MAYAH COMMUNICATIONS

Application Note 27

SIP Server Installation (Mayah example)

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1 General

This document describes how we at MAYAH have set up a public SIP server.
There were two reasons for installing a SIP server at Mayah Communications head office in Hallbergmoos:

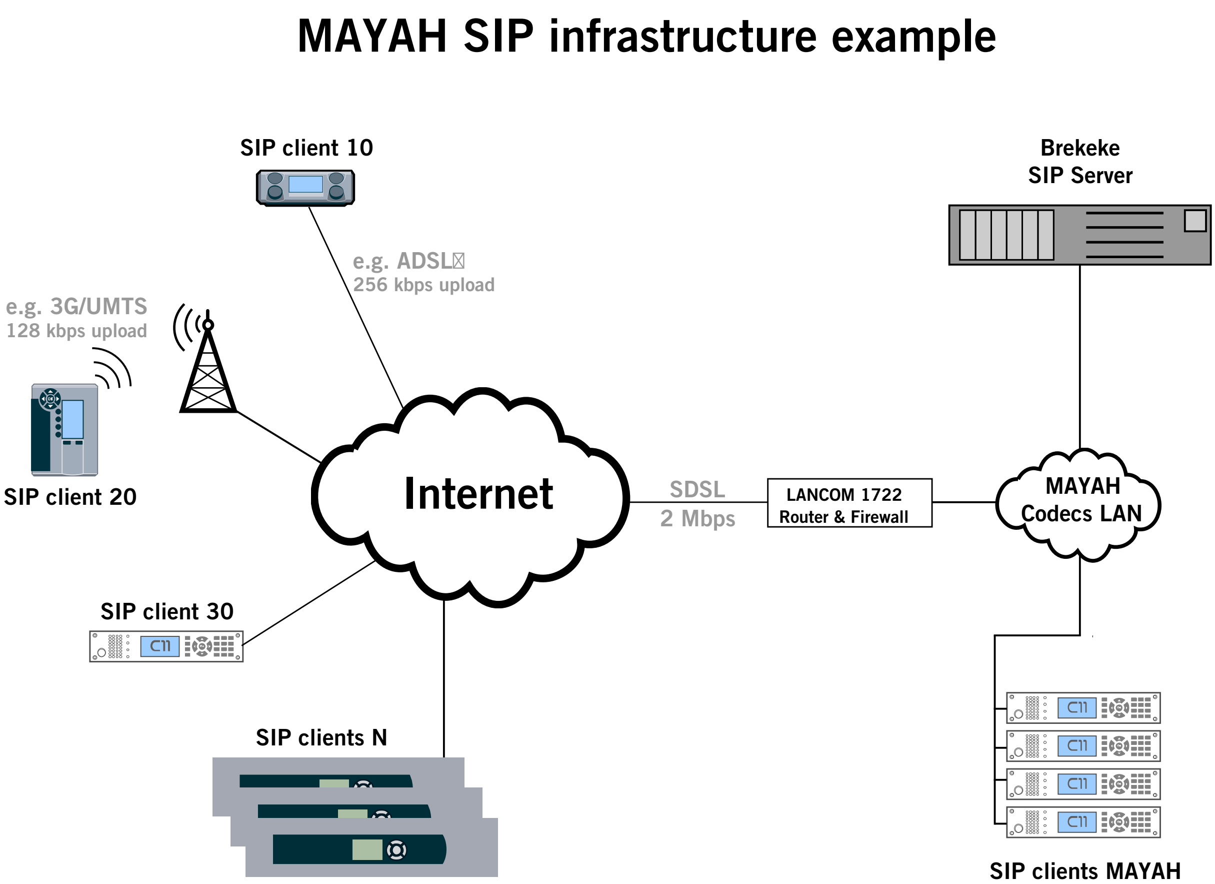
1) To enable users of MAYAH audio equipment to test SIP
2) To provide a template for installing a simple SIP service

Note:
MAYAH SIP Server is not available all the time. For real tests the access data and availability time slot must be requested. Please send your requests per e-mail at info@mayah.com

2 Environment

The installation described in this document is of course adapted to the given network structure here at MAYAH. Your environment might look different and therefore not all settings described here might be applicable. For security reasons we decided to put the SIP server not in a DMZ, but behind a firewall/router. Therefore port forwarding had to be configured in the router.

2.1 Schematic diagram



2.2 Used Equipment

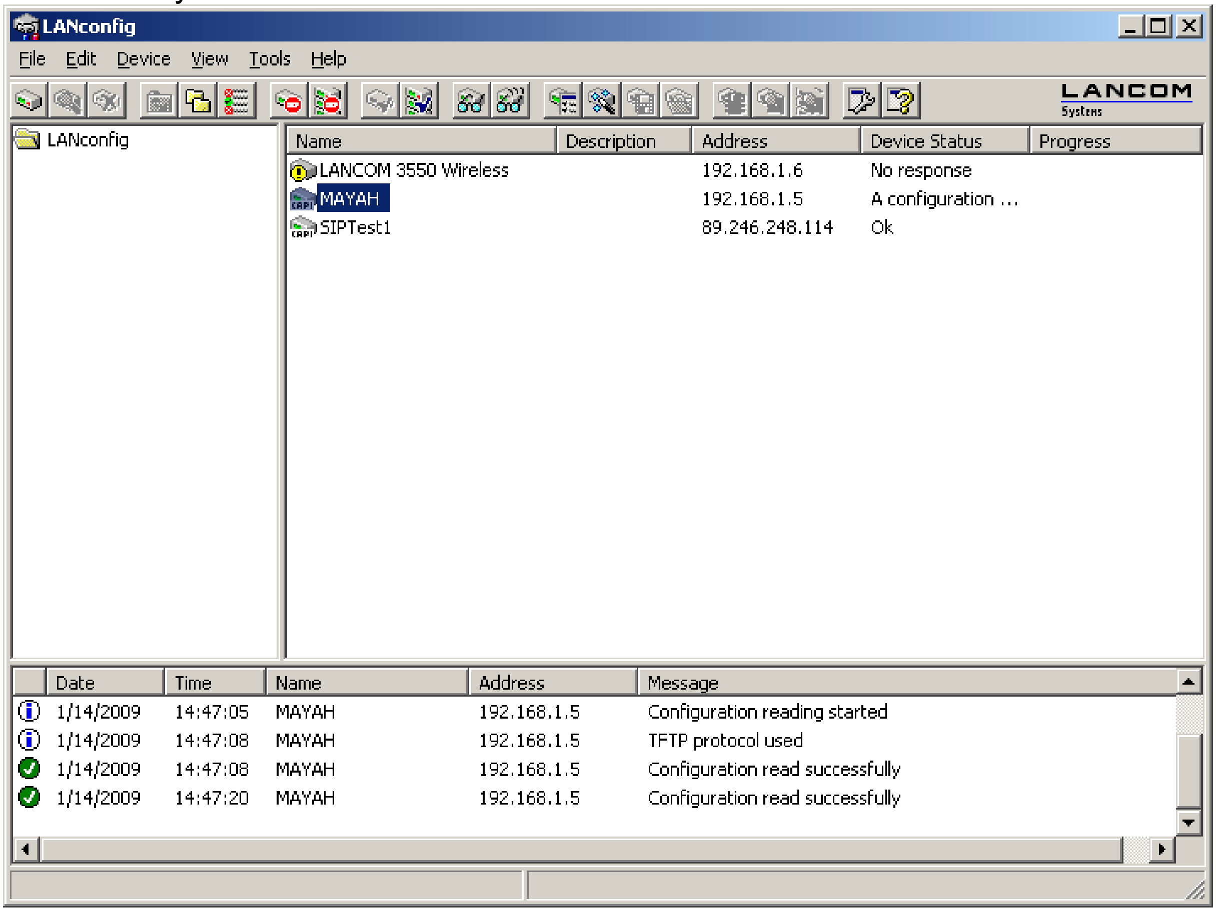
1) LANCOM 1722 (Firmware 7.28) as Router
2) Windows PC with
   - Windows 2000 (Service Pack 4)
   - Java Virtual Machine (Version 1.5.0)
   - Brekeke SIP Server Standard (Version 2.2.5.8)

2.3 Used Infrastructure

1) 2 Mbps SDSL Internet access (with fixed IP addresses)
2) Fast Ethernet LAN

3 Configuration of LANCOM 1722

LANCOM 1722 Firmware version 7.28.0031 (2/6/2008) and Windows configuration tool LANconfig 7.20.0018 (8/7/2007) was used.
Usually the LANconfig tool detects all LANCOM devices in the local network automatically.

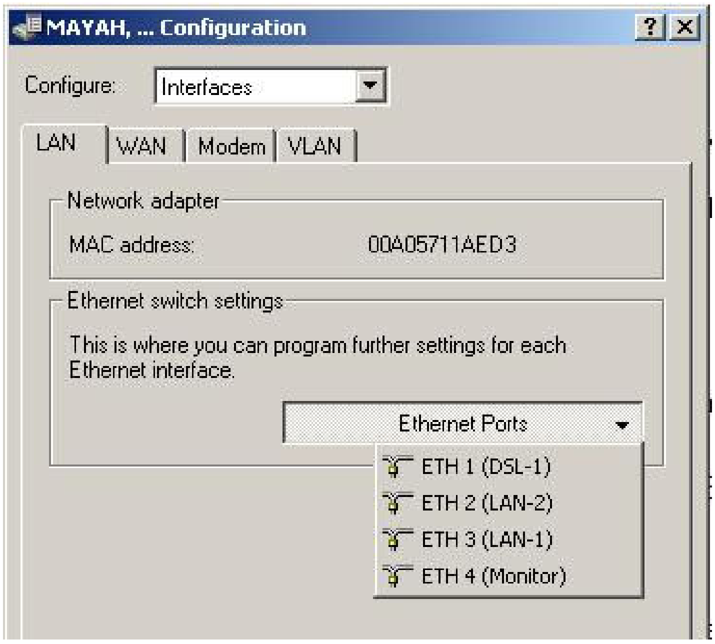


For configuration just double on the referring LANCom 1722 router in the list.

3.1 *Configuration ot the physical Ports of LANCOM 1722*

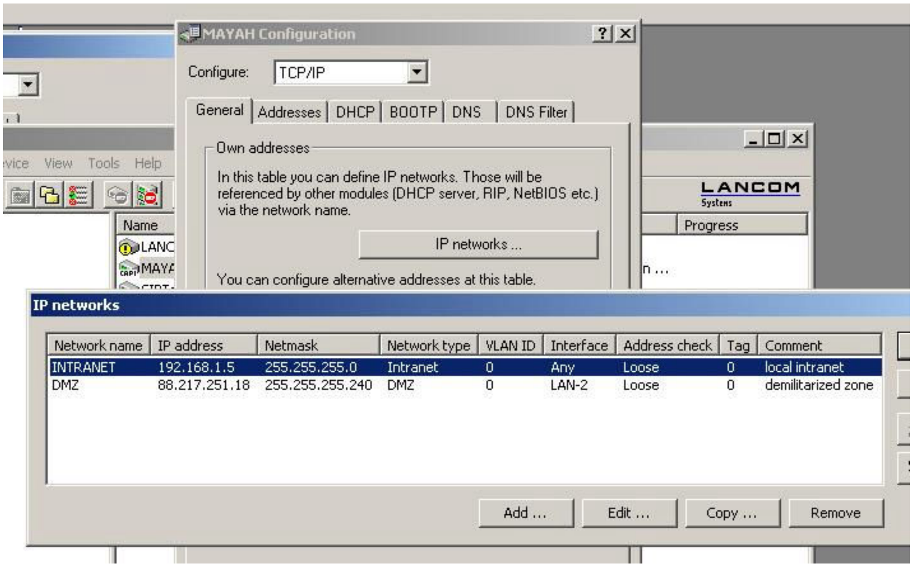
The 4 physical ports of the LANCOM 1722 can be configured via the configuration items

- Interfaces with button ‘Ethernet Ports’



and

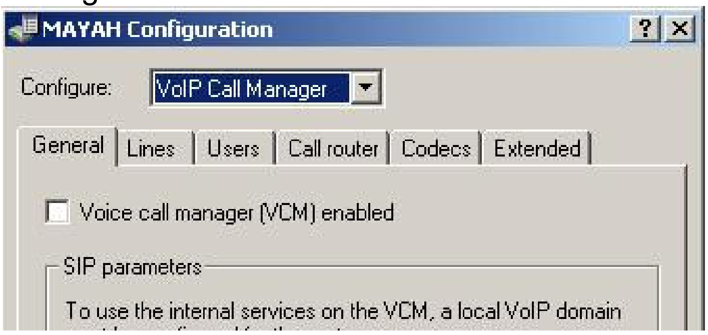
- TCP/IP with button ‘IP networks’



As you can see the physical port 3 is configured as a DMZ. However, the described below Brekeke SIP server should be protected by the firewall of LANCom 1722 and therefore is located behind the pyhsical port 3.

3.2 Disabling of internal SIP server of LANCOM 1722

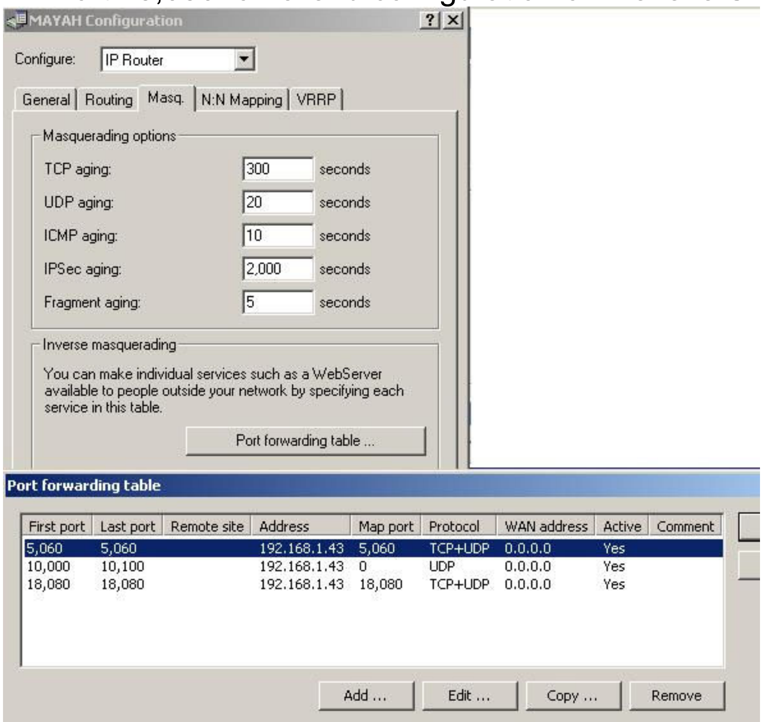
Since all VOIP functionality is done by the Brekeke SIP server the internal VoIP call manager of the LANCOM 1722 should be switched off.



3.3 Configuration of Port forwarding

Via configuration item IP router with button ‘Port forwarding’ the following ports must be forwarded to th PC hosting the Brekeke SIP server (in this case IP address 192.168.1.43)

- Port 5,060 for SIP exchange
- Port 18,080 for farend configuration of Brekeke SIP server



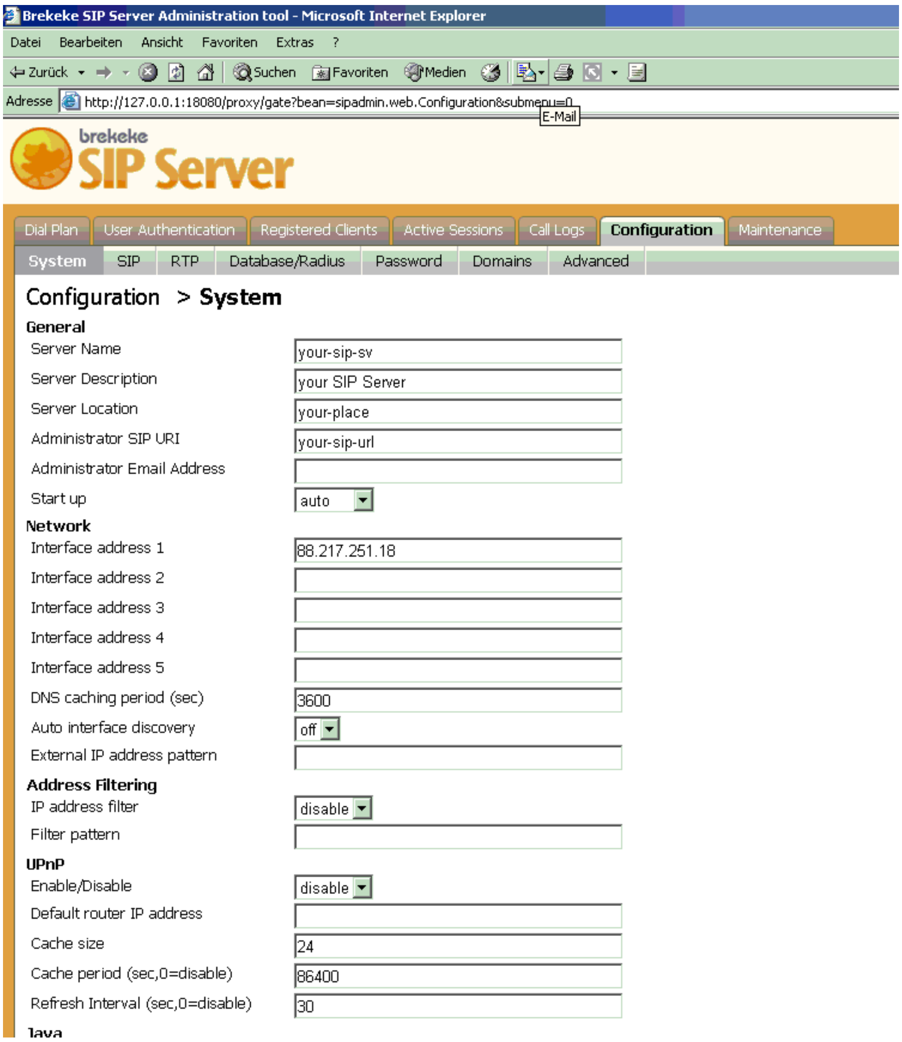
Note:
Forwarding ports 10,000 to 10,100 like in the picture is not.

4 Settings on Brekeke SIP Server

Brekeke SIP server version 2.2.5.8 is used. Where you can download Brekeke SIP server is described in chapter 6.2 (Links).
The Brekeke SIP Server provides many enhanced NAT traversal features, which unfortunately proved to be counterproductive in this case. These features had to be disabled in this scenario, since the server did not “know” that port forwarding was used and the (usually helpful) NAT traversal features prevented calls form being successful.

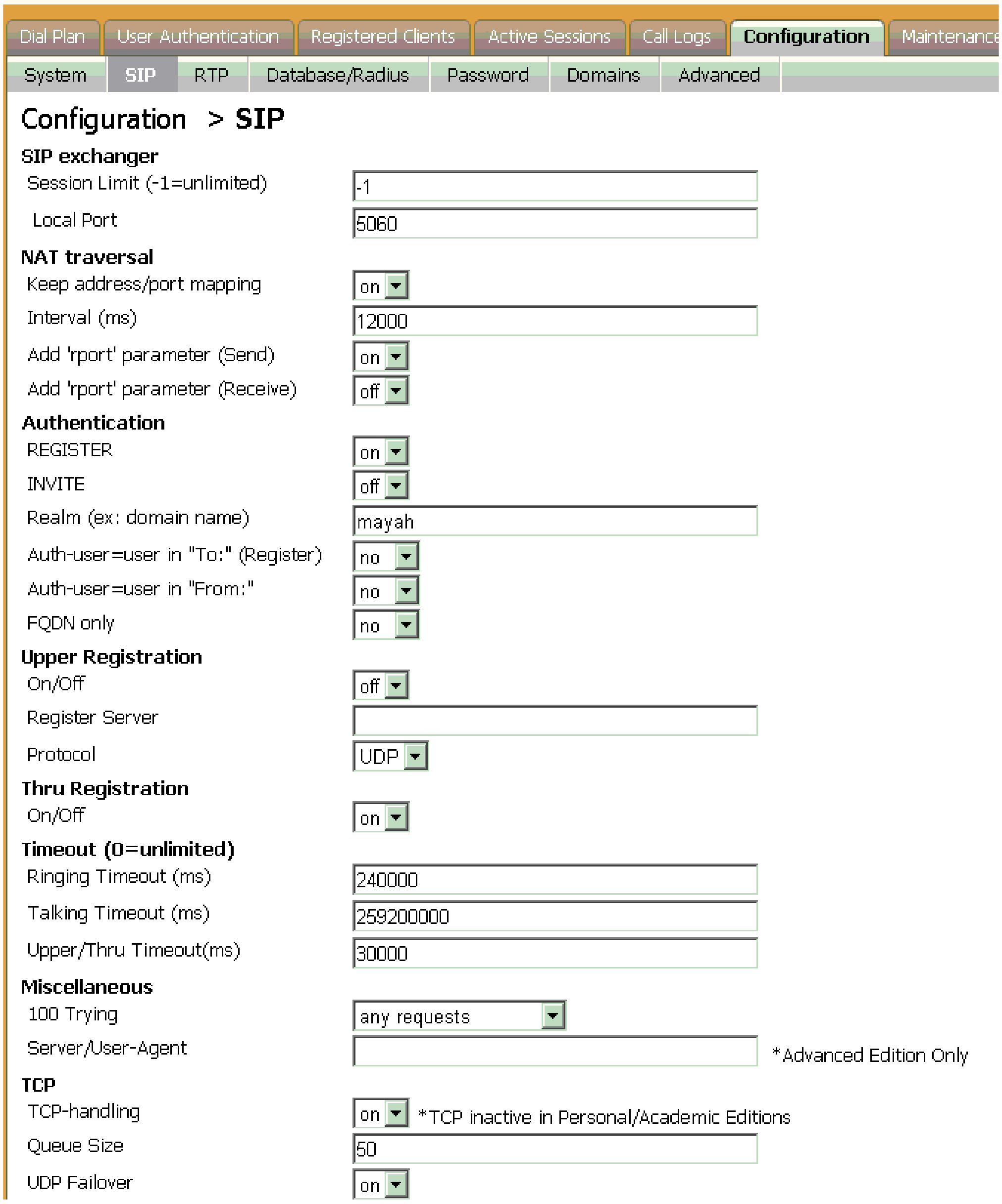
4.1 General Configuration

At Brekeke SIP server the general configuration is done via item > Configuration > System. At item ‘Interface address 1’ you must enter the public IP-address of the gateway used by the PC where Brekeke SIP server is installed on. This has to be done because of the port forwarding mechanism.
In this application gateway is the LANCom router 1722 (described in chapter 3). In this example it uses the public IP-address 88.217.251.18.
This IP address must be entered at your Mayah codecs as SIP registrar when you want use the Brekeke SIP server located in Mayah premises.
All other system configurations should be identical to the one in the picture below.



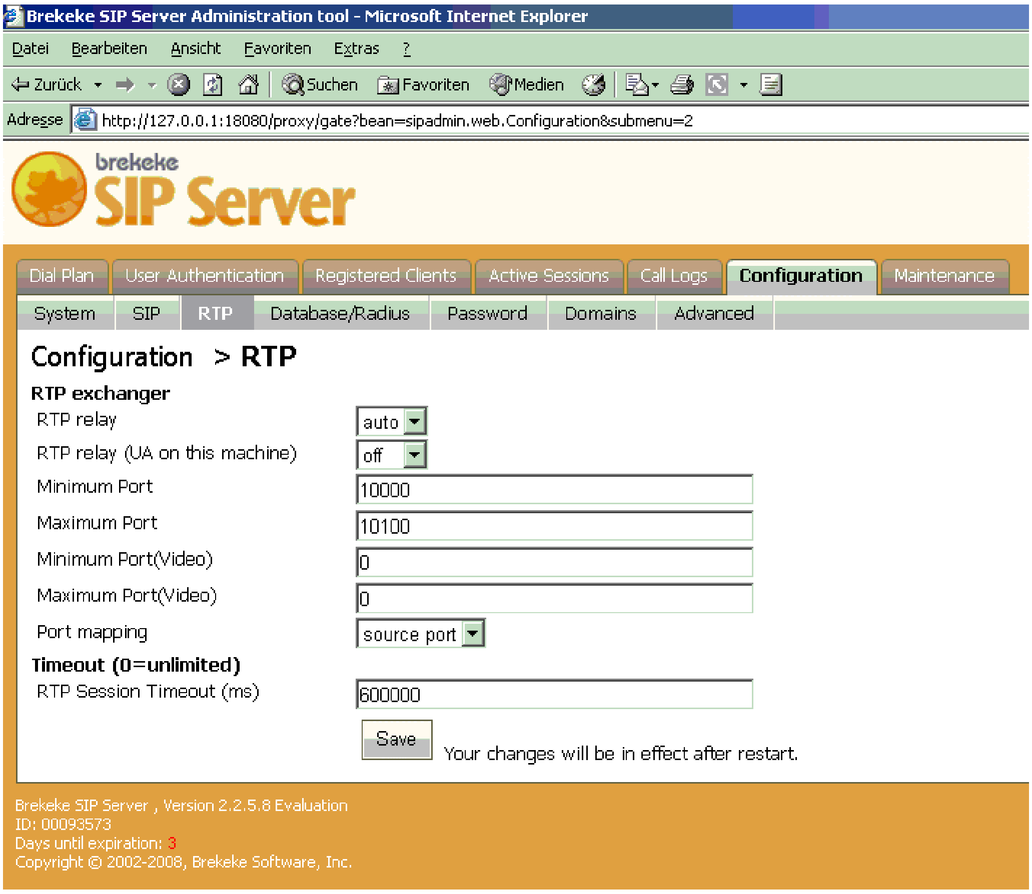
4.2 SIP Configuration

SIP configuration must be done via item > Configuration > SIP as described in picture below. The "realm" parameter may of course differ.



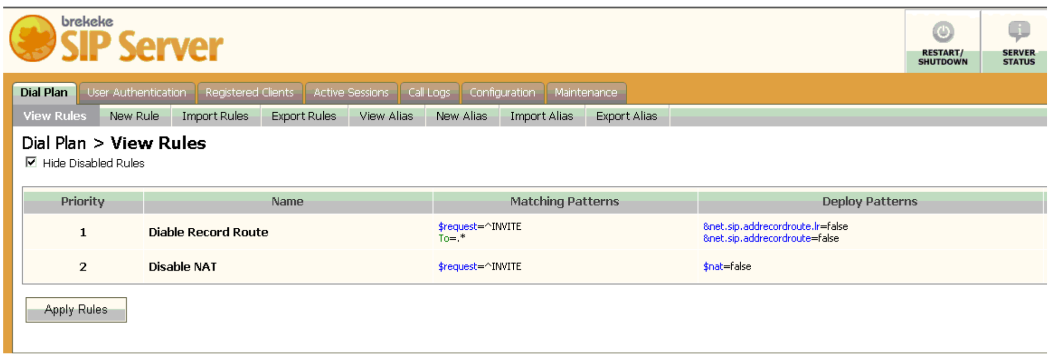
4.3 RTP configuration

RTP configuration must be done via item Configuration > RTP as described in picture below.



4.4 Dial plan

A new dial plan can be configured via item > Dial plan > New Rule.



4.4.1 Reasons for dial plans

Dial plans in the Brekeke Server can be seen as small scripts that are run when a call comes in. We use this mechanism for doing extended configuration.

4.4.1.1 Disable Record Route

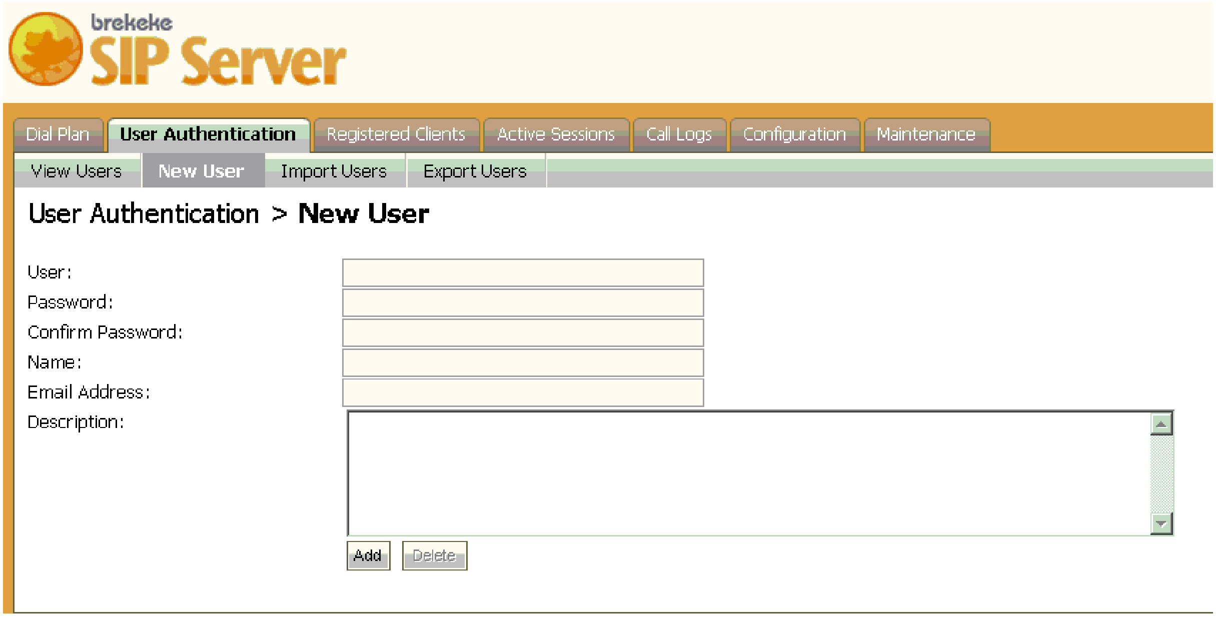
This is done since we have Codecs in the same LAN as the SIP server. Without this set the SIP ‘Bye’ messages might not be routed correctly resulting in callnot being hung-up.

4.4.1.2 Disable NAT

Furthermore Network Address Translation (NAT) should be prevented.
By default the Brekeke SIP server does not just handle the establishing of connections but also the audio over IP transfer. This is not recommendable for most applications.
For instance:
Your Brekeke SIP server is located somewhere in Europe but the audio over IP transmission should be done between two codecs in Australia. Then it makes no sense that the whole audio transfer between both Australian codecs is routed via Europe since this means an additional bottle neck and delay.
To prevent this dial plan item 2 as described in picture above must be entered.

4.5 Adding clients

Finally the data of clients using your SIP service must entered. This can be done via item > User Authentication > New user.



SIP registration data:

- User (mandatory data):
  Name of SIP account
- Password:
  Password protection of a SIP account is recommended but not mandatory.
- Name (mandatory data):
  Usually the SIP phone number to dial to the referring device. However, this data can be alphanumeric.
- E-mail address:
  Optional info to contact SIP user.
- Description:
  Here optional comments can be added.

All existing SIP accounts are listed at item > User Authentication > View Users.
Currently registered SIP clients are listed at > Registered Clients > View Clients.

5 How to use MAYAH SIP server for tests

If you don't want to set up your own SIP infrastructure for your test you can also register at MAYAH SIP server. To get your SIP account information please contact MAYAH (see chapter 6.1).

Disclaimer:
Such accounts on MAYAH SIP Server can only be used for test purposes and cannot be regarded as permanent solution for your Audio-over-IP transmissions. MAYAH Communications reserves the right to cancel any of the test accounts at any time.

5.1 SIP Access Data

- SIP Registrar: 88.217.251.18
- STUN Server: stun.t-online.de or stunserver.org
  Note:
  Since STUN Server is given as URL a DNS-Server must be entered. Otherwise the IP addresses of the above STUN servers must be entered.

- Account Phone Number must be requested from MAYAH
- Account User Name must be requested from MAYAH
- Account Password must be requested from MAYAH

5.2 Entering SIP Access Data

5.2.1 C11, SPORTY, FLASHMAN II

SIP access data can be entered via front panel menu item
CODEC / SETUP / INTERFACE / Ethernet / SIP
Note:
SIP Access Data can only be entered or changed if SIP account is inactive, i.e. "off".
Please check the menu item
CODEC / SETUP / INTERFACE / Ethernet / SIP / Account active.

5.2.2 CENTAURI II

SIP access data can be entered via front panel menu item
SETUP / INTERFACE / NETWORK / SIP
Note:
SIP Access Data can only be entered or changed if SIP account is inactive, i.e. "off".
Please check the menu item
SETUP / INTERFACE / NETWORK / SIP / Account active.

5.3 How to setup the SIP connection

For successful SIP registration at MAYAH SIP Server:

- SIP Access Data (see above) must be entered correctly

- SIP account must be active, i.e. “on”
- Internet connection must be available

After successful SIP registration C11, SPORTY and FLASHMAN II display the following symbol in the status bar of the main screens.

CENTAURI II displays a following message
“SIP: successfully registered”

Now you can establish a SIP connection to another device registered at MAYAH SIP Server.

To establish a SIP connection via Direct Dial
- protocol must be set to SIP
- Account Phone Number provided by MAYAH must be used as destination.

6 Additional Info

6.1 Contacts

1) Werner Ludwig: Mayah audio products support engineer
   E-mail: wludwig@mayah.com
   Tel.: +49 (0) 811 5517-0
2) Uwe Flatter: Mayah sales manager
   Sales and Rental for Mayah audio products (even for test and demo use)
   E-mail: uflatter@mayah.com
   Tel.: +49 (0) 811 5517-0

6.2 Links

1) Brekeke: www.brekeke.com/download/download_sip_2_0.php
2) LANCOM: www.lancom-systems.de
3) Java: www.java.com/de/download/