OnDO SIP Server

Version 1.5

Administrator's Guide

Brekeke Software, Inc.

Version

OnDO SIP Server v1.5 Administrator's Guide

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1. Introduction

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

1.1. What is the OnDO SIP Server?

It is a SIP server which provides proxy, registrar, NAT traversal and authentication created by Brekeke Software, Inc. With the OnDO SIP Server, you can use any hard phones, soft phones, VoIP Gateways and applications that are SIP compliant for communications. The OnDO SIP Server has the following main functions:

- **Call Routing**

  The OnDO SIP Server will route SIP requests from a SIP UA or other servers to the most appropriate SIP URI address based on its registrar database. By specifying desired Call Routing setting in Dial Plan, you can also prioritize your call routing. If the routing resolves successfully on the server, a user can establish a call 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 packet to route calls. For example, users can set a prefix for each location with Dial Plan settings. Such settings are especially useful for multi-location office usage of an OnDO SIP Server.

- **Registrar**

  The OnDO SIP Server will receive REGISTER requests from SIP UAs, and update 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 OnDO 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, OnDO SIP Server will relay RTP packets to prevent losing voice or media data. The NAT traversal feature on the OnDO SIP Server supports both Near-End NAT (the server and SIP UAs located within the same firewall) and Far-End NAT (SIP UAs located on the other side of a firewall of a remote network).

- **Upper/Thru Registration**

  This is a unique feature of the OnDO SIP Server that allows easy configuration of parallel users of preexisting or other SIP servers. By forwarding REGISTER requests to specified SIP servers, these features allow users to register their SIP UAs at the other SIP server and the OnDO SIP Server simultaneously. For example, with this feature, users can register their SIP UAs at an ITSP, thus users under the OnDO SIP Server can talk with other users in the ITSP or receive calls to PSTN.

2. System Requirements

The OnDO SIP Server supports the following platforms:

OS	Microsoft Windows, Linux, Mac OS X, Solaris
Java	JDK 1.4 or later
Memory	At least 256 MB

✓ *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 J2SE

Download the Java platform from the following website:

http://java.sun.com/products/

Search for Java2, Standard Edition (J2SE) for Windows. Then download and install the latest version of J2SE.

3.2. Step 2: Installation of the OnDO SIP Server

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

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

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

1. Open **[Control Panel]>[Administrative tools]>[Service]**. Select **[OnDO SIP Server]** and start the service.
2. Restart your computer. The OnDO SIP Server HTTP service will automatically start.

3.4. Step 4: Starting the OnDO SIP Server Admintool

1) Select **[Start]>[Program]>[Brekeke]>[OnDO SIP Server 1.3]>[OnDO SIP Server Admintool]**. A web browser will open and you will see the OnDO SIP Server Admintool Login page, enter User ID and Password and click **[Login]**.

✓ *Default Administrator User ID and initial password: User ID: sa Password: sa*

2) After the login, click the menu item **[Start/Shutdown]**. If the Status is **[active]**, the OnDO SIP Server has started successfully.

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

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

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

4.1. Step 1: Installation of J2SE SDK

Download the Java platform from the following website:

http://java.sun.com/products/

Search for Java2, Standard Edition (J2SE) for your OS. Then download and install the latest version of J2SE.

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 OnDO SIP Server

1) Obtain the file *binary 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:8080.

   (If you chose a port number other than "8080" 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 OnDO SIP Server Admintool

1) Open a web browser and input the URL http://localhost:8080/proxy. (If you chose a port number other than "8080" when installing Tomcat. Please change the port number in the URL above to the number specified during installation.) Then you will see the Login page of OnDO SIP Server Admintool.

✓ *Default Administrator's User ID and initial password: User ID: sa Password: sa*

After the log-in, click the menu item **[Start/Shutdown]**. If the Status is **[active]**, the SIP server has started successfully. If the Status is **[inactive]**, the server hasn't started successfully. The error should be shown above the Status.

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

5. OnDO SIP Server Administration Tool

5.1. Start / Shutdown

The administrator can start or shutdown OnDO SIP Server. While the server is running, the word "Active" is displayed. When it is not running, "Inactive" is displayed.

While the server is inactive, the menu item **[Status]**, **[Registered List]**, and **[Sessions]** are disabled.

Button	**Explanation**
Start	Starts OnDO SIP Server
Shutdown	Stops OnDO 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.

5.2. Server Status

Display server information and status.

Server Status	
Field Name	**Explanation**
server-product	Server product name
server-ver	Server's version number
server-name	Server name (*1
server-description	Server description (*1
server-location	Server location (*1
server-startup-time	Time the server was started
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 port (*1
interface	Network interface address(es) used by the server (*1
startup-user	User name that started OnDO SIP Server service or Tomcat (this information is taken from the OS)
work-directory	The directory that the OnDO SIP Server is running from
session-active	The number of currently active sessions
session-total	The total number of sessions processed after startup
command-active	The number of management commands currently being processed on the OnDO SIP Server
command-total	Total number of management commands processed on the OnDO SIP Server after startup
sip-packet-total	The total number of SIP packets received after startup
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 (*1
admin-mail	Administrator's e-mail address (*1

✓ *(*1 modifiable in the [Configuration] page*

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

5.3. Registered List

Display the records that are in the registrar database. When the OnDO SIP Server accepts a REGISTER request, the records are updated automatically. Here you may remove records from the database, or create a new record in the database manually.

Item	Default value	Explanation
User	-	User name that receives a contact from other UAs.
Contact URI	-	User's contact SIP URI
Expires	3600 (sec)	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. Default is set for 3600 seconds (=1 hour).
Priority	1000	Priority to contact clients, ranging from 100 - 1000.
Requester	-	The IP address that this REGISTER request was sent from.
Time Update		Timestamp of the latest update of this record

Button	Explanation
Unregister	Delete the specified record from the registrar database.
Register	Register a new record in the registrar database.

Link	Explanation
Refresh	Refresh the registered list.

5.4. Session List

5.4.1. Display Current Sessions

Click each item’s [Session ID] to display the Session Detail Screen

<table>
<tr><th>Button</th><th>Explanation</th></tr>
<tr><td>Session ID</td><td>Display detailed session information by clicking on the link.</td></tr>
<tr><td>From</td><td>Caller's SIP URI and its IP address</td></tr>
<tr><td>To</td><td>Callee(destination)'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>
<tr><td>Refresh</td><td>Refresh the session list.</td></tr>
</table>

Link	Explanation
Refresh	Refresh the Session list.

5.4.2. Session Detail Screen

The session detail screen displays detailed information for the selected session.

<table>
<tr><th>Button</th><th>Explanation</th></tr>
<tr><td>from-url</td><td>Caller's SIP URI</td></tr>
<tr><td>from-ip</td><td>Caller's IP address</td></tr>
<tr><td>from-if</td><td>Network Interface of caller's side</td></tr>
<tr><td>to-url</td><td>Callee(destination)'s SIP URI</td></tr>
<tr><td>to-ip</td><td>Callee(destination)'s IP address</td></tr>
<tr><td>to-if</td><td>Network Interface of Callee(destination)'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 this session</td></tr>
<tr><td>sip-packet-total</td><td>Total number of received SIP packets</td></tr>
<tr><td>listen-port</td><td>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>phase</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</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>Time of talking</td></tr>
<tr><td>rtp-relay</td><td>RTP relay status (on/off)</td></tr>
</table>

When session status shows [rtp-relay]=on, the information below is displayed. This information shows status of RTP stream of both [rtp-srcdst] (Caller—>Callee) and [rtp-dstsrc] (Callee –> Caller).

Item	Explanation
media	Media type (audio, video)
payload	Payload type
status	Status (active = streaming, hold)
listen-port	Port number at SIP server for receiving RTP packets
send-port	Port number at SIP server 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
Refresh	Refresh the display
Disconnect	Disconnects the session
Back	Go back to [Session List] page

5.5. Dial Plan

5.5.1. Dial Plan

The Dial Plan page, which is in the **[Dial Plan]** menu of OnDO SIP Server Admintool, 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 the rule and changing the priority.

By pressing the **[Update]** button, you can apply the new rules or modified rules even when the server is running.

Item	Explanation
Rule name	The name of Dial Plan rule
Matching Patterns	Defined condition
Deploy Patterns	How the call should be processed

Button	Explanation
New	Append a new rule.
Edit	Edit the rule
Rename	Rename the rule
Delete	Delete the rule
Copy	Copy the rule
Up	Raise the priority of the rule
Down	Lower the priority of the rule
Update	Save and applies changes

5.5.2. Edit Dial Plan

[Dial Plan] menu > **[New]** button or **[Edit]** button.

Refer to “Section 6, Dial Plan”, for information on how to write dial plan rules.

Item	Default value	Explanation
Rule name		Name of the rule You cannot use the existing rule name.
Description		Description of the rule
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		Matching Patterns − A value of the variable that should match Deploy Patterns − The value that will be assigned to the variable

Button	**Explanation**
insert	Insert the specified definition in [Variable] and [Value] options into the given list box.
delete	Delete the selected definition. The deleted definition is displayed in [Variable] and [Value] options.
up	Move the selected rule up
down	Move the selected rule down
Save	Save selected rule and move back to the [Dial Plan] page
Cancel	Cancel selected rule and move back to the [Dial Plan] page

5.6. Authentication

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

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

The setting for changing the password for the access of OnDO SIP Server admintool is at **[Configuration]** page. Refer to the section "5.8.5 Login Password" for the details.

5.6.1. Edit Users

[Authentication] menu > **[Edit Users]**

This page is for adding/ searching/ editing users.

1) **Search**

Search a user from the user directory database. By setting a search condition and click **[Search]** button, the list of corresponding users will be shown.

Item	Default value	Explanation
filter		Search keywords
max	100	Number of results to display

Button	Explanation
search	Execute the search

2) Edit (*required fields)

Edit a user's information.

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

Button	Explanation
Modify	Modify the contents
Add	Add the contents
New	Clear the entry field for adding a new user
Delete	Delete the specified user

5.6.2. Import

Using the import function you can add multiple users' authentication information at the same time. Write in a file with CSV (Comma Separated Value) format. Add one line per user in the following order:

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

- Select the checkboxes for your desired options.
- Click the [Browse] button to select a CSV file.
- Click the [Upload] button to upload the records from the file. The authentication information will be updated.

Checkbox Options

Item	Default value
Overwrite existing records	Unchecked
Delete all existing records	Unchecked
Ignore the first line (header record in the file)	Unchecked

Button	Explanation
Browse	Select a file to import from
Upload	Execute upload

5.7. Log

Click the **[Log]** menu in the OnDO SIP Server Admintool for **[Log]** page.

This page shows number of sessions by date.

Click the desired date to display that date's session log.

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

5.7.1. Display Session Log

A detailed session log will be displayed in a new window.

Item	Explanation
sid	Session ID
from-url	Caller's SIP URI
to-url	Callee (routing destination)'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.)

Button	Explanation
Close	Close the window

5.8. Configuration

Click the **[Config]** menu to open the **[Configuration]** page.

This page is for editing parameter settings, passwords, and upgrading the software.

Changes will take effect when the server is restarted.

5.8.1. System

System is a required field.

Configure system administrator's information and network. These settings will be shown in the **[Server Status]** page.

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, OnDO SIP Server will automatically start when web server (Tomcat) is started.

Network			
	Item	**Default value**	**Explanation**
	Interface address 1-5		IP address(es) or domain name(s) (FQDNs) to be used as interface address(es) by OnDO 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 OnDO SIP Server is installed. In a Windows OS environment, OnDO 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. Time unit as seconds.
*	Auto interface discovery	off	When it is set for "on", interface address will be updated automatically.

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

Button	Explanation
Save	Save changes. Changes will take effect when the server is restarted.

5.8.2. SIP (General)

SIP (General) is a required field.

Configure General SIP settings, NAT traversal and Authentications.

Refer to the section “9. NAT traversal” for the details of NAT traversal.

To add/edit user accounts for authentication, please go to [Authentication] menu (Refer to the section “5.6 Authentication”.

SIP exchanger			
	Item	**Default value**	**Explanation**
*	Session Limit	-1	Maximum number of sessions the server will handle concurrently. "-1" specifies an unlimited number of sessions.
*	Local Port	5060	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 OnDO SIP Server will send keep-alive packets to SIP UAs that are behind the NAT at specified intervals. This is so that NAT will not close the external port used by the OnDO SIP Server to send packets to SIP UAs that are behind the NAT.
	Interval	120000(ms)	Interval for above setting. Default is set for 120,000 milliseconds (=2 minutes).

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

Button	Explanation
Save	Save changes. Changes will take effect when the server is restarted.

5.8.3. SIP (Advanced)

SIP (Advanced) is a required field.

Upper Registration, Thru Registration and Various Timeout Settings

Refer to the section “7. Upper Registration” for the details of Upper Registration and the section “8. Thru Registration” for the details of Thru Registration.

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 (Set "0" for unlimited)			
	Item	**Default value**	**Explanation**
*	Ringing Timeout	120000 (ms)	Timeout for Ringing
*	Talking Timeout	259200000 (ms)	Timeout for talking time
*	BYE Timeout	60000 (ms)	Timeout for responding to a BYE request
*	Upper/Thru Timeout	30000 (ms)	Timeout for waiting the response for a REGISTER request to Upper Registration/Thru Registration destination

Button	Explanation
Save	Saves changes Changes will take effect when the server is restarted.

5.8.4. RTP

RTP is a required field.

RTP Settings

If NATs are involved in the SIP communications, OnDO 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 OnDO SIP Server. When set to "auto", OnDO SIP Server will decide whether or not to relay RTP automatically. (RTP packets are automatically handled when OnDO 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”, OnDO SIP Server will not relay RTP packets for the clients running on the server.
*	Minimum Port	10000	The minimum port number to transmit RTP packets from.
*	Maximum Port	10999	The maximum port number to transmit RTP packets from.
*	Minimum Port (Video)	0	The minimum port number to transmit RTP packets for video stream from.
*	Maximum Port (Video)	0	The maximum port number to transmit RTP packets for video stream from.
*	Port mapping	source port	Selects a destination port number for the OnDO SIP Server to send RTP packets to clients behind Far-End NAT. Designates whether to use the source port from the IP header of RTP packet (when set to “Source Port”) or the RTP port specified in SDP within the SIP packet (when set to “sdp”).

Timeout (Set "0" for unlimited)			
	Item	**Default value**	**Explanation**
*	RTP Session Timeout	600000 (ms)	The timeout for detecting RTP packets when relaying RTP. Default is set for 600,000 milliseconds (=10 minutes).

Button	Explanation
Save	Saves changes Changes will take effect when the server is restarted.

5.8.5. Login Password

Change the login password for the OnDO SIP Server Admintool.

To set the password for authenticating SIP requests, please use **[Authentication]** page. Refer to the section [5.6 Authentication].

Administrator's default user id is "sa" and its password is "sa".

Item	Default value	Explanation
Type a new password	sa	Password for accessing OnDO SIP Server Admintool

Button	Explanation
Apply	Accept the password change

5.8.6. Upgrade Software

This field is for upgrading the OnDO SIP Server.

1) To upgrade, first shutdown the OnDO SIP Server from **[Start/Shutdown]** page.
2) Click the **[Browse]** button.
3) Select a file for updating and push **[Upload]** button.

6. Dial Plan

6.1. What is the Dial Plan?

The OnDO SIP Server's Dial Plan defines rules for call 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 brief summary of some of the OnDO SIP Server's Dial Plan functions. For more detailed information please refer to the *OnDO SIP Server Tutorial-Dial Plan*.

To add a new rule:

1) To add a new rule, go the Dial Plan page, and push **[New]**. You will see a new page for defining dial plan rules.
2) Enter the name of your rule in the **[Rule Name]** field.
3) Enter the description of your rule in the **[Description]** field.
4) If you would like to disable your rule, check the **[Disable]** checkbox.

Below is the list of functionalities that you can achieve by setting a Dial Plan:

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

6.2. Edit

To edit the Dial Plan, open **[Dial Plan]** menu from the Admintool. Refer to the section [5.5 Dial Plan].

You can also edit Dial Plan files. Your changes will be in effect after you restart. Dial Plan files are located under install directly:

\webapps\proxy\WEB-INF\work\sv\etc\dialplan.tbl

6.3. Syntax

When all conditions set in the Matching Patterns are true, the actions defined in Deploy Patterns are executed.

6.3.1. [Matching Patterns] field

Define conditions for applying the rule.

Conditions can be defined using a pair of the following: the name of the SIP header, condition variable, 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 () 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 are as follows:

Symbols	Meaning
^	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) 0 times or more
X{n}	Match the preceding element (X, in this case) n times
X{n,}	Match the preceding element (X, in this case) n times or more
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 digitth buffer in Deploy Pattern, `use %<digit> (for example %1)`

To add a condition in [Matching Patterns] section, click […] button (which is between [Variable] field and [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 [Value] field and then, click [insert] button. Refer to the section [5.5.2 Edit Dial Plan].

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

Environment variable

A variable name which starts with ‘&’ is an environmental variable. 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.

2) 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.

例:

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

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.

$date

Meaning:

Current Year/Month/Date

Syntax:

$date

$date(format)

format − Date format

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 compare it with the string “06-03-2010”.

$geturi

Meaning:

Get the string of the URI

Syntax:

$geturi(str)

str − Text string

Explanation:

Gets the SIP URI part from the text string “str”.

Example:

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

Gets the SIP URI part from From: header and compare 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”.

$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, OnDO SIP Server checks if the source IP address 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

$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 OnDO 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, OnDO SIP Server checks if the Request URI is outbound or not.

For example, if OnDO 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

<u>$port</u>

Meaning:

Source port number

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 OnDO 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, OnDO 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:

The contact IP address registered to OnDO SIP Server

Syntax:

$registeredaddr

$registeredaddr (str)

str－SIP URI　or a user name

Explanation:

Returns the contact IP address registered in the OnDO SIP Server's 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 user can not be found, the condition will not be fulfilled.

Example:

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

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

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

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

`$registeredaddr( “user” ) = ^192\.168\.0\.`

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

$registereduri

Meaning:

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

Syntax:

$registereduri

$registereduri (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 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:

`$registereduri= sip:100@host`

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

`$registereduri ( “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 text string in 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.

$time

Meaning:

Current time

Syntax:

$time

$time (format)

format − Time format

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 0-9 o’clock.

`$time( “SSSS” ) = [02468]$`

If the millisecond is a even number.

6.3.2. [Deploy patterns] field

The **[Deploy patterns]** field defines actions that will be taken for when a rule’s conditions are fulfilled. 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 [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 **[insert]** 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, OnDO 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 is **not** 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 **[Configuration]** page.

Syntax:

&environment_variable_name = a setting value

Example:

&net.sip.timeout.ringing = 10000

Set the value of ringing timeout to 10000.

(The environment variable is net.sip.timeout.ringing)

&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 OnDO 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", OnDO SIP Server continues to check the next rule below.

If "false", OnDO 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 , OnDO 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 (callee).

Syntax:

$ifdst = IP address or FQDN

Explanation:

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

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 call originator (caller).

Syntax:

$ifsrc = IP address or FQDN

Explanation:

OnDO SIP Server’s interface address used for sending/receiving the packets to/from the call originator (caller).

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", OnDO 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, OnDO 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”, OnDO SIP Server will decide whether to replace the headers or not automatically.

The default value is "auto".

For example, if this feature is enabled, OnDO 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 OnDO SIP Server. If "false", RTP packets will not be relayed through OnDO SIP Server. If “auto”, OnDO SIP Server will decide whether to relay RTP packets or not automatically (For example, OnDO 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 UAs send REGISTER requests to the registrar server (upper server) specified in the **[Register Server (IP or FQDN)]** field via the OnDO SIP Server. Using this feature, SIP UA 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 OnDO SIP Server, there are no special settings necessary at users' SIP UAs.

✓ *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(Advanced)]**, set Upper Registration as follows.

Item	**Setting value**	**Explanation**
On/Off	on	Enable Upper Registration
Register Server (IP or FQDN)	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	OnDO SIP Server's IP address
Registrar	OnDO SIP Server's IP address
Outbound Proxy	OnDO 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 OnDO 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.

✓ *Please note that using Upper Registration feature, users can specify only one upper server at the OnDO 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	OnDO 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	OnDO 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. OnDO SIP Server Behind NAT (Near-End NAT traversal)

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

1) Interface setting at the OnDO SIP Server

   Go to **[Config] menu**>**[System]**. Put your NAT router's public IP address to one of **[Interface Address 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 OnDO SIP Server's listening ports for SIP and RTP are forwarded to the OnDO SIP Server's IP address. If your environment uses firewall to filter packets, make sure to open the following ports which are used by the OnDO SIP Server.

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

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

   The OnDO 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 OnDO SIP Server is located, you would need to set the Near-End NAT setting as well.

1) Keep Alive Setting at the OnDO 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 OnDO 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 OnDO SIP Server's "keep alive" setting is set in the following places:
Go to OnDO 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 OnDO SIP Server, precise setting at the SIP User Agent is necessary.

10.1. Prepare Appropriate SIP UA

Prepare the appropriate SIP UA to meet your requirements and environment. Commonly used SIP UAs are SIP softphones, SIP hardphones, VoIP Gateways, Analog Telephone Adaptor (ATA), and Instant Messenger (IM). Some SIP UAs are free to try or use.

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	OnDO 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	OnDO SIP Server's IP address
Outbound Proxy	OnDO SIP Server's IP address
Domain	OnDO SIP Server's IP address
Realm	OnDO 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 SIP UA is properly set, you can confirm registration status at **[Registered List]** on the OnDO 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

  The front-end tool to manage OnDO 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 the section "5. OnDO SIP Server Administration tool"

- Client

  A 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. OnDO SIP Server mediates the connection between those clients.

  - Refer to the section "10. SIP UA 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 the section "5.5.2 Edit Dial Plan", "6.3.2 Deploy Patterns"
  - Related words: Dial Plan, Rule, Matching Patterns

- Dial Plan

  Dial Plan is one of the methods that OnDO 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 the section "6. Dial Plan".

  - Refer to the section "1.1 What is the OnDO SIP Server?", "5.5 Dial Plan page", "6 Dial Plan"
  - Related words: Rule, Deploy Pattern, Matching Patterns

- Environment Variable

  The variables for setting OnDO 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 the 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 OnDO SIP Server.

  - Refer to the section "1.1 What is the OnDO SIP Server?" "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 the section "5.5.2 Edit 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), OnDO SIP Server connects those clients using its proprietary NAT traversal feature. RTP packets will be relayed through OnDO SIP Server depending on the network environment. OnDO SIP Server's NAT traversal features supports both Far-End NAT and Near-End NAT.

  - Refer to the section "1.1 What is the OnDO SIP Server?" "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 OnDO SIP Server.

  - Refer to the section "1.1 What is the OnDO SIP Server?" "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. OnDO 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 OnDO SIP Server admintool > [Registered List] page.

  - Refer to the section "1.1 What is the OnDO SIP Server?", "5.3 Registered List"
  - 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 the 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 OnDO SIP Server). But if it is difficult for those UAs to directly communicate with each other depending on the network environment, OnDO SIP Server will relay RTP packets. OnDO SIP Server use the port 10000-10999 (by default) for RTP relay.

  - Refer to the section "1.1 What is the OnDO SIP Server?", "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 the section "5.5.2 Edit Dial Plan", "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 the section “5.4 Session List page”
  - Related words: SIP

- Session ID or SID

  A unique id assigned for each session.

  - Refer to the section ” 5.4 Session List page”
  - Related words: Session

- Server

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

  - Refer to the section “1.1 What is the OnDO 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.
  OnDO 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 the section “1.1 What is the OnDO 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 OnDO SIP Server’s address, OnDO SIP Server will forward the REGISTER request to the address specified in the request URI.

  - Refer to the section “1.1 What is the OnDO SIP Server?”, “5.8.3 SIP(Advanced)” “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 OnDO SIP Server, user information needs to be added to the user directory database in advance.
    - Refer to：“5.6 Authentication page”“5.8.2 SIP(General)”

- Upper Registration

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