



# **Brekeke R14**

## SIP Trunk Provisioning Guide

Brekeke v2.  
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## Brekeke R14 SIP Trunk Provisioning Guide

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### ABSTRACT

Brekeke is a java-based PBX solution that includes and embedded/bundled SIP proxy and SIP registrar server. Supported on all major operating systems—including Microsoft Windows, Solaris, FreeBSD, and Linux—Brekeke version 2.x requires Apache Tomcat 5.5.12 or later and the Sun Java 6 runtime environment.

IMPORTANT: Sun Java is highly recommended

This document covers Brekeke v2.x PBX deployed downstream of a third-party firewall and downstream of an Edgemark SIP application-layer gateway.

NOTE: The Broadsoft Identity/Device profile required for Brekeke v2.x PBX and the Brekeke v2.x PBX-based systems listed is the “Generic SIP Trunk Single Registration” Identity/Device profile.

For further information regarding Brekeke v2.x features, see <http://www.brekeke.com/>.

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## Brekeke R14 SIP Trunk Provisioning Guide

### INTRODUCTION

The following document covers MegaPath R14 SIP trunk configuration settings for the Brekeke v2.x PBX platform. This document covers MegaPath R14 SIP trunk configurations in conjunction with a third-party firewall and with an Edgemark 4500-series SIP application-layer gateway.

The R14 Identity/Device profile required for Brekeke v2.x PBX, as well as the Brekeke v2.x PBX-based systems mentions is the "Generic SIP Trunk Single Registration" Identity/Device profile.

The Brekeke v2.x PBX is bundled with the Brekeke SIP proxy and registrar server; given the presence of the embedded SIP proxy and registrar server, care should be taken in noting the end-to-end network topology to ensure that all network elements in the end-to-end call path that are involved in SIP-header manipulation of any type are accounted for in the PBX and SIP configuration files of the Brekeke PBX platform.

### 1. STANDARD FIREWALL

This configuration features a Brekeke v2.x PBX build deployed behind a standard, third-party firewall. The firewall is configured to forward the SIP and the RTP range from the firewall WAN IP address to the internal IP address of the Brekeke v2.x PBX server NATd behind the firewall.



Figure 1.1 Brekeke PBX LAN topology, third-party firewall

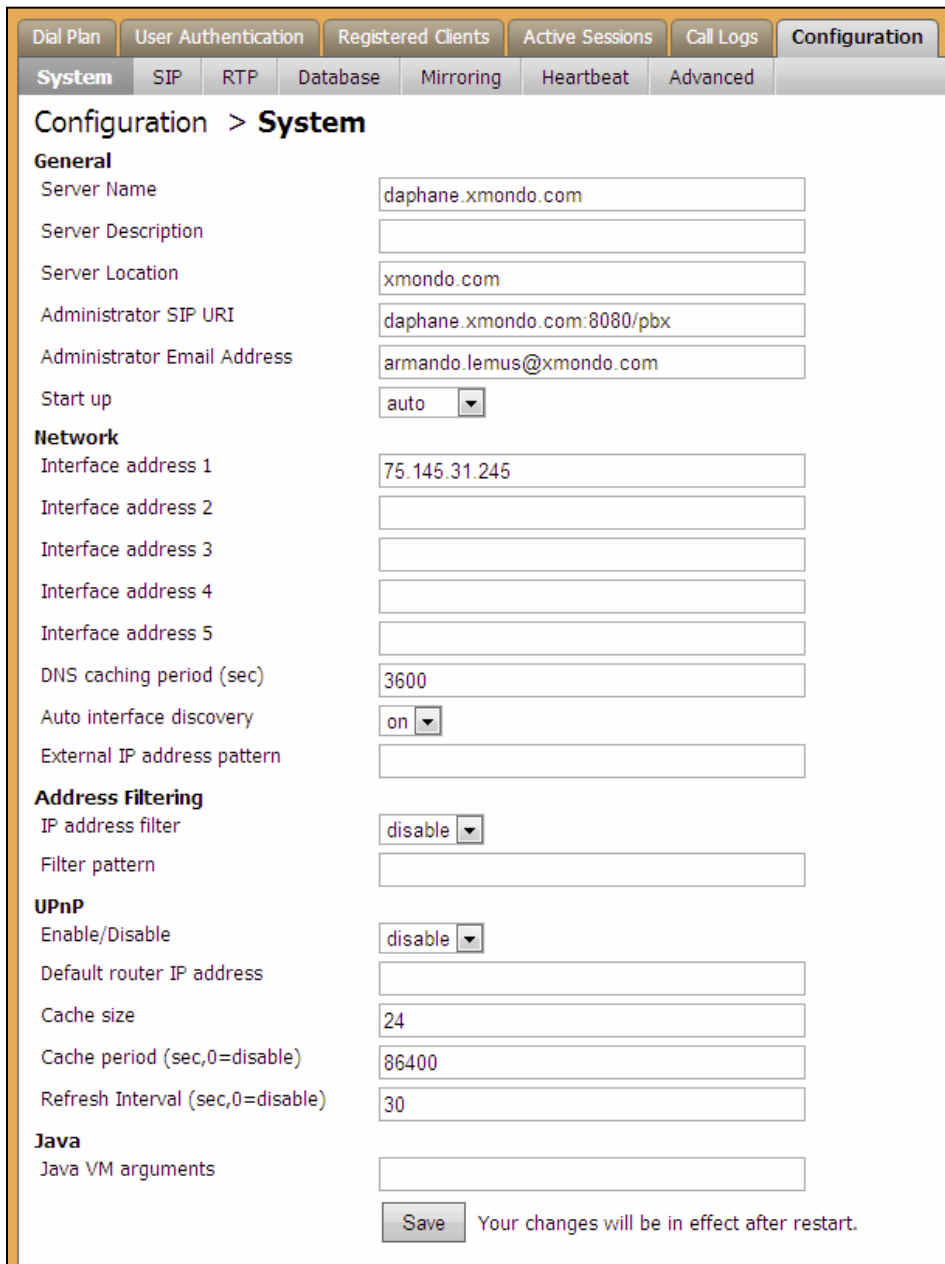
### IMPORTANT NOTES:

- The Brekeke PBX RTP/UDP port range (highly customizable) MUST be port-forwarded for inbound RTP to the Brekeke PBX; SIP/UDP 5060 MUST be port-forwarded for inbound SIP transactions to the Brekeke PBX.
- The third-party firewall SHOULD support static NAT for all outbound and inbound UDP traffic to mitigate one-way audio.
- If one-way audio is still an issue with a firewall known to support static NAT, then ensure that all SIP helper or SIP proxy services are disabled in the firewall.

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### 1.1 SIP Server Configuration

1. Open a Web browser and log in to the Brekeke platform (http://<hostname>:8080/pbx).
2. Go to the SIP server configuration section.
3. Select the **System** tab.
4. In the **Network>Interface address 1** field, enter the IP address of the WAN-side interface of the third-party firewall; this is the WAN-side interface that all incoming SIP and RTP traffic will traverse to reach the Brekeke SIP server.



The screenshot displays the 'Configuration > System' page in the Brekeke management console. The page is organized into several sections with corresponding configuration fields:

- General:**
  - Server Name: daphane.xmondo.com
  - Server Description: (empty)
  - Server Location: xmondo.com
  - Administrator SIP URI: daphane.xmondo.com:8080/pbx
  - Administrator Email Address: armando.lemus@xmondo.com
  - Start up: auto
- Network:**
  - Interface address 1: 75.145.31.245
  - Interface address 2: (empty)
  - Interface address 3: (empty)
  - Interface address 4: (empty)
  - Interface address 5: (empty)
  - DNS caching period (sec): 3600
  - Auto interface discovery: on
  - External IP address pattern: (empty)
- Address Filtering:**
  - IP address filter: disable
  - Filter pattern: (empty)
- UPnP:**
  - Enable/Disable: disable
  - Default router IP address: (empty)
  - Cache size: 24
  - Cache period (sec,0=disable): 86400
  - Refresh Interval (sec,0=disable): 30
- Java:**
  - Java VM arguments: (empty)

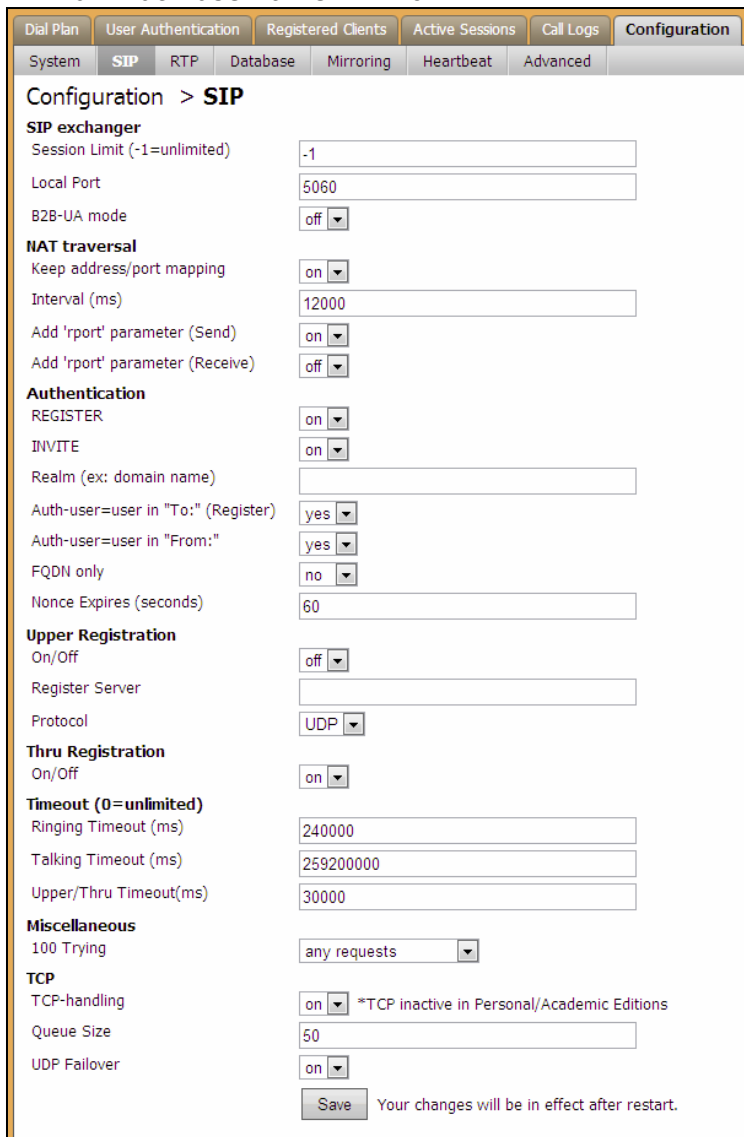
At the bottom of the form, there is a 'Save' button and a message: 'Your changes will be in effect after restart.'

Figure 1.1.1 SIP Server System Configuration

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5. Under the SIP tab, the following parameters should be enabled:

- NAT traversal
  - Keep address/port mapping
  - Add 'rport' parameter (Send)
- Authentication
  - REGISTER
  - INVITE
  - Auth-username in "To:" (Register)
  - Auth-username in "From:"



The screenshot shows the 'Configuration > SIP' settings page. The 'SIP' tab is selected, and various parameters are configured as follows:

Section	Parameter	Value
SIP exchanger	Session Limit (-1=unlimited)	-1
	Local Port	5060
	B2B-UA mode	off
NAT traversal	Keep address/port mapping	on
	Interval (ms)	12000
	Add 'rport' parameter (Send)	on
	Add 'rport' parameter (Receive)	off
Authentication	REGISTER	on
	INVITE	on
	Realm (ex: domain name)	
	Auth-user=user in "To:" (Register)	yes
	Auth-user=user in "From:"	yes
	FQDN only	no
	Nonce Expires (seconds)	60
Upper Registration	On/Off	off
	Register Server	
	Protocol	UDP
Thru Registration	On/Off	on
Timeout (0=unlimited)	Ringing Timeout (ms)	240000
	Talking Timeout (ms)	259200000
	Upper/Thru Timeout(ms)	30000
Miscellaneous	100 Trying	any requests
TCP	TCP-handling	on *TCP inactive in Personal/Academic Editions
	Queue Size	50
	UDP Failover	on

At the bottom of the configuration page, there is a 'Save' button and a note: "Your changes will be in effect after restart."

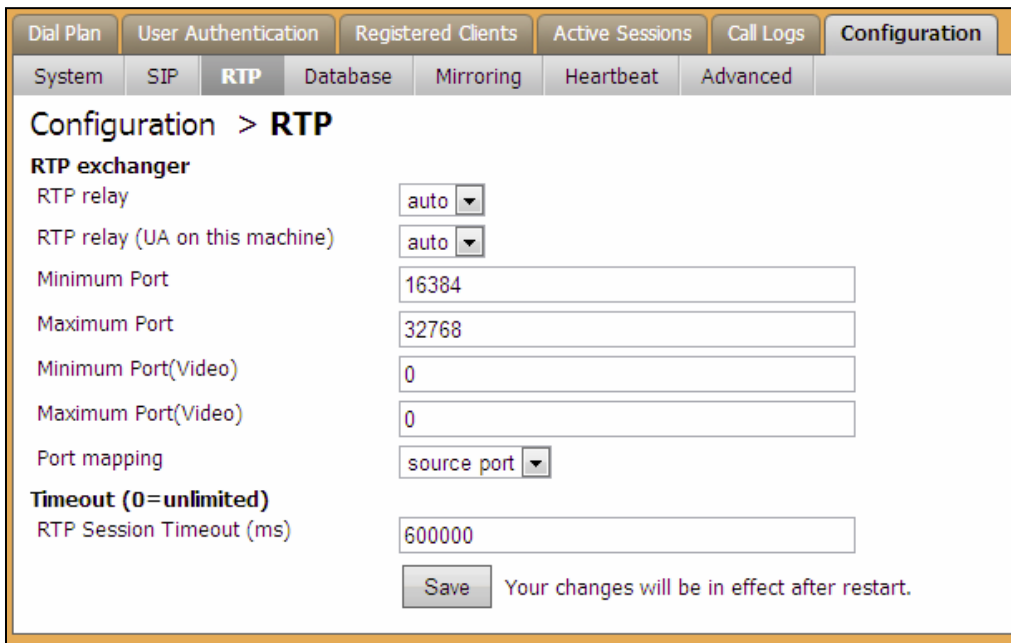
Figure 1.1.2 SIP Server SIP Configuration

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6. Under the **RTP** tab, ensure all RTP relay functions are set to **auto**
7. Set the **Minimum** and **Maximum** port numbers for the RTP port range.

IMPORTANT: Ensure that the same RTP port range is specified in the port-forwarding RTP rule of the third-party firewall.

8. Click **Save** to save all settings.
9. Restart the SIP server from the main screen.



The screenshot shows the configuration interface for the SIP server. The 'Configuration' tab is selected, and the 'RTP' sub-tab is active. The settings are as follows:

Configuration > RTP	
<b>RTP exchanger</b>	
RTP relay	auto
RTP relay (UA on this machine)	auto
Minimum Port	16384
Maximum Port	32768
Minimum Port(Video)	0
Maximum Port(Video)	0
Port mapping	source port
<b>Timeout (0=unlimited)</b>	
RTP Session Timeout (ms)	600000
<input type="button" value="Save"/> Your changes will be in effect after restart.	

Figure 1.1.2 SIP Server RTP Configuration

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### 1.2 SIP Trunk Registration and Auto-Route Selection Rules

The R14 SIP trunk credentials and route selections for inbound and outbound calls (filtered based on regular expressions) are configured in the PBX under the **ARS** tab.

1. Access the PBX configuration tabs from the main page.
2. Select the **ARS** tab.
3. Go to the **Settings** page to input data into the **Route Template**.
4. In the **General** section, enter the following parameters:
  - **Register URI:** sip:username@hostname
    - For example, sip:demotrunk02@lab-1-siptrunk-a.voice.speakeasy.net
  - **Proxy Address:** hostname
    - For example, lab-1-siptrunk-a.voice.speakeasy.net
  - **User:** username
    - For example, demotrunk02
    - Password: username password
  - **Password:** username password
    - For example, Uyt6E3z12
  - **Realm:** speakeasy.net

### ARS > Route Template

Note: The information on this page is read only. Please go to the Settings page to make changes.

**General**

<b>Route name</b>	<input type="text" value="Megapath"/>		
<b>Description</b>	<div style="border: 1px solid gray; height: 60px;"></div>		
<b>Registration Status</b>		Registered	<input type="button" value="Unregister"/>

<b>Type</b>	<input type="text" value="Type A"/>	<b>Group</b>	<input type="text"/>
<b>Register URI</b>	<input type="text" value="sip:demotrunk02@lab-1-siptrunk-a"/>	<b>Realm</b>	<input type="text" value="speakeasy.net"/>
<b>Proxy Address</b>	<input type="text" value="lab-1-siptrunk-a.voice.speakeasy.r"/>	<b>Register Expire (sec)</b>	<input type="text" value="3600"/>
<b>User</b>	<input type="text" value="demotrunk02"/>	<b>Register Update Period (%)</b>	<input type="text" value="90"/>
<b>Password</b>	<input type="text" value="...."/>	<b>Session interval (ms)</b>	<input type="text"/>
<b>LineKey</b>	<input type="checkbox"/>		

Figure 1.2.1 ARS Settings, Route Template, General section

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- Configure the **Patterns – IN** section. This example assumes that the user “1500” and extension “1500” have been configured in the system and that a SIP phone is registered against the Brekeke v2.x embedded SIP server.

Patterns - IN		
IN - 1	Matching patterns	Deploy patterns
Priority <input type="text" value="50"/>	From <input @"="" type="text" value="sip:(.+)"/>	From <input type="text"/>
Max Sessions <input type="text" value="2"/>	To <input type="text" value="3125334875"/>	To <input type="text" value="1500"/>
	Plugin <input type="text"/>	Custom <input type="text"/>
	Param <input type="text"/> Return	
	<input type="checkbox"/> Apply to Request URI instead of To	
<b>Parameters</b>		
	RTP relay <input type="text" value="default"/>	Codec Priority <input type="text"/>
	Use Remote Preferred Codec <input type="text" value="default"/>	Block SIP INFO (DTMF) <input type="text" value="no"/>
	Send RTCP <input type="text" value="off"/>	SDP 18x <input type="text" value="default"/>

Figure 1.2.2 ARS Settings, Route Template, Patterns – IN section

In this example, we are accepting all calls from all providers. The inbound DID in the **To** field is a DID associated with the SIP trunk.

In the **Matching patterns** section:

- From: sip:(.+)"@
- To: 3125334875

In the **Deploy patterns** section:

- To: 1500

NOTE: Leave the **From** field blank to allow all caller IDs.

The SIP trunk registration status can also be assessed in a secure shell or console session by issuing the following command at the command prompt to access the Asterisk command-line interface:

```
[Brekeke IP PBX]root # asterisk -vvvr
```



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6. Configure the **Patterns – OUT** section.

Patterns - OUT			
OUT - 1	Matching patterns		Deploy patterns
Priority <input type="text" value="50"/>	From <input type="text"/>	From <input type="text" value="&lt;sip:3125334875@lab-1-siptrunk-a.voice.s"/>	
Max Sessions <input type="text" value="5"/>	To <input type="text" value="sip:([0-9]{11,25})@"/>	To <input type="text" value="sip:\$1@lab-1-siptrunk-a.voice.speakeasy.r"/>	
	User <input text"="" type="text" value="^.+\$/td&gt; &lt;td&gt;Target &lt;input type="/>		
	Plugin <input type="text"/>	DTMF <input type="text"/>	
	Param <input type="text"/> Return	Confirm <input type="text"/> Key <input type="text" value="5"/>	
	<input type="text"/>	Custom <input type="text"/>	
<b>Parameters</b>			
RTP relay	<input type="text" value="default"/>	Codec Priority	<input type="text"/>
Block SIP INFO (DTMF)	<input type="text" value="no"/>	Send RTCP	<input type="text" value="off"/>
Session Timer(sec, 0=disable)	<input type="text" value="0"/>	100rel	<input type="text" value="off"/>
Next route on failure	<input type="text" value="no"/>	Disable on registration failure	<input type="text" value="no"/>
Response timeout (ms)	<input type="text" value="-1"/>	Error codes	<input type="text" value="500"/>
Recovery time (ms)	<input type="text" value="0"/>	Disable on failure	<input type="text" value="This route"/>

Figure 1.2.3 ARS Settings, Route Template, Patterns –OUT section

7. In the **Matching patterns** section:
- **To:** sip:([0-9]{11,25})@  
NOTE: This is a catch-all regular expression pattern to capture local and toll-free dial patterns.
  - **User:** ^.+\$/td>
- NOTE: This is a catch-all regular expression pattern to capture all callers.
8. In the **Deploy patterns** section:
- **From:** <sip:caller-ID number@hostname>  
NOTE: The < and > brackets must be included.
    - For example, <sip:312534875@lab-1-siptrunk-a.voice.speakeasy.net>
  - **To:** sip:\$1@hostname  
NOTE: This is a catch-all for all called numbers.
    - For example, sip:\$1@lab-1-siptrunk-a.voice.speakeasy.net
9. Save and apply the Route template.

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Once the ARS route is applied, you should be able to see a successful SIP trunk registration under the **ARS>View>General>Registration Status** field.

**ARS > Route Template**

Note: The information on this page is read only. Please go to the Settings page to make changes.

**General**

<b>Route name</b>	Megapath
<b>Description</b>	<div style="border: 1px solid gray; height: 40px;"></div>
<b>Registration Status</b>	Registered <input type="button" value="Unregister"/>

Figure 1.2.4 ARS View, Route Template, General section, Registration Status

### 1.3 SIP Server Dial Plan

After configuring the auto-route selection policy, the SIP server dial plan must be modified for the SIP server to route inbound calls from MegaPath to the IP PBX. In this section, for the given network topology, we will make use of the IP address of the MegaPath session border controller and the WAN IP address of the third-party firewall.

**IMPORTANT:**

- The SIP server uses regular expressions to match inbound traffic and as such, all "." characters in an IP address must be backslash-escaped in the following manner:  
     64\.17\.101\.137
- Brekeke v2.x SIP server dial plan only supports hard-coded IP addresses in the SIP server dial plan; however, any number of IP addresses can be piped together with the "|" sign, or "pipe" sign. For example,  
     64\.17\.101\.137|12\.24\.56\.122|192\.168\.1\.1

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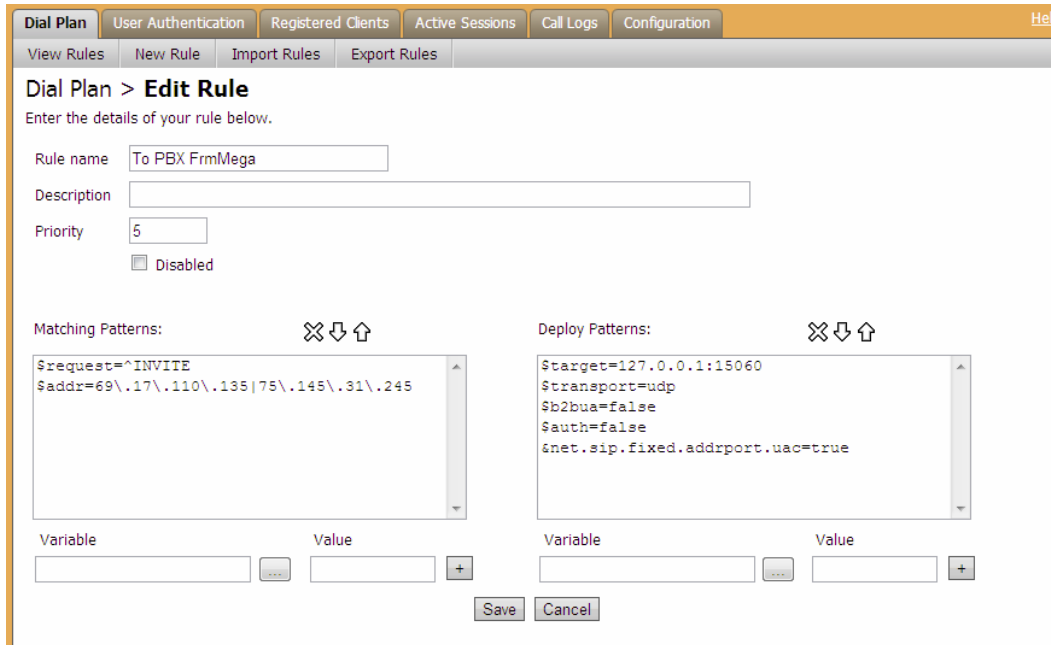


Figure 1.3.1 SIP server, Dial Plan, Edit Rule

Create a new rule by copying any of the existing **To PBX...** rules, renaming the copied rule to something recognizable in relation to MegaPath.

Move this new **To PBX...** up in order of priority such that it is evaluated before all other **To PBX...** rules.

In the **Matching Patterns** section, make sure that only the following patterns are present:

- \$request=^INVITE
- \$addr=IP ADDRESS OF MEGAPATH SBC(s) | WAN IP ADDRESS OF FIREWALL

For example,

```
$request=^INVITE
$addr=69\.17\.110\.135|66\.92\.30\.168|75\.145\.31\.245
```



Figure 1.3.2 IP address scheme for SIP server Dial Plan

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In the **Deploy Patterns** section, the following defaults will suffice:

NOTE: In the deploy pattern rules, IP address . characters DO NOT need to be backslash-escaped.

```
$target=127.0.0.1:15060
$transport=udp
$b2bua=false
$auth=false
&net.sip.fixed.addrport.uac=true
```

Save and apply the new **To PBX...** rule. Restart the SIP server from the main menu screen.

### 2. EDGEMARC SIP ALG



Figure 2.1 Brekeke LAN topology, Edgemarc SIP ALG

IMPORTANT: With and Edgemarc SIP ALG, the Brekeke SIP server and the Edgemarc are configured in a peer-to-peer relationship, especially in terms of auto-route selection for inbound and outbound call patterns and the Brekeke SIP server dial plan.

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### 2.1 SIP Server Configuration

1. Open a Web browser
2. Log in to the Brekeke platform (http://<hostname>:8080/pbx) and go to the SIP server configuration section.
3. Select the **System** tab.
4. Under **Network>Interface address 1**, input the IP address of the WAN-side interface of the Edgemarc 14 SIP ALG; this is the WAN-side interface that all incoming SIP and RTP traffic will traverse to reach the Brekeke SIP server.

Configuration > System	
<b>General</b>	
Server Name	<input type="text" value="daphane.xmondo.com"/>
Server Description	<input type="text"/>
Server Location	<input type="text" value="xmondo.com"/>
Administrator SIP URI	<input type="text" value="daphane.xmondo.com:8080/pbx"/>
Administrator Email Address	<input type="text" value="armando.lemus@xmondo.com"/>
Start up	<input type="text" value="auto"/> ▼
<b>Network</b>	
Interface address 1	<input type="text" value="75.145.31.243"/>
Interface address 2	<input type="text"/>
Interface address 3	<input type="text"/>
Interface address 4	<input type="text"/>
Interface address 5	<input type="text"/>
DNS caching period (sec)	<input type="text" value="3600"/>
Auto interface discovery	<input type="text" value="on"/> ▼
External IP address pattern	<input type="text"/>

Figure 2.1.1 SIP Server System Configuration

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5. Under the **SIP** tab, the following parameters should be enabled:
  - **NAT traversal**
    - Keep address/port mapping
    - Add **'rport'** parameter (Send)
  - **Authentication**
    - REGISTER
    - INVITE
    - Auth-username in **To** (Register)
    - Auth-username in **From**

Dial Plan	User Authentication	Registered Clients	Active Sessions	Call Logs	Configuration	
System	<b>SIP</b>	RTP	Database	Mirroring	Heartbeat	Advanced
<b>Configuration &gt; SIP</b>						
<b>SIP exchanger</b>						
Session Limit (-1=unlimited)	<input type="text" value="-1"/>					
Local Port	<input type="text" value="5060"/>					
B2B-UA mode	<input type="button" value="off"/>					
<b>NAT traversal</b>						
Keep address/port mapping	<input type="button" value="on"/>					
Interval (ms)	<input type="text" value="12000"/>					
Add 'rport' parameter (Send)	<input type="button" value="on"/>					
Add 'rport' parameter (Receive)	<input type="button" value="off"/>					
<b>Authentication</b>						
REGISTER	<input type="button" value="on"/>					
INVITE	<input type="button" value="on"/>					
Realm (ex: domain name)	<input type="text"/>					
Auth-user=user in "To:" (Register)	<input type="button" value="yes"/>					
Auth-user=user in "From:"	<input type="button" value="yes"/>					
FQDN only	<input type="button" value="no"/>					
Nonce Expires (seconds)	<input type="text" value="60"/>					
<b>Upper Registration</b>						
On/Off	<input type="button" value="off"/>					
Register Server	<input type="text"/>					
Protocol	<input type="button" value="UDP"/>					
<b>Thru Registration</b>						
On/Off	<input type="button" value="on"/>					
<b>Timeout (0=unlimited)</b>						
Ringing Timeout (ms)	<input type="text" value="240000"/>					
Talking Timeout (ms)	<input type="text" value="259200000"/>					
Upper/Thru Timeout(ms)	<input type="text" value="30000"/>					
<b>Miscellaneous</b>						
100 Trying	<input type="button" value="any requests"/>					
<b>TCP</b>						
TCP-handling	<input type="button" value="on"/> **TCP inactive in Personal/Academic Editions					
Queue Size	<input type="text" value="50"/>					
UDP Failover	<input type="button" value="on"/>					
<input type="button" value="Save"/> Your changes will be in effect after restart.						

Figure 2.1.2 SIP Server SIP Configuration

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6. Under the **RTP** tab, set ensure all RTP relay functions are set to **auto**.
7. Set the **Minimum** and **Maximum** port numbers for the RTP port range.
8. Save all settings and restart the SIP server from the main screen.

Dial Plan	User Authentication	Registered Clients	Active Sessions	Call Logs	Configuration	
System	SIP	<b>RTP</b>	Database	Mirroring	Heartbeat	Advanced

Configuration > RTP

**RTP exchanger**

RTP relay

RTP relay (UA on this machine)

Minimum Port

Maximum Port

Minimum Port(Video)

Maximum Port(Video)

Port mapping

**Timeout (0=unlimited)**

RTP Session Timeout (ms)

Your changes will be in effect after restart.

Figure 2.1.3 SIP server RTP Configuration

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### 2.2 SIP Trunk Registration and Auto-Route Selection Rules

1. In **Route Template>General**, enter the following parameters:
  - **Register URI:** sip: username@LAN IP ADDRESS OF EDGEMARC
    - For example, sip:demotrunk02@172.16.248.1
  - **Proxy Address:** LAN IP ADDRESS OF EDGEMARC
    - For example, 172.16.248.1
  - **User:** username
    - For example, demotrunk02
  - **Password:** username password
    - For example, Uyt6E3z12
  - **Realm:** speakeasy.net

ARS > **Route Template**

General [Edit Variables](#)

<b>Route name</b>	Megapath		
<b>Description</b>			
<b>Disabled</b>	<input type="checkbox"/>		

<b>Type</b>	Type A ▾	<b>Group</b>	
<b>Register URI</b>	sip:demotrunk02@172.16.248.1	<b>Realm</b>	speakeasy.net
<b>Proxy Address</b>	172.16.248.1	<b>Register Expire (sec)</b>	3600
<b>User</b>	demotrunk02	<b>Register Update Period (%)</b>	90
<b>Password</b>	....	<b>Session interval (ms)</b>	
<b>LineKey</b>	<input type="checkbox"/>		

Figure 2.2.1 ARS Settings>Route Template>General section



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2. Configure the **Patterns – IN** section.

This example assumes that the user “1500” and extension “1500” have been configured in the system and that a SIP phone is registered against the Brekeke v2.x embedded SIP server.

In this example, we are accepting all calls from all providers. The inbound DID in the **To** field is a DID associated with the SIP trunk.

- **Matching patterns:**
  - **From:** sip:(.+)
  - **To:** 3125334875
- **Deploy patterns:**
  - **From:** Leave this field blank to allow all caller IDs.
  - **To:** 1500

Patterns - IN			
IN - 1	Matching patterns		Deploy patterns
Priority <input type="text" value="50"/>	From <input 2"="" type="text" value="sip:(.+)&lt;/input&gt;&lt;/td&gt; &lt;td colspan="/> From <input type="text"/>		
Max Sessions <input type="text" value="2"/>	To <input type="text" value="3125334875"/>	To <input type="text" value="1500"/>	
	Plugin <input type="text"/>	Custom <input type="text"/>	
	Param <input type="text"/> Return <input type="text"/>		
	<input type="checkbox"/> Apply to Request URI instead of To		
Parameters			
	RTP relay	<input type="text" value="default"/>	Codec Priority <input type="text"/>
	Use Remote Preferred Codec	<input type="text" value="default"/>	Block SIP INFO (DTMF) <input type="text" value="no"/>
	Send RTCP	<input type="text" value="off"/>	SDP 18x <input type="text" value="default"/>

Figure 2.2.2 ARS Settings, Route Template, Patterns –IN section

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3. Configure the **Patterns – OUT** section.
  - **Matching patterns** section:
    - **To:** sip:([0-9]{11,25})@  
NOTE: This is a catch-all regular expression pattern to capture local and toll-free dial patterns.
    - **User:** ^.+  
NOTE: This is a catch-all regular expression pattern to capture all callers.
  - **Deploy patterns** section:
    - **From:** <sip:caller-ID number@LAN IP ADDRESS BREKEKE PBX>  
NOTE: "<" and ">" brackets must be included.  
For example, <sip:312534875@172.16.248.165>
    - **To:** sip:\$1@IP ADDRESS OF EDGEMARC  
NOTE: This is a catch-all for all called numbers.  
For example, sip:\$1@172.16.248.1

OUT - 1		Matching patterns		Deploy patterns	
Priority <input type="text" value="50"/>		From <input type="text"/>		From <input type="text" value="&lt;sip:3125334875@172.16.248.165&gt;"/>	
Max Sessions <input type="text" value="5"/>		To <input type="text" value="sip:([0-9]{11,25})@"/>		To <input type="text" value="sip:\$1@172.16.248.1"/>	
<input type="checkbox"/> Disabled		User <input type="text" value="^.+"/>		Target <input type="text"/>	
<input type="button" value="Copy"/> <input type="button" value="Delete"/>		Plugin <input type="text"/>		DTMF <input type="text"/>	
		Param <input type="text"/> Return <input type="text"/>		Confirm <input type="text"/> Key <input type="text" value="5"/>	
				Custom <input type="text"/>	
Parameters					
RTP relay	<input type="text" value="default"/>	Codec Priority	<input type="text"/>		
Block SIP INFO (DTMF)	<input type="text" value="no"/>	Send RTCP	<input type="text" value="off"/>		
Session Timer(sec, 0=disable)	<input type="text" value="0"/>	100rel	<input type="text" value="off"/>		
Next route on failure	<input type="text" value="no"/>	Disable on registration failure	<input type="text" value="no"/>		
Response timeout (ms)	<input type="text" value="-1"/>	Error codes	<input type="text" value="500"/>		
Recovery time (ms)	<input type="text" value="0"/>	Disable on failure	<input type="text" value="This route"/>		

Figure 2.2.3 ARS Settings, Route Template, Patterns –OUT section

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- Save and apply the Route template. Once the ARS route is applied, you should be able to see a successful SIP trunk registration under the **ARS>View>General>Registration Status** field.

**ARS > Route Template**

Note: The information on this page is read only. Please go to the Settings page to make changes.

**General**

<b>Route name</b>	Megapath
<b>Description</b>	<div style="border: 1px solid gray; height: 50px;"></div>

**Registration Status**      Registered

Figure 2.2.4 ARS View, Route Template, General section, Registration Status.

### 2.3 SIP Server Dial Plan

After configuring the auto-route selection policy, the SIP server dial plan must be modified for the SIP server to route inbound calls from MegaPath to the IP PBX. In this section, for the given network topology, we will make use of the IP address of the MegaPath session border controller and the WAN IP address of the third-party firewall.

**IMPORTANT:**

- The SIP server uses regular expressions to match inbound traffic and as such, all "." characters in an IP address must be backslash-escaped in the following manner:

64\.17\.101\.137

- Brekeke v2.x SIP server dial plan only supports hard-coded IP addresses in the SIP server dial plan; however, any number of IP addresses can be piped together with the "|" sign, or "pipe" sign. For example,

64\.17\.101\.137|12\.24\.56\.122|192\.168\.1\.1

5	To PBX from SBC	<pre>\$request=^INVITE \$addr=69\.17\.110\.135 75\.145\.31\.243 172\.16\.248\.1</pre>	<pre>\$target=127.0.0.1:15060 \$transport=udp \$b2bua=false \$auth=false &amp;net.sip.fixed.addrport.uac=true</pre>
---	-----------------	---	---

Figure 2.3.1 SIP server, Dial Plan, Edit Rule

Create a new rule by copying any of the existing **To PBX...** rules, renaming the copied rule to something recognizable in relation to MegaPath.

Move this new **To PBX..."** up in order of priority such that it is evaluated before all other "To PBX..." rules.

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In the "Matching Patterns:" section, make sure that only the following patterns are present:

- \$request=^INVITE
- \$addr=IP ADDRESS OF MEGAPATH SBC(s) | WAN IP ADDRESS OF EDGEMARC | LAN IP ADDRESS OF EDGEMARC

For example,

```
$request=^INVITE
$addr=69.17.110.135|66.92.30.168|75.145.31.243|172.16.248.1
```

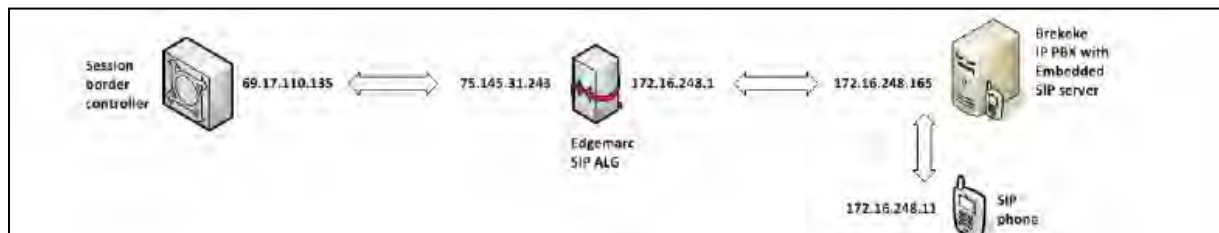


Figure 2.3.2 IP address scheme for SIP server Dial Plan

In the **Deploy Patterns** section, the following defaults will suffice:

NOTE: In the deploy pattern rules, IP address "." characters DO NOT need to be backslash-escaped.

```
$target=127.0.0.1:15060
$transport=udp
$b2bua=false
$auth=false
&net.sip.fixed.addrport.uac=true
```

Save and apply the new **To PBX...** rule. Restart the SIP server from the main menu screen.

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### 2.4 Edgemark SIP ALG SIP Trunk Configuration

1. Log in to the Edgemark SIP ALG. Under the **VoIP ALG** menu option, select the **SIP** option.
2. In the **SIP Server Address** field under **SIP Settings**, input the fully-qualified domain name or IP address of the SIP gateway or session border controller assigned to you by your VoIP service provider.
3. In the **SIP Server Port** field, input the SIP server port number.
4. Enable **Use Custom Domain**.
5. In the **SIP Server Domain** field, enter "**speakeasy.net**" and then click **Submit**.

[Help](#)

### **SIP Settings**

SIP protocol settings.

---

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

SIP Server Address:

SIP Server Port:

Use Custom Domain:

SIP Server Domain:

List of SIP Servers:

Enable Multi-homed Outbound Proxy Mode:

Enable Transparent Proxy Mode:

Limit Outbound to listed Proxies / SIP Servers:

Limit Inbound to listed Proxies / SIP Servers:

---

### **Stale Timer**

The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period.

Stale client time (m):

---

Registration Rate-Pacing parameters are available on the [Survivability page](#).

Figure 2.4.1 VoIP ALG, SIP Settings

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6. From the **SIP** sub-menu, select **Trunking**.
7. In the **Add a trunking device** menu box:
  - Set the **Action:** field to **Add new trunking device**.
  - In the **Name** field, input a recognizable name for the trunking device.
  - In the **Address** field, input the IP address of the Brekeke IP PBX server.
  - In the **Port** field, input the SIP port number of the Brekeke IP PBX server.
  - Click **Commit** to create the SIP trunking device

[Help](#)

### SIP Trunking

Configuration of SIP trunking devices.

---

### SIP Trunking devices

A SIP trunking device can be a PSTN gateway, or similar device, that does not issue REGISTER messages. Calls will be forwarded to the device based on the dial-plan rules below.

If VLANS are enabled, the SIP trunking device needs to be in the same VLAN as defined in the VoIP ALG page.

SIP Trunking Devices			
Select: <a href="#">All</a> <a href="#">None</a>		Action: <input type="button" value="Delete"/>	
	Address	Port	Name
<input type="checkbox"/>	172.16.248.165	5060	IPPBX

#### Add a trunking device

Action:

Name:

Address:

Port:

Figure 2.4.2 SIP Trunking, SIP Trunking devices

1. Under **SIP Trunking**, go to the **Rules** section to add an inbound rule.

NOTE: For a single SIP registration, we will be using a Default Rule to send all inbound call traffic to the LAN-SIDE Brekeke IP PBX server.

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2. In the **Add a rule** menu box:
  - Set the **Action** field to **Add a new rule**.
  - Set the **Type** field to **Inbound**.
  - Set the **Trunking device** field to the recognizable name you created for the target trunking device in the last section.
  - Select **Commit** to create the Default Rule.

**Rules**

Rules are used to forward and/or modify incoming and outgoing calls. There are 3 types of rules:

- Inbound: from server to trunking device
- Outbound: from trunking device to server
- Redirect: from local phone to trunking device (w/o routing to server)

Outbound rules can match against and/or modify either the calling or called number. Inbound and redirect rules operate on the called number only. Stripped and added digits always apply to the left-most digits of the DID.

Dial Rules									
Select: <a href="#">All</a> <a href="#">None</a>								Action: <a href="#">Delete</a>	
	Type	Party	PRIOR	Pattern-match	Strip	Add	b2bua	ss	Trunking device
<input type="checkbox"/>	Inbound			Default Rule					IPPBX (172.16.248.165:5060)

**Add a rule**

Action:

Type:

Call Party:

Default rule:

Priority (inbound & redirect only):

Pattern-match (if not default):

Strip digits:

Add string:

Use B2BUA:

Use SIP proxy as secondary target:

Trunking device:

Figure 2.4.3 SIP Trunking>Rules>Add a rule