Enterprise SIP Server

Version 1.2

Tutorial - Dial Plan

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1. Introduction

This document is one part of a collection of documents that explain the operational functionalities of the Enterprise SIP Server (“ESS”). This particular document is a tutorial on how to use a Dial Plan.

Using this Dial Plan, you can manage SIP sessions flexibly as you like such as for routing calls to desired destination, for filtering specific sip sessions, for load balancing SIP devices, etc.

For every session request, Enterprise SIP Server will first check if the destination user (SIP-URI) is in its register database. If the destination is in the database, Enterprise SIP Server will route the request to the most appropriate SIP-URI based on the database. When a user cannot be located in the database, the Dial Plan setting will be used. Enterprise SIP Server checks from the first listed condition in the Dial Plan rules, which are shown in the “Dial Plan” page of the Enterprise SIP Server Administration tool. If the condition isn’t fulfilled, the Enterprise SIP Server checks the second line, and so on. Once the Enterprise SIP Server finds the matching condition, it will execute the process defined by that rule and finish the decision operation.

For each rule in the Dial Plan, only when all conditions set in “Matching Patterns” are true, the processes defined in “Deploy Patterns” are executed.

2. Rule Definitions for Dial Plan

2.1. Matching Patterns

In Matching Patterns, you can set conditions for call session control settings. The “header field names” and the “condition variable names” below can be used here.

Condition Variable Name	Definition
`$addr`	Caller’s IP Address. ex_ `$addr=192\.168\.1\.100`
`$date`	Date on which the call request was received. ex_ `$date=2003/12/01`
`$localhost`	Whether the call is from localhost (Enterprise SIP Server) or not. ex_ `$localhost=true`
`$outbound`	Whether the callee is outside of the local network where the Enterprise SIP Server resides. ex_ `$outbound=false`
`$port`	Caller’s port number. ex: `$port=5060`
`$registered`	Whether the callee is registered or not. ex_ `$registered=false`
`$request`	Request ex_ `$request=INVITE sip:11@192.168.1.200:5060 SIP/2.0`
`$sid`	Session ID (Session ID number used for internal maintenance) ex_ `$sid=23`
`$time`	Time at which the call request was received. ex_ `$time=15:02:30`

2.2. Deploy Patterns

Define deploy actions for when the condition is fulfilled. The “header field names” and the “handling variable names” below can be used here.

Handling Variable Name	Definition
`$action`	Response code which will be sent to the caller or register action at Enterprise SIP Server ex_ `$action=404` `$action=register`
`$continue`	If `$continue=true` is set, checking procedures of Matching Patterns won’t be finished after the deploy patterns are executed. Checking procedure will be continued from the line right after this rule. ex_ `$continue=true`
`$ifdst`	Network Interface (IP address) used to send packets to the callee. ex_ `$ifdst=192.168.1.100`
`$ifsrc`	Network Interface (IP address) used to send packets to the caller. ex_ `$ifsrc=192.168.2.1`
`$nat`	Whether or not to use NAT Traversal function. ex_ `$nat=true`
`$replaceurl`	Whether SIP-URI in the SIP packet is replaced or not. ex_ `$replaceurl=false`
`$rtp`	Whether RTP Stream Tunneling is established or not. ex_ `$rtp=auto`
`$target`	Specifies the callee’s address. ex_ `$target=192.168.0.2`

2.3. Main SIP Header Fields

These are the most commonly used header fields and it is a good idea to know them when writing Dial Plan. For other header fields, please refer to the RFC3261.

Header Field	Definition
TO	The SIP-URI of the callee or routing sip server
FROM	SIP-URI of the caller
Max-Forwards	Maximum number of hops that a SIP request can pass through till its destination
User-Agent	Caller's user agents name

2.4. Main Methods of SIP Request

Method	Definition
REGISTER	Registers the contact information of the user agent (e.g. current position)
INVITE	Invitation to participate in the session. Requests an opening session
ACK	Acknowledgement to a response that an INVITE request was received
CANCEL	Cancels pending transactions in the session
INFO	Gives session information
OPTIONS	Queries the server's abilities

2.5. Regular Expressions

Following Expressions can be used in Matching Patterns.

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)

- ✓ *There are other expressions that you can use. For details, please refer to Java2 Platform document (http://java.sun.com/j2se/1.4.2/docs/api/java/util/regex/Pattern.html).*
- ✓ *You can see many examples of using these regular expressions in the later chapters in this document.*

3. Filtering

3.1. Filtering based on the Caller's IP Address

The conditional variable $addr has the caller's IP Address.

3.1.1. Call from IP Address 192.168.0.1 is rejected

Matching Patterns	Deploy Patterns
$request=^INVITE **$addr=192\.168\.0\.1$**	$action=603

✓ *Depending on the value $action is set as, the relevant response code will be sent to the caller. For more information refer to the "5. Error Exit" section in this document.*

3.1.2. Routing a call from an IP Address with the prefix 192.168. to sip:ivr@domain

Matching Patterns	Deploy Patterns
$request=^INVITE **$addr=192\.168\.**	to=sip:ivr@domain

3.1.3. Refusing a call from IP Address 192.168.0.1 - 192.168.0.5

Matching Patterns	Deploy Patterns
$request=^INVITE **$addr=192\.168\.0\.[12345]$**	$action=603

3.1.4. Accepting calls from only the IP Address 192.168.0.1

Matching Patterns	Deploy Patterns
$request=^INVITE **$addr=192\.168\.0\.1$** to=sip:(.+)[>;]*	to=sip:%1
$request=^INVITE $addr=.+	$action=603

✓ *When adding the condition $outbound=true, specify the routing SIP server by "to=". Then, when the destination SIP-URI is outside of the network, Enterprise SIP Server forwards the packets to the specified SIP server.*

3.2. Filtering based on the Caller's Port Number

The caller's port number is included in conditional variable `$port` field.

3.2.1. To refuse calls from Port Number 5061

Matching Patterns	Deploy Patterns
$request=^INVITE $port=5061$	$action=603

3.2.2. To treat calls from Port Number 5060 – 5069 as error 404 (not found)

Matching Patterns	Deploy Patterns

$request=^INVITE **$port=506.$**	$action=404

3.2.3. To refuse calls if the caller's Port Number is 6060 and the corresponding IP Address is 192.168.0.100

Matching Patterns	Deploy Patterns
$request=^INVITE **$port=6060$** **$addr=192\.168\.0\.100**	$action=603

3.3. Filtering based on whether the Callee is Registered in the register database or not.

Whether the callee is registered or not can be found in the conditional variable `$registered`.

When a callee is registered `$registered=true`
When a callee is not registered `$registered=false`

3.3.1. Registered callee's calls will be routed through sip:ivr@domain

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`$registered=true`**	`to=sip:ivr@domain`

3.3.2. To refuse calls to unregistered

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`$registered=false`**	`$action=603`

3.3.3. If the callee is registered and correspondingly the SIP-URI prefix is "1", the caller will hear a busy signal

Matching Patterns	Deploy Patterns

`$registered=true`
`$request=^INVITE`
`to=sip:1.+`
	`$action=686`

3.4. Filtering based on whether the Caller is Calling from a localhost.

Whether the caller is calling from a localhost or not can be found in the conditional variable `$localhost`.

Call originating from a localhost	`$localhost =true`
Call originating from a non-localhost	`$localhost =false`

3.4.1. A call originating from localhost is routed through sip:ivr@domain

Matching Patterns	Deploy Patterns
$localhost=true $request=^INVITE	to=sip:ivr@domain

3.4.2. Refusing a call that originated from a non- localhost

Matching Patterns	Deploy Patterns
$localhost=false $request=^INVITE	$action=603

3.4.3. Routing a call to sip:ivr@domain, when the call is from a localhost and it came from Port Number 15060

Matching Patterns	Deploy Patterns
$localhost=true **$port=15060**	to=sip:ivr@domain

3.5. Filtering based on whether the Call is an Outbound Connection

When the domain address in the destination URI is not the Enterprise SIP Server's address, Enterprise SIP Server will recognize it as an outbound call. Whether the call is outbound or not, can be found in the conditional variable `$outbound`.

The call is an outbound connection	`$outbound =true`
The call is not an outbound connection	`$outbound =false`

3.5.1. If the call is outbound connection, route the call to sip:ivr@domain

Matching Patterns	Deploy Patterns
`$outbound=true` `$request=^INVITE`	`to=sip:ivr@domain`

3.5.2. If a call is not outbound, it will be treated as error 404 (not found)

Matching Patterns	Deploy Patterns
`$outbound=false` `$request=^INVITE`	`$action=404`

3.5.3. If the call is not outbound and callee SIP-URI's prefix is "0", the prefix will be removed from the user name and the call will be routed through the "domain" using the extracted user name

Matching Patterns	Deploy Patterns

`$outbound=false`
`$request=^INVITE`
`to=sip:0(.+)@`
	`to=sip:%1@domain`

3.5.4. If the call is outbound, the SIP-URI inside the packet won't be changed and the call will be routed to 192.168.0.5

Matching Patterns	Deploy Patterns
$outbound=true $request=^INVITE	$target=192.168.0.5

✓ *There are 2 methods for setting the routing destination in the Dial Plan.*

Setting the callee in "to":

By assigning an assigned value for the To: header field in the SIP request, the callee can be set.

ex: `to="sip:newuser@domain"`

Setting the callee in $target::

The IP Address of the routing destination is set. The SIP-URI information within the SIP request won't be changed.

ex: `$target=192.168.0.100`

3.6. Filtering based on the Caller's SIP-URI

The caller's SIP-URI can be found in the From: header field.

3.6.1. If the caller's SIP-URI is sip:user@domain, refuse the call

Matching Patterns	Deploy Patterns
$request=^INVITE **from=sip:user@domain**	$action=603

3.6.2. If the caller's User Name is 1000, route the call to sip:iver@domain

Matching Patterns	Deploy Patterns
$request=^INVITE **from=sip:1000@**	to=sip:ivr@domain

3.6.3. If the caller's Domain Name is gw.domain, the call will be treated as unauthorized

Matching Patterns	Deploy Patterns
$request=^INVITE **from=@gw.domain[>;]***	$action=601

3.6.4. If the caller's User Name is not between 1000 – 2000 refuse the call

Matching Patterns	Deploy Patterns
$request=^INVITE **from=sip:[12]000@** to=sip:(.+)[>;]*	to=sip:%1
$request=^INVITE	$action=603

3.7. Filtering based on the Destination SIP-URI

The destination SIP-URI can be found in the To: header field.

3.7.1. If the destination SIP-URI is sip:user@domain, refuse the call

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:user@domain`**	`$action=603`

3.7.2. If the destination domain is “gw1”, route the call to “gw2”

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:(.+)@gw1[ >;]*`**	`to=sip:%1@gw2`

3.7.3. If the destination User Name is 1000, route the call to sip:ivr@domain

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:1000@`**	`to=sip:ivr@domain`

3.7.4. If the destination User Name’s prefix is 650, the prefix is removed from the User Name and the call is routed to the “domain”

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:650(.+)@`**	`to=sip:%1@domain`

3.8. Filtering based on the Caller's User Agent (SIP Client)

The caller's User Agent Name can be found in the User-Agent header field.

3.8.1. If the caller's User Agent Name is "SAMPLE", refuse the call

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`user-agent=^SAMPLE$`**	`$action=603`

3.8.2. If the User Agent Name is "Gateway", the maximum number of forwards (maximum number of Server Hops) is set to 10

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`user-agent=^GATEWAY$`**	`max-forwards=10`

3.9. Filtering based on Request.

Request can be found in the `$request` conditional variable.

3.9.1. When REGISTER request is received, return as an error

Matching Patterns	Deploy Patterns
`$request=^REGISTER`	`$action=400`

3.9.2. Routing MESSAGE Request to sip:user@domain

Matching Patterns	Deploy Patterns
`$request=^MESSAGE`	`to=sip:user@domain`

3.10. Filtering based on the Number of Forwards (Number of Hops)

The number of forwards value decreases at every server. The number of forwards value can be found in the Max-Forwards: header field.

3.10.1. If the number of forwards left is less than 5, refuse the call

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`max-forwards=^[0-5]$`**	`$action=603`

3.10.2. If the number of forwards left is between 50 and 99, the value will be set to 10

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`max-forwards=^[5-9].$`**	`max-forwards=10`

3.11. Filtering based on Time

The time which the server receives the call can be found in `$time`. The format is “hh:mm:ss”.

3.11.1. Route all calls from 7 pm to 10 am of the following day to sip:ivr@domain

Matching Patterns	Deploy Patterns

`$request=^INVITE`
`$time=^1[0-8]:`
`to=sip:(.+)[ >;]*`
	`to=sip:%1`

`$request=^INVITE`	`to=sip:ivr@domain`

3.11.2. Change the router destination between gw1 and gw2, every 30 minutes

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`$time=^..:[0-2].:..$`**	`$target=gw1`
`$request=^INVITE`	`$target=gw2`

3.12. Filtering based on Date

The date the server received the call can be found in `$date`. The format is “yyyy/mm/dd”

3.12.1. If it is the 1st of the month, refuse the call

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`$date=/01$`**	`$action=603`

4. Routing

4.1. Setting the Destination SIP-URI

The callee's SIP-URI can be changed in the To: header field. During a session, routing is based on the SIP-URI shown in the To: header field.

4.1.1. Setting the callee's SIP-URI as "sip:user@domain" (when callee is a "user")

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:user@`	**`to=sip:user@domain`**

4.1.2. Setting the callee's Domain as "gw2" (when the callee's domain is "gw1")

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:(.+)@gw1[ >;]*`	**`to=sip:%1@gw2`**

4.2. Setting the Destination Address

The callee's address is set in the handling variable `$target`. Routing will occur to the address set in the `$target`. When both `$target` and To: header field are set in the Deploy Patterns, the value in `$target` will be used for routing.

4.2.1. Setting the destination address as "pbx". (all calls)

Matching Patterns	Deploy Patterns
`$request=^INVITE`	**`$target=pbx`**

4.2.2. Setting the destination address as 192.168.0.2 . (When the callee isn't registered with the database)

Matching Patterns	Deploy Patterns
`$request=^INVITE` `$registered=false`	**`$target=192.168.0.2`**

5. Error Exit

Responses for error exit are set in the handling variable $action. Depending on the value $action is given, the relevant response code will be sent to the caller.

Examples of Response Codes_

400 = Bad Request
401 = Unauthorized
403 = Forbidden
404 = Not Found (destination cannot be found)
406 = Not Acceptable
486 = Busy Here (busy signal)
603 = Decline (refused, rejected)

5.2.1. If a MESSAGE request is received, return the code 400 (bad request)

Matching Patterns	Deploy Patterns

`$request=^MESSAGE`	**`$action=400`**

5.2.2. If a REGISTER request from the IP Address starts from "192.168" return the code 401 (unauthorized)

Matching Patterns	Deploy Patterns
`$request=^REGISTER` `$addr=192\.168\.`	**`$action=401`**

5.2.3. If the callee's Domain is "gw", return the code 403 (forbidden)

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=@gw[ >;]*`	**`$action=403`**

5.2.4. If the time a request is received is between 12:00 pm and 1:59 pm, return the code 404 (destination cannot be found)

Matching Patterns	Deploy Patterns
`$request=^INVITE` `$time=^1[23]:`	**`$action=404`**

5.2.5. If the number of forwards allowed is less than 5, return the code 406 (not acceptable)

Matching Patterns	Deploy Patterns
`$request=^INVITE` `max-forwards=^[0-5]$`	**`$action=406`**

5.2.6. If the callee isn't registered with the database, return the code 486

Matching Patterns	Deploy Patterns
`$request=^INVITE` `$registered=false`	**`$action=486`**

5.2.7. If the callee's User Name is "user", return the code 603 (refused)

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:user@`	**`$action=603`**

6. Prefix

The routing destination is set based on the callee's SIP-URI User Name prefix.

6.2.1. If the prefix is "0", route the call to "gw"

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:0(.+)@`**	`to=sip:%1@gw`

6.2.2. If the prefix is "6"- "9", route the call to "gw"

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`to=sip:[6-9](.+)@`**	`to=sip:%1@gw`

7. Load Balancing

With Enterprise SIP Server, Load Balancing is possible through various methods.

7.1. Load Balancing based on the Caller's IP Address

7.1.1. Load Balancing based on the last digit of the caller's IP Address

Matching Patterns	Deploy Patterns
$request=^INVITE **$addr=[0-4]$** to=sip:(.+)@	to=sip:%1@gw1
$request=^INVITE **$addr=[5-9]$** to=sip:(.+)@	to=sip:%1@gw2

- ✓ *If the last (rightmost) digit of the caller's IP Address is between 0 and 4, the call will be routed to "gw1". If the last (rightmost) digit is between 5 and 9, the call will be routed to "gw2".*

7.2. Load Balancing based on the Caller's SIP-URI

7.2.1. Load Balancing based on the caller's Domain Name

Matching Patterns	Deploy Patterns
$request=^INVITE **from=sip:(.+)@domain1**	to=sip:%1@gw1
$request=^INVITE **from=sip:(.+)@domain2**	to=sip:%1@gw2

- ✓ *If the caller's SIP-URI Domain Name is "domain 1", the call is routed to "gw1". If the Domain Name is "domain 2", the call is routed to "gw2".*

7.2.2. Load Balancing based on the caller's User Name

Matching Patterns	Deploy Patterns
$request=^INVITE **from=sip:[0-4].+@**	to=sip:%1@gw1
$request=^INVITE **from=sip:[5-9].+@**	to=sip:%1@gw2

- ✓ *If the caller's SIP-URI User Name prefix is between 0 and 4, route the call to "gw1". If the prefix is between 5 and 9, route the call to "gw2".*

7.3. Load Balancing based on Time

7.3.1. Load Balancing by switching 3 destinations every second

Matching Patterns	Deploy Patterns
$request=^INVITE **$time=^(..:..:.[0369])**	$target=pbx1
$request=^INVITE **$time=^(..:..:.[147])**	$target=pbx2
$request=^INVITE **$time=^(..:..:.[258])**	$target=pbx3

7.3.2. Load Balancing depending on whether the time is between 10 am and 6 pm or another time slice

Matching Patterns	Deploy Patterns
$request=^INVITE **$time=^(1[0-7]:..:..)"**	to=sip:ivr1@domain
$request=^INVITE	to=sip:ivr2@domain

7.4. Load Balancing based on the Session ID

The Session ID is stored in the server and a unique number is created each time a new session is started. The Session ID can be found in the conditional variable `$sid`.

7.4.1. Load Balancing based on whether the Session ID is odd or even

Matching Patterns	Deploy Patterns

`$request=^INVITE`
`$sid=[13579]$`
`to=sip:(.+)@`
	`to=sip:%1@gw1`

`$request=^INVITE`
`$sid=[24680]$`
`to=sip:(.+)@`
	`to=sip:%1@gw2`

7.5. Load Balancing based on the Caller's Port Number

7.5.1. Load Balancing based on whether the caller's Port Number is odd or even

Matching Patterns	Deploy Patterns
`$request=^INVITE` **`$port=[13579]$`**	`$target=pbx1`
`$request=^INVITE` **`$port=[24680]$`**	`$target=pbx2`

7.6. Load Balancing based on the Callee's SIP-URI

7.6.1. Load Balancing based on the callee's Domain Name

Matching Patterns	Deploy Patterns
$request=^INVITE **to=sip:(.+)@domain1**	to=sip:%1@gw1
$request=^INVITE **to=sip:(.+)@domain2**	to=sip:%1@gw2

7.6.2. Load Balancing based on the callee's User Name prefix

Matching Patterns	Deploy Patterns
$request=^INVITE **to=sip:(911)@**	to=sip:%1@urgent
$request=^INVITE **to=sip:(81.+)@**	to=sip:%1@gwjp
$request=^INVITE **to=sip:(1.+)@**	to=sip:%1@gwus

7.6.3. Load Balancing based on the callee's User Name prefix

Matching Patterns	Deploy Patterns
$request=^INVITE **to=sip:1(.+)@**	to=sip:%1@office1
$request=^INVITE **to=sip:2(.+)@**	to=sip:%1@office2
$request=^INVITE **to=sip:3(.+)@**	to=sip:%1@office3

7.7. Load Balancing based on the Number of Forwards (Hop number)

7.7.1. Load Balancing based on the number of forwards allowed

Matching Patterns	Deploy Patterns

`$request=^INVITE`
`max-forwards=^[0-9]$`
`to=sip:(.+)@`
	`to=sip:%1@proxy1`

`$request=^INVITE` `to=sip:(.+)@`	`to=sip:%1@proxy2`

- ✓ *If the number of forwards allowed is 9 or less, calls will be routed to proxy1, otherwise calls will be routed to proxy2.*

8. NAT Traversal Functionality

Enterprise SIP Server offers NAT Traversal Functionalities for SIP. When 2 clients each on a different network form a connection, the NAT functionality will automatically be set active and you don't need to set anything. You can also set the NAT Traversal Functionality for each routing in the Dial Plan.

When the NAT traversal functionality is active, information within a SIP packet is changed and the Enterprise SIP Server hides each client's IP Address. Tunneling of RTP and RTCP packets for sending voice will also occur.

Therefore, NAT Traversal functionality not only resolves the network connection for the call, but also provides a level of security by making it difficult to identify individual clients.

8.1. Changing between NAT Traversal ON/OFF

Even though NAT Traversal functionality is automatically executed, it is possible to set the NAT functionality in the Dial Plan for each call.

8.1.1. If the callee's Domain is "domain", set the NAT Traversal functionality to active

Matching Patterns	Deploy Patterns
`$request=^INVITE to=@domain[ >;]*`	**`$nat=true`**

8.1.2. If the caller's IP Address is 192.168.0.5, set the NAT Traversal functionality to inactive

Matching Patterns	Deploy Patterns
`$request=^INVITE $addr=192\.168\.0\.5$`	**`$nat=false`**

8.2. Setting up the Network Interface

The Enterprise SIP Server can automatically acquire the Network Interface found by the OS, setup by the user isn't necessary. If the Enterprise SIP Server is executed on a Unix-type OS, the network information will be taken from the /etc/hosts file.

If routing requires port forwarding or you don't want to use the interfaces found by the OS, setup using the items shown must be set.The NAT traversal functionality will select the most optimal interface from the list of interfaces that are set. If a network interface is set in the Dial Plan, it will have priority.

The interface for any call should be set within the "Network settings" menu in the Enterprise SIP Server administration tool window.

Parameter	Setup Value
Interface address 1_3	Interface to use

✓ *For individual connections, you need to assign the network interface in the Dial Plan individually.*

8.2.1. If the callee's domain is "gw", the interface 192.168.1.100 will be used for sending packets to the callee

Matching Patterns	Deploy Patterns
`$request=^INVITE to=@gw[ >;]*`	**`$ifdst=192.168.1.100`**

8.2.2. If the caller's IP Address is 192.168.0.5, the receiving interface will use the settings of "Interface address 2"

Matching Patterns	Deploy Patterns
`$request=^INVITE $addr=192\.168\.0\.5$`	**`$ifdst=&2`**

8.2.3. If the callee's User Name prefix is 10, the network interface will use 192.168.3.2 is used for sending packets to the caller

Matching Patterns	Deploy Patterns
`$request=^INVITE to=sip:10\.`	**`$ifsrc=192.168.3.2`**

8.3. RTP Stream Tunneling

By using RTP stream (RTP and RTCP packets) tunneling (goes through a SIP server), you can talk through different networks. When NAT Traversal is set active, RTP stream tunneling will occur automatically. Also each RTP stream tunneling can be switched with every connection.

The setting for any call is set in the "RTP exchanger" inside the "Configuration" menu in the "Enterprise SIP Server Administration Tool" menu.

Parameter	Setup Value
RTP relay	ON/OFF (RTP Stream Tunnel is active or inactive)

✓ *For individual connections, you need to set the RTP stream tunneling in the Dial Plan individually.*

8.3.1. If the caller's IP Address is 192.168.0.5, the RTP Stream Tunneling will be established

Matching Patterns	Deploy Patterns

`$request=^INVITE` `$addr=192\.168\.0\.5$`	**`$rtp=true`**

8.3.2. If the callee's Domain is "domain", the RTP Stream Tunneling won't be established

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=@domain[ >;]*`	**`$rtp=false`**

8.4. Changing the SIP-URI

Routing failure may occur if each local SIP-URI is exposed to external network when calls are connected across different local network. You may need to hide the SIP-URI to avoid this and for security reasons.

8.4.1. If the callee's Domain is "domain", change the SIP-URI

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:(.+)@domain1[ >;]*`	**`$replaceurl=true`** `to=sip:%1@domain2`

8.4.2. If the callee's SIP-URI is "sip:ivr@domain", don't change the SIP-URI

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:ivr@domain[ >;]*`	**`$replaceurl=false`**

9. Authentication

The authentication settings in Enterprise SIP Server Admintool >[Config]menu >[SIP(General)] >[Authentication] is applied for all SIP requests.
You can change the setting for specific session using Dial Plan.

Variables:

`$auth=true`	Enable authentication
`$auth=false`	Disable authentication

9.4.1. If the request is INVITE and if the destination number starts from 0, the authentication is disabled.

Matching Patterns	Deploy Patterns
`$request=^INVITE` `to=sip:0.+@`	`$auth = false`

10. Changing [Config] value for specific sessions

The values set in Enterprise SIP Server Admintool ->[Config] menu are applied to all sessions. You can change them using Dial Plan by specifying the session type.

10.4.1. For the all sessions from the user "999", set 10 seconds for INVITE timeout.

Matching Patterns	Deploy Patterns
`$request=^INVITE from=sip:999@`	`&net.sip.timeout.inviting=10000`

Following parameters can be used in Deploy Patterns.

Handling Variable Name	Corresponding field in [Config] menu	Value
&net.sip.addreordroute	[SIP(General)]->[Add Record-Route header]	on/off
&net.registrar.upper.allow	[SIP(Advanced)]->[Upper Registration]->[On/Off]	on/off
&net.registrar.thru.allow	[SIP(Advanced)]->[Thru Registration]->[On/Off]	on/off
&net.sip.timeout.inviting	[SIP(Advanced)]->[Timeout]->[INVITE Timeout (ms)]	ms
&net.sip.timeout.ringing	[SIP(Advanced)]->[Timeout]->[Ringing Timeout (ms)]	ms
&net.sip.timeout.talking	[SIP(Advanced)]->[Timeout]->[Talking Timeout (ms)]	ms
&net.sip.timeout.bye	[SIP(Advanced)]->[Timeout]->[BYE Timeout (ms)]	ms
&net.registrar.upper.timeout	[SIP(Advanced)]->[Timeout]->[Upper/Thru Timeout (ms)]	ms
&net.rtp.responseport	[RTP]->[RTP exchanger] Specify a RTP listening port	
&net.rtp.session.timeout	[RTP]->[Timout]->[RTP Session Timeout(ms)]	ms
&net.sip.addreordroute.lr	Set "off" when connecting the other SIP Server which can't handle "lr".	on/off

11. Others

11.1. Changing the Forward Number

The allowed forward number can be changed in the Max-Forwards: header field. Calls (Sessions) can go through its number of proxies or similar methods between clients. The number of Max-Forwards: will be decreased at every method (proxy server). If it reaches 0, an error will be returned.

11.1.1. If the callee isn't registered, the allowed forward number is set to 15

Matching Patterns	Deploy Patterns
`$request=^INVITE` `$registered=false`	**`max-forwards=15`**

11.2. Changing the User Agent

The User Agent Name can be changed in the User-Agent: header field. Normally a User Agent Name contains the caller's Client Name and version number.

11.2.1. If the caller's User Name is "pbx", the User Agent Name is set as "user"

Matching Patterns	Deploy Patterns
`$request=^INVITE` `from=sip:pbx@`	**`user-agent=user`**

11.2.2. If the caller's User Agent Name is "Gateway", the User Agent Name will be removed

Matching Patterns	Deploy Patterns
`$request=^INVITE` `user-agent=^GATEWAY$`	**`user-agent=`**

11.3. Adding a Record-Route:

You can set to add Record-Route header field for any session. Also you can set to add it for individual connection.

"SIP exchanger" within the "Configuration" menu in the "Enterprise SIP Server Administration Tool" menu is used for any session, if set to "ON", the SIP Server IP Address is entered in the Record-Route: header field.

Parameter	Setup Information
Add Record-Route header	Record-Route: / Route: ON/OFF _Add or not_

✓ *For some connections which do not need Record-Route header, set this in the Dial Plan.*

11.3.1. If the callee's Domain is "gw", Record-Route isn't added

Matching Patterns	Deploy Patterns
$request=^INVITE to=@gw[>;]*	**record-route=**