Brekeke SIP Server

Version 2.1

Administrator's Guide

Brekeke Software, Inc.

Version
Brekeke SIP Server v2.1 Administrator's Guide
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1. Introduction

This document explains the installation and configuration settings of the Brekeke SIP Server.

1.1. Editions

The Brekeke SIP Server comes in several editions to meet the needs of different levels of users: Personal, Academic, Standard, and Advanced.

The Advanced Edition is new in Version 2.1. The Brekeke SIP Server Advanced Edition has the following advantages:

- No Dial Plan restrictions. Academic and Personal Editions are limited to 5 rules.
- Unlimited Multiple Domains. Standard, Academic, and Personal Editions are restricted to 2 domains.
- Multiple Targets Failover.
- Aliases can be managed through the browser-based Administrator GUI.
- User-Agent/Server headers can be changed from "Brekeke SIP Server."

1.2. What is the Brekeke SIP Server?

The Brekeke SIP Server is a SIP Registrar and Proxy Server, it registers and authenticates users, and routes calls between user agents. The product has original NAT traversal (SIP NAT) functionality as well as flexible control routing functions. With Brekeke SIP Server, you can use SIP hardphones, SIP softphones, and SIP-PSTN Gateways for SIP communications.

The Brekeke SIP Server has the following main functions:

♦ **Call Routing**

The Brekeke SIP Server will route SIP requests from a SIP user agent (or other servers) to the most appropriate SIP URI address based on its registrar database. By specifying desired call routing settings in a Dial Plan, you can also prioritize your call routing. If the routing resolves successfully on the server, you can establish a session even when the final SIP URI address is unknown to the caller. Using regular expressions, you can easily create a Dial Plan that will analyze the headers or the IP address of SIP packets to route calls. For example, you can set a prefix for each location with Dial Plan settings. Such settings are especially useful for multi-location office usage of the Brekeke SIP Server

♦ **Registrar**

The Brekeke SIP Server receives REGISTER requests from SIP user agents, and updates its database appropriately. SIP URI in the REGISTER request will be added in the Register database as a user's address. Using the registrar function, you will be able to receive calls from any SIP UA using your unique SIP URI.

- **NAT Traversal**

  When caller and callee are located on different networks, the Brekeke SIP Server can connect calls by rewriting SIP packets appropriately. It is common to have private local IP addresses within a LAN environment, thus NAT traversal service is necessary when a local user is establishing a connection with another user in the global IP network (Internet). Depending upon the situations, Brekeke SIP Server will relay RTP packets to prevent losing voice or media data. The NAT traversal feature on the Brekeke SIP Server supports both Near-End NAT (the server and SIP user agents located within the same firewall) and Far-End NAT (SIP user agents located on the other side of a firewall of a remote network).

- **Upper/Thru Registration**

  Upper/Thru Registration is a unique feature of the Brekeke SIP Server that allows easy configuration of parallel users of pre-existing or other SIP servers. By forwarding REGISTER requests to specified SIP servers, these features allow users to register their SIP user agents at the other SIP server and the Brekeke SIP Server simultaneously. For example, with this feature, users can register their SIP user agents at an ITSP, thus users under the Brekeke SIP Server can talk with other users in the ITSP or receive calls to PSTN.

2. System Requirements

The Brekeke SIP Server supports the following platforms:

OS	Microsoft Windows, Linux, Mac OS X, Solaris
Java	1.4 or later (recommend 1.5)
Memory	At least 256 MB

Note: Apache Tomcat (v.4.1.2 or later) is required for operating systems other than Windows.

3. Installation for Windows OS

3.1. Step 1: Installation of Java SE

Download the Java platform from the following website:

http://java.sun.com/javase/downloads/

Search for Java2, Standard Edition (Java SE) for Windows. Then download and install the latest version of Java SE.

3.2. Step 2: Installation of the Brekeke SIP Server

1. Obtain the installer program from Brekeke Software, Inc. You will receive a Product ID via email.
2. Start the installer.
3. Continue the installation by following the installer's instructions. The Brekeke SIP Server will be installed automatically. If you check the **[Run Brekeke SIP Server]** box at the last stage of the installation and click the **[Finish]** button, the Brekeke SIP Server HTTP service will start automatically.

3.3. Step 3: Starting the Brekeke SIP Server HTTP service

If you did not check **[Run Brekeke SIP Server]** at the last stage of the installation, please start the Brekeke SIP Server HTTP Service by the following steps.

1. Open **[Control Panel]>[Administrative tools]>[Service]**.
2. Select **[Brekeke SIP Server]** and start the service.
3. Restart your computer.

   The Brekeke SIP Server HTTP service will automatically start.

3.4. Step 4: Starting the Brekeke SIP Server Admintool

1. Select **[Start] > [Program] > [Brekeke] > [Brekeke SIP Server] > [Brekeke SIP Server Admintool]**. A web browser will open and you will see the License Agreement page. Enter the Product ID you received via email in the **[Product ID]** field. Click **[I agree]** button and then on **[Activate]** button to activate the License.

Note: You will need to activate the Product ID only when you are freshly installing version 2.x.x.x or upgrading the SIP Server from version.1.5.x.x to the new Brekeke SIP Server v.2.x.x.x. For all other Brekeke SIP Server upgrade from v.2.x.x.y to other v.2.x.a.b do not need Product ID to be activated.

2. At the Admintool Login page, enter User ID and Password and click **[Login]**.

Note: Default Administrator User ID and initial password: User ID = sa Password = sa

3. After the login, click the menu item **[Restart/Shutdown]**. If the Status is **[Active]**, the Brekeke SIP Server has started successfully.

   If the Status is **[Inactive]**, the server has not started successfully. The error should be shown above the Status.

*Note: When the Brekeke SIP Server's port number (default port 5060) is already in use by another application, the Brekeke SIP Server status will be shown as **[inactive]**. For example, if you attempt to start the Brekeke SIP Server while another SIP UA is running on the same machine, the Brekeke SIP Server may fail to start. In this case, please stop the other SIP UA, and click the **[start]** button on the Brekeke SIP Server Admintool's **[Restart/Shutdown]** page.*

4. Installation for Windows, Linux, Mac OS X, and Solaris

4.1. Step 1: Installation of Java SE SDK

Download the Java platform from the following website:

http://java.sun.com/javase/downloads/

Search for Java SE for your OS. Then download and install the latest version of Java SE.

4.2. Step 2: Installation of Tomcat

Download Tomcat can be downloaded from the following website:

http://jakarta.apache.org/tomcat/

Download and install Tomcat version 4.1.2 or later for the type of OS you are running.

4.3. Step 3: Installation of the Brekeke SIP Server

1. Obtain the file *binary file (.war file)* from Brekeke Software, Inc.
2. Copy this file into the directory *webapps*, which is located under the Tomcat installation directory.

4.4. Step 4: Starting Tomcat

1. Start Tomcat
2. Open a web browser and input the URL http://localhost:18080. (If you chose a port number other than "18080" when installing Tomcat. Please change the port number in the URL above to the number specified during installation.)If Tomcat has started successfully, you will see the Apache Jakarta Project page.

4.5. Step 5: Starting the Brekeke SIP Server Admintool

1. Open a web browser and input the URL http://localhost:18080/proxy

   (If you chose a port number other than "18080" when installing Tomcat change the port number in the URL above to the number specified during installation).
2. Then you will see the License Agreement page.
3. Enter the Product ID you received via email in the **[Product ID]** field.
4. Click on **[I agree]** button and then on **[Activate]** button to activate the License.

Note: You will need to activate the Product ID only when you are freshly installing version 2.x.x.x or upgrading the SIP Server from version.1.5.x.x to the new Brekeke SIP Server v.2.x.x.x. For all other Brekeke SIP Server upgrade from v.2.x.x.x do not need Product ID to activate.

5. At the Admintool Login page, enter User ID and Password and click **[Login]**.

Note: Default Administrator User ID and initial password: User ID = sa Password = sa

6. After the login, click the menu item **[Restart/Shutdown]**. If the Status is **[Active]**, the Brekeke SIP Server has started successfully. If the Status is **[Inactive]**, the server hasn't started successfully. The error should be shown above the Status.

*Note: When the Brekeke SIP Server's port number (default port 5060) is already in use by another application, Brekeke SIP Server status will be shown as **[Inactive]**. For example, if you attempt to start the Brekeke SIP Server while another SIP UA is running on the same machine, the Brekeke SIP Server may fail to start. In this case, please stop the other SIP UA, and click the **[start]** button on the Brekeke SIP Server Admintool's **[Restart/Shutdown]** page.*

5. Brekeke SIP Server Administration Tool

5.1. Server Status

The Server Status page shows the version information and current status of the server.

Server Status	
Field Name	**Explanation**
Status	Server's status. If the server is running, the status is "ACTIVE". Otherwise, the status is "INACTIVE".
server-product	Server product name
server-ver	Server's version and revision number
server-name	Server name
server-description	Server description
server-location	Server location
server-startup-time	Time the server was started
server-current-time	Current time
server-life-length	Length of time the server has been running for
machine-name	Name of the machine the server is running on (host name)
listen-port	Server's SIP listen UDP port
multiple-domains	Multiple Domains mode
Interface	Network interface address(es) used by the server
startup-user	User name that started the server
work-directory	The directory path that server is running from
session-active	The number of currently active sessions
session-total	The total number of sessions processed after startup
session-peak	The number of peak sessions
registered-record	Number of registered records
os-name	Name of OS
os-ver	OS version
java-ver	Java version
admin-sip	Administrator's SIP URI
admin-mail	Administrator's e-mail address

Database Status	
Field Name	**Explanation**
registered-database	Status of the connection with Register database
userdir-database	Status of the connection with User Directory database

5.2. Restart / Shutdown

An administrator is able to start or shutdown the Brekeke SIP Server. While the server is running, the word "Active" is displayed. When it is not running, "Inactive" is displayed.

Field Name	Explanation
Status	Server's status If the server is running, the status is "ACTIVE". Otherwise, the status is "INACTIVE".
Listen-Port	Server's SIP listen UDP port
Active Sessions	The number of currently active sessions

Button	Explanation
Start	Starts Brekeke SIP Server
Restart	Restarts Brekeke SIP Server
Shutdown	Stops Brekeke SIP Server. A message to confirm the shutdown command will appear if there are any active sessions. Selecting [Force Shutdown] will terminate all active sessions and shutdown the server.

5.3. Dial Plan

The Dial Plan, which is in the **[Dial Plan]** menu of Brekeke SIP Server Admintool, is for adding and/or editing a Dial Plan rule. In the Advanced Edition, an administrator can manage Alias Database.

Please refer to "Section 6, Dial Plan", for more information about the Dial Plan.

5.3.1. View Rules

The View Rules page shows the list of existing Dial Plan rules. The rule in the higher position in the list has the higher priority. Disabled rules are shown in grey. The buttons at the right side of each rule are for editing, copying and deleting the rule. By pressing the **[Apply Rules]** button, you can apply the new rules or modified rules even when the server is running.

Field Name	Explanation
Priority	Priority of the Dial Plan rule
Name	The name of the Dial Plan rule
Matching Patterns	Defined condition
Deploy Patterns	How the SIP request should be processed

Button	Explanation
Edit	Edit the Dial Plan rule
Copy	Copy the Dial Plan rule
Delete	Delete the Dial Plan rule
Apply Rules	Save and applies changes

5.3.2. New Rule/ Edit Rule

New Rule option helps an administrator to create a new Dial Plan rule. Edit Rule option helps an administrator to modify the Dial Plan rule.

Please refer to “Section 6, Dial Plan”, for information on how to write a Dial Plan rule.

	Item	Default value	Explanation
*	Rule name		Name of the rule You cannot use the existing rule name.
	Description		Description of the rule
*	Priority		Priority of the Dial Plan rule can be set here
	Disabled	Unchecked	When it is checked, the rule is disabled.
*	Matching Patterns		List of Matching Patterns
*	Deploy Patterns		List of Deploy Patterns
	Variable		The name of variable By pressing [⋯], most of the variables are displayed for you to choose from.
	Value		For Matching Patterns, a value of the variable that should match For Deploy Patterns, the value that will be assigned to the variable

(* is a required field.)

Button	Explanation
Insert	Insert the specified definition in [Variable] and [Value] fields into the given list box.
Delete	Delete the selected definition. The deleted definition is displayed in [Variable] and [Value] fields.
Down	Move the selected definition down
Up	Move the selected definition up
Save	Save the Dial Plan rule and return to the [View Rules] page
Cancel	Cancel the changes and return to [View Rules] page

5.3.3. Import Rules

You can import and upload new Dial Plan rules with the Import Rules option.

Select a Dial Plan table file to import Dial Plan rules in the CSV format from [Browse…] button and then click the [Upload] button to upload Dial Plan rules.

5.3.4. Export Rules

You can export the existing Dial Plan rules to another location using the Export Rules option. The rules will be saved in the CSV format.

5.3.5. View Aliases

The View Aliases page shows the list of alias records. The buttons at the right side of each record are for editing, copying and deleting the record.

Note: Alias feature is available in the Advanced Edition only.

Field Name	**Explanation**
Alias Name	Alias name of the record
Group ID	Optional ID for a group of Alias records
Entity Name	Entity name of the record

Button	**Explanation**
Edit	Edit the record
Delete	Delete the record

Filter		
Item	**Default value**	**Explanation**
Pattern		Search keywords
On Field	Alias Name	By: Alias Name, Group ID, Entity Name
Maximum Rows	100	Number of results to display

5.3.6. New Alias/ Edit Alias

New Alias option helps an administrator to create a new alias record. Edit Alias option helps an administrator to modify the alias record.

Note: Alias feature is available in the Advanced Edition only.

	Item	Default value	Explanation
*	Alias		Alias name of the record
	Group ID		Optional ID for a group of Alias records
*	Entity		Entity name of the record

(* is a required field.)

Button	Explanation
Modify	Modify the contents
Add	Add the contents
Delete	Delete the specified user

5.3.7. Import Alias

You can import and upload new alias records with the Import Alias option.

The file format:

```
Alias_Name, [Group_ID], Entity_Name
```

Note: Alias feature is available in the Advanced Edition only.

5.3.8. Export Alias

You can export the existing alias records to another location using the Export Alias option. The records will be saved in the CSV format.

Note: Alias feature is available in the Advanced Edition only.

5.4. User Authentication

The User Authentication, which is in the **[User Authentication]** menu of Brekeke SIP Server Admintool, is for adding and/or editing a user for authentication.

The setting for enabling authentication is at **[Configuration]** page. Refer to Section 5.8.2 "SIP General" for the details.

5.4.1. View Users

The View Users page shows the list of users for authentication. The buttons at the right side of each user are for editing and deleting the user.

Field Name	Explanation
User	User Name for authentication
Name	User's name
Email Address	User's e-mail address
Description	Misc. User information

Button	Explanation
Edit	Edit the user
Delete	Delete the user

Filter		
Item	**Default value**	**Explanation**
Pattern		Search keywords
On Field	User	By:User, Name, Email Address, Description
Maximum Rows	100	Number of results to display

5.4.2. New User/ Edit User

New User option helps an administrator to create a user. Edit User option helps an administrator to modify the user for authentication.

	Item	Default value	Explanation
*	User		User Name for authentication
*	Password		Password
*	Confirm Password		Reenter Password (confirm)
	Name		User's name
	Email Address		User's e-mail address
	Description		Misc. User information

(* is a required field.)

Button	Explanation
Modify	Modify the contents
Add	Add the contents
Delete	Delete the specified user

5.4.3. Import Users

You can import and upload new user information with the Import Users option.

The file format:

```
User, [Password], [Name], [Email Address], [Description]
```

5.4.4. Export Users

You can export the existing user information to another location using the Export Users option. The records will be saved in the CSV format.

5.5. Registered Clients

The Registered Clients, which is in the **[Registered Clients]** menu of Brekeke SIP Server Admintool, is for showing and adding a SIP client.

5.5.1. View Clients

Display the records that are in the registrar database. When the Brekeke SIP Server accepts a REGISTER request from a SIP client, the database is updated automatically. The button at the right side of each record is for deleting the record.

<table>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td>User</td><td>User name</td></tr>
<tr><td>Contact URI</td><td>User's contact SIP URI</td></tr>
<tr><td>Detail</td><td>Details of the registration record
<table>
<tr><th>Variable</th><th>Explanation</th></tr>
<tr><td>Expires</td><td>Expiration period of the record [seconds]</td></tr>
<tr><td>Priority</td><td>Priority of the record (100 - 1000)</td></tr>
<tr><td>User Agent</td><td>Name of the client</td></tr>
<tr><td>Requester</td><td>The IP address that this REGISTER request was sent from.</td></tr>
<tr><td>Time Update</td><td>Timestamp of the latest update of the record</td></tr>
</table></td></tr>
</table>

Button	Explanation
Delete	Delete the registration record

Filter		
Item	**Default value**	**Explanation**
Pattern		Search keywords
On Field	User	By: User, Contact URI, Requester

5.5.2. New Client

New Client option helps an administrator to create a registration record manually.

	Item	Default value	Explanation
*	User		User name that receives a contact from other UAs.
*	Contact URI		User's contact SIP URI
*	Expires	3600	The length in seconds that a record will be stored in the registrar database. Records will be deleted after the specified time passes. While the record is stored in the registrar database, registered users can receive contacts from other SIP UAs through the specified SIP URI that user set up in the "User" setting in Registered List. [seconds]
*	Priority	1000	Priority of the record (100 - 1000).

(* is a required field.)

Button	Explanation
Register	Register a new record in the registrar database.

5.6. Active Sessions

The Active Sessions, which is in the **[Active Sessions]** menu of Brekeke SIP Server Admintool, is for showing active SIP sessions and their details.

5.6.1. Active Sessions

The Active Sessions page shows the list of current active SIP sessions. The buttons at the right side of each session is for showing the details of the session.

<table>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td>Session ID</td><td>Session ID</td></tr>
<tr><td>From</td><td>UAC’s SIP URI and its IP address</td></tr>
<tr><td>To</td><td>UAS’s SIP URI and its IP address</td></tr>
<tr><td>Time</td><td>Session start time</td></tr>
<tr><td>Status</td><td>Session Status
<table>
<tr><th>Status</th><th>Explanation</th></tr>
<tr><td>Initializing</td><td>Session initializing</td></tr>
<tr><td>Inviting</td><td>Sending request</td></tr>
<tr><td>Provisioning</td><td>Preparing for setting up a session</td></tr>
<tr><td>Ringing</td><td>Ringing</td></tr>
<tr><td>Accepted</td><td>Session established</td></tr>
<tr><td>Talking</td><td>Talking</td></tr>
<tr><td>Closing</td><td>Closing a session</td></tr>
</table>
</td></tr>
</table>

Filter		
Item	**Default value**	**Explanation**
From		UAC’s SIP URI or its IP address
To		UAS’s SIP URI or its IP address
Time Range	00:00 to 24:00	Time period
Status	All	Session status

5.6.2. Session Details

The session details page displays detailed information for the selected SIP session.

<table>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td>From-uri</td><td>UAC's SIP URI</td></tr>
<tr><td>From-ip</td><td>UAC's IP address</td></tr>
<tr><td>From-if</td><td>Network interface address of UAC's side</td></tr>
<tr><td>To-uri</td><td>UAS's SIP URI</td></tr>
<tr><td>To-ip</td><td>UAS's IP address</td></tr>
<tr><td>To-if</td><td>Network interface address of UAS's side</td></tr>
<tr><td>Call-ID</td><td>Call-ID</td></tr>
<tr><td>rule</td><td>Dial Plan rules which are applied for the session</td></tr>
<tr><td>plug-in</td><td>Plug-in used to handle the session</td></tr>
<tr><td>sip-packet-total</td><td>Total number of received SIP packets</td></tr>
<tr><td>listen-port</td><td>UDP port number which server uses for send/receive SIP packets</td></tr>
<tr><td>sip-packet-stacked</td><td>Total number of unprocessed SIP packets</td></tr>
<tr><td>session-status</td><td>Session Status
<table>
<tr><th>Status</th><th>Explanation</th></tr>
<tr><td>Initializing</td><td>Session initializing</td></tr>
<tr><td>Inviting</td><td>Sending request</td></tr>
<tr><td>Provisioning</td><td>Preparing for setting up a session</td></tr>
<tr><td>Ringing</td><td>Ringing</td></tr>
<tr><td>Accepted</td><td>Session established</td></tr>
<tr><td>Talking</td><td>Talking</td></tr>
<tr><td>Closing</td><td>Closing a session</td></tr>
</table>
</td></tr>
<tr><td>time-inviting</td><td>Session start time</td></tr>
<tr><td>time-talking</td><td>Talking start time</td></tr>
<tr><td>length-talking</td><td>Length of talking</td></tr>
<tr><td>rtp-relay</td><td>RTP relay status (on/off)</td></tr>
</table>

When RTP relay is enabled, the [rtp-relay] shows “on” and the information below is displayed. This information shows status of RTP streams of both [rtp-srcdst] (UAC to UAS) and [rtp-dstsrc] (UAS to UAC).

Field Name	**Explanation**
media	Media type (audio, video)
transport	Transport type
payload	Payload type
status	Status (active, hold)
listen-port	UDP port number for receiving RTP packets
send-port	UDP port number for sending RTP packets
target	Destination of RTP packets
packet-count	The number of packets
packet/sec	The number of packets per seconds
current size	Packet size (bytes) of RTP packet sent most recently
buffer size	Buffer size (bytes)

Button	**Explanation**
Disconnect	Disconnects the SIP session
Back	Go back to [Active Sessions] page

5.7. Call Logs

The Call Logs, which is in the **[Call Logs]** menu of Brekeke SIP Server Admintool, is for showing the call logs.

5.7.1. Call Logs

The Call Logs page shows the calendar with the number of sessions by date. Please click the desired date to display that date's session log.

Check Box	Explanation
HTML	Clicking a date will display that day’s session log in a new browser window.
CSV	Clicking a date will save that day’s session log in a CSV file.

Button	Explanation
Save	Specify a term to save logs for. Logs older than the specified term will be deleted automatically.

5.7.2. Daily Log

A detailed session log will be displayed in a new window. You can filter the call logs by stating the From-Url to To-Url.

Field Name	Explanation
sid	Session ID
from-url	UAC’s SIP URI
to-url	UAS’s SIP URI
talking-length	Talking time
invite-start-time	Session start time
talk-start-time	Talking start time
end-time	Session end time
result	Result
error	Error Code "-1" indicates a normally ended call. For irregularly ended calls, a SIP error response code will be displayed.

Filter		
Item	**Default value**	**Explanation**
From-Url		UAC's SIP URI
To-Url		UAS's SIP URI
Time Range	00:00 to 24:00	Time period
Maximum Rows	100	Number of results to display

5.8. Configuration

The Configuration, which is in the **[Configuration]** menu of Brekeke SIP Server Admintool, is for editing parameter settings, passwords, and updating the software. Changes will take effect when the server is restarted. "*" is a required field.

5.8.1. System

The System page allows an administrator to configure a system and general network settings.

General			
	Item	**Default value**	**Explanation**
	Server Name	your-sip-sv	Name of the server
	Server Description	your SIP Server	Description for the server
	Server Location	your-place	Location of the server
	Administrator SIP URI	your-sip-uri	Administrator's SIP URI
	Administrator Email Address		Administrator's e-mail address
*	Start up	auto	When "auto" is set, Brekeke SIP Server will automatically start when the web server (Tomcat) is started.

Network			
	Item	**Default value**	**Explanation**
	Interface address 1-5		IP address(es) or FQDNs to be used as interface address(es) by Brekeke SIP Server. IP addresses which can be used as interface addresses are the IP addresses assigned to the Network Interface Cards of the machine where Brekeke SIP Server is installed. In a Windows OS environment, Brekeke SIP Server will automatically get the local IP addresses if not specified here. When the server is located behind a NAT, the global IP address or its FQDN of NAT must be specified here. When the global IP address of your NAT dynamically changes, please set its FQDN.
*	DNS caching period	3600	A period which result of DNS name resolution will be kept. When "-1" is set, the record will be kept forever and the cache will not be refreshed. [seconds]
*	Auto interface discovery	off	When it is set for "on", interface address will be updated automatically.
	External IP address pattern		Regular-expression based IP address pattern which should be treated as an external IP address.

Address Filtering			
	Item	**Default value**	**Explanation**
*	IP address filter	disable	When it is set for “allow”, SIP Server will accept SIP packets only from the IP address specified in the Filter Pattern field. When it is set for “block”, SIP Server will accept all other IP address SIP packets other than the IP address specified in the Filter Pattern field.
	Filter pattern		Specify the desired IP address pattern by regular expressions.

UPnP			
	Item	**Default value**	**Explanation**
*	Enable/Disable	disable	When it is set for “enable”, SIP Server will use UPnP to discover a router, to recognize the global IP address, and to manage a port-forwarding.
	Default router IP address		Default router’s IP address
	Cache size		Size of the internal port-mapping cache table.
	Cache period		Cache period of the port-mapping. When "0" is set, the caching will be disabled. [seconds]
	Refresh Interval		Refreshing interval period of the UPnP. When "0" is set, the refreshing will be disabled. [seconds]

Java		
Item	**Default value**	**Explanation**
Java VM arguments		Specify parameters (excluding classpath) that will be passed to the Java VM.

5.8.2. SIP

Configure SIP settings, NAT traversal, Authentications and various timeout settings. Refer to Section “9. NAT Traversal” for the details of NAT traversal. To add/edit user accounts for authentication, please go to [User Authentication] menu.

SIP exchanger			
	Item	**Default value**	**Explanation**
*	Session Limit	-1	Maximum number of SIP sessions the server will handle concurrently. "-1" specifies an unlimited number of SIP sessions.
*	Local Port	5060	UDP port number to send/receive SIP packets. Please use this default value 5060 if you don’t have any specific reason for changing this port.

NAT traversal			
	Item	**Default value**	**Explanation**
	Keep address/port mapping	on	When set to "on", the Brekeke SIP Server will send keep-alive packets to SIP UAs that are behind NAT at specified intervals. This is so that NAT will not close the external port used by the Brekeke SIP Server to send packets to SIP UAs that are behind NAT.
	Interval	12000	Interval for above setting.　[milliseconds]
	Add ’rport' (Send)	on	When "on" is set , the server adds “rport” in Via header of a outgoing request packet so that the server can detect its own port number.
	Add ’rport' (Receive)	off	When "on" is set , the server adds “rport” with the value of the sender’s source port in Via header of an incoming request packet .

Authentication			
	Item	**Default value**	**Explanation**
	REGISTER	off	When set to "on", the Brekeke SIP Server authenticates REGISTER requests.
	INVITE	off	When set to "on", the Brekeke SIP Server authenticates INVITE requests.
	Realm		This is set as the "realm" value.
	Auth-user=user in "To:" (Register)	no	When set to "yes", the Brekeke SIP Server will authenticate REGISTER requests only when authentication user name matches the user name in the To header. When set to "no", the Brekeke SIP Server will authenticate all REGISTER requests.
	Auth-user=user in "From:"	no	When set to "yes", the Brekeke SIP Server will authenticate requests only when authentication user name matches the user name in the From header. When set to "no", the Brekeke SIP Server will authenticate all requests.
	FQDN only	no	When set to "yes", only SIP URIs that contain an FQDN will be accepted. SIP URIs that contains IP addresses will not be accepted.

Upper Registration			
	Item	**Default value**	**Explanation**
	On/Off	off	Enable/disable Upper Registration
	Register Server		IP address or FQDN of a register server to be used as the Upper Registration destination

Thru Registration			
	Item	**Default value**	**Explanation**
*	On/Off	off	Enable/disable Thru Registration

Timeout			
	Item	**Default value**	**Explanation**
*	Ringing Timeout	120000	Timeout for Ringing [milliseconds]
*	Talking Timeout	259200000	Timeout for talking time [milliseconds]
*	Upper/Thru Timeout	30000	Timeout for waiting the response for a REGISTER request to Upper Registration/Thru Registration destination [milliseconds]

Miscellaneous			
	Item	**Default value**	**Explanation**
*	100 Trying	any requests	When “any requests” is selected, SIP Server will send back 100 Trying response for any initial request. When “only for initial INVITE” is selected, SIP Server will send back 100 Trying response for initial INIVTE request only.
	Server/User-Agent		The specified name will be shown in Server and User-Agent headers. Note: This feature is available in the Advanced Edition only.

5.8.3. RTP

The RTP page allows an administrator to configure RTP settings. If NATs are involved in the SIP communications, Brekeke SIP Server will relay RTP packets so that the RTP packets reach the SIP clients which are behind NAT.

RTP exchanger			
	Item	**Default value**	**Explanation**
*	RTP relay	auto	When set to "on", RTP packets will be handled through the Brekeke SIP Server. When set to "auto", Brekeke SIP Server will decide whether or not to relay RTP automatically. (RTP packets are automatically handled when Brekeke SIP Server does NAT traversal).
*	RTP relay (UA on this machine)	auto	If "auto", the server will decide automatically whether to relay RTP or not. If “off”, Brekeke SIP Server will not relay RTP packets for the clients running on the server machine.
*	Minimum Port	10000	The minimum UDP port number to transmit RTP packets from.
*	Maximum Port	10999	The maximum UDP port number to transmit RTP packets from.
	Minimum Port (Video)	0	The minimum UDP port number to transmit RTP packets for video stream from. If “0”, the server uses the same port range as Audio.
	Maximum Port (Video)	0	The maximum UDP port number to transmit RTP packets for video stream from. If “0”, the server uses the same port range as Audio.
*	Port mapping	source port	Selects a destination port number for the Brekeke SIP Server to send RTP packets to clients behind Far-End NAT. Designates whether to use the source port from RTP packet (when set to “Source Port”) or the RTP port specified in SDP (when set to “sdp”).

Timeout			
	Item	**Default value**	**Explanation**
*	RTP Session Timeout	600000	The timeout for detecting RTP packets when relaying RTP. [milliseconds]

5.8.4. Database

The Database page allows an administrator to configure database settings.

Here is the list of the databases which the server uses.

Database Name	Purpose
Registered Database	Registered Table This table stores the data of registered user agents. The data will be updated by REGISTER requests and used for the session routing.
Users Database	Users Table This table stores authentication data of users.
Alias Database	Alias Table This table stores alias data. Note: Alias Database is available in Advanced Edition only.

Each database can use Embedded or Third-Party database system. Please refer to "Using a Third-Party Database Tutorial" for more information about using of Third-Party database system.

Embedded Database			
	Item	**Default value**	**Explanation**
	Port Number	9001	TCP port number used by the Embedded database system. If no port is specified, TCP port 9001 is used by default.

Thirdparty Registered Database			
	Item	**Default value**	**Explanation**
*	On/Off	Off	Enable or disable to use the third party database system for Registered Database.
	Registered Database URL		URL for the Registered Database (ex. jdbc:mysql://localhost/db)
	Registered Database Driver		JDBC Driver for the Registered Database. (ex. com.mysql.jdbc.Driver)
	User Name		User name for the Registered Database.
	Password		Password for the Registered Database.

Thirdparty Users Database			
	Item	**Default value**	**Explanation**
*	On/Off	Off	Enable or disable to use the third party database system for Users Database.
*	Encrypt Users Passwords	true	Enable or disable the user password encryption.
	Users Database URL		URL for the Users Database (ex. jdbc:mysql://localhost/db)
	Users Database Driver		JDBC Driver for the Users Database. (ex. com.mysql.jdbc.Driver)
	User Name		User name for the Users Database.
	Password		Password for the Users Database.

Thirdparty Alias Database			
	Item	**Default value**	**Explanation**
*	On/Off	Off	Enable or disable to use the third party database system for Alias Database.
	Alias Database URL		URL for the Alias Database (ex. jdbc:mysql://localhost/db)
	Alias Database Driver		JDBC Driver for the Alias Database. (ex. com.mysql.jdbc.Driver)
	User Name		User name for the Alias Database.
	Password		Password for the Alias Database.

Note: Alias Database is available in Advanced Edition only.

5.8.5. Password

The Password page allows an administrator to change the login password for the Brekeke SIP Server Admintool. Administrator's default user id is "sa" and its password is "sa".

To set the password for authenticating SIP requests, please use **[User Authentication]** page and refer to the section 5.4 "User Authentication".

5.8.6. Domains

The Domains page allows an administrator to manage multiple domains. With the Multiple Domains Mode, Brekeke SIP Server can host multiple domains on one server.

The buttons at the right side of each domain are for editing and deleting the domain.

	Item	**Default value**	**Explanation**
*	Multiple Domains mode	Unchecked	When it is checked, the server can host multiple domains.

<table>
<tr><th>Field Name</th><th>Explanation</th></tr>
<tr><td>Domain</td><td>Name of the domain</td></tr>
<tr><td>Authentication</td><td>Authentication policy
<table>
<tr><th>policy</th><th>Explanation</th></tr>
<tr><td>Realm</td><td>This is set as the "realm" value.</td></tr>
<tr><td>REGISTER</td><td>If it is “on”, the server authenticates REGISTER requests.</td></tr>
<tr><td>INVITE</td><td>If it is “on”, the server authenticates REGISTER requests</td></tr>
<tr><td>Auth-user=user in To (REGISTER)</td><td>The server will authenticate REGISTER requests only when authentication user name matches the user name in the To header.</td></tr>
<tr><td>Auth-user=user in From</td><td>The server will authenticate requests only when authentication user name matches the user name in the From header.</td></tr>
</table>
</td></tr>
</table>

5.8.7. New Domain/ Edit Domain

New Domain option helps an administrator to add a new domain. Edit Domain option helps an administrator to modify the domain.

	Item	Default value	Explanation
*	Domain		Name of the domain
	Disabled	Unchecked	When it is checked, the domain is disabled.
*	Admin-Password		The password for the domain
	Realm		The realm for the domain
*	Authentication		
*	REGISTER	off	When set to "on", the Brekeke SIP Server authenticates REGISTER requests.
*	INVITE	off	When set to "on", the Brekeke SIP Server authenticates INVITE requests.
*	Auth-user=user in To (REGISTER)	off	When set to "yes", the Brekeke SIP Server will authenticate REGISTER requests only when authentication user name matches the user name in the To header. When set to "no", the Brekeke SIP Server will authenticate all REGISTER requests.
*	Auth-user=user in From	off	When set to "yes", the Brekeke SIP Server will authenticate requests only when authentication user name matches the user name in the From header. When set to "no", the Brekeke SIP Server will authenticate all requests.

(* is a required field.)

5.8.8. Advanced

The Advanced page allows an administrator to edit property variables.

5.9. Maintenance

The Maintenance, which is in the **[Maintenance]** menu of Brekeke SIP Server Admintool, is for updating the software and for activating the license.

5.9.1. Back Up

You can back-up the existing settings using the Back Up option. The settings will be saved in the sst file.

5.9.2. Restore

With the Restore option, you can restore the backuped settings from the sst file.

5.9.3. Update Software

This page is for updating the Brekeke SIP Server. Please specify an update file and push [Upload] button. After updating the software, please restart the computer.

5.9.4. Activate License

This page is for updating the Brekeke SIP Server Product ID.

6. Dial Plan

6.1. What is the Dial Plan?

The Brekeke SIP Server's Dial Plan defines rules for routing. The Dial Plan can also be used for setting up filtering by defining appropriate conditions, setting environment variables and modifications of selected SIP headers. Regular expressions are used for defining those rules.

This document provides a reference of the Dial Plan functions. For more detailed information please refer to the *Brekeke SIP Server Tutorial-Dial Plan*.

By setting a Dial Plan, you can achieve the following functions:

- Routing
- Filtering
- Modifications (add/delete/replace) of SIP headers
- Load Balancing
- Setting the server's environment variables
- RTP relay settings
- Load Session Plug-ins
- Load Dial Plan Plug-ins

6.2. Edit

To edit the Dial Plan, open **[Dial Plan]** menu from the Admintool. Please refer to the section "5.3. Dial Plan" for more details.

You can also edit Dial Plan files directly. Your changes will be in effect after you restart Brekeke SIP Server. Dial Plan file is located under install directly:

```
<INSTALL_DIR>\webapps\proxy\WEB-INF\work\sv\etc\dialplan.tbl
```

6.3. Syntax

When all conditions set in the Matching Patterns are satisfied, the actions defined in the Deploy Patterns are applied.

6.3.1. Matching Patterns

Define conditions for applying the rule.

Conditions can be defined using a pair of the following: the name of the SIP header, condition functions, system environment variables, source IP address, or the source port number, and the string pattern for matching. By defining multiple pairs, you can make the conditions more specific. Regular Expressions are used for defining string matching patterns. The string between brackets () in the right side can be referred to in Matching Patterns and Deploy Patterns.

Matching Patterns Syntax:

SIP_header_field = string pattern

&environment_variable_name = string pattern

$condition_function_name = string pattern

$condition_function_name(arguments) = string pattern

Main regular expressions which can be used in Matching Pattern's right side are as follows:

Symbols	Meaning
`!`	If '!' is placed in the front of the string pattern, it means NOT.
`^`	Match the beginning of the line
`$`	Match the end of the line
`[abc]`	Match any character listed between brackets. In this case, a or b or c.
`[^abc]`	Math any character except those listed between the brackets. In this case, any characters except a, b and c.
`.`	Match any character except new line
`X+`	Match the preceding element (X, in this case) one or more times
`X*`	Match the preceding element (X, in this case) zero or more times
`X{n}`	Match the preceding element (X, in this case) n times
`X{n,}`	Match the preceding element (X, in this case) at least n times
`X{n,m}`	Match the preceding element (X, in this case) at least n times, but no more than m times
`(chars)`	The characters between the brackets will be put in a buffer. To refer to the n-th digit buffer in Deploy Pattern, use %<digit> (for example %1)

To add a condition in the Matching Patterns section, click **[…]** button (which is between the Variable field and the Value field) and select a variable name from the pull-down list or type a variable name directly in the Variable field. Type a string pattern to the Value field and then, click the **[+]** button. Refer to Section 5.3.2, "Edit Dial Plan" for more information.

1) SIP Header Field Name

To use a SIP header as a condition, specify a pair of a SIP header name and a string pattern.

Syntax:

SIP header field name = a string pattern

Example:

`From = sip:user@domain/.com[>;]*`

If the SIP URI in From: header is "sip:user@domain.com"

`To = sip:11@`

If the SIP user name in To: header field is "11"

`To = sip:9(.+)@`

If the SIP user name in To: header field starts with 9

`To = sip :(....)@`

If the SIP user name in To: header field contains only 4 characters

`Supported = timer`

If Supported: header field contains the string "timer",

`Expires = ^[0-5]$`

If the value of Expires: header field is in the range 0-5

`Contact = sip:[A-Za-z]+@`

If the user name in Contact header contains only alphabet

2) Environment Variable

The environment variable is a variable name which starts with ‘&’. The variable name is not case sensitive.

Syntax:

```
&variable_name = a string pattern
```

Example:

`&sv.name = ^main-sv$`

If the value of the server name (Environment variable: sv.name) is “main-sv”.

`&net.sip.timeout.ringing = ^5[0-9][0-9][0-9]$`

If the value of Ringing Timeout (Environment variable: net.sip.timeout.ringing）is in the range 5000-5999.

3) Conditional Function

The variable that starts with ‘$’ is treated as a conditional function. The variable name is not case sensitive. Some conditional functions can have an argument.

Syntax:

```
$conditional_function_name = a string pattern
$conditional_function_name(argument) = a string pattern
```

How to call functions:

Function_name(SIP header field name)

If a SIP header field is set as an argument, the value of the SIP header field will be passed to the function.

Example: `$func( From )`

The value of From: header will be passed to the function “func”.

Function_name(&Environment_variable_name)

If an environment variable is set as an argument of a function, the corresponding value of the variable will be passed to the function. The prefix ‘&’ should be added before an environment variable name.

- Environment variable can be set at Dial Plan’s Deploy Pattern or in the property file.

Example: `$func(&net.sip.timeout.ringing )`

The value of environment variable net.sip.timeout.ringing will be passed to the function “func”.

Function_name($Conditional_function_name)

If a conditional function name is set as an argument of a function, the return value of the conditional function will be passed to the function which called the conditional function. The prefix ‘$’ should be added before a conditional function name.

Example: `$func1( $func2 )`

The return value of the function “func2” will be passed to the function “func1”.

Example: `$func1( $func2( $func3 ) )`

The return value of the function “func3” will be passed to the function “func2” and the return value of the function “func2” will be passed to the function “func1”.

Example: `$func( $func( To ) )`

The contents of To: header field will be passed to the function “func” and its return value will be passed to the function “func” again.

Function_name(“Text_String”)

If a text string is set as an argument, the text string is passed to the function. The text string should be enclosed in double quotes.

Example: `$func( “string” )`

The string “string” will be passed to the function “func”.

$addr

Meaning:

Source IP address

Syntax:

`$addr`

Explanation:

Returns the source IP address of the packet.

Example:

`$addr = ^127\.0\.0\.1$`

If the source IP address of the packet is the loopback address (127.0.0.1).

`$addr = ^192\.168\.`

If the source IP address of the packet starts with “192.168.”.

`$addr = ^172\.16\.0\.[1-5]$`

If the source IP address is in the range 172.16.0.1-172.16.0.5.

$body

Meaning:

Match in the message body

Syntax:

`$body( regex )`

regex − regular expression

Explanation:

Gets the matched string from the message body such as SDP. The regular expression should contain a pair of brackets for defining the matched string.

Example:

`$body( "m=audio (.+) RTP/AVP" ) = ^2000$`

If the audio RTP port is 2000.

$date

Meaning:

Current Year/Month/Date

Syntax:

`$date`

`$date( format )`

format − Date format

`$date( format, timezone )`

format − Date format

timezone − Time Zone

Explanation:

Returns the text string of current year/month/date.

Date format should be specified as an argument. The default format is `”yyyy/MM/dd”.`

Date format can consist of the following characters.

Character	Meaning		Character	Meaning
y	Year		m	Minute
M	Month		s	Second
d	Day		S	Millisecond
H	Hour			

Example:

`$date = 2010/06/03`

If the date is June 3rd, 2010.

`$date = [15]$`

If the last digit of the day is 1 or 5, i.e. the day of the month is 1,5,11,15, 21, 25, 31.

`$date( “MM-dd-yyyy” ) = 06-03-2010`

Gets the current date with the format “MM-dd-yyyy“ and compares it with the string “06-03-2010”.

`$date( “yyyy/MM/dd”, " JST" ) = (.+)`

Gets the current date based on the time zone “JST”.

$geturi

Meaning:

Get the string of the SIP URI

Syntax:

`$geturi( str )`

str − text string

Explanation:

Gets the SIP URI part from the specified string.

Example:

`$geturi( From ) = sip:user@domain/.com$`

Gets the SIP URI part from From header and compares with”sip:user@domain.com”.

This condition has the same meaning as the following condition.

`From = sip:user@domain/.com[>;]*`

`$geturi( &sv.admin.sip ) = sip:admin@host$`

Gets the SIP URI part from the value of the environment variable sv.admin.sip and compare it with the text string ”sip:admin@host”.

`$geturi( $request ) = sip:1234@192/.168/.0/.1$`

Gets the SIP URI part from the request-line (the return value of the conditional function ”$request”) and compare it with the string ”sip:1234@192.168.0.1”.

$globaladdr

Meaning:

If global address or not

Syntax:

`$globaladdr( str )`

str −IP address or FQDN

Explanation:

Checks if the address or FQDN specified as an argument is a global address or not.

If it is a global address, “true” will be returned. If not, “false” will be returned.

Example:

`$globaladdr( “192.168.0.200” ) = false`

If 192.168.0.200 is not a global address.

$headerparam

Meaning:

The header parameter

Syntax:

`$headerparam( string )`

str − string

`$headerparam( string, key )`

str − string

key − header parameter variable name

Explanation:

Returns the value of the header parameter variable from the specified string.

Example:

`$headerparam( Contact )= (.+)`

Get all header parameters from Route header.

`$headerparam( To, "para" ) = (.+)`

Get the “para”’s value from To header’s header parameters.

It is the same as `$param($headerparam( To ),“para”)`

$istalking

Meaning:

If talking or not

Syntax:

`$istalking`

`$istalking( str )`

str − SIP URI

Explanation:

Checks if the SIP URI specified as an argument is talking or not.

If it is talking, “true” will be returned. If not, “false” will be returned.

If no argument is set, Brekeke SIP Server checks if the Request URI is talking or not.

Example:

`$istalking = true`

If the request URI is talking.

`$istalking( “sip:user@192.168.0.2” ) = true`

If the sip:user@192.168.0.2 is talking.

$localhost

Meaning:

If localhost or not

Syntax:

`$localhost`

`$localhost( str )`

str – SIP URI or IP address or FQDN

Explanation:

Checks if the SIP URI or address specified as an argument is the localhost or not.

If it is localhost, "true" will be returned. If not, "false" will be returned.

If no argument is specified, Brekeke SIP Server checks if the source IP address of the packet is localhost or not.

- The addresses set in network interface settings in [Configuration] page will also be treated as "localhost".

Example:

`$localhost = true`

If the source of the packet is localhost

`$localhost( $addr ) = true`

If the source of the packet is localhost.（This is same as the case you didn't specify any argument.）

`$localhost( From ) = false`

If the SIP URI in From header is not localhost

`$localhost( "192.168.0.100" ) = true`

If 192.168.0.1 is localhost

$mydomain

Meaning:

If my domain or not

Syntax:

`$mydomain( str )`

str – domain name

Explanation:

Checks if the domain name specified as an argument is hosted by this server or not.

If it is my domain, “true” will be returned. If not, “false” will be returned.

The domain hosted by the server should be listed in the Domains page. Please refer to the section “5.8.6. Domains” for more details.

Example:

`$mydomain( “sip.domain.com” ) = true`

If the “sip.domain.com” is hosted by this server.

$outbound

Meaning:

If outbound or not

Syntax:

`$outbound`

`$outbound( str )`

str – SIP URI or IP address or FQDN

Explanation:

Checks if the SIP URI or address set as an argument is outbound (IP address/port number which is not Brekeke SIP Server’s IP address/port) or not.

If it is outbound, “true” will be returned. If not, “false” will be returned.

If no argument is set, Brekeke SIP Server checks if the Request URI is outbound or not.

For example, if Brekeke SIP Server’s IP address is 192.168.0.1:5060, the IP address 192.168.0.2 or 192.168.0.1:6060 is considered as “outbound”.

Example:

`$outbound = true`

If the request URI contains an outbound address

`$outbound( $request) = true`

If the request URI contains an outbound address. (This is same as the case you didn’t specify any argument.)

`$outbound( To ) = false`

If the SIP URI in To header is not outbound.

$outbound (“sip:user@host”) = true

If “host” is outbound

$param

Meaning:

The parameter value

Syntax:

`$param( str, key )`

str − string

key − parameter variable name

Explanation:

Returns the value of the parameter variable from the specified string.

Example:

$param("sip:bob@192.168.0.1;expires=3600; q=1.0”,“expires”)= ^300$

If the expires’s value is 300.

$param(Via, "branch") = (.+)

Get the branch’s value.

$port

Meaning:

Source port of the incoming SIP packet

Syntax:

`$port`

Explanation:

Returns the source port number of the packet.

Example:

$port = ^5060$

If the source port number of the packet is 5060.

$port = ^50[0-9][0-9]$

If the source port number of the packet is in the range 5000-5099.

$registered

Meaning:

If registered or not

Syntax:

```
$registered
$registered( str )
```

str − SIP URI or a user name

Explanation:

Checks the SIP URI or the user name specified as an argument is registered in the Brekeke SIP Server's register database.

If the corresponding user is registered, "true" will be returned. If not, "false" is returned.

If no argument is specified, Brekeke SIP Server checks if the user in the request URI is registered or not.

Example:

`$registered = true`

If the user in the request URI is registered.

`$registered( $request ) = true`

If the user in the request URI is registered. (This is same as the case you didn't specify any argument.)

`$registered( From ) = true`

If the caller (The user in From header) is registered.

`$registered( "user" ) = false`

If the user "user" is not registered.

$registeredaddr

Meaning:

See $regaddr.

$registereduri

Meaning:

See $reguri.

$regaddr

Meaning:

The contact IP address registered in Brekeke SIP Server's register database

Syntax:

`$regaddr`

`$regaddr( str )`

str – SIP URI or a user name

Explanation:

Returns the contact IP address registered in the register database for the SIP URI or user name specified as an argument. If no argument is specified, the registered IP address for the user in the request URI will be returned.

If any corresponding record can not be found, the condition will not be fulfilled.

Example:

`$regaddr = ^192\.168\.0\.1$`

If the user in the request URI is registered from the IP address 192.168.0.1.

`$regaddr( From ) = ^192\.168\.0\.200$`

If the caller (the user in From header) is registered from the IP address 192.168.0.200.

`$regaddr( “user” ) = ^192\.168\.0\.`

If the user ”user” registered from the IP address 192.168.0.x.

$reguri

Meaning:

Contact SIP URI registered in the Brekeke SIP Server's register database

Syntax:

`$reguri`

`$reguri( str )`

str – SIP URI or a user name

Explanation:

Returns the contact SIP URI registered in the register database for the SIP URI or a user name specified as an argument. If no argument is specified, the registered contact SIP URI for the user in the request URI will be returned.

If any corresponding user can not be found, this condition will not be fulfilled.

Example:

`$reguri = sip:100@host`

If the user's contact SIP URI in the request URI is "100@host".

`$reguri( "user" ) = sip:admin@`

If the user "user"' s contact SIP URI's user part is "admin".

$request

Meaning:

SIP request Line

Syntax:

`$request`

Explanation:

Returns the SIP request line in the packet.

Example:

`$request = sip:100@host`

If the request URI is "sip:100@host".

`$request = ^INVITE`

If the request is INVITE.

$sid

Meaning:

A session ID

Syntax:

`$sid`

Explanation:

Returns the session ID.

Session ID is a unique number assigned to each session.

Example:

`$sid = ^100$`

If the session ID is 100.

`$sid = [02468]$`

If the session id is an even number.

$sessionnum

Meaning:

The number of current sessions

Syntax:

```
$sessionnum
```

Explanation:

Returns the number of current sessions.

Example:

`$sessionnum = ^1000$`

If the number of current sessions reaches 1000.

$soapget

Meaning:

Match in the message body

Syntax:

```
$soapget( http-uri, namespace, method [,param [,param..]] )
```

http-uri − web site's address

namespace− name space

method − method name

param − input parameter

Explanation:

Gets the information from the web service by SOAP.

Note: This method is available in Advanced Edition only.

Example:

`$soapget(“http://192.168.0.1”,”ns”,"getUserProperty","in0=4002", "in1=Email" ) = (.+)`

$subparam

Meaning:

The subscriber parameter

Syntax:

```
$subparam( str )
```

str − string

```
$subparam( str, key )
```

str − string

key − subscriber parameter variable name

Explanation:

Returns the value of the subscriber parameter variable from the specified string.

Example:

`$subparam( To )= (.+)`

Get all subscriber parameters from To header.

`$subparam(“sip:user;para=1@foo.com”, “para” ) = ^test$`

If “para”’ s value is test.

It is the same as `$param($subparam(“sip:user;para=1@foo.com"),“para”)`

$time

Meaning:

Current time

Syntax:

`$time`

`$time( format )`

format − Time format

`$time( format, timezone )`

format − Time format

timezone − Time Zone

Explanation:

Returns current time.

Time format should be specified as an argument. The default format is “HH:mm:ss“. For the details of the format, please refer to the part “$date”.

Example:

`$time = 09:26:40`

If current time is 09:26:40.

`$time = ^0[0-9]:`

If current time is from 0 to 9 o’clock.

`$time( “SSSS” ) = [02468]$`

If the millisecond is an even number.

`$time( “HH:mm:ss”, "PDT" ) = (.+)`

Get the current time based on the time zone “PDT”.

$uriparam

Meaning:

The URI parameter

Syntax:

`$uriparam( str )`

str − string

`$uriparam( str, key )`

str − string

key − URI parameter variable name

Explanation:

Returns the value of the URI parameter variable from the specified string.

Example:

`$uriparam( $request )= (.+)`

Get all URI parameters from the request URI.

`$uriparam( To, "para" ) = (.+)`

Get the “para”’ s value from To header’ s URI parameters.

It is the same as `$param($uriparam( To ),“para”)`

$webget

Meaning:

Match in the web page

Syntax:

`$webget( http-uri, regex )`

http-uri − web site’ s address

regex − regular expression

Explanation:

Gets the matched string from the specified web site. The regular expression should contain a pair of brackets for defining the matched string.

Note: This method is available in Advanced Edition only.

Example:

`$webget( "http://www.foo.com/", "<B>(.+)</B>" ) = (.+)`

Get the string enclosed with <B> and </B> from the specified web site.

6.3.2. Deploy Patterns

The Deploy Patterns field defines actions that will be taken when a rule's conditions defined in in the Matching Pattern are met. At Deploy Patterns, you can define SIP header, routing destination IP address, environment variables, plug-in to load, and whether to perform RTP relay or not. Action is defined with a pair of "Handling variable name", SIP header name or environment variable and "value". You can define multiple actions in one rule.

In the Value field, matched string in Matching patterns can be used to define Deploy Patterns. When '%n' (n=numbers) was defined in value, the character string that locates in "n"th number of parenthesis ( ) in Matching Patterns will be inserted at the Deploy Patterns field.

Deploy Patterns Syntax:

SIP_header_field = a setting value
&environment_variable_name = a setting value
$handling_variable_name = a setting value

To add a definition to the Deploy Patterns field, push the **[...]** button between the Variable and Value fields. A drop-down menu will appear in the Variable field. Select a variable from the menu or type variable name for Variable field, and type its value in the Value field. To complete the steps, press the **[+]** button.

1) SIP Header Field Name

By specifying a SIP header field name in variable field, you can replace, add or delete the value of the SIP header. If the specified SIP header field exists in a SIP packet, Brekeke SIP Server will replace the value of the header to the specified value. If setting value is empty, the SIP header will be removed from the SIP packet.

The SIP routing destination will be decided depending on the setting for the SIP header field "To" as follows:

If **To = sip:username@host** is set,
the sip session will be routed to the address "host".

If **To = sip:username@** is set,

the sip session will be routed to the contact address for the registered user “username” in in the server’s register database.

Syntax:

SIP header field name = setting value

Example:

From = sip:admin@192.168.0.1

From: header will be replaced with “sip:admin@192.168.0.1”.

To = sip:boss@192.168.0.100

To: header will be replaced with “sip:boss@192.168.0.100”. The session will be routed to the address “192.168.0.100”.

To = sip:sales@

The session will be routed to the contact address of the registered user “sales”.

From = “Ted” <sip:1650111@domain>

From: header’s SIP URI will be replaced with <sip:1650111@domain>. Caller’s display name will be set as “Ted”.

Expires = 300

The value of Expires: will be set as 300.

User-Agent =

User-Agent: header will be deleted.

Refer-To = sip:user@server

Refer-To: header field will be replaced with “user@server”.

2) Environment Variable

The variable which starts with ‘&’ is treated as an environment variable. The environment variable name isn’t case sensitive.

This setting will be applied only for the session that matches with matching patterns. To configure the environment variables for the whole system, please set them in the property file or in the Configuration page.

Syntax:

&environment_variable_name = a setting value

Example:

`&net.sip.timeout.ringing = 10000`

Set the value of ringing timeout `to 10000.`

(Set the environment variable `net.sip.timeout.ringing = 10000`)

`&net.sip.addrecordroute = false`

Don't add Record-Route: header.

(Set the environment `variable net.sip.addrecordroute = false`)

`&net.rtp.audio.payloadtype = 0`

Change the audio payload type in SDP to PCMU.

(Set the environment `variable net.rtp.audio.payloadtype = 0`)

3) Handling Variable

The variable which starts with '$' is treated as a handling variable. Handling variables are not case sensitive.

Syntax:

$handling_variable_name = a setting value

$action

Meaning:

Response to send

Syntax:

`$action` = SIP response number

Explanation:

This sets a SIP response number for a specified request.

If a response is returned to the request sender, the request will not be routed to the request destination.

Example:

`$action = 200`

Returns the response 200 OK.

`$action = 603`

Returns the response 603 Decline.

$auth

Meaning:

Whether to authenticate or not

Syntax:

`$auth` = true or false

Explanation:

This sets whether to authenticate the request or not.

If "true", the authentication will be enabled. If "false", the authentication will be disabled. The default value is the value which is set in [Configuration] page.

Example:

`$auth = true`

Authenticate the request

$continue

Meaning:

Whether Brekeke SIP Server continues checking the rule or not.

Syntax:

`$continue` = true or false

Explanation:

This is a variable to make the server handle multiple rules.

If "true", Brekeke SIP Server continues to check the next rule below.

If "false", Brekeke SIP Server will not continue checking the next rules. The default is "false".

As long as the Matching Patterns conditions are fulfilled and Deploy Patterns contains $continue=true, Brekeke SIP Server continues checking rules.

Example:

`$continue = true`

Continues checking the next rule.

$ifdst

Meaning:

Interface address used for sending/receiving packets to/from the session destination

Syntax:

`$ifdst` = IP address or FQDN

Explanation:

Brekeke SIP Server's interface address used for sending/receiving the packets to/from the session destination (callee side).

This address is used for the values in Via, Record-Route headers.

Example:

`$ifdst = 192.168.0.100`

Set 192.168.0.100 as an interface address for the sending packets to the session destination.

$ifsrc

Meaning:

Interface address for sending/receiving the packets to/from the session originator.

Syntax:

`$ifsrc` = IP address or FQDN

Explanation:

Brekeke SIP Server's interface address used for sending/receiving the packets to/from the session originator (caller side).

This address is used for the values in Via:, Record-Route: headers.

Example:

`$ifsrc = 192.168.1.200`

Sets 192.168.1.200 as a interface address for communicating with the caller side.

$nat

Meaning:

Whether to handle NAT traversal

Syntax:

`$nat` = true or false

Explanation:

Whether to handle NAT traversal or not.

If "true", NAT traversal will be handled. If false, NAT traversal will not be handled. If "auto", Brekeke SIP Server will automatically decides whether to handle NAT traversal.

The default value is "auto".

If this NAT traversal feature is enabled, RTP relay (the variable $rtp) will also be enabled.

If this NAT traversal feature is disabled, Brekeke SIP Server will not handle NAT traversal even in the case NAT traversal should be necessary.

Example:

`$nat = true`

Handle NAT traversal.

$replaceuri

Meaning:

Whether to replace From and To header to appropriate addresses

Syntax:

`$replaceuri` = true or false

Explanation:

Sets whether to replace From and To headers to appropriate addresses.

If "true", it is enabled. If "false", it is disabled. If "auto", Brekeke SIP Server will decide whether to replace the headers or not automatically.

The default value is "auto".

For example, if this feature is enabled, Brekeke SIP Server will not include local IP addresses in the packets sent to outside of the NAT.

Example:

`$replaceuri = false`

From and To header will not be replaced.

$rtp

Meaning:

Whether to relay RTP packets

Syntax:

`$rtp` = true or false

Explanation:

Sets whether the server relay RTP packets.

If "true", RTP packets will be relayed through Brekeke SIP Server. If "false", RTP packets will not be relayed through Brekeke SIP Server. If "auto", Brekeke SIP Server will decide whether to relay RTP packets or not automatically (For example, Brekeke SIP Server relays RTP packets for the UAs behind NAT). The default value is the value set in [Configuration] page.

Example:

`$rtp = true`

Enable RTP relay.

$session

Meaning:

Load a session plug-in.

Syntax:

`$session` = a session plug-in name

Explanation:

Specifies the name of session plug-in to use.

Example:

`$session = com.sample.radius.proxy.RadiusAcct`

Set the com.sample.radius.proxy.RadiusAcct class as a session plug-in.

$target

Meaning:

Routing destination

Syntax:

`$target` = IP address or FQDN

Explanation:

Sets the session’s routing destination.

Example:

`$target = provider.domain`

Routes the session to provider.domain.

7. Upper Registration

Upper Registration is a function that all SIP user agents send REGISTER requests to the registrar server (upper server) specified in the Register Server (IP or FQDN) field via the Brekeke SIP Server. Using this feature, SIP user agents will be registered at the upper server, and users can receive calls from the upper server with simple settings. Since users can specify the upper server's address at the Brekeke SIP Server, there are no special settings necessary at users' SIP user agents.

Note: Please note that using Thru Registration feature, users need to set up the upper server at their SIP UAs.

To activate Upper Registration, please use the following settings:

1. In the **[Configuration]** page > **[SIP]**, set Upper Registration as follows:

Item	Setting value	Explanation
On/Off	on	Enable Upper Registration
Register Server	The address of the other register server	Specify an IP address or FQDN as the Upper Registration destination

2. Client Set up

Item	Setting value
SIP proxy server	Brekeke SIP Server's IP address
Registrar	Brekeke SIP Server's IP address
Outbound Proxy	Brekeke SIP Server's IP address
User Name	When authentication is set at the upper server, set the user name that is assigned by the upper server here.
Password	When authentication is set at the upper server, set the password that is assigned by the upper server here.

8. Thru Registration

Thru Registration is the function to forward REGISTER requests to the register server (upper server) specified in request URI through the Brekeke SIP Server. Using this feature, SIP UA will be registered with the upper server, and users can receive calls from the upper server. Since the Thru Registration feature requires setup of the other SIP proxy server on each SIP UAs, each SIP UA can register at different servers.

Note: Please note that using Upper Registration feature, users can specify only one upper server at the Brekeke SIP Server, where as with Thru Registration, users can set up different upper servers at their SIP UA settings.

Please use the following settings for Thru Registration:

1. In the **[Configuration]** page> **[SIP(Advanced)]**, set Thru Registration as follows:

Item	Value	Explanation
On/Off	on	Enable Thru Registration

2. Client Set up

Item	Setting value
SIP proxy server	Brekeke SIP Server’s IP address In the case where “Outbound Proxy” setting is available, you would need to set the upper server’s address here.
Registrar	Register server’s address (as upper server)
Outbound Proxy	Brekeke SIP Server’s IP address
User Name	When authentication is set at the upper server, set the user name that is assigned by the upper server here.
Password	When authentication is set at the upper server, set the password that is assigned by the upper server here.

9. NAT Traversal

9.1. Brekeke SIP Server Behind NAT (Near-End NAT traversal)

If you are using the Brekeke SIP Server behind NAT, but need to communicate with SIP clients outside the NAT, please use the following settings:

1. Interface setting at the Brekeke SIP Server.

   Go to **[Config] menu** > **[System]**. Direct your NAT router's public IP address to one of the Interface Addresses 1–5.

2. Port forwarding at NAT router.

   Setting Port forwarding at NAT router is required to ensure NAT traversal to work properly. With proper setting at NAT router, the Brekeke SIP Server's listening ports for SIP and RTP are forwarded to the Brekeke SIP Server's IP address. If your environment uses a firewall to filter packets, make sure to open the following ports which are used by the Brekeke SIP Server.

Below is the port number that is used by the Brekeke SIP Server:

Protocol	**Port Number (Default)**	**Set at**
SIP (UDP)	5060	**[Configuration]** > **[SIP(general)]**
RTP (UDP)	10000-10999	**[Configuration]** > **[RTP]**

The Brekeke SIP Server's listening ports are set in the following places:

SIP listening port: **[Config]** menu > **[SIP (General)]** > **[SIP Exchanger]** > **[Local Port]**

RTP ports: **[Config]** menu > **[RTP]** > from **[Minimum Port]** to **[Maximum Port]**

9.2. For Clients Behind NAT over the Internet (Far-End NAT traversal)

To communicate properly with SIP UAs located behind a firewall over the Internet, Far-End NAT traversal feature is applied to the call. If you have a firewall in the same network where the Brekeke SIP Server is located, you would need to set the Near-End NAT setting as well.

1. Keep Alive Setting at the Brekeke SIP Server

   Far-End NAT requires maintaining port mapping at the router that is located at the same network with SIP UA. SIP packets from the server will be undeliverable when port mapping has been cleared. To ensure maintaining the port mapping at the router, the Brekeke SIP Server will send dummy SIP packets periodically; this feature is called Keep Alive. The interval of “keep alive” needs to be set short to prevent port mapping being cleared.

   For some routers, this “keep alive” feature does not work to maintain port mapping. For such a case, we recommend that you use the port forwarding setting at the router instead.

   The Brekeke SIP Server's "keep alive" setting is set in the following places:

   Go to Brekeke SIP Server Admintool>**[Config]menu**>**[SIP(General)]**>**[NAT traversal]**

   Set **[keep address/port mapping]**=on

Item	Value	Explanation
Keep address/port mapping	on	Enable keep alive feature
Interval (ms)	(Depends on network environments.)	This is the interval to send dummy SIP packets. Default is set as 120,000 milliseconds (=2 minutes). Shorter interval is recommended to ensure maintaining port mapping at the router.

2. Port Forwarding Setting at the Router or Firewall Located at the Same Network with UA

   In addition to the “keep alive” feature, there is another way to establish communications with a SIP UA located behind a firewall over the Internet. When the communication cannot be established, even with “keep alive” settings, it is necessary to set port forwarding settings on the router located on the same network with SIP UA. For port forwarding, you can set the port number that SIP UA is using on the router. If a firewall is used to filter packets, similarly open ports that SIP UA is using at firewall setting. Please refer to the configuration screen or document of SIP UA for the port numbers to set at these settings.

10. SIP User Agent Setup

To have proper communications using the Brekeke SIP Server, precise setting at the SIP User Agent is necessary.

10.1. Prepare Appropriate SIP User Agent

Setting up the SIP user agent (UA) begins with preparing an appropriate SIP user agent to meet your requirements and environment. Commonly used SIP user agents are SIP softphones, SIP hardphones, VoIP Gateways, Analog Telephone Adaptor (ATA), and Instant Messenger (IM). Some SIP user agents are free to try or use and readily available by download through the internet.

10.2. Setting Items

Below is a basic setting item for SIP UAs. Some SIP UAs may not have the same settings.

Item	Setting value
SIP proxy server	Brekeke SIP Server’s IP address ✓ Set server port for 5060 if is applicable. If you wish to modify the port number, please refer to the section 5.8.2 SIP (General).
Registrar	Brekeke SIP Server’s IP address
Outbound Proxy	Brekeke SIP Server’s IP address
Domain	Brekeke SIP Server’s IP address
Realm	Brekeke SIP Server’s IP address ✓ Set the same Realm which is set to the SIP Server if the server does authentication.
User Name	Assign
Authentication User Name	Assign ✓ Set the authentication user name registered with the server if the server does authentication
Password	Set the authentication user’s password registered with the server if the server does authentication
STUN	STUN can sometimes solve NAT traversal when SIP UA is located behind NAT. However, there are some cases when using STUN causes failure of NAT traversal. For such cases, this setting needs to be disabled.

10.3. Confirming Registration

If a SIP UA is properly set, you can confirm registration status from the Registered List on the Brekeke SIP Server Admintool screen. For more details of how to confirm registration status, please refer to the Section 5.3, Registered List.

Appendix A: Glossary

- Admintool, Administrative tool or Administration tool

  Front-end tool to manage Brekeke SIP Server. Because it is web-based, you can access the tool either from locally or remotely. You can start/shutdown the server, check the server's status, and configure the environment.

  - Refer to Section 5 "Administration Tool"

- Client

  Software or a hardware used for starting/receiving a session. The client should support SIP protocol. For example, soft phones, IM clients, IP phones are clients. Brekeke SIP Server mediates the connection between those clients.

  - Refer to Section 10 "SIP User Agent Setup"
  - Related words: Server, SIP, UA

- Deploy Patterns or Action Patterns

  The patterns defined by you that determine the actions in Dial Plan. You can define to replace the SIP headers contents, to set the destination of a SIP packet, etc.

  - Refer to Section 5.3.2 "Edit Rule", "6.3.2 Deploy Patterns"
  - Related words: Dial Plan, Rule, Matching Patterns

- Dial Plan

  Dial Plan is one of the methods that Brekeke SIP Server uses to decide the routing destination of a session. Dial Plan can consist of multiple rules. Each rule is defined with the pair of Matching Patterns and Deploy Patterns. Only when the session matches with the conditions in Matching Patterns, the actions defined in Deployed Patterns will be handled.

  You can view and edit the Dial Plan rules at Admintool > **[Dial Plan]** page. For the details, refer to Section 6 "Dial Plan".

  - Refer to Section 1.1 "What is Brekeke SIP Server?", "5.5 Registered Clients", "6 Dial Plan"
  - Related words: Rule, Deploy Pattern, Matching Patterns

- Environment Variable

  The variables for setting Brekeke SIP Server's behavior and administration information, various internal parameters. You can set the values of the environment variables in the property file. Or you can set some parts of those environment variables in [Configuration] page.

  To set different an environment variable for each session, you need to specify it using Dial Plan's Deploy Patterns.

  - Refer to Section 5.8 "Configuration ", "6 Dial Plan"
  - Related words: Deploy Pattern

- Far-End NAT traversal

  NAT traversal of the UA (client) which is behind a NAT which exists far from Brekeke SIP Server.

  - Refer to Section 1.1 "What is Brekeke SIP Server?", and Section 9 "NAT Traversal".
  - Related words: NAT traversal, Near-End NAT traversal.

- ITSP

  Abbreviation of Internet Telephony Service Provider.

- Matching Patterns or conditions patterns

  Conditions in Dial Plan rules. You can use regular expressions for defining conditions using SIP headers, source IP address of the packets.

  - Refer to Section 6.2 "Editing a Dial Plan", "6.3.1. Matching Patterns"
  - Related words: Dial Plan, Rule, Deploy Patterns

- NAT (Network Address Translation) Traversal

  When each client in the same session is behind a different NAT (firewall), Brekeke SIP Server connects those clients using its proprietary NAT traversal feature. RTP packets will be relayed through Brekeke SIP Server depending on the network environment. Brekeke SIP Server's NAT traversal features supports both Far-End NAT and Near-End NAT.

  - Refer to Section 1.1 "What is Brekeke SIP Server?", and Section 9 "NAT Traversal".
  - Related words: Near-End NAT traversal, Far-End NAT traversal, RTP relay

- Near-End NAT Traversal

  NAT traversal of the UA (client) which is behind a NAT and which is in the same LAN as Brekeke SIP Server.

  - Refer to Section 1.1 "What is Brekeke SIP Server?", and Section 9 "NAT Traversal".
  - Related words: NAT traversal, Far-End NAT traversal

- Register database

  The database that the client addresses are recorded based on the data in REGISTER requests sent from the clients. Brekeke SIP Server will look up the client's registered address from the database for deciding the session's routing destination, when needed.

  You can view the list of registered clients at Brekeke SIP Server admintool > [Registered List] page.

  - Refer to Section 1.1 "What is Brekeke SIP Server?", and 5.5 "Registered Clients"
  - Related words: Thru Registration, Upper Registration

- RTP

  Abbreviation of Real-time Transport Protocol. It is the protocol that clients use for sending/receiving media (voice, video, etc.). For the details, refer to RFC1889,1890.

  - Refer to Section 5.8.4 "RTP"
  - Related words: SIP, RTP relay

- RTP relay or RTP tunnel

  RTP packets are usually transmitted directly between clients (not through Brekeke SIP Server). But if it is difficult for those UAs to directly communicate with each other depending on the network environment, Brekeke SIP Server will relay RTP packets. Brekeke SIP Server use the port 10000-10999 (by default) for RTP relay.

  - Refer to Section "1.1 What is Brekeke SIP Server?", and 5.8.4 "RTP"
  - Related words: NAT traversal, RTP

- Rule or Dial Plan rule

  A rule is a pair of Matching Patterns and Deploy Patterns for setting Dial Plan.

  - Refer to Section 6.2 "Editing a Dial Plan", and 6.3 "Syntax"
  - Related words: Dial Plan, Deploy Patterns, Matching Patterns

- Session

A session is initiated by an INVITE request. For the voice conversation, 1 session is usually used for a call. A session remains until a BYE request is processed or an error response is processed. Sessions status can be checked at admintool > **[Session List]** page.

   - Refer to Section 5.4 "Active Sessions"
   - Related words: SIP

- Session ID or SID

A unique id assigned for each session.

   - Refer to Section 5.4 "Active Sessions"
   - Related words: Session

- Server

Server means Brekeke SIP Server in this document unless otherwise noted.

   - Refer to Section 1.1 "What is Brekeke SIP Server?"

- SIP

Abbreviation of Session Initiation Protocol. It is a protocol that clients and servers use for setting up sessions or for controlling calls, etc. For the details, refer to RFC3261.
Brekeke SIP Server will send SIP packets sent from a client to an appropriate destination. The Server edits the SIP packets before sending to the destination as needed. The Server uses the port number 5060 (by default) for SIP.

   - Refer to Section 1.1 "What is Brekeke SIP Server?"
   - Related words: RTP, Session, Server, Client

- Thru Registration

If the request URI in the REGISTER request sent from a client doesn't include Brekeke SIP Server's address, Brekeke SIP Server will forward the REGISTER request to the address specified in the request URI.

   - Refer to Section 1.1 "What is the Brekeke SIP Server?", 5.8.3 "SIP(Advanced)", and 8. "Thru Registration"
   - Related words: Register database, Upper registration

- UA or User Agent
  - Related words: Client

- User directory database

  The database that holds the records of user information such as user name, password, etc. for authenticating SIP requests. You can view and edit the user information at Admintool > **[Authentication]** page.

  To authenticate users using Brekeke SIP Server, user information needs to be added to the user directory database in advance.
  - Refer to Section 5.4 "User Authentication", and 5.8.2 "SIP(General)"

- Upper Registration

  This feature forwards REGISTER requests sent from clients to another server as configured at Brekeke SIP Server. A client can send just one REGISTER request to Brekeke SIP Server to register itself both at Brekeke SIP Server and at other server.
  - Refer to Section 1.1 "What is Brekeke SIP Server?", and 5.8.3 "SIP(Advanced)", and 7. "Upper Registration"
  - Related words: Register database, Thru registration