

## 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:

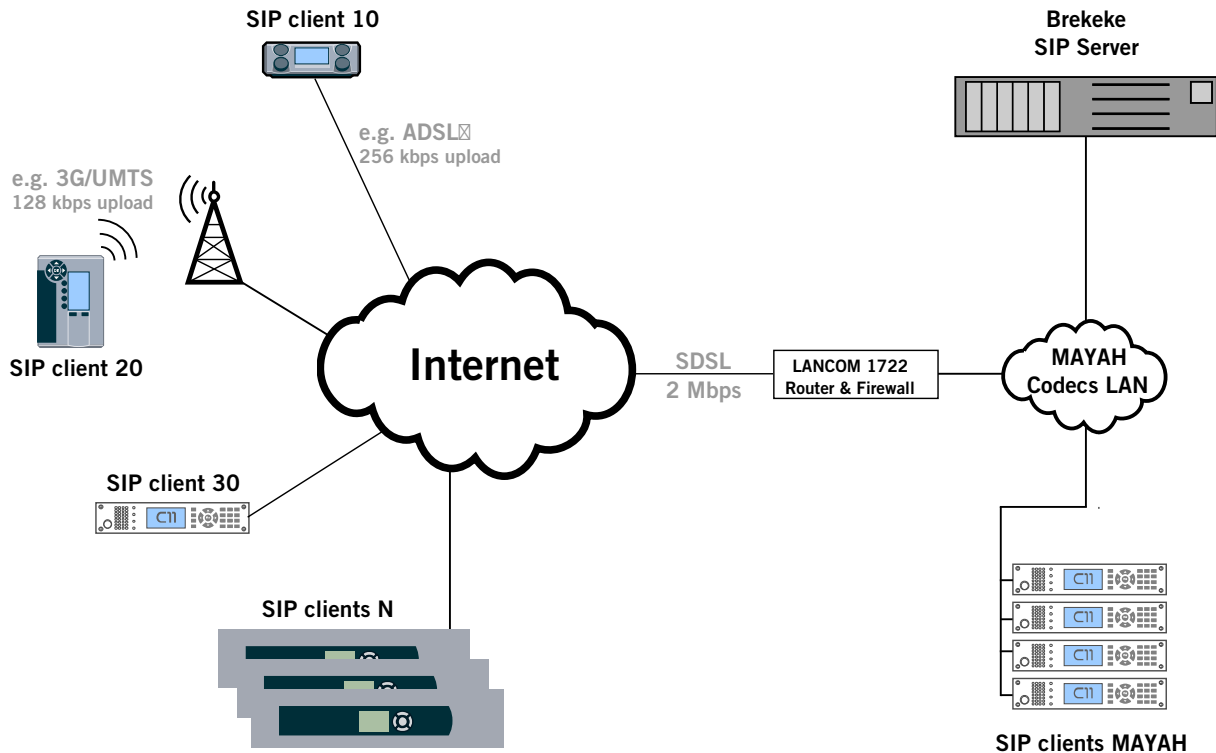
MAYAH SIP Server is not available all the time. For real tests the access data and availability time slot must be requested. Please send your requests per e-mail at [info@mayah.com](mailto:info@mayah.com)

# 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

### MAYAH SIP infrastructure example



## **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)

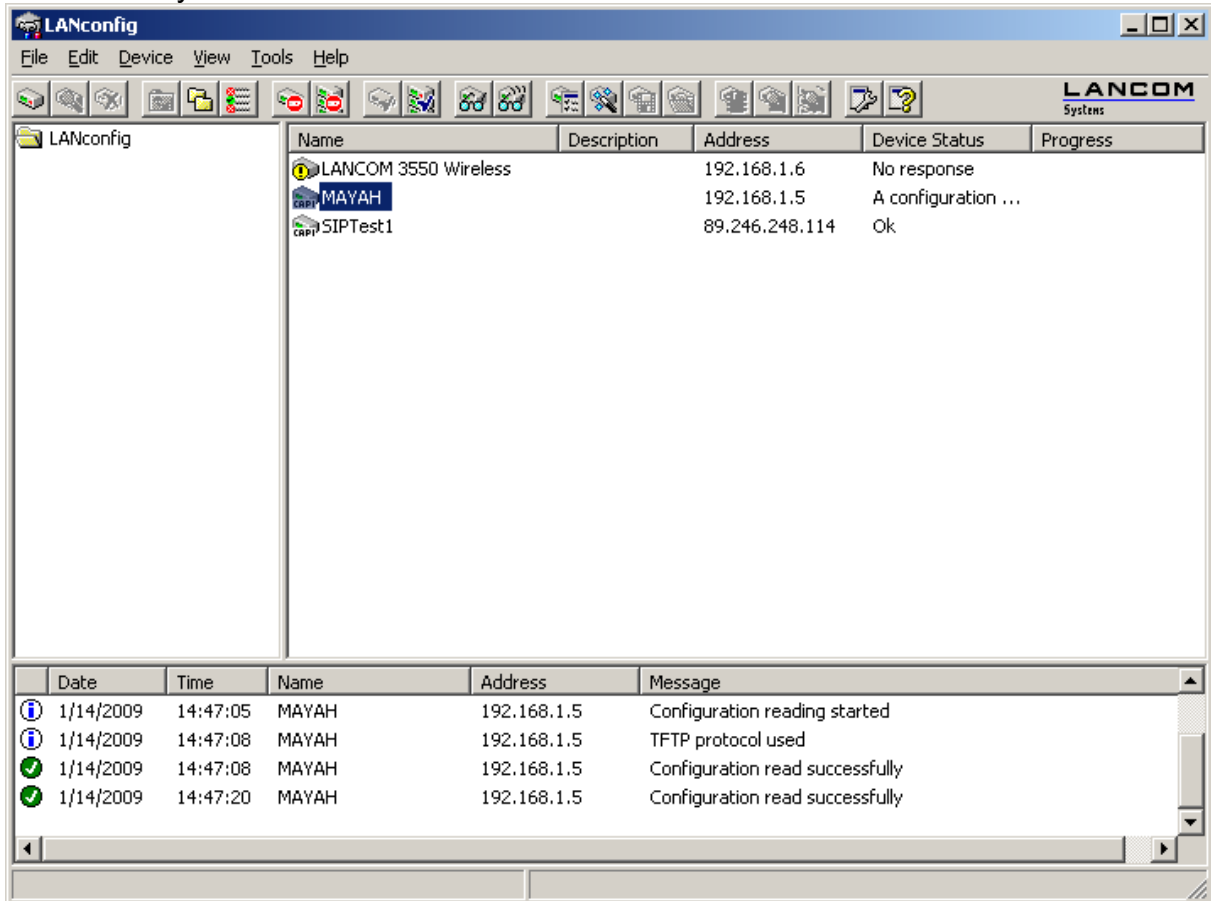
## **2.3 Used Infrastructure**

- 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.

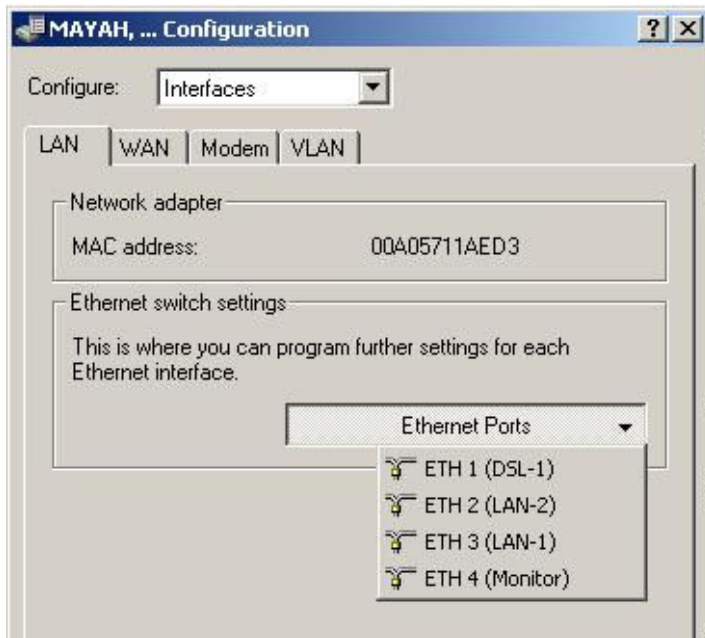


For configuration just double on the referring LANCom 1722 router in the list.

### 3.1 Configuration of the physical Ports of LANCOM 1722

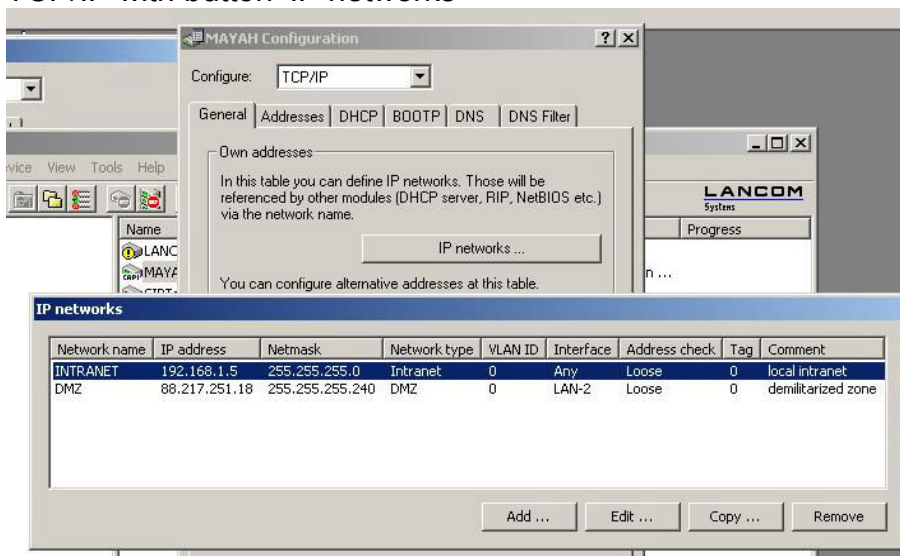
The 4 physical ports of the LANCOM 1722 can be configured via the configuration items

- 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 physical port 3.

### 3.2 Disabling of internal SIP server of LANCOM 1722

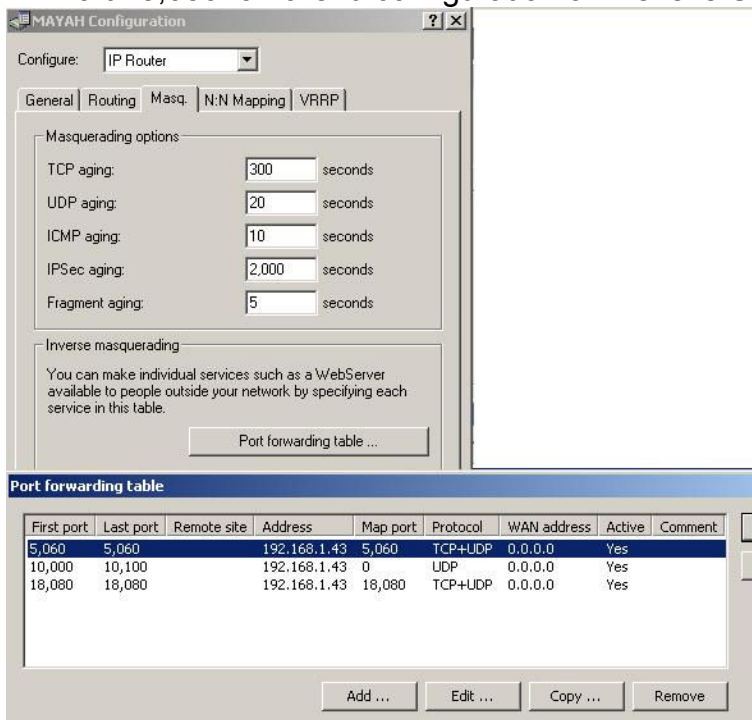
Since all VOIP functionality is done by the Brekeke SIP server the internal VoIP call manager of the LANCOM 1722 should be switched off.



### 3.3 Configuration of Port forwarding

Via configuration item IP router with button 'Port forwarding' the following ports must be forwarded to the 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 from being successful.

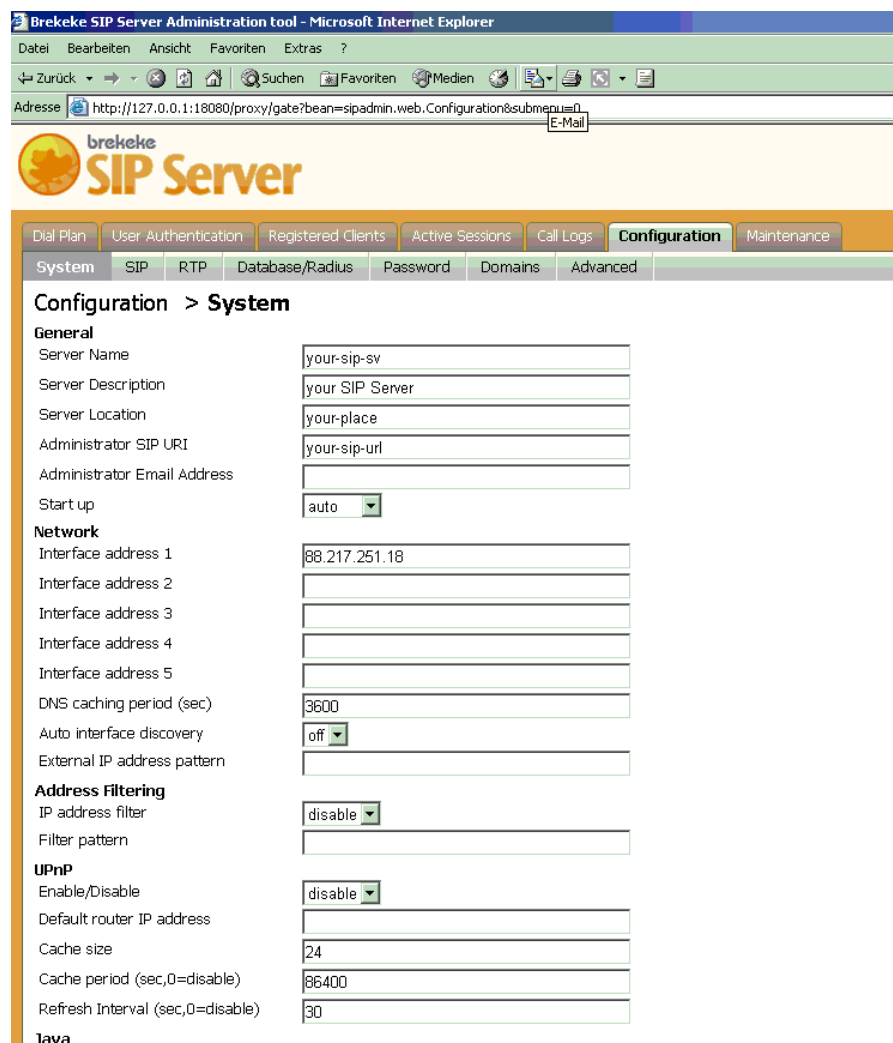
### 4.1 General Configuration

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.



The screenshot shows the Brekeke SIP Server Administration tool web interface in Microsoft Internet Explorer. The browser address bar shows the URL: `http://127.0.0.1:18080/proxy/gate?bean=sipadmin.web.Configuration&submenu=0`. The page title is "Brekeke SIP Server". The navigation menu includes "Dial Plan", "User Authentication", "Registered Clients", "Active Sessions", "Call Logs", "Configuration", and "Maintenance". The "Configuration" menu is expanded to show "System", "SIP", "RTP", "Database/RADIUS", "Password", "Domains", and "Advanced". The "System" configuration page is displayed, showing the following settings:

Configuration > System	
<b>General</b>	
Server Name	your-sip-sv
Server Description	your SIP Server
Server Location	your-place
Administrator SIP URI	your-sip-url
Administrator Email Address	
Start up	auto
<b>Network</b>	
Interface address 1	88.217.251.18
Interface address 2	
Interface address 3	
Interface address 4	
Interface address 5	
DNS caching period (sec)	3600
Auto interface discovery	off
External IP address pattern	
<b>Address Filtering</b>	
IP address filter	disable
Filter pattern	
<b>UPnP</b>	
Enable/Disable	disable
Default router IP address	
Cache size	24
Cache period (sec,0=disable)	86400
Refresh Interval (sec,0=disable)	30

Java

## 4.2 SIP Configuration

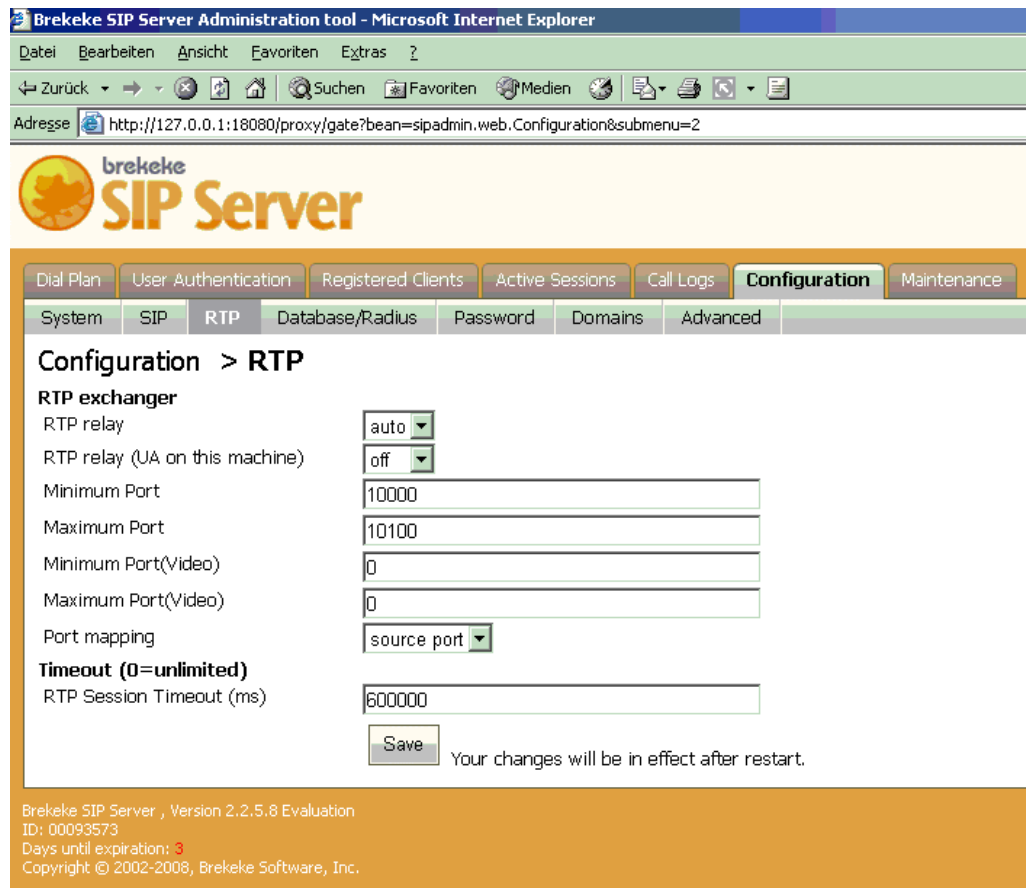
SIP configuration must be done via item > Configuration > SIP as described in picture below. The "realm" parameter may of course differ.

Dial Plan	User Authentication	Registered Clients	Active Sessions	Call Logs	Configuration	Maintenance
System	SIP	RTP	Database/Radius	Password	Domains	Advanced
<b>Configuration &gt; SIP</b>						
<b>SIP exchanger</b>						
Session Limit (-1=unlimited)	<input type="text" value="-1"/>					
Local Port	<input type="text" value="5060"/>					
<b>NAT traversal</b>						
Keep address/port mapping	<input type="text" value="on"/>					
Interval (ms)	<input type="text" value="12000"/>					
Add 'rport' parameter (Send)	<input type="text" value="on"/>					
Add 'rport' parameter (Receive)	<input type="text" value="off"/>					
<b>Authentication</b>						
REGISTER	<input type="text" value="on"/>					
INVITE	<input type="text" value="off"/>					
Realm (ex: domain name)	<input type="text" value="mayah"/>					
Auth-user=user in "To:" (Register)	<input type="text" value="no"/>					
Auth-user=user in "From:"	<input type="text" value="no"/>					
FQDN only	<input type="text" value="no"/>					
<b>Upper Registration</b>						
On/Off	<input type="text" value="off"/>					
Register Server	<input type="text"/>					
Protocol	<input type="text" value="UDP"/>					
<b>Thru Registration</b>						
On/Off	<input type="text" value="on"/>					
<b>Timeout (0=unlimited)</b>						
Ringing Timeout (ms)	<input type="text" value="240000"/>					
Talking Timeout (ms)	<input type="text" value="259200000"/>					
Upper/Thru Timeout(ms)	<input type="text" value="30000"/>					
<b>Miscellaneous</b>						
100 Trying	<input type="text" value="any requests"/>					
Server/User-Agent	<input type="text"/>					
<small>*Advanced Edition Only</small>						
<b>TCP</b>						
TCP-handling	<input type="text" value="on"/>					
<small>*TCP inactive in Personal/Academic Editions</small>						
Queue Size	<input type="text" value="50"/>					
UDP Failover	<input type="text" value="on"/>					



## 4.3 RTP configuration

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



The screenshot shows the Brekeke SIP Server Administration tool interface. The browser title is "Brekeke SIP Server Administration tool - Microsoft Internet Explorer". The address bar shows the URL: `http://127.0.0.1:18080/proxy/gate?bean=sipadmin.web.Configuration&submenu=2`. The main navigation bar includes tabs for "Dial Plan", "User Authentication", "Registered Clients", "Active Sessions", "Call Logs", "Configuration", and "Maintenance". Under the "Configuration" tab, there are sub-tabs for "System", "SIP", "RTP", "Database/Radius", "Password", "Domains", and "Advanced". The "RTP" sub-tab is selected, and the page title is "Configuration > RTP".

The "RTP exchanger" section contains the following configuration fields:

- RTP relay: auto
- RTP relay (UA on this machine): off
- Minimum Port: 10000
- Maximum Port: 10100
- Minimum Port(Video): 0
- Maximum Port(Video): 0
- Port mapping: source port
- Timeout (0=unlimited): RTP Session Timeout (ms): 600000

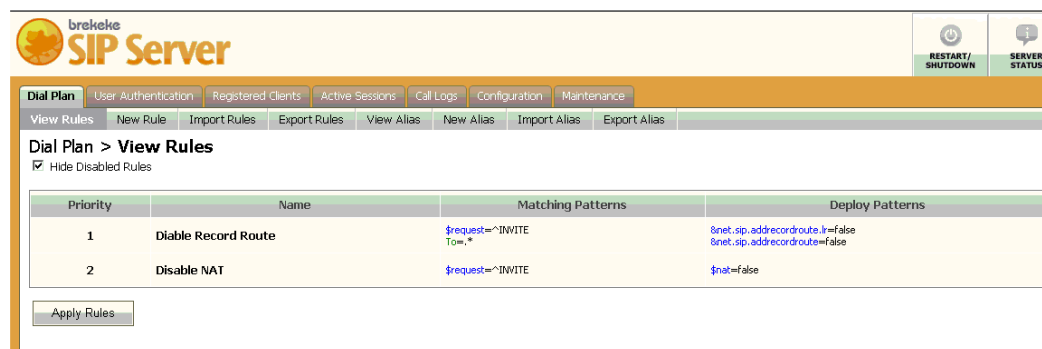
A "Save" button is located below the fields, with the text "Your changes will be in effect after restart." below it.

At the bottom of the page, the following information is displayed:

Brekeke SIP Server , Version 2.2.5.8 Evaluation  
ID: 00093573  
Days until expiration: 3  
Copyright © 2002-2008, Brekeke Software, Inc.

## 4.4 Dial plan

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



The screenshot shows the Brekeke SIP Server Administration tool interface. The browser title is "brekeke SIP Server". The main navigation bar includes tabs for "Dial Plan", "User Authentication", "Registered Clients", "Active Sessions", "Call Logs", "Configuration", and "Maintenance". Under the "Dial Plan" tab, there are sub-tabs for "View Rules", "New Rule", "Import Rules", "Export Rules", "View Alias", "New Alias", "Import Alias", and "Export Alias". The "View Rules" sub-tab is selected, and the page title is "Dial Plan > View Rules".

The "View Rules" section contains a table with the following data:

Priority	Name	Matching Patterns	Deploy Patterns
1	Disable Record Route	<code>\$request=~INVITE To=*</code>	<code>\$net.sip.addrrecordroute=false \$net.sip.addrrecordroute=false</code>
2	Disable NAT	<code>\$request=~INVITE</code>	<code>\$nat=false</code>

An "Apply Rules" button is located below the table.

### 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.

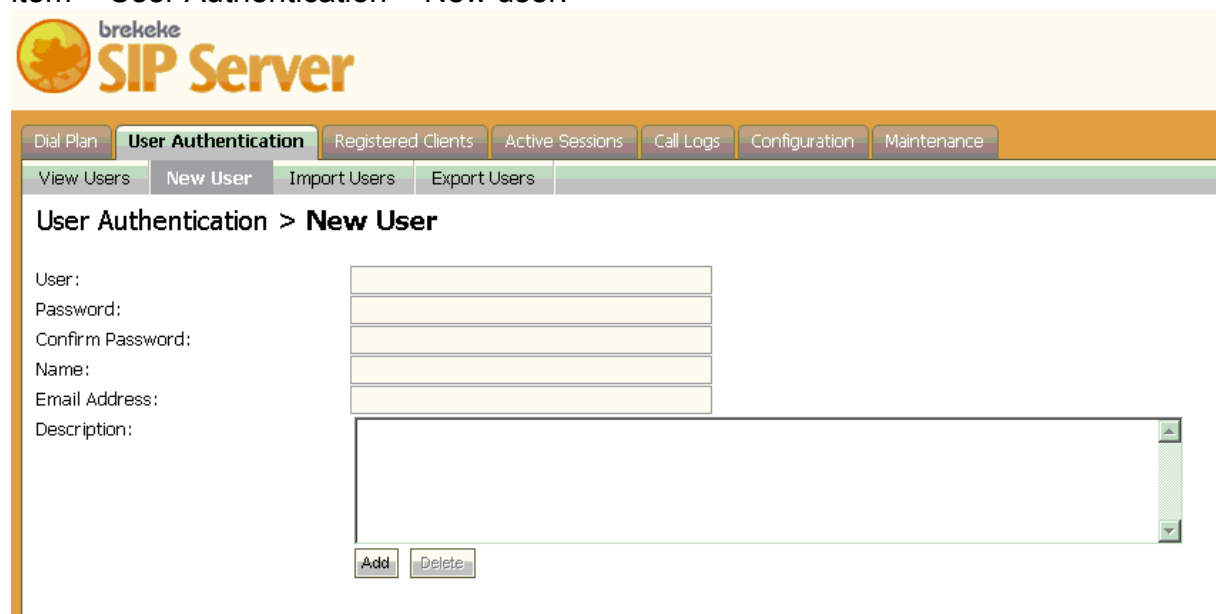
For instance:

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.



The screenshot shows the Brekeke SIP Server web interface. At the top, there is a navigation bar with tabs for 'Dial Plan', 'User Authentication', 'Registered Clients', 'Active Sessions', 'Call Logs', 'Configuration', and 'Maintenance'. Below this, there is a sub-navigation bar with tabs for 'View Users', 'New User', 'Import Users', and 'Export Users'. The main content area is titled 'User Authentication > New User' and contains a form with the following fields: 'User:', 'Password:', 'Confirm Password:', 'Name:', 'Email Address:', and 'Description:'. The 'Description' field is a large text area. At the bottom of the form, there are 'Add' and 'Delete' buttons.

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](mailto: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](mailto:uflatter@mayah.com)  
Tel.: +49 (0) 811 5517-0

### **6.2 Links**

- 1) Brekeke: [www.brekeke.com/download/download\\_sip\\_2\\_0.php](http://www.brekeke.com/download/download_sip_2_0.php)
- 2) LANCOM: [www.lancom-systems.de](http://www.lancom-systems.de)
- 3) Java: [www.java.com/de/download/](http://www.java.com/de/download/)