
      UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
      RX packets:797562 errors:0 dropped:0 overruns:0 frame:0
      TX packets:35206 errors:20 dropped:0 overruns:0 carrier:20
      collisions:7447 txqueuelen:1000
      RX bytes:141351732 (134.8 MiB)  TX bytes:19431622 (18.5 MiB)
      Interrupt:11 Base address:0xc000

eth1  Link encap:Ethernet  HWaddr 00:50:C2:83:D0:01  this is our berofix PCI card as you can see on the MAC-Address
      UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
      RX packets:35124 errors:0 dropped:0 overruns:0 frame:0
      TX packets:276362 errors:0 dropped:0 overruns:0 carrier:0
      collisions:0 txqueuelen:1000
      RX bytes:20636164 (19.6 MiB)  TX bytes:28931756 (27.5 MiB)
      Interrupt:11 Base address:0xa000

lo    Link encap:Local Loopback
      inet addr:127.0.0.1  Mask:255.0.0.0
      UP LOOPBACK RUNNING  MTU:16436  Metric:1
      RX packets:9 errors:0 dropped:0 overruns:0 frame:0
      TX packets:9 errors:0 dropped:0 overruns:0 carrier:0
      collisions:0 txqueuelen:0
      RX bytes:720 (720.0 b)  TX bytes:720 (720.0 b)
```

In this example eth1 is the berofix PCI card and eth0 is our default network interface which is connected to our LAN. We want to assign the IP Address 192.168.0.2 to our LAN interface and the IP Address 192.168.0.3 to our berofix device. After we have setup the Network Bridge we want to access the berofix PCI card from all machines in our LAN, which are located in our network 192.168.0.1-254. To establish the Network Bridge, you have to use a few commands like described in the following example:

```
berofix:# ifconfig eth0 0.0.0.0 up
berofix:# ifconfig eth1 0.0.0.0 up
berofix:# brctl addbr br0
berofix:# brctl addif br0 eth0
berofix:# brctl addif br0 eth1
berofix:# ifconfig br0 192.168.0.3 up
```

These commands will first remove the IP-Addresses from the network devices and then will create the Network Bridge and add your standard network card (eth0) as well as the berofix PCI card (eth1) to this newly created Network Bridge (br0). In the example above we applied the 192.168.0.3 to the Network Bridge. The default IP settings of the berofix PCI/PCIe cards are 10.0.0.2 netmask 255.0.0.0. To make the berofix reachable from our LAN the IP-Address of the berofix device has to be changed. This can be done using the bfdetect tool (which is described in the next chapter) or through the webinterface of the berofix device. So we assume the berofix IP Address is changed to 192.168.0.3 with Netmask 255.255.255.0 . If everything was setup as it should, you should be able to reach the berofix device(192.168.0.3) from any machine in your Lan. Unfortunately this described way is not persistent and will be deleted after rebooting the machine. To setup this Network Bridge in a persistent way, you can modify your "/etc/network/interfaces" file on debian based systems (like ubuntu):

```
auto lo
iface lo inet loopback

auto br0
iface br0 inet static
address 192.168.0.2
netmask 255.255.255.0
gateway 192.168.0.1
bridge_ports eth0 eth1
```

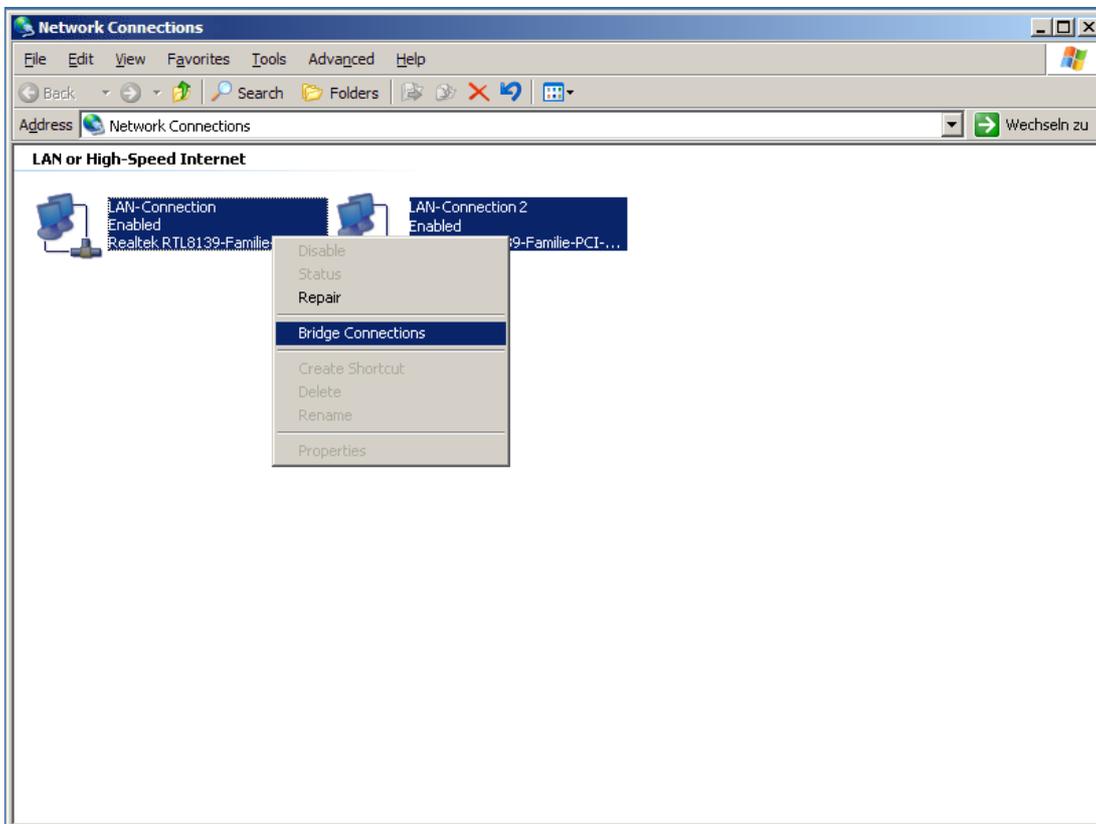
After that you have to restart your networking:

```
berofix:#/etc/init.d/networking restart
```

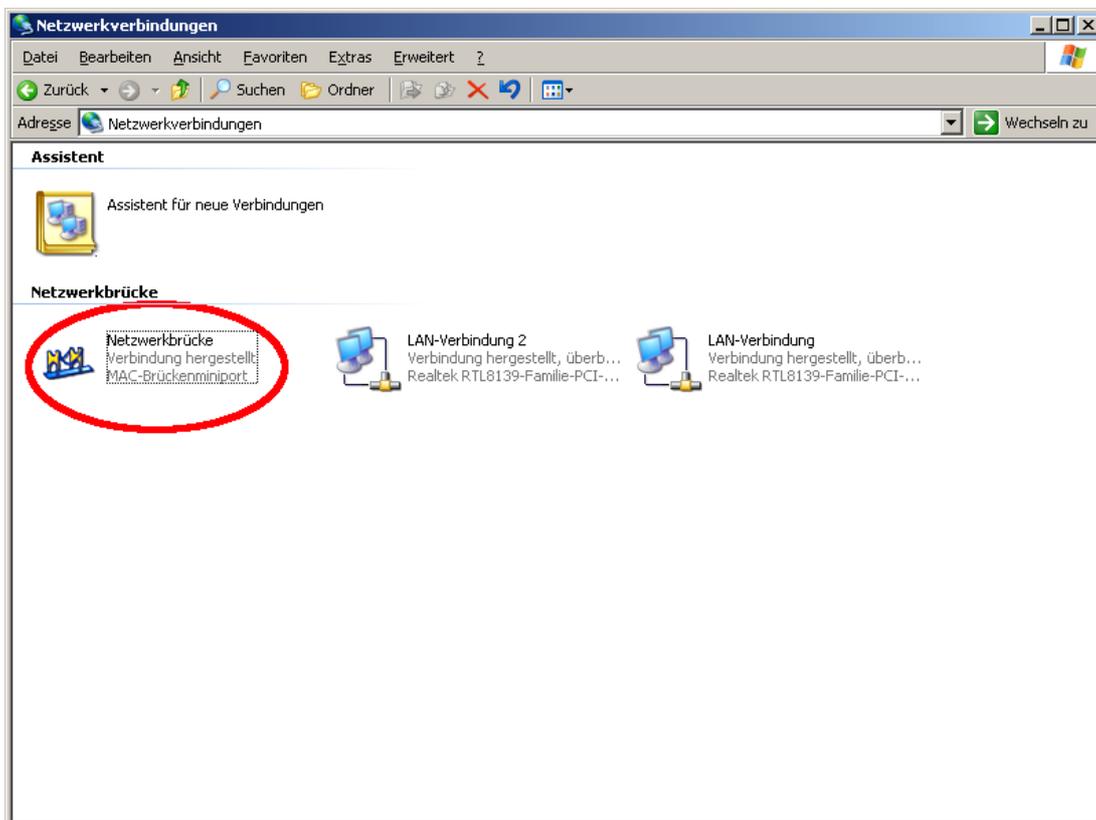
After everything is up and running, you should be able to reach the berofix device (192.168.0.3) from any machine in your LAN.

Windows network bridge

To use the Network Bridge under Windows based Systems (Windows SBS Versions don't seem to support Network Bridging) open "network connections" Window. As already seen in the chapter IP-Forwarding, you will see two Network controllers. Your standard Host Network Controller ("Local Area Network Connection 1") and the berofix Network Controller ("Local Area Network Connection 2"). To create the Network Bridge please mark both connections and click the right mouse button. The following context menu will appear:



Click on "Bridge connections" and a third Controller will appear, our Network Bridge:



As you might notice the Network Controllers don't have IP Address's any more. You can only assign an IP-Address to the Bridge Controller. By default the Bridge is in DHCP mode. If you please mark the Bridge, click the right mouse button, choose "Preferences" and configure an IP-Address. At that moment the Network Bridge is reachable throughout the LAN - you can also reach the berofix web-interface. For example:

Let's assume your LAN is 172.20.0.0/255.255.0.0, your bridge got an IP 172.20.5.194 and you configured berofix internal IP address to 172.20.70.11. Through the configured bridge, berofix is reachable from the whole network now, if your firewall is not blocking it!

Using the bfdetect tool to detect berofix devices

The bfdetect tool is a small command line tool, runnable under Linux and Windows, which helps you to detect any berofix device in your network. It can also be used to configure some basic network settings, like IP-Address etc. before starting to use the Web-Interface. The bfdetect tool can be downloaded from our Website:

For Linux:

```
http://www.beronet.com/downloads/berofix/tools/bfdetect.tar.gz
```

For Windows:

```
http://www.beronet.com/downloads/berofix/tools/bfdetect_win_x86.zip
```

In Windows based environments bfdetect_win_x86.zip will include a bfdetect.exe, which can be executed in the "command prompt", after extracting the archive. Under Linux based systems the archive has to be compiled after extracting it. See the following example:

```
beropbx:#tar xzf bfdetect.tar.gz
beropbx:#cd bfdetect
beropbx:#make install (just in case, don't forget to change permissions of the file to be executable)
```

To detect any berofix device in your network, the bfdetect tool will send an UDP Packet to the broadcast address of your default network adapter. Any berofix device in the network will reply to this request. If your berofix device is not located behind your default network adapter (e.g. berofix PCI/PCIe card), you have to specify the network adapter, which should be used to send out the Broadcast packet. See the following example:

```
beropbx:#bfdetect 10.0.0.1
Scanning network for bero*fix devices...
[1] mac:00:50:C2:83:D3:36 ip:10.0.0.2
[2] mac:00:50:C2:83:D2:AE ip:172.20.22.110
[3] mac:00:50:C2:83:D8:31 ip:172.20.22.203
[4] mac:D8:DF:0D:00:01:86 ip:172.20.22.101
[5] mac:00:50:C2:83:D5:42 ip:172.20.0.104

Select a device: 1

Configuration of selected device:

mac: 00:50:C2:83:D3:36
ip: 10.0.0.2
netmask: 255.0.0.0
gateway: 10.0.0.1
dhcp: no

[i] change ip
[n] change netmask
[g] change gateway
[d] switch dhcp on
[s] save and quit
[q] quit

What would you like to do?
```

All the rest is menu driven and pretty much explains itself. To get more help about the bfdetect tool, you can use the "-h" option.

Berofix web-interface (GUI)

If you have successfully setup the network as described in chapter 2, you can completely configure the berofix device via the web interface. Simply enter the IP-address of berofix in your favorite browser, for instance:

```
Default IP-Address
http://10.0.0.2
```

As already mentioned above, the default login username and password for berofix devices is:

```
Default UserName / Password
username: admin
password: admin

Note: You should change the password for security reasons immediatly after the first login
```

ISDN status					
Port	Type	Prot.	L2Link	L1Link	Restart
1	TE	PTP	●	●	Restart
2	TE	PTP	●	●	Restart
3	TE	PTP	●	●	Restart
4	TE	PMP	●	●	Restart
5	NT	PMP	●	●	Restart
6	NT	PMP	●	●	Restart

The first screen that you see is the status screen. It shows the Port status of each configure ISDN, Analog and GSM Port. In ISDN PMP Configuration it is very likely that the L2 and sometimes even the L1 Link status goes down and so the Lamp will be red. This is normal behaviour. The L2 is dynamic and will be pulled up when a call is established.

Each technology has additional information that can be seen when hovering over the Lamp. In ISDN the CRC Errors can be seen, in Analog the Line-Voltage and in GSM the Signal-Strength.

The berofix WebInterface consists of the following major menu-points:

- Dialplan (manage Dialplan related stuff)
- SIP+ (manage SIP Accounts like registrations and proxy's)
- PSTN (manage PRI/BRI/ANALOG/GSM Ports)
- Hardware (manage Module related settings)
- Preferences (manage Preferences like network and time -setting, logging etc.)
- Management (Small Statistics, Debug, Firmware-Update, Reboot/Reset etc.)
- Logout (Logout)

But before going in details with the above mentioned menu points, there is one more general thing you should know.

Activating changes

When Settings in the Webinterface are modified and saved, they don't apply immediately. A small "activate" Link appears on the top. The changes you have made take effect after pressing this "activate" button. Take a look at the following picture.

ISDN status					
Port	Type	Prot.	L2Link	L1Link	Restart
1	TE	PTP	●	●	Restart

Analog status				
Port	Type	Status	Line voltage	
1	FXS	FXS_IDLE	●	
2	FXS	FXS_IDLE	●	

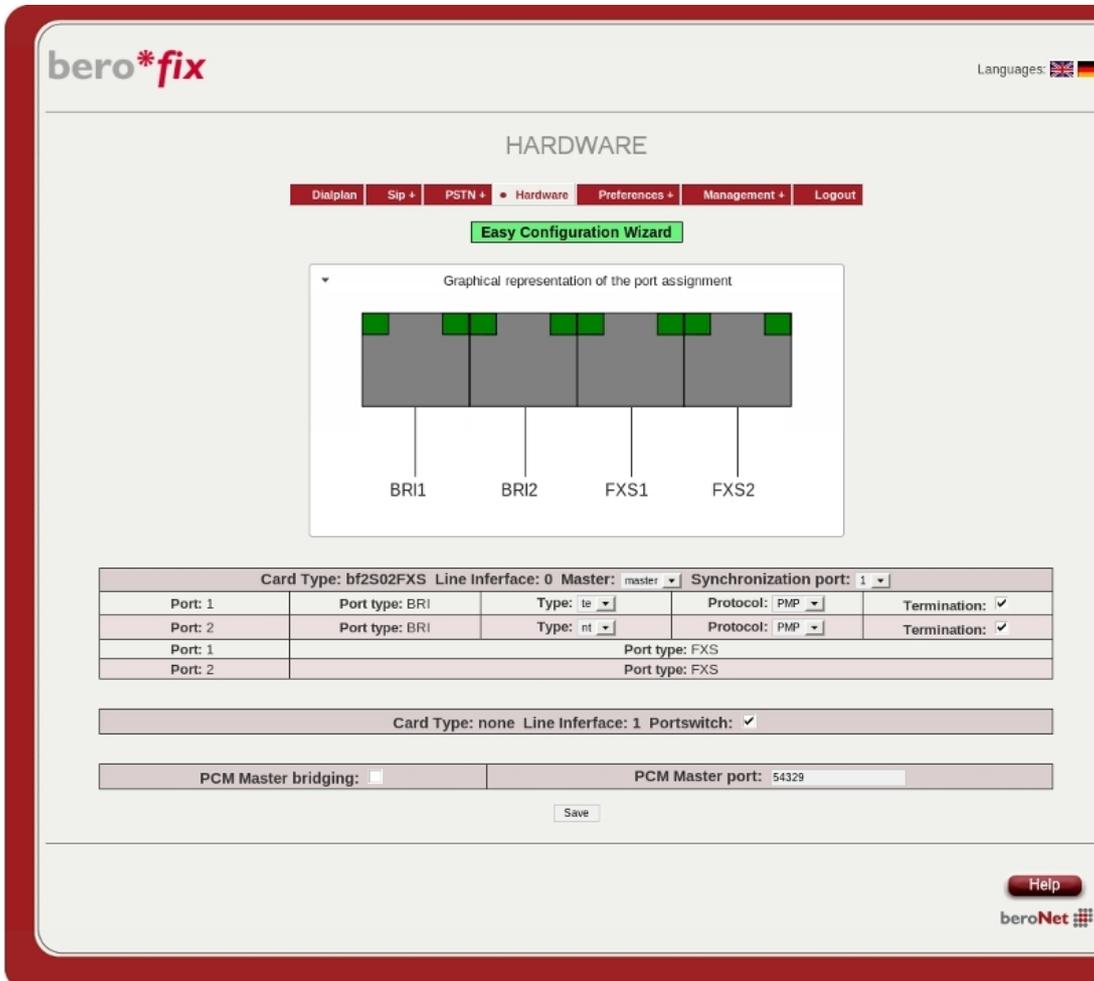
The "activate" button is available in three colors with the following meanings:

- **Yellow** Reload of the Configuration. No influence to current calls.
- **Orange** Parts of the Gateway-Software has to be restarted. All Current calls are going to be dropped.
- **Red** Reboot berofix. All current calls are going to be dropped. This can take up to 30sec..

This Document will describe every details of the GUI per Menu-point.

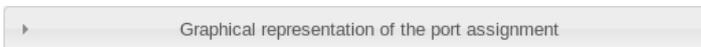
Hardware

The menu point "Hardware", will give you an overview about all modules, plugged on your berofix device. Before going into detail, take a look at the following picture, it shows an overview of the Hardware section:

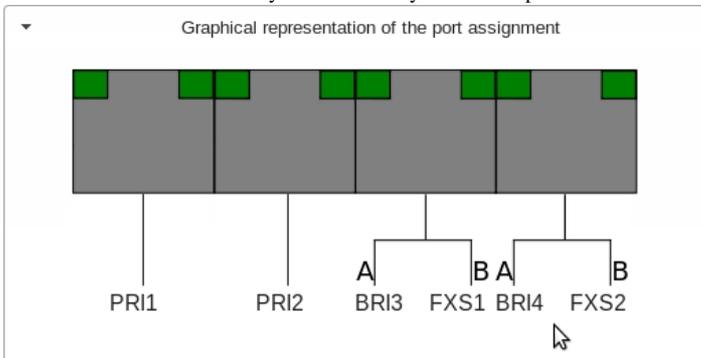


As you can see in the example above there are two Line Interface slots (LIS), LIS_0 and LIS_1. On LIS_0 is a BF2S02FXS modules shown, indicated by "Card Type", while LIS_1 shows an activated "Portswitch Option". Further more you can see the options "MASTER" and "Synchronization Port" on LIS_0. The "MASTER" Option has two possible values 'master' and 'slave'. If it is set to "master", this module will generate its own clock and can provide this to all slave modules (and also via PCM to another berofix device). In the case a BRI or PRI module is connected and master is selected, you can also configure the additional option "Synchronization Port". If this option is set to a valid port, the module will retrieve its clock from the selected port (usually an external PSTN Line) instead of generating it itself. It is always good or even necessary to set the first module to master (in case of ISDN also with "Synchronization Port") connected to the PSTN (external Line), and the second one to slave. This configuration will assure you an interconnection between the modules, to enable hardware bridging for transparent voice, data and fax transmissions.

Graphical representation



The graphical representation is an useful source of information, it shows you how the Ports of the module are connected to the 4 RJ45 Ports of your berofix device. This overview will also show you the necessity of BFTAdapters or of a BFBRIDGE. The next picture should make this more clear.



Modules and line interface slots

Below the graphical representation bar, you can see a summary of the modules which are connected to their respective Line Interface Slots (LIS). The following examples will show you all specialized settings for each module.

▪ BF4S0 / BF2S02FXS ISDN BRI modules

Card Type: bf4S0 Line Interface: 0 Master: master Synchronization port: 1				
Port: 1	Port type: BRI	Type: te	Protocol: PTP	Termination: <input checked="" type="checkbox"/>
Port: 2	Port type: BRI	Type: nt	Protocol: PTP	Termination: <input checked="" type="checkbox"/>
Port: 3	Port type: BRI	Type: te	Protocol: PMP	Termination: <input checked="" type="checkbox"/>
Port: 4	Port type: BRI	Type: nt	Protocol: PMP	Termination: <input checked="" type="checkbox"/>

Card Type: bf2S02FXS Line Interface: 0 Master: master Synchronization port: 1				
Port: 1	Port type: BRI	Type: te	Protocol: PMP	Termination: <input checked="" type="checkbox"/>
Port: 2	Port type: BRI	Type: nt	Protocol: PMP	Termination: <input checked="" type="checkbox"/>
Port: 1	Port type: FXS			
Port: 2	Port type: FXS			

When using ISDN BRI modules (i.e. BF4S0 or BF2S02FXS) you have the following options as seen in the above picture:

*TYPE: TE / NT (Terminal Endpoint / Network Termination) In general TE mode is used to connect your berofix to external lines provided by your provider, and NT Mode is used to connect phones or PBX's.

*Protocol: PTP / PTMP (Point-to-Point or Point-to-Multipoint) In general PTP is used for DID based connections and PTMP for MSN based.

*Termination: 0 / 1 (activates or deactivate 100 Ohm Linetermination) In general Linetermination 100 Ohm has to be turned on.

▪ BF1E1 / BF2E1 ISDN PRI modules

Card Type: bf2E1 Line Interface: 0 Master: slave			
Port: 1	Port type: PRI	Type: te	Protocol: PTP
Port: 2	Port type: PRI	Type: te	Protocol: PTP

When using ISDN PRI modules (i.e. BF1E1 or BF2E1) you have the following options as seen in the above picture:

*TYPE: TE / NT (Terminal Endpoint / Network Termination) In general TE mode is used to connect your berofix to external lines provided by your provider, and NT Mode is used to connect internal PBX's.

*Protocol: PTP (Point-to-Point) PRI modules supports only PTP.

*Line termination 120 Ohms is selectable for each port by DIP switches on the module itself.

▪ BF4FXO Analog module

Card Type: bf4FXO Line Interface: 0 Master: master	
Port: 1	Port type: FXO
Port: 2	Port type: FXO
Port: 3	Port type: FXO
Port: 4	Port type: FXO
Country: 1TR110_DE	

When using FXO Analog modules (i.e. BF4FXO) you have the following options as seen in the above picture:

*Country Setting: Each Country has its own settings regarding signals and tones.

▪ BFBridge module

Card Type: none Line Interface: 1 Portswitch: <input checked="" type="checkbox"/>

The BFBridge is not really a module but it behaves like one. The BFBridge is useful when you only have a single module BF4S0, BF2S02FXS, BF4FXO or BF4XS and want to use all 4 RJ45 of your berofix device, instead of the BFTAdapters. If you use one of this modules (BF4S0, BF2S02FXS, BF4FXO or BF4XS) on LIS 0 while LIS 1 is empty (expect the BFBridge itself), berofix will provide the "Portswitch Option" on LIS 1 as you can see in the above picture. You have to enable this option, when you want to use the BFBridge.

Activating this option without plugging the BFBridge or vice versa will result in unpredictable behavior.

PCM bridging

Since berofix Firmware 2.X berofix devices can be interconnected via a PCM cable to enable hardware bridging for transparent voice, data and fax transmissions across two berofix devices. That means that the Audio part is transmitted in TDM over the PCM cable, and not send using RTP over the network. The signaling is still send via SIP.

PCM-master

To activate this feature log into the first berofix device which we call PCM-Master. As mentioned before (section Hardware) one module on the PCM-Master has to be set to 'master'. That means that this module will generate the clock for all other slaves.

PCM Master bridging: <input checked="" type="checkbox"/>	PCM Master port: 54329
--	------------------------

All you have to do to configure the PCM-Master is to activate the "PCM Master bridging" option and select a Port for the "PCM Master port" option as seen in the above picture.

PCM-slave

It is very similar to configure the second berofix device, which we call the PCM-Slave. Again as mentioned before (Section Hardware) all modules on the PCM-Slave have to be configured as "slave" to retrieve the clock. If you do so you will see following configuration options.

PCM master IP-Address: 172.20.70.41	PCM Master port: 54329
-------------------------------------	------------------------

In the Field "PCM master IP-Address" you have to enter the IP-Address of your PCM-Master and set the "PCM Master Port" to the same value as you have entered on the PCM Master.

PCM-overview

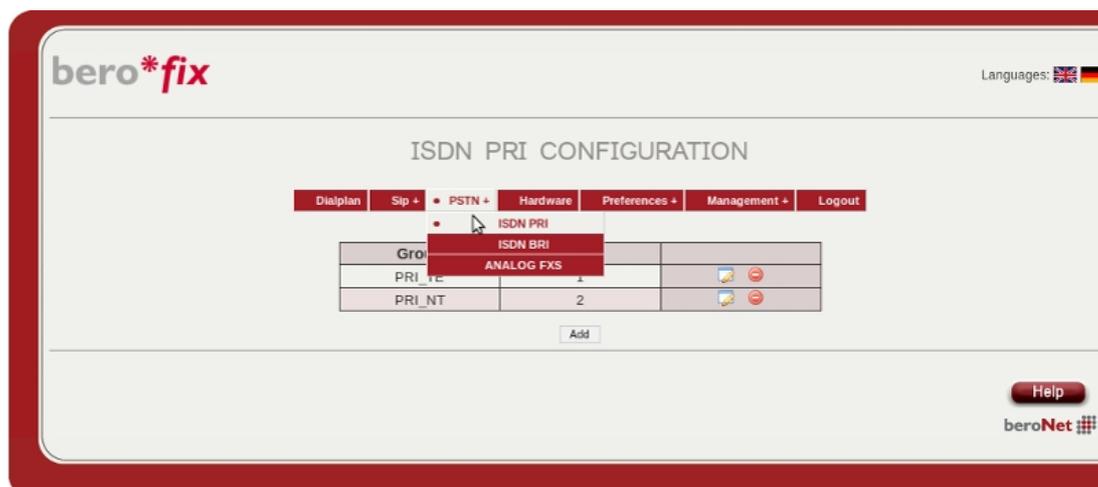
To check if the PCM connection is configured correctly, you can use the menu point 'State' which you can find in the management section. Depending on which berofix you are logged in you can see the following pictures.

PCM Server status:	●
IP Address (client):	172.20.70.42
PCM Client status:	●
IP Address (server):	172.20.70.41

In this example our PCM-Master has the IP-Address 172.20.70.41 and the PCM-Slave has the IP-Address 172.20.70.42. The green light on this screens indicates that communication between the PCM-Master and PCM-Slave has been established and is operational.

PSTN+

The menu point PSTN+ gives you an overview about the Ports which are provided by the plugged modules you are using on this particular berofix. For each technology you will find a sub menu point, like 'ISDN PRI' / 'ISDN BRI' / 'Analog FXO' / 'Analog FXS' / GSM. The sub menu entries are dynamic, and you will only see these which are provided by your modules. These sub menu points or technologies can be grouped together in so called 'Port-Groups'. As you can see in the next picture, you can add, modify and delete 'Port-Groups' by clicking on the corresponding button.



You can add Ports to the 'Port-Group' by selecting available available Ports. A Port which is already assigned to a 'Port-Group' will be displayed grey as shown in the next picture.

PSTN											
Group Name:	Telecom										
Ports:	<table border="1"> <tr> <td>Li0(bf4S0)</td> <td>Li1(bf2S02FXS)</td> </tr> <tr> <td>Port 1 <input checked="" type="checkbox"/></td> <td>Port 5 <input type="checkbox"/></td> </tr> <tr> <td>Port 2 <input checked="" type="checkbox"/></td> <td>Port 6 <input type="checkbox"/></td> </tr> <tr> <td>Port 3 <input type="checkbox"/></td> <td></td> </tr> <tr> <td>Port 4 <input type="checkbox"/></td> <td></td> </tr> </table>	Li0(bf4S0)	Li1(bf2S02FXS)	Port 1 <input checked="" type="checkbox"/>	Port 5 <input type="checkbox"/>	Port 2 <input checked="" type="checkbox"/>	Port 6 <input type="checkbox"/>	Port 3 <input type="checkbox"/>		Port 4 <input type="checkbox"/>	
Li0(bf4S0)	Li1(bf2S02FXS)										
Port 1 <input checked="" type="checkbox"/>	Port 5 <input type="checkbox"/>										
Port 2 <input checked="" type="checkbox"/>	Port 6 <input type="checkbox"/>										
Port 3 <input type="checkbox"/>											
Port 4 <input type="checkbox"/>											

Each configured 'Port-Group' has technology specific settings. The settings on the 'Port-Groups' will apply to all Ports, which are member of this 'Port-Group'. Furthermore if you Dial out on a 'Port-Group' the system will automatically find the next free channel in this 'Port-Group', regardless if it is a BRI/PRI, Analog or GSM Port. You need to put all the Ports you want to be able to configure into a 'Port-Group' (even if you only want or have one).

As mentioned before, depending of the technologies you are using, different settings will be available. In the next chapters we are going to explain this in detail:

ISDN PRI / BRI options

This Chapter will explain the ISDN configuration possibilities and ISDN specific settings.

General ISDN PRI / BRI settings

The picture below will show you ISDN PRI / BRI 'Port-Groups' specific basic settings:

The screenshot shows a configuration window with the following fields and options:

- ChanSel:** A dropdown menu set to 'standard'.
- Tones:** A dropdown menu set to '[de]'.
- Interdigit timeout:** A text input field containing the number '3'.
- Interdigit timeout initial:** A text input field containing the number '15'.
- Overlap Dialing:** A checkbox that is currently unchecked.
- QSIG support:** A checkbox that is currently unchecked.
- Link Down behaviour:** A dropdown menu set to 'Pull Link Up (2s)'.

At the bottom of the configuration area, there is a red text link labeled 'more...'.

Group Name

Unique name of the 'Port-Group'

Ports

Ports which are member of the 'Port-Group'.

ChanSel

Channel Selection schemes: (standard / Random / Round Robin) default: standard

- Standard - selects the next free channel in ascending order
- Random - selects the next free channel at random
- Round Robin - selects the next free channel base on the round-robin principle

Tones

ISDN Tone sets which are categorized by country

Interdigit timeout

(default is 3 sec. For every incoming call a inter digit collect timer will be started. After this specified timeout, without getting a digit, the call will be processed to the Dialplan. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Interdigit timeout initial

(default is 15sec.)This Timer is the initial inter digit timer, that means before we got any digit. This Timer will be stopped after the first digit and the above mentioned Interdigit timeout Timer will apply. During this time a Dialtone will be generated. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Overlap Dialing

This Option will activate real Overlap Dialing, for instance in ISDN Environments. By activating this option the 'Interdigit' as well as the 'Interdigit timeout initial' Timer will be deactivated.

QSIG support

Enable or disable QSIG support.

Link Down behavior

LinkDownBehavior:[Nothing/Pull Link Up (2s)/Pull Link Up (once)]

- Nothing
- Pull Link Up (2s) will try to get UP Links up to 2 seconds
- Pull Link Up (once) will try to get UP Links once

In some countries like Cyprus the behavior of ISDN PTP Pots regarding Layer1 and Layer2 are different. They deactivate Layer 1 and Layer 2 after a while of inactivity. With this option you can solve this issue.

Advanced ISDN PRI / BRI settings (more)

Behind the more button you will find more advanced ISDN PRI / BRI settings as shown in the picture below.

EC
0=off,1=on (default value is 1=on) this will activate or deactivate the onboard Hardware Echocanceler

EC tail length
EC tail length [0=8ms,1=16ms,....,15=128ms] (default value is 15=128ms) 16 tabs each 8ms.

dnumplan
dnumplan [0=unknown, 1=international, 2=national, 4=subscriber). dnumplan is the 'Type of Number' in terms of ISDN for the Destination Address (DAD). This options :defines the number format of the DAD for an outgoing call. Be aware that the remote end has to also support this feature.

onumplan
onumplan [0=unknown, 1=international, 2=national, 4=subscriber). onumplan is the 'Type of Number' in terms of ISDN for the Originating Address (OAD). This options :defines the number format of the OAD for an outgoing call. If you want to use 'CLIP_NO_SCREENING' you have to set this to national,international or subscriber, :depending on how you are going to send your OAD.

screening
screening (default is 0 = off) these are the exact ISDN screening and presentation indicators.
Screening=0 and presentation=0 means callerID presented but not screened(the remote end does see the callerID).
Screening=1 and presentation=1 means callerID presented but screened(the remote end does not see the callerID)

presentation
presentation (default is 0=off) see option screening

Bearer
ISDN Bearer capability to use for outbound calls on this 'Port-Group'

- speech (default for standard Voice calls)
- Audio_3.1K (useful for outbound Fax calls)
- Audio_7K
- Video
- Digital_Unrestricted (useful for ISDN digital calls)
- Digital_Restricted
- Digital_Unrestricted_Tones

Call deflection (default is disabled)

Call deflection / Partial Routing is used to redirect calls on the provider site, without using B-Channels on the berofix. If 'Call deflection' is enabled you can use :a SIP 302 'Move temporarily' to Redirect the call on the provider site.

CLIR on OAD

CLIR on OAD (default=empty) to dynamically hide the CallerID in direction of this 'Port-Group' (ISDN). For instance if berofix detects a call to this 'Port-Group' with an OAD=CLIR_on_OAD (after the call left the dialplan), the CallerID will be hidden, that means the remote end doesn't see it. For calls originated from this

'Port-Group' with an empty OAD or with screening and presentation set to "1", the CallerID will be replaced by the CLIR_ON_OAD value.

Dialplan Source

The Dialplan Source is used as 'Source' for matching in the 'Dialplan'. That means if a call is initiated from this 'Port-Group' you can use 'Dialplan Source', to tell the Dialplan, which value should be used for the 'Source' in the Dialplan. Dialplan Source can have the following values

- OAD (by default it is set to OAD)
- OAD2 (OAD2 in case you have 2 OAD's you can choose with which value you want to go in the Dialpan)
- Qsigname (Qsigname to use it at source in the Dialplan)
- Redirectednr

Caller ID Mapping

+oad (Caller ID)

oad (Caller ID) gives you the possibility to tell berofix, which Field should be used for the OAD for calls to this 'Port-Group'. oad (Caller ID) could be applied to the following fields

- new_source (use new_source from the Dialplan as OAD for calls to this 'Port-Group')
- from_user (use SIP from_user as OAD for calls to this 'Port-Group')
- from_display (use SIP from_display as OAD for calls to this 'Port-Group')
- pai_all (use P-Asserted-Identities as OAD for calls to this 'Port-Group')
- pai_user
- pai_display
- ppi_all
- ppi_user
- ppi_display
- none (use nothing for the OAD)

-manual (use a constant string for as OAD for calls to this 'Port-Group')

+oad2 (second Caller ID)

oad2 (second Caller ID) (no default value). The oad2 defines which Field should be used for the second OAD (oad2(second Caller ID)). The oad2(second Caller ID) could be applied to the same fields already mentioned at oad(CallerID) above.

+qsigname

qsigname (no default value). This defines which Field should be used for the qsigname. The qsigname could be applied to the same fields already mentioned at oad(CallerID) above.

+Redirected Nr (default is none) Redirected Nr defines what the Redirected Nr should contain

Redirected Nr (no default value). This defines which Field should be used for the Redirected Nr . Redirected Nr could be applied to the same fields already mentioned at oad(CallerID) above

The above mentioned settings are mostly used and they are directly outputted through the WebInterface. But berofix has a lot of more settings, which are used in very special scenarios. These settings can be found at the additional configuration options (see picture below).



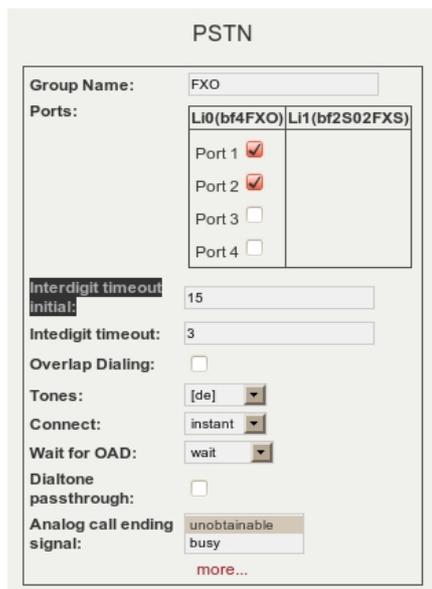
The lower Box 'Additional configuration options description' contains the possible settings including a small description. For instance if you want to use the settings nationalprefix=0 and internationalprefix=00, you have to enter these in the upper Box Line by Line as shown in the above picture.

Analog FXO options

This Chapter will explain analog FXO configuration possibilities and analog FXO specific settings.

General analog FXO settings

The picture below will show you Analog FXO 'Port-Group' specific basic settings:



Group Name

Unique name of the 'Port-Group'

Ports

Ports which are member of the 'Port-Group'.

Interdigit timeout

(default is 3 sec. For every call from FXO an inter digit collect timer will be started. After this specified timeout, without getting a digit, the call will be processed to the Dialplan. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Interdigit timeout initial

(default is 15sec.) This Timer is the initial inter digit timer, that means before we got any digit. This Timer will be stopped after the first digit and the above mentioned Interdigit timeout Timer will apply. During this time a Dialtone will be generated. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Overlap Dialing

This Option will activate real Overlap Dialing. By activating this option the 'Interdigit' as well as the 'Interdigit timeout initial' Timer will be deactivated.

Tones

Tone sets which are categorized by country

Connect

(default is instant)

How berofix should detect a FXO connect

- instant (default. After Dialing the state will immediately change to 'connect')
- polarity (the opposite site send a polarity reversal to detect a 'connect')

Wait for CLIP

- wait (default: wait 2sec. to detect the CLIP)
- dontwait (will immediately process without waiting for the CLIP)

Dialtone passthrough

Dialtone passthrough:(default is disabled)

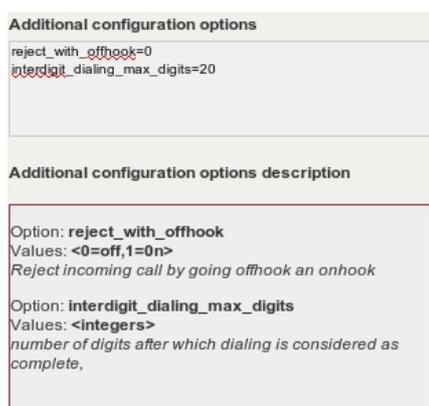
Analog call ending signal

How berofix should detect the end of a call

- unobtainable (default unobtainable)
- busy

Advanced analog FXO settings (more)

Behind the more button you will find more advanced Analog FXO settings as shown in the picture.



EC

0=off,1=on (default value is 1=on) this will activate or deactivate the onboard Hardware Echocanceler

EC tail length

EC tail length [0=8ms,1=16ms,....,15=128ms] (default value is 15=128ms) 16 tabs each 8ms.

CLIR on CLIP

CLIR on CLIP (default=empty) to dynamically hide the CallerID in direction of this 'Port-Group' (FXO). For instance if berofix detects a call to this 'Port-Group' with a CLIP=CLIR_on_CLIP (after the dialplan), the CallerID will be hidden, that means the remote end doesn't see it.

Caller ID Mapping

+CLIP

CLIP gives you the possibility to tell berofix, which Field should be used as CLIP for calls to this 'Port-Group'. CLIP could be applied to the following fields:

- new_source (use new_source from the Dialplan for the CLIP for call to this 'Port-Group')
- from_user (use SIP from_user as CLIP for calls to this 'Port-Group')
- from_display (use SIP from_display as CLIP for calls to this 'Port-Group')
- pai_all (use P-Asserted-Identities as CLIP for calls to this 'Port-Group')
- pai_user
- pai_display
- ppi_all
- ppi_user
- ppi_display
- none (use nothing as CLIP)
- manual (use a constant string as CLIP for calls to this 'Port-Group')

+CNIP

CNIP (no default value). The CNIP defines which Field should be used as CNIP. The CNIP could be applied to the same fields already mentioned at CLIP above.

The above mentioned settings are mostly used and are directly outputted through the WebInterface. But berofix has a lot of more settings, which are used in very special scenarios. These settings can be found at additional configuration options (see picture below).

Additional configuration options

```
reject_with_offhook=0
interdigit_dialing_max_digits=20
```

Additional configuration options description

Option: **reject_with_offhook**
 Values: <0=off,1=0n>
 Reject incoming call by going offhook an onhook

Option: **interdigit_dialing_max_digits**
 Values: <integers>
 number of digits after which dialing is considered as complete,

Analog FXS options

This Chapter will explain analog FXS configuration and analog FXS specific settings.

General analog FXS options

The picture below will show you Analog FXS 'Port-Group' specific basic settings:

PSTN

Group Name:

Ports:

Li0()	Li1(bf2S02FXS)
	Port 1 <input checked="" type="checkbox"/>
	Port 2 <input type="checkbox"/>

Interdigit timeout initial:

Interdigit timeout:

Overlap Dialing:

Tones: ▼

CLIP:

CNIP:

[more...](#)

Group Name

Unique name of the 'Port-Group'

Ports

Ports which are member of the 'Port-Group'.

Interdigit timeout

(default is 3 sec. For every call from FXS an inter digit collect timer will be started. After this specified timeout, without getting a digit, the call will be processed to the Dialplan. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Interdigit timeout initial

(default is 15sec.) This Timer is the initial inter digit timer, that means before we got any digit. This Timer will be stopped after the first digit and the above mentioned Interdigit timeout Timer will apply. During this time a Dialtone will be generated. Note this Timer is only started if 'Overlap Dialing' is deactivated.

Overlap Dialing

This Option will activate real Overlap Dialing. By activating this option the 'Interdigit' as well as the 'Interdigit timeout initial' Timer will be deactivated.

Tones

Tone sets which are categorized by country

CLIP

CLIP of this Trunk. Specify the Trunk Extension or any Number that you can later use in the Dialplan as Destination.

CNIP

CNIP of this Trunk. Specify the Trunk Extension or any Number that you can later use in the Dialplan as Destination

Advanced analog FXS settings (more)

Behind the more button you will find more advanced Analog FXS settings as shown in the picture.

Advanced Configuration	
EC	<input checked="" type="checkbox"/>
EC tail length	<input type="text" value="15"/>
CLIR on CLIP	<input type="text"/>
Caller ID Mapping	
CLIP (Caller ID)	<input type="text" value="new_source"/>
CNIP (Caller Name)	<input type="text" value="none"/>

EC
0=off,1=on (default value is 1=on) this will activate or deactivate the onboard Hardware Echocanceller

EC tail length
EC tail length [0=8ms,1=16ms,...,15=128ms] (default value is 15=128ms) 16 tabs each 8ms.

CLIR on CLIP
CLIR on CLIP (default=empty) to dynamically hide the CallerID in direction of this 'Port-Group' (FXS). For instance if berofix detects a call to this 'Port-Group' with a CLIP=CLIR_on_CLIP (after the dialplan), the CallerID will be hidden, that means the remote end doesn't see it.

Caller ID Mapping

+CLIP (CallerID)

CLIP gives you the possibility to tell berofix, which Field should be used as CLIP for calls to this 'Port-Group'. CLIP could be applied to the following fields:

- new_source (use new_source from the Dialplan for the CLIP for call to this 'Port-Group')
- from_user (use SIP from_user as CLIP for calls to this 'Port-Group')
- from_display (use SIP from_display as CLIP for calls to this 'Port-Group')
- pai_all (use P-Asserted-Identities as CLIP for calls to this 'Port-Group')
- pai_user
- pai_display
- ppi_all
- ppi_user
- ppi_display
- none (use nothing as CLIP)
- manual (use a constant string as CLIP for calls to this 'Port-Group')

+CNIP (Caller Name)

CNIP (no default value). The CNIP defines which Field should be used as CNIP. The CNIP could be applied to the same fields already mentioned at CLIP above.

The above mentioned settings are mostly used and are directly outputted through the WebInterface. But berofix has a lot of more settings, which are used in very special scenarios. These settings can be found at additional configuration options (see picture below).

Additional configuration options
reject_with_offhook=0 interdigit_dialing_max_digits=20
Additional configuration options description
Option: reject_with_offhook Values: <0=off,1=on> Reject incoming call by going offhook an onhook
Option: interdigit_dialing_max_digits Values: <integers> number of digits after which dialing is considered as complete,

GSM options

GSM has 3 Different Setting Pages:

- | | |
|----------------|---|
| 1) GSM | (Port Groups, like in ISDN or ANALOG) |
| 2) GSM General | (PIN and SMSC Settings per Port and SMS to Mail Settings) |
| 3) SMS | (SMS GUI Frontend) |

1) GSM

The GSM Module behaves like all the other modules, which means it's ports need to be grouped, so that they can be used in the dialplan.

2) GSM General

Every GSM Port has some unique configuration which is done in this Setting Page. These configurations are for example the PIN of the SIM Card or the SMSC (SMS Center) for the sim card.

NOTE: The PIN can be left blank if no PIN is stored for the Sim Card.

The SMSC needs to be configured and is different for each provider. Lists can be downloaded on the internet, here are some german providers SMSCs:

```

o2 +491760000443
d1 +491710760000
o2 +491722270000

```

This information is supplied without liability, you should contact your mobile provider.

3) SMS

The SMS GUI shows the 3 different SMS Directories in berofix (in, out, failout). It has also an option to send an SMS directly by choosing the GSM Port, entering a Destination Phone Number and the Text of the SMS.

Incoming SMS are displayed in the "inbox" table and can be viewed and deleted. Failed SMS will be stored in the "failout box", the reason and the content of each failed SMS can be viewed.

SIP+

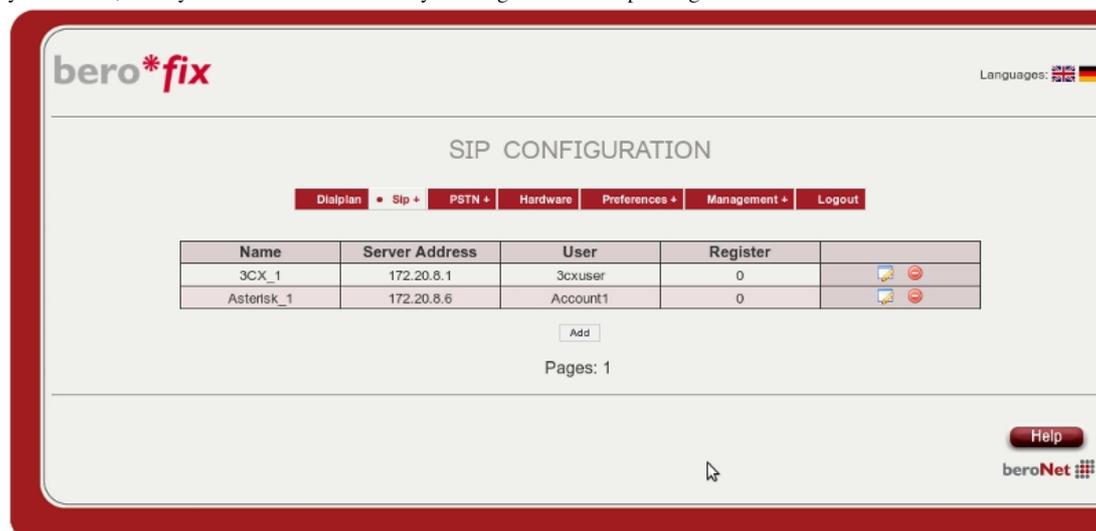
By selecting the menu point SIP, you will be directed to the page containing all options regarding SIP like for instance 'Registrar'/'Proxy', RTP-Port-Range and many others. There are 2 menu-points, SIP and SIP General which will be explained in detail in the next chapters.

SIP

In this chapter we will explain SIP specific configuration and Settings

General SIP settings

Under the menu point SIP you will get an overview about all SIP accounts, as you can see in the next picture. As already mentioned in the chapter 'Port-Groups', you can add, modify and delete SIP accounts by clicking on the corresponding button.



A SIP accounts consist always of the following settings, which you can see in the next picture.

Name:	<input type="text"/>
Server Address:	<input type="text"/>
User:	<input type="text"/>
Secret:	<input type="text"/>
Register:	<input checked="" type="checkbox"/>
Registration interval:	<input type="text"/>
Register option:	<input type="text" value="validate"/>

[more...](#)

Name

Name you want to use for this SIP account (berofix internal)

Server address

IP address of the SIP server (for instance your 3CX or Asterisk Server). You can also set an alternative udp Port in the format **sip.beronet.com:5061** if required (default is 5060)

User

SIP Username used for SIP authentication. This username will be used as User Part in the SIP From Header. If you need to specify a different SIP From URL (sometimes called fromdomain) you can specify the SIP User in the form:

from_user@from_url

by default the from_url is simply the Server address.

Secret

SIP authentication password

Register

When activated, berofix will register at the SIP Server by sending out SIP REGISTER Messages.

Note: If you enable this option, berofix changes the following settings automatically:

-Dialplan Source = from_display (Dialplan uses from_displayname for matching for call originated by this SIP account)

-From User part = account_username (use account username as From_User for Calls in SIP direction)

-From Display part = new_source (use NewSource from the Dialplan as From Displayname for the CallerID Transmission in SIP direction)

Note2: If you deactivate this option, berofix changes the following settings automatically:

-Dialplan Source = from_user (Dialplan uses from_displayname for matching for call originated by this SIP account)

-From User Part = new_source (use NewSource as From_User for Calls in SIP direction)

-From Display Part = new_source (use NewSource from the Dialplan as From Displayname for the CallerID Transmission in SIP direction)

Registration Interval

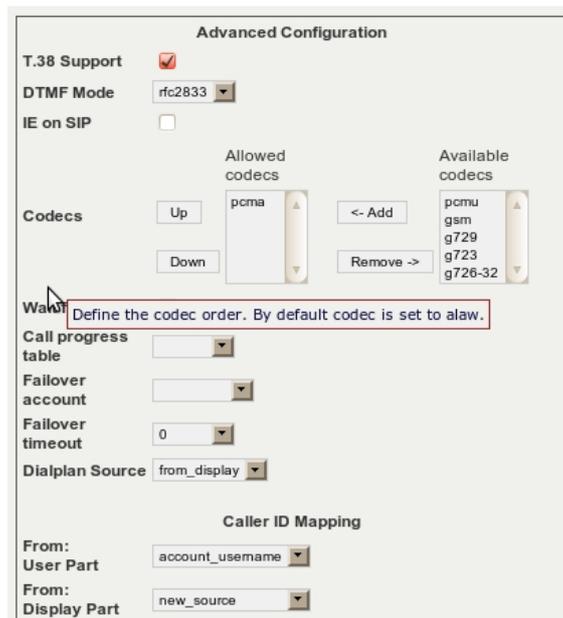
Frequency interval in seconds, to refresh Registration by sending register messages.

Register option

If the Registration should be validated or not. Only available if Registration is checked

Advanced SIP settings (more)

Behind the more button you will find more advanced SIP settings as shown in the picture.



T38 support

Check for Fax Tones and try a T.38 reinvite to make a reliable Fax-Over-IP Connection. The SIP Device which is connected to berofix must support T.38. Most ATA's and some SIP Softpbxes support T.38. Default is active.

DTMF Mode

DTMF Mode over SIP (default is rfc2833).

-**rfc2833** DTMF tones are transmitted outband via RTP packet

-**inband DTMF** tones are transmitted inband n(inside the audio channel)

-**info** DTMF tones are transmitted outband via SIP-Info packet

IE on SIP

IE on SIP activates the possibility to send additional ISDN Information Elements over SIP-Headers. If set to active berofix will encode ISDN Information Elements like the 'Bearer Capability', 'Type of Number' or the 'Release Cause' etc. as X-BF SIP Headers. berofix will also look for X-BF Headers on incoming SIP Messages to decode them into ISDN Information Elements. See Howto to use X-BF Headers for more details. Default=0.

Codecs

Define which codecs are allowed and in which order they should be offered . By default codecs is set to "G.711 Alaw".

Wait for Cancel

When the PSTN Releases a call with a proper reason and with inband information, this setting will be reviewed. If set to active berofix will not immediately send the corresponding SIP response, instead it will make the inband audio information available, so that the user can hear this message. berofix will wait until the user cancels this call while providing the inband-audio (wait_for_cancel). If set to deactivate berofix will immediately finish the call by sending the corresponding SIP response that is mapped for the PSTN Release reason. See berofix ISDN Cause/SIP Response map for details of this mapping. Default is 1.

Call Progress Table

You can define which Call Progress Table should be used for this SIP account. By default (when left empty) the berofix build-in Call-Progress Table would be used.

Failover Account

Failover Account

Failover Timeout

Failover Timeout

Dialplan Source

The Dialplan Source is used as 'Source' for matching in the 'Dialplan'. That means if a call is originated from this SIP account you can use 'Dialplan Source', to tell the Dialplan, which field should be used for the 'Source' in the Dialplan. Dialplan Source can have the following values

-from_user (use SIP from_user as as source)

-from_display (use SIP from_display as source)

-pai_all (use P-Asserted-Identities as source)

-pai_user

```
-pai_display
-ppi_all
-ppi_user
-ppi_display
```

CallerID Mapping

The CallerID Mapping is used to assign several possible values to the SIP From_User part and the SIP From_Display part for flexible transmission of the CallerID in direction SIP. With this option you can change the standard CallerID transmission behaviour of the berofix and adjust it to the needs of your SIP-Server. Find more information here: Callerid-Mapping (http://wiki.beronet.com/index.php/Howto_modify_what_goes_into_From/P-Preferred-Identity/Callerid_and_so_on).

Note: A SIP account which should register at the SIP Server uses account_username for SIP from_user and and new_source as from_displayname, while an account which doesn't register uses new_source as from_user and from_displayname.

From: User part This setting is used to tell the berofix which value should be used for the from_user Part of the SIP FROM Header. From: User Part could contain the following values:

```
-new source (use new source from the Dialplan as SIP from_user)
-oad (use OAD as SIP from_user)
-oad2 (use OAD2 as SIP from_user)
-qsigname (use the QSigName as SIP from_user)
-account_username (use account_username from this SIP-account as SIP from_user)
-manual (enter a manual value to be used as SIP from_user)
-none
```

From: Display part This setting is used to tell the berofix which value should be used for the from_display Part of the SIP FROM Header. 'From_Display part' could contain the following values:

```
-oad (use OAD as SIP from_displayname)
-oad2 (use OAD2 as SIP from_displayname)
-qsigname (use the QSigName as SIP from_displayname)
-account_username (use account_username from this SIP-account as SIP from_displayname)
-manual (enter a manual value to be used as SIP from_displayname)
-none
```

SIP general

Under SIP General settings you can set the following values as you can see in the next picture:

Sip General	
Bind Port:	5060
RTP Port Range:	5000-5059,5062-6000
Extern IP:	
TOS RTP:	160
TOS SIP:	160
SIP transport:	udp

Bind port

The Port on which the berofix device should listen for SIP traffic. Default is 5060

RTP port range

RTP port range which should be used for RTP Traffic. Default is (5000-5059,5062-6000)

Extern IP

If you are behind a firewall or NAT, you can enter the external IP address (for instance your external IP of your router).

General form:

```
externip=<IP-Address>[.ic=1][.rc=1]
```

the IP-Address is the IP that is used in the sip headers. if not set, the bind ip address is used.

ic=1 means that the external ip is also used in the contact header for outgoing invites

rc=1 means that the external ip is also used in the contact header for outgoing registers

TOS RTP

Type of service for RTP traffic. Useful for prioritization of the RTP traffic. Default is 160

TOS SIP

Type of service for SIP traffic. Useful for prioritization of the SIP traffic. Default is 160

SIP transport

The beroFix devices support the following SIP transport modes:

```
-udp (SIP Transport via UDP)
-tcp (SIP Transport via TCP)
-tls (SIP Transport via TCP with TLS and Certificate)
```

The dial plan

The dial plan is one of the most important things to set up during the configuration of a beroFix device. The dial plan defines rules concerning how calls should be routed under certain circumstances. The beroFix dial plan engine is based on regular expressions ('Howto RegEx' (<http://www.zytrax.com/tech/web/regex.htm>)) and reads the dial plan entries from the top to the bottom. After the first match beroFix will leave the dial plan and execute the rule. Therefore it is important to know that special dial plan rules should be placed on top while the general ones should be placed below them. This can be done by using the position arrows. The

next picture will show you the dial plan overview.

DIALPLAN								
• Dialplan Sip + PSTN + Hardware Preferences + Management + Logout								
Direction: all		Search:			Entries per page: 15			
Direction	From ID	To ID	Destination	New destination	Source	New source	Position	
sip-isdn	.*	g:BRI_TE_Ports	0(*)	\1	(.*)	\1	1	   
isdn-sip	g:BRI_TE_Ports	p:3CX_1	(.*)	\1	(.*)	\1	2	   
analog-isdn	g:FXS_1	g:BRI_TE_Ports	(.*)	\1	(.*)	\1	3	   
isdn-analog	g:BRI_TE_Ports	g:FXS_1	.*(125)	\1	(.*)	\1	4	   

Add

Pages: 1

As you can see you can add, modify, copy and delete a dial plan rule by pressing on it's corresponding button on the right in the table. Furthermore you access the advanced options for a particular dial plan rule by pressing the 'tool icon' also on the right-hand side in the table. Above the table you can set some Filters for the dial plan table like 'direction' or 'entries per page' or even search for a special character inside all fields of a dial plan rule. As you can see that there are several columns. Before explaining the meaning of each column, it is important to know that some of these columns are matching criteria and the others are executing ones. Matching columns are taken into account to decide if a particular dial plan rule matches while the executing columns would be executed when a dial plan rule has matched. Only if all matching columns are true a particular dial plan rule will it be executed.

Matching Columns:

- **Direction:** The direction of the call. Here you can set the direction a call has to be from for this rule to qualify and to where the call is to be routed. For instance in the first row you see that the "call" has to originate from SIP and will then be routed to ISDN. In the the second row it is vice versa.
- **FromID** The ID from where the call is originated. Depending of the direction that is chosen the FromID could be the name of a 'SIP-account', a 'PSTN-port-group' or a manual value like an IP-address.
- **Destination** Also called "CalledID" or DAD. This is the number which the caller dialed.
- **Source** Also called "CallerID" or OAD. This is the number of the device of the caller.

Execution Columns:

- **ToID:** The ID where the call should be routed to. Depending on the direction this could be a particular 'PSTN-port' a 'PSTN-port-group', the name of a SIP-account or an IP-address
- **New Destination** The new destination after the dial plan rule has been executed (see 'Destination' above).
- **New Source** The new source after the dial plan rule has been executed (see 'Source' above).

New dial plan rule

You can add, modify or copy a dial plan rule by pressing it's corresponding buttons. The next picture will show you the input form.

DIALPLAN	
From direction: SIP	To direction: ISDN
Match type: From user	To ID: g:BRI_TE_Ports
From ID: 3CX_1	
Destination: (*)	New destination: \1
Source: (*)	New source: \1
Comments: All Call from 3CX should be routed to ISDN	
First position: <input type="checkbox"/>	
Save Close	

On the left-hand side you will find all of the matching fields and on the right all of the executing fields. Each matching field has one executing field, that means there is always a pair of both.

From and to direction

The first pair is the direction pair, consisting of the fields **From direction** and **To direction**. From direction is the matching field and it checks from where the call originated while the "To direction" field (executing field) makes necessary changes so that the call can be routed to where you would like it to go. In the picture above we have chosen the direction SIP-to-ISDN. Depending on which modules are plugged in, the fields 'From direction' and 'To direction' could have the following values::

- SIP
- Analog (FXS/FXO)
- ISDN (PRI/BRI)
- GSM

In other words if haven't plugged in a GSM module, you will not see the GSM option in 'From direction' and 'To direction' pair.

Note: You can combine every technology direction with each other except the direction SIP-to-SIP.

From and To ID

The next pair are the **From ID** and **To ID** fields. Depending on what is chosen in the 'From direction' field, you will have different values in the 'From ID Box'.

From ID

From Direction is set to PSTN: (ISDN / Analog / GSM):

In this case the From_ID is a list of all configured PSTN port-groups indicated by their names. For instance if you select ISDN you will only see the ISDN port-groups in the list. Alternatively to the port-group you could also select a single port, identifiable by it's port number. But this port has to be a member of a port-group anyway, otherwise you won't be able to select it. All calls which originated from this single port or the port-group would match the criteria 'From_ID'.

From Direction is set to SIP:

If 'From Direction' is set to SIP, the 'From ID' is a list of all SIP-accounts, configured on the beroFix, indicated by their 'Names'. In this case the a new field called 'Match type' would be added. The 'Match type' give you the possibility to choose how beroFix should match for the From_ID. The match type has the following options and meanings:

-From_IP: By choosing this option, the IP-address from where the call is originated has to match to the 'Server address' of the SIP account, which has been chosen in the From_ID Field.

-From User: By choosing this option, additionally to the IP-Adress (see From_IP), the SIP From_user part, of the a call has to match to the the 'user' field of the SIP-account, which has been choosen in the From_ID field.

-Manual Address: By choosing this option, you can set a manual address (IP-address) which should be used instead of a existing SIP-account. Matching is the same like described in 'From_IP' above. This field also could contain regular expressions like our example below:

```
[172.20.0.2|172.20.0.3]
```

To ID

The To ID field is a bit easier, because it is the execution part. If 'To direction' set to PSTN (ISDN / Analog / GSM) a list of the corresponding configured PSTN ports groups or PSTN ports is shown and you can select the desired direction. If 'To direction' is set to SIP you can select one of the configured SIP accounts or enter a manual address as described above.

Destination / New Destination

This pair 'Destination / New Destination' is one of the most important matching criterias. The destination is the calledID (the number which was originally dialed) also called DAD, this means the number which enters the dial plan. While the new destination is the calledID, when we leave the dial plan. Both destination and new destination are based on regular expressions.

Source / New Source

This pair 'Source / New Source' very similar to the pair 'Destination / New Destination' but apply to the callerID (OAD)

Comments

For each dial plan rule you can leave a small comment. You can see the comment as a tooltip Box in the 'Dialplan overview', when you hover over a dial plan rule with the mouse.

First position

If you enable option while creating a new dial plan rule, the rule will be added onto the top of the dial plan list instead of the bottom.

Active

Each dial plan rule can be active or inactive. You can see a deactivated rule in the dial plan overview, it will be indicated as inactive by a grey background.

Advanced dial plan options

When you have created a dial plan rule you can customize it to be more advanced by clicking on the tool icon on the right-hand side of the dial plan overview page (the fourth icon on the right). Depending on what is set for 'From' and 'To Direction' you will have corresponding buttons to reach the more advanced options for ISDN / Analog / GSM and SIP.

Option	Value	Default
EC	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
EC tail length	15	<input checked="" type="checkbox"/>
dnumplan	unknown	<input checked="" type="checkbox"/>
onumplan	unknown	<input checked="" type="checkbox"/>
screening	<input type="checkbox"/>	<input checked="" type="checkbox"/>
presentation	<input type="checkbox"/>	<input checked="" type="checkbox"/>
Bearer	SPEECH	<input checked="" type="checkbox"/>
Call deflection	<input type="checkbox"/>	<input checked="" type="checkbox"/>

The picture above shows you the advanced option's in this case for ISDN. These settings are exact the same like described in the Chapter 'Advanced ISDN PRI / BRI options' , which are attached to the 'ISDN Port Group'. In the dial plan we have the possibility to make special settings for one particular rule. This means on the 'ISDN-Port-Group' we have the default settings while in the dial plan we can overwrite one or more values for a specific dial plan rule. In the right column of the above picture you can see the default button. By default this is activated. For instance if we check the option EC (Echo canceling) you will see that the value EC itself and the default column are activated. This means that the default on the ISDN-port-group for the option EC is activated and the dial plan inherits this value. To change this particular value you have to deactivate the 'default' option and then deactivate the value EC, as you can see in the next picture.

Option	Value	Default
EC	<input type="checkbox"/>	<input type="checkbox"/>
EC tail length	15	<input checked="" type="checkbox"/>

This should illustrate how you can make very specific dial plan settings to particular rules.

Note: The dial plan always has the highest priority. For instance in the above mentioned example EC is deactivated in the dial plan but activated in the 'ISDN-port-group'. If the dial plan rule matches, the value EC would be 'off' although this value is set to be 'on' by the ISDN-port-group.

All other options (Analog / GSM / SIP) follow the same mechanism as described here. For more information please read their respective Part in Chapter 3.

Examples of dial plan rules

The best way to explain the berofix device handles dial plan rules, is to show you some examples. And before we start showing you examples, this might help to understand them (in case you didn't read the 'Howto RegEx' (<http://www.zytrax.com/tech/web/regex.htm>)):

```
( ) will make everything contained by parentheses referable.
You can refer to it by the use of the <DIGIT> i.e. \1 for the first parentheses \2 for the next etc.
. is a special symbol that matches any single symbol
* is a multiplier that changes the statement after which it stands
(i.e. .* would mean <any_symbol>AND<any_number_of> which would mean literally any string of any length)
therefore (.*?) would cause anything to match and make it referable by it's respective <DIGIT>
```

Example1: Incoming call from SIP with the following settings

```
SourceIP: 172.20.0.1
CallerID: 2593890
CalledID: 025938912
```

Dialplan entry values:

```
Direction: "SIP->ISDN"
FromID: "(.*)" matches any IP-address
ToID: "(g:te)" routed to ISDN-port-group g:te
Destination: "0(.*)" matches any calledID starting with '0'
New Destination: "\1 " \1 will reference to parameter 1 ( \1 the value in the first parenthesize of destination )
Source: "(.*)" matches any callerID
New Source: "\1 " \1 is the value in the first parenthesize of source
```

With these settings the "Call" will be routed to ISDN-port-group "g:te". The calledID will be changed to 25938912, that means the first '0' will be stripped while the callerID will routed transparent and still is 2593890.

Example2: Incoming call from SIP with the following settings

```
SourceIP: 172.20.0.1
CallerID: 12
CalledID: 0176242XXXXX.
```

Dialplan entry values:

```

Direction:      "SIP->ISDN"
FromID:         "172.20.0.1"    matches if source IP-address is 172.20.0.1
ToID:          "1"             routed to ISDN-port 1
Destination:   "0176(.*)"     matches if calledID is starting with 0176
New Destination: "0049176\1"  will cut 0176 from calledID and add 0049176 to the calledID, followed by reference to parameter 1
Source:        "(.*)"         matches callerIDs with exact 2 digits.
New Source:    "25938912"     CallerID will be overwritten by 25938912

```

With these settings the the "Call" would be routed to ISDN-port 1". The calledID will be modified to 0049176XXXXX. The callerID will be changed to 25938912.

Example3: Incoming call from ISDN with the following settings

```

ISDNPort:      g:teports
CallerID:      0176242XXXX
CalledID:      25938912

```

Dialplan entry values:

```

Direction:      "ISDN->SIP"
FromID:         "g:teports"    matches if the call is originated from the ISDN-port-group named 'teports'
ToID:          "p:mysipserver" routed to the SIP account named 'mysipserver'
Destination:   "259389([0-8][0-9])" matches all numbers starting with 259389 followed by 2 digits in the range from [00-89]
New Destination: "\1"         will cut 259389 from calledID and add the 2 digits referenced by parameter 1
Source:        "(.*)"         matches any callerID
New Source:    "\1 "         \1 is the value in the first parenthesis of 'Source'

```

With these settings the the "Call" would be routed to SIP account p:mysipserver. The calledID will be changed to 12. The callerID will routed transparent and still is 0176242XXXX.

Example4: Incoming call from ISDN with the following settings

```

ISDNPort:      1
CallerID:      12
CalledID:      02593890

```

Dialplan entry values:

```

Direction:      "ISDN->SIP"
FromID:         "1"           matches the call is originated from the ISDN-port 1
ToID:          "p:mysipserver" routed to the SIP account named 'mysipserver'
Destination:   "0([2-9])(.*)" matches all numbers starting with 0. The second digit has to be in range[2-9] followed by the any digits.
New Destination: "\1\2"     will cut 0 from calledID and add the parameter 1 followed by parameter 2
Source:        "(.*)"         will match any callerID
New Source:    "\1 "         \1 is the value in the first parenthesis of 'Source' field

```

With these settings the the "Call" would be routed to SIP account p:mysipserver. The calledID will be changed to 2593890. The callerID is untouched.

With the above examples you should be able to handle almost every situation in the real world. If this is not enough and you need some special things, feel free to implement more complex regular expressions. More informations about this can be found in the corresponding chapters.

Preferences

This chapter will explain the possible settings you can make in the preferences part of the web-interface.

Network settings

By selecting the menu point "Network" you can configure all network options like IP-address, netmask as well as the default gateway, as you can see in the next picture.

Network Settings	
<input type="radio"/> DHCP <input checked="" type="radio"/> Static	
IP Address:	172.20.70.41
NETMASK:	255.255.0.0
GATEWAY:	172.20.0.1
MTU Size:	1500
Name Server:	
VLAN Enable:	<input checked="" type="checkbox"/>
VLAN Id:	
VLAN Priority:	0
<input type="button" value="Reset"/> <input type="button" value="Save"/>	

DHCP / Static IP

Choose to manually configure the beroFix network settings or use DHCP to get them from a DHCP server.

IP-Address

If you using static IP you can enter the IP-address the beroFix device should use.

NETMASK

If you using static IP you can enter the subnetmask the beroFix device should use.

Gateway

If you using static IP you can enter the default gateway the beroFix device should use.

MTU size

Maximum transmission unit. A value between 0-1500. Default is 1500.

VLAN Enable

If you want you add the berofix device to a VLAN then enable this option.

VLAN ID

To set the ID of the VLAN the berofix device is to be part of.

VLAN Priority

Here you can set the flag of the VLAN packet priority. High priority can result in better transmission because the packets are handled with priority over other packets.

Time settings

This is what the time settings look like:

Time Settings	
NTP Host:	<input type="text"/>
Timezone:	(GMT +1:00) Berlin, Brussels, Copenhagen, Madrid, Paris <input type="button" value="v"/>
Summer/Winter time change:	<input type="checkbox"/>
Time from ISDN:	Disabled <input type="button" value="v"/>
<input type="button" value="Reset"/> <input type="button" value="OK"/>	

NTP Host

If you want to get the time for your berofix device from a server using hosting ntp, enter it's IP address here.

Timezone

Please select the time zone you are using the berofix device in.

Summer / Winter change

Enable this option if you want the berofix device to automatically adjust the systemtime to summer / winter changes.

Time from ISDN

If you want the berofix device to get it's systemtime from ISDN, you can do so by enabling this option. You have to choose a specific port. All incoming ISDN calls are checked for the time and if it differs from the current system time, the system time is set to the new time.

Provisioning

This is what the provisioning interface looks like:

Provisioning Settings	
Provisioning options:	tftp <input type="button" value="v"/>
TFTP Host:	<input type="text"/>
TFTP URL:	<input type="text"/>
Use boot TFTP:	<input type="checkbox"/>
API Password:	admin
enable UserAppfs:	<input type="checkbox"/>
<input type="button" value="Save"/>	

If you want to use the provisioning feature, you can choose either TFTP or HTTP provisioning.

TFTP / HTTP Host

Here you have to enter the IP-address of the respective TFTP / HTTP host. (The host that provides the configuration.)

TFTP / HTTP URL

Here you have to enter the URL of the configuration file in question of the respective TFTP / HTTP server.

Use boot TFTP / HTTP

If this option is activated, berofix will try to start provisioning process during the boot phase.

API password

The password to the berofix API of the berofix device. You can access the API via HTTP 'http://IP-Address/app/api/berofix.php'. The default username and password is 'admin'.

enable UserAppfs

This setting controls if berofix apps should be started or not.

berofos heartbeat

This is what the berofos heartbeat interface looks like:

berofos Heartbeat settings	
IP-Address	<input type="text"/>
MAC-Address	<input type="text"/>
Interval (sec.)	<input type="text"/>
Heartbeat on boot	<input type="checkbox"/>
<input type="button" value="Save"/>	
berofos Heartbeat state	
●	<input type="button" value="Activate"/>

IP-Address

If you want to use the berofos heartbeat feature, you need to enter the IP-address of the berofos device here.

MAC address

If you want to use the berofos heartbeat feature, you need to enter the MAC-address of the berofos device here.

Interval (sec.)

Here you can set the interval in which the berofix device sends a heartbeat signal to the berofos device.

Heartbeat on boot

Enable this option if you want the berofix device to send a heartbeat during and after booting automatically. (Alternatively you can also manually start the heartbeat)

berofos heartbeat state

Here you can see what the current heartbeat setting is.

Logging

This is what the logging interface looks like:

Logging	
Logging server:	127.0.0.1
Logging server port:	54323
Logging active:	<input checked="" type="checkbox"/>
Log level:	0
Syslog facility:	16 - local 0
Save	

Logging server

The IP-address where the berofix device should send it's logging information to.

Logging server port

The UDP port of the logging server

Logging Active

Here you can enable / disable the logging feature.

Log level

Here you can set the logging level. Log level 1 is very low while log level 9 is very high.

Syslog facility

Syslog facility is an information field associated with a syslog message. It is defined by the syslog protocol. It is meant to provide a very through clue from what part of a system the message originated from. LOCAL_0 to LOCAL_7 facilities, are traditionally reserved for administrator and application use. The facility can be very helpful to define rules that split messages for example to different log files based on the facility level. syslog_facility=[16-23] correlate to local0-local7.

Security

This is what the security interface looks like:

Security settings	
Old password:	•••••
New password:	
Confirm new password:	
Disable sessions:	<input type="checkbox"/>
ACL	
Telnet ACL:	127.0.0.1 / 32
HTTP ACL:	0.0.0.0 / 0
SIP ACL:	0.0.0.0 / 0
bfdetect ACL:	0.0.0.0 / 0
Save	

Security:

Old password

If you want to change your password, you must first enter the current one here as authentication.

New password

If you want to change your password, you can enter the new one here.

Confirm new password

If you want to change your password, you need to enter it again for verification.

Disable sessions

By default the berofix web-GUI uses session management. If you need to you can deactivate sessions here.

Access control list

Telnet ACL

Enter an IP-address or an IP-address range, to allow access to the telnet interface.

HTTP ACL

Enter an IP-address or an IP-address range, to allow access to the HTTP interface.

SIP ACL

Enter an IP-address or an IP-address range, to allow access to the berofix SIP listening Port.

bfdetect ACL

Enter an IP-address or an IP-address range, to allow responses to the bfdetect requests

SSH ACL

Enter an IP-address or an IP-address range, to allow access via SSH.

Note: By default the Network connected to eth0 will be added as an allowed Network i.e. all access via the lan connected to eth0 will be allowed.

Some Examples:

- ACL 127.0.0.1 / 32 will limit the access to the IP-address 127.0.0.1 (localhost)
- ACL 192.168.1.0 / 24 will limit the access to the network 192.168.1.1-192.168.1.254
- ACL 172.20.0.0 / 16 will limit the access to the network 172.20.0.1-172.20.254.254

Causes map

Modifying the causes map This is an example what you could see when clicking on the modify button on the right-hand side (in the row of the cause that you want to modify):

Call direction

Here you can choose the direction of call that a call has to be from this cause to be relevant.

To cause (SIP)

Here you choose a cause to which the original cause should be translated to.

From cause (ISGW)

Here you choose which cause is to be converted to the different cause set above.

SIP response message

Here you can set a message which is also sent as part of SIP transmission if the cause applies. Very useful if you know of some error in the SIP transmission process but can't locate it.

Please find more detailed info here: [BeroFix_ISDN_Cause/SIP_Response_map](#)

Call progress table

Here you can define what should happen when ISDN events occur. What you can configure depends on the modules in use. So far this feature only supports SIP - ISDN and vice versa. Here is an example to make it more clear:

te	ALERTING	any_state	183	Ringing	 
----	----------	-----------	-----	---------	---

Interface type

Here you can choose the type of interface that this call progress table should be applied to.

ISDN event

Here you can choose which ISDN event is meant, by this call progress table entry.

ISDN channel details

Here you can define a certain state the ISDN channels have to be in for this call progress table entry to apply.

SIP response

Here you can define to what SIP response the ISDN event should be translated.

Response text

Here you can add a response text. It is also sent to the respective SIP device for instance an 3cx telephone system, which could then forward the packet to some SIP phone which would then display this response e.g. "Ringing" (depending on ISDN event is being translated).

Please find more info here: [BeroFix Call Progress Tables](#)

Management

Under the menu point 'Management' you can reach the state view, activate debugging as well as reboot or reset the BeroFix to its factory defaults. In the following chapters we will go into details for each menu point.

State

The menu point "State" will give you an overview about the PSTN, SIP and PCM interconnection state. As you can see in the next picture you will get the state overview of the PSTN-Ports.

ISDN status					
Port	Type	Prot.	L2Link	L1Link	
1	TE	PTP			Restart
2	TE	PTP			Restart

Analog status				
Port	Type	Status	Line voltage	
1	FXO	FXO_IDLE		
2	FXO	FXO_IDLE		
3	FXO	FXO_IDLE		
4	FXO	FXO_IDLE		
5	FXS	FXS_IDLE		
6	FXS	FXS_IDLE		

The first Box in the picture above shows you the ISDN Ports which are members of an ISDN-port-group. Furthermore you can see that the state for layer 1 and layer 2 is down for ISDN-port 1 and 2. You can get more informations regarding the state of these ports by hovering over the L1 link and L2 link field. The second box in the above shown picture shows you a list of all analog ports which are members of an analog-port-group. On FXO and FXS ports you can also see the line voltage.

Registration status:			
Account name	Registrar	User	Status
Asterisk	172.20.0.1	test	

PCM Server status:	
---------------------------	--

As you can see in the section ISDN status, that ports 1-3 are configured and the state for layer 1 and layer 2 is up.

Active calls

Currently "active channels" on the beroFix device.

port statistics

Minimum call statistics containing information about port, direction, calls, connected, avrg. connected time and less than 10 Sec. calls

Backup and restore

You can backup the configuration of a beroFix device to an external file by downloading it. These backup files could be used to restore the settings of a beroFix device. Be aware that this could depend on the beroFix firmware version. So if you try to restore a backup file from an older version of the firmware, the restore process might fail. This should only be the case for big fireware version leaps (e.g. 1.x -> 2.x)

You can also untar the backup and restore single files, by just uploading the pure config file in the restore field. E.g. you only want to restore the dialplan, then just uploade the file "isgw.dialplan".

Firmware update

By selecting the menu point "Update Tool" you can update the firmware of a beroFix device. Under the following URL

```
http://www.beronet.com/downloads/berofix/
```

You can always find the newest available firmwares.

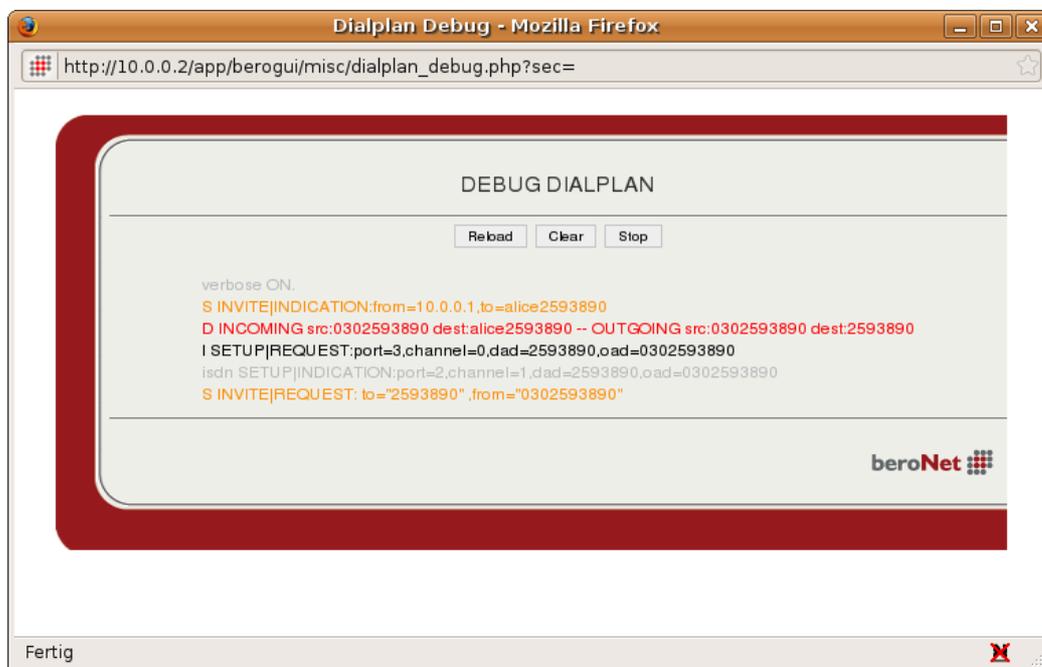
The firmware update tool can be found under "Management->Firmware Update" and looks like:



The rest should be straight forward and self-explanatory.

Dial plan debug

This page is to give the user the opportunity to debug the dial plan if something is not working as expected.



Full trace

The menu point "Full Trace" will give you the opportunity to create a trace of all interactions of the berofix device. This is useful when you want to report a problem to us that you're having with a berofix device in a standard manner. This trace will help us to reproduce, analyse and fix your problem quickly. On all ISDN-ports you can start a so called "Fulltrace" by pressing the button "Start". By starting the debug process the system will start tracing all ISDN and SIP protocol elements. After you have finished your tests you can press "stop" and download the trace file. All relevant information about the callflow are stored in this debug file. Send us this debug file to support@beronet.com with a small description about your test and we will be able to reproduce your problem.

Info

This Info page give you some general Informations about the beroFix device as you can see in the following picture.

The screenshot shows the 'INFO' page of the beroFix device management interface. The page has a red header with the 'berofix' logo and language selection options (UK and DE). A navigation menu is located below the header, with 'Management' selected. The main content area is divided into several sections, each with a title bar and a table of data:

- Serial Number**: A single text field containing a redacted serial number.
- VERSIONS**: A table with four rows:

appfs	appfs-2.2.tar.gz
rootfs	Version 4
FPGA	Version 8
HW Revision	2.20
- MAC Address**: A single text field containing a redacted MAC address.
- Net Configuration**: A table with three rows:

IP Address:	[redacted]
Netmask:	[redacted]
Gateway:	[redacted]
- Date and Time**: A single text field showing '21-September-2012-12:44:19'.
- Line Interfaces**: A single text field showing 'ii0: BF4FXS'.
- Uptime**: A table with two rows:

System	13:44:19 up 1:35
ISGW	00:01:05

In the bottom right corner, there is a 'Help' button and the 'beroNet' logo.

CDR

This Option enables CDR logging or not. The gathered information can be downloaded by clicking on its respective download button.

Reboot / reset

By select the menu point "Reboot / Reset", you can reboot the device or reset it to "factory defaults". This is the so called **Software factory reset**

Hardware reset

If you don't have access to the GUI anymore you can perform a Hardware Factory Reset:

Comment faire un faire un reset usine (hardware factory reset) d'une passerelle beroNet

Remote management

Cloud enable

Activates the remote management via cloud feature

Cloud Key

The key that is used by the beroFix device when it tries to register itself with the cloud.

Cloud username

If you want to use the remote management feature you need to enter your cloud username here.

Cloud password

If you want to use the remote management feature you need to enter your cloud password here.

SNMP enable

Here you can enable or disable the SNMP service. This service provides useful information, which can be read with a SNMP client. Below you will find the necessary definitions file, which is used by your SNMP client to interpret the gathered information, so that it can be displayed in a meaningful manner.

Assembly Guide

- [Assembly-Guide-Gateways \(http://www.beronet.com/downloads/berofix/docs/Assembly-Guide-Gateways.pdf\)](http://www.beronet.com/downloads/berofix/docs/Assembly-Guide-Gateways.pdf)

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