FreeSWITCH R14

SIP Trunk Provisioning Guide

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FreeSWITCH R14 SIP Trunk Provisioning Guide

ABSTRACT
FreeSWITCH is a freely distributed soft switch that can be configured as an IP PBX; it is supported by a wide variety of operating systems to include MS Windows, FreeBSD, Solaris, and all Linux distributions. This document covers FreeSWITCH 1.2 interoperation with MegaPath R14 SIP trunks. For the purpose of instruction, a brief explanation of the installation process is provided for the Debian 6.0 Linux distribution. This document covers FreeSWITCH deployed downstream of a third-party firewall and of an Edgemarc SIP ALG.

INTRODUCTION

FreeSWITCH is a freely distributed softswitch that can be configured as IP PBX. FreeSWITCH has grown in popularity among the open source community and has become the prevalent SIP telephony engine in many SIP server open source projects. FreeSWITCH runs a wide variety of operating systems to include MS Windows, FreeBSD, Solaris, and all Linux distributions.

For the purpose of instruction, a brief explanation of the installation process is provided for the Debian 6.0 Linux distribution.

This documents provides the basic configuration parameters for MegaPath R14 SIP trunks FreeSWITCH 1.2 deployed downstream of a third-party firewall and downstream of an Edgemarc SIP application-layer gateway.

It should be noted that all FreeSWITCH configurations files are XML files.

For additional information regarding FreeSWITCH, see http://www.freeswitch.org/.

1. INSTALLATION

This installation guide assumes a fresh install of Debian 6.0 (squeeze).

1. From the command line with root privileges, install the following Debian packages:
   - autoconf
   - automake
   - g++
   - git-core
   - libjpeg62-dev
   - libncurses5-dev
   - libtool
   - make
   - python-dev
   - gawk
   - pkg-config
   - dnsutils
   - psmisc
   - tcpdump

2. Change to the directory /usr/local/src and issue the following commands to download and bootstrap the latest 1.2 source:

   ```
   git clone git://git.freeswitch.org/freeswitch.git
   cd freeswitch
   ./bootstrap.sh
   ```

3. Compile the source:

   ```
   ./configure
   Make
   ```

4. Install FreeSWITCH:

   ```
   make all install cd-sounds-install cd-moh-install
   ```

5. Start FreeSWITCH:

   /usr/local/freeswitch/bin/freeswitch-nc

6. Once FreeSWITCH is running in the background, issue the following command to access the FreeSWITCH CLI:

   /usr/local/freeswitch/bin/fs_cli

2. STANDARD FIREWALL

For this deployment, ensure that the third-party firewall is configured for static NAT and to port-forward SIP 5080/UDP and the RTP/UDP range 16384:32768 to the downstream FreeSWITCH IP PBX server.

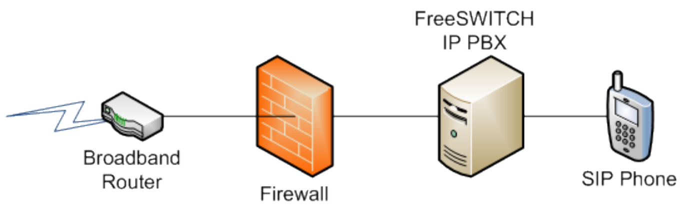

Figure 2.1 FreeSWITCH IP PBX behind a third-party firewall.

IMPORTANT: Static NAT is required to mitigate one-way audio.

2.1 Dial Plan

2.1.1 Outbound Dial Plan

The following section describes a dial plan for local, toll-free, and long-distance outbound calls, using regular expression for dial-pattern matching.

NOTES:

- The gateway all calls are routed to is the gateway named "speakeasy" and this gateway name is configured later on in the SIP profile XML file used to configure the R14 SIP trunk.
- The authentication username associated with the R14 trunk is "demotrunk02."
- The authentication password associated with the R14 trunk is "h3g3m0nyn0w."
- The DID associated with the R14 SIP trunk in this example is 312-533-4875.

/usr/local/freeswitch/conf/dialplan/default/00_dialplan.xml

```
<include>

  <extension name="local">
    <condition field="destination_number" expression="^([0-9]{7})$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/312$1"/>
    </condition>
  </extension>

  <extension name="tollfree">
```

```
    <condition field="destination_number"
expression="^(1(8(0{2}|8{2}|7{2}|6{2})\d{7}))$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/$2"/>
    </condition>
  </extension>

  <extension name="longdistance">
    <condition field="destination_number" expression="^(1{0,1}\d{10})$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/$1"/>
    </condition>
  </extension>

</include>
```

2.1.2 Inbound Dial plan

The inbound dial plan is located in the /usr/local/freeswitch/conf/dialplan/public directory. In this example, the inbound dial plan is configured in the “00_inbound_did.xml” XML file under that directory.

```
</include>
  <extension name="public_did_0">
    <condition field="destination_number" expression="^(3125334875)$">
      <action application="set" data="domain_name=$${domain}"/>
      <action application="transfer" data="9664 XML default"/>
    </condition>
  </extension>
</include>
```

When the number 312-533-4875 is called from an outside source, FreeSWITCH transfers the incoming extension indicated by “data” value in the “transfer” action, which in this instance is the extension 9664.

2.2 SIP Profile

IMPORTANT: The Broadsoft Identity/Device Profile used for the R14 SIP trunk is “Generic SIP trunk single SIP registration.”

In the following example, the R14 SIP trunk is configured in the XML file “00_speakeasy.xml” located in the /usr/local/freeswitch/conf/sip_profiles/external directory.

```
<include>
      <gateway name="speakeasy">
            <param name="username" value="demotrunk02"/>
            <param name="realm" value="speakeasy.net"/>
            <param name=”from-domain” value=”speakeasy.net”/>
            <param name="password" value="h3g3m0nyn0w"/>
            <param name="proxy" value="lab-1-siptrunk-a.voice.speakeasy.net"/>
            <param name="expire-seconds" value="3600"/>
            <param name="register" value="true"/>
            <param name="register-transport" value="udp"/>
            <param name="retry-seconds" value="10"/>
      </gateway>
</include>
```

2.3 Access Control

Access for local area network SIP phones and traffic from external sources by way of the WAN address of the firewall is configured in the “acl.conf.xml” file located in the /usr/local/freeswitch/conf/autoload_configs directory.

In this example, the LAN network address is 172.16.1.0/24 and the WAN IP address of the firewall is 67.101.126.34 /32 .

```
<configuration name="acl.conf" description="Network Lists">
  <network-lists>

    <list name="lan" default="allow">
      <node type="allow" cidr="10.0.0.0/8"/>
      <node type="allow" cidr="172.16.1.0/24"/>
    </list>

    <list name="domains" default="deny">
      <node type="allow" domain="$${domain}"/>
      <node type="allow" cidr="67.101.126.34/32"/>
    </list>

  </network-lists>
</configuration>
```

2.4 Directory (User Extensions)

The following is an example of a user extension, extension 1500. All user extensions must be set up as a pointer in the default.xml file located under /usr/local/freeswitch/conf/directory.

/usr/local/freeswitch/conf/directory/default.xml

```
<include>
  <!--the domain or ip (the right hand side of the @ in the addr-->
  <domain name="$${domain}">
    <params>
      <param name="dial-string"
value="{sip_invite_domain=${dialed_domain},presence_id=${dialed_user}@${dialed_do
main}}${sofia_contact(${dialed_user}@${dialed_domain})}"/>
    </params>

    <variables>
      <variable name="record_stereo" value="true"/>
      <variable name="default_gateway" value="$${default_provider}"/>
      <variable name="default_areacode" value="$${default_areacode}"/>
      <variable name="transfer_fallback_extension" value="operator"/>
    </variables>

    <groups>
      <group name="default">
        <users>
          <X-PRE-PROCESS cmd="include" data="default/*.xml"/>
        </users>
      </group>

      <group name="x-labs">
        <users>
          <user id="1000" type="pointer"/>
          <user id="1001" type="pointer"/>
          <user id="1002" type="pointer"/>
          <user id="1003" type="pointer"/>
          <user id="1004" type="pointer"/>
          <user id="1005" type="pointer"/>
          <user id="1006" type="pointer"/>
          <user id="1007" type="pointer"/>
          <user id="1008" type="pointer"/>
          <user id="1009" type="pointer"/>
          <user id="1010" type="pointer"/>
          <user id="1011" type="pointer"/>
          <user id="1500" type="pointer"/>
          <user id="1600" type="pointer"/>
        </users>
      </group>
```

```
  </groups>

 </domain>
</include
```

Once setup as a pointer, the user extension itself can be configured; all user extensions by default are under /usr/local/freeswitch/conf/directory/default.

IMPORTANT: Ensure that the caller-ID number is set to a DID associated with the R14 SIP trunk, as the proper caller-ID number is required by the R14 SIP trunk in order to establish an outbound call.

/usr/local/freeswitch/conf/directory/default/1500.xml

```
<include>
 <user id="1500">
  <params>
   <param name="password" value="$${default_password}"/>
   <param name="vm-password" value="1500"/>
  </params>
  <variables>
   <variable name="user_context" value="default"/>
   <variable name="effective_caller_id_name" value="1500"/>
   <variable name="effective_caller_id_number" value="3125334875"/>
   <variable name="outbound_caller_id_name" value="$${outbound_caller_name}"/>
   <variable name="outbound_caller_id_number" value="$${outbound_caller_id}"/>
   <variable name="callgroup" value="x-labs"/>
  </variables>
 </user>
</include>
```

2.5 Checking the SIP Trunk Registration

1. From the command-line of the FreeSWITCH server, issue the following command to access the FreeSWITCH CLI:

   /usr/local/freeswitch/bin/fs_cli

2. Issue the following command to see the SIP trunk registration status:

   sofia status

The following line will appear in the status output, indicating the SIP trunk registration status:

```
external:<gateway name>   gateway  sip:<username>@<hostname or host IP address>
        REGED
```

3. EDGEMARC SIP APPLICATION-LAYER GATEWAY

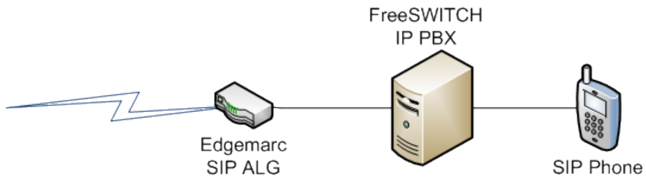

Figure 3.1 FreeSWITCH IP PBX behind and Edgemarc SIP ALG.

3.1 Dial Plan

3.1.1 Outbound Dial Plan

The following section describes a dial plan for local, toll-free, and long-distance outbound calls, using regular expression for dial-pattern matching.

NOTES:

- The gateway all calls are routed to is the gateway named “speakeasy” and this gateway name is configured later on in the SIP profile XML file used to configure the R14 SIP trunk.
- The authentication username associated with the R14 trunk is “demotrunk02.”
- The authentication password associated with the R14 trunk is “h3g3m0nyn0w.”
- The DID associated with the R14 SIP trunk in this example is 312-533-4875.

/usr/local/freeswitch/conf/dialplan/default/00_dialplan.xml

```
<include>

  <extension name="local">
    <condition field="destination_number" expression="^([0-9]{7})$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/312$1"/>
    </condition>
  </extension>

  <extension name="tollfree">
    <condition field="destination_number"
expression="^(1(8(0{2}|8{2}|7{2}|6{2})\d{7}))$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/$2"/>
    </condition>
```

```
  </extension>


  <extension name="longdistance">
    <condition field="destination_number" expression="^(1{0,1}\d{10})$">
      <action application="set"
data="effective_caller_id_number=${effective_caller_id_number}"/>
      <action application="set"
data="effective_caller_id_name=${effective_caller_id_name}"/>
      <action application="set" data="no_media=true"/>
      <action application="bridge" data="sofia/gateway/speakeasy/$1"/>
    </condition>
  </extension>

</include>
```

3.1.2 Inbound Dial plan

The inbound dial plan is located in the /usr/local/freeswitch/conf/dialplan/public directory. In this example, the inbound dial plan is configured in the "00_inbound_did.xml" XML file under that directory.

```
</include>
  <extension name="public_did_0">
    <condition field="destination_number" expression="^(3125334875)$">
      <action application="set" data="domain_name=$${domain}"/>
      <action application="transfer" data="9664 XML default"/>
    </condition>
  </extension>
</include>
```

When the number 312-533-4875 is called from an outside source, FreeSWITCH transfers the incoming extension indicated by "data" value in the "transfer" action, which in this instance is the extension 9664.

3.2 External SIP profile

In this instance, the LAN-side IP address of the Edgemarc SIP ALG is 172.16.1.1. The R14 SIP trunk will be configured as show in the example in the XML file /usr/local/freeswitch/conf/sip_profiles/external/00_speakeasy.xml

```
<include>
        <gateway name="speakeasy">
                <param name="username" value="demotrunk02"/>
                <param name="realm" value="speakeasy.net"/>
                <param name=”from-domain” value=”speakeasy.net”/>
                <param name="password" value="h3g3m0nyn0w"/>
```

```
            <param name="proxy" value="172.16.1.1"/>
            <param name="expire-seconds" value="3600"/>
            <param name="register" value="true"/>
            <param name="register-transport" value="udp"/>
            <param name="retry-seconds" value="10"/>
        </gateway>
</include
```

3.3 Access Control

Access for local area network SIP phones and traffic from external sources by way of the LAN address of the Edgemarc SIP ALG is configured in the “acl.conf.xml” file located in the /usr/local/freeswitch/conf/autoload_configs directory.

In this example, the LAN network address is 172.16.1.0/24 and the LAN IP address of the Edgemarc SIP ALG is 172.16.1.1 .

```
<configuration name="acl.conf" description="Network Lists">
  <network-lists>

    <list name="lan" default="allow">
      <node type="allow" cidr="10.0.0.0/8"/>
      <node type="allow" cidr="172.16.1.0/24"/>
    </list>

    <list name="domains" default="deny">
      <node type="allow" domain="$${domain}"/>
      <node type="allow" cidr="172.16.1.1/32"/>
    </list>

  </network-lists>
</configuration>
```

3.4 Directory (User extensions)

The following is an example of a user extension, extension 1500. All user extensions must be set up as a pointer in the default.xml file located under /usr/local/freeswitch/conf/directory.

/usr/local/freeswitch/conf/directory/default.xml

```
<include>
  <!--the domain or ip (the right hand side of the @ in the addr-->
  <domain name="$${domain}">
    <params>
      <param name="dial-string"
value="{sip_invite_domain=${dialed_domain},presence_id=${dialed_user}@${dialed_do
main}}${sofia_contact(${dialed_user}@${dialed_domain})}"/>
```

```
    </params>

    <variables>
      <variable name="record_stereo" value="true"/>
      <variable name="default_gateway" value="$${default_provider}"/>
      <variable name="default_areacode" value="$${default_areacode}"/>
      <variable name="transfer_fallback_extension" value="operator"/>
    </variables>

    <groups>
      <group name="default">
        <users>
          <X-PRE-PROCESS cmd="include" data="default/*.xml"/>
        </users>
      </group>

      <group name="x-labs">
        <users>
          <user id="1000" type="pointer"/>
          <user id="1001" type="pointer"/>
          <user id="1002" type="pointer"/>
          <user id="1003" type="pointer"/>
          <user id="1004" type="pointer"/>
          <user id="1005" type="pointer"/>
          <user id="1006" type="pointer"/>
          <user id="1007" type="pointer"/>
          <user id="1008" type="pointer"/>
          <user id="1009" type="pointer"/>
          <user id="1010" type="pointer"/>
          <user id="1011" type="pointer"/>
          <user id="1500" type="pointer"/>
          <user id="1600" type="pointer"/>
        </users>
      </group>

    </groups>

  </domain>
</include
```

Once setup as a pointer, the user extension itself can be configured; by default, all user extensions are under /usr/local/freeswitch/conf/directory/default.

IMPORTANT: Ensure that the caller-ID number is set to a DID associated with the R14 SIP trunk, as the proper caller-ID number is required by the R14 SIP trunk in order to establish an outbound call.

/usr/local/freeswitch/conf/directory/default/1500.xml

```
<include>
  <user id="1500">
    <params>
      <param name="password" value="$${default_password}"/>
      <param name="vm-password" value="1500"/>
    </params>
    <variables>
      <variable name="user_context" value="default"/>
      <variable name="effective_caller_id_name" value="1500"/>
      <variable name="effective_caller_id_number" value="3125334875"/>
      <variable name="outbound_caller_id_name" value="$${outbound_caller_name}"/>
      <variable name="outbound_caller_id_number" value="$${outbound_caller_id}"/>
      <variable name="callgroup" value="x-labs"/>
    </variables>
  </user>
</include>
```

3.5 Edgemarc SIP ALG SIP Trunking

1. Log in to the Edgemarc SIP ALG. Under the "VoIP ALG" menu option.
2. Select the "SIP" option.
3. Under "SIP Settings," in the "SIP Server Address:" field, 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.
4. Input the SIP server port number in the "SIP Server Port:" field.
5. Enable "Use Custom Domain:".
6. Enter "speakeasy.net" in the "SIP Server Domain:" field.

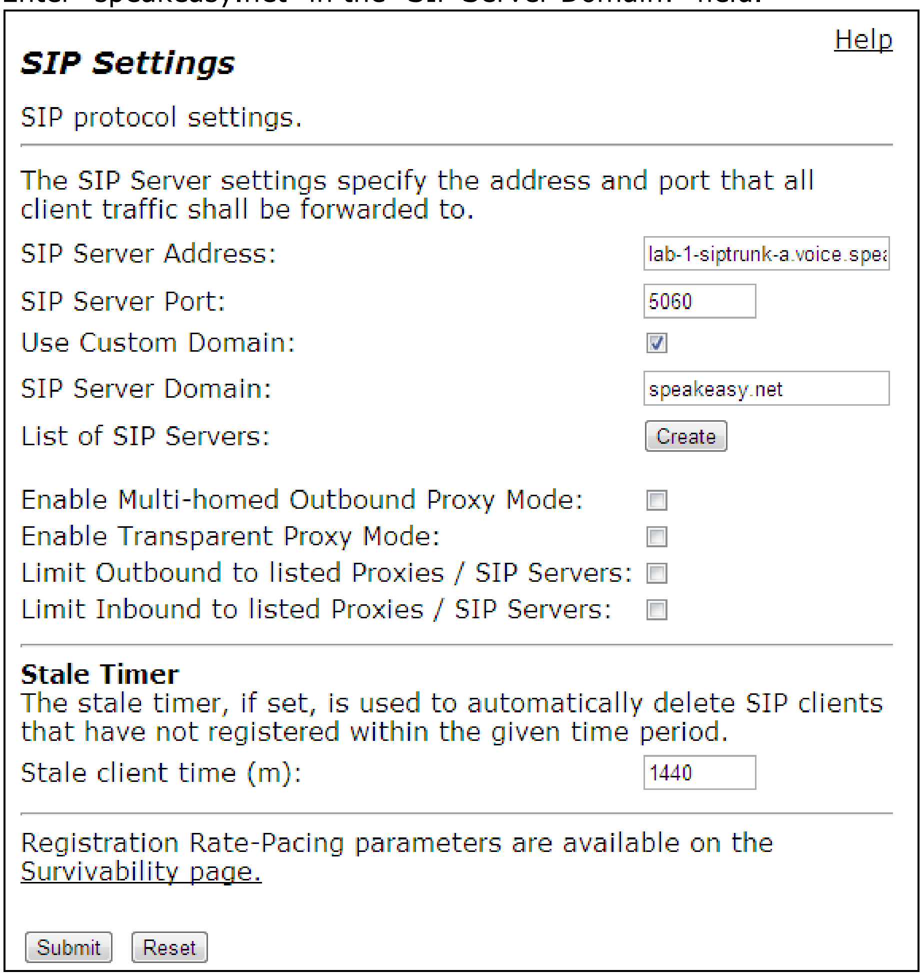

Figure 3.5.1 VoIP ALG, SIP Settings.

7. Submit changes.
8. From the "SIP" sub-menu, select "Trunking."

9. In the "Add a trunking device" menu box:
    - Set the "Action:" field to "Add new trunking device."
    - Input a recognizable name for the trunking device in the "Name:" field.
    - Input the IP address of the FreePBX server in the "Address:" field.
    - Input the SIP port number of the FreePBX server in the "Port:" field.
    - Select the "Commit" button to create the SIP trunking device.

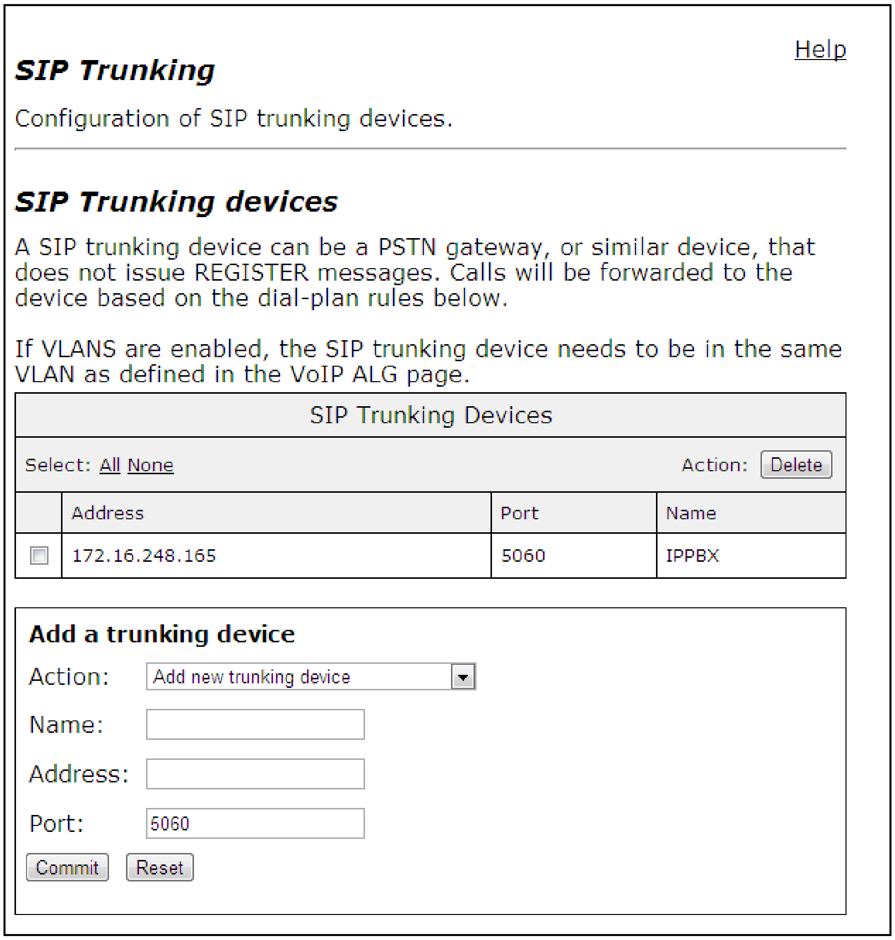

Figure 3.5.2 SIP Trunking, SIP Trunking devices.

10. 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 FreePBX server.

11. 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.

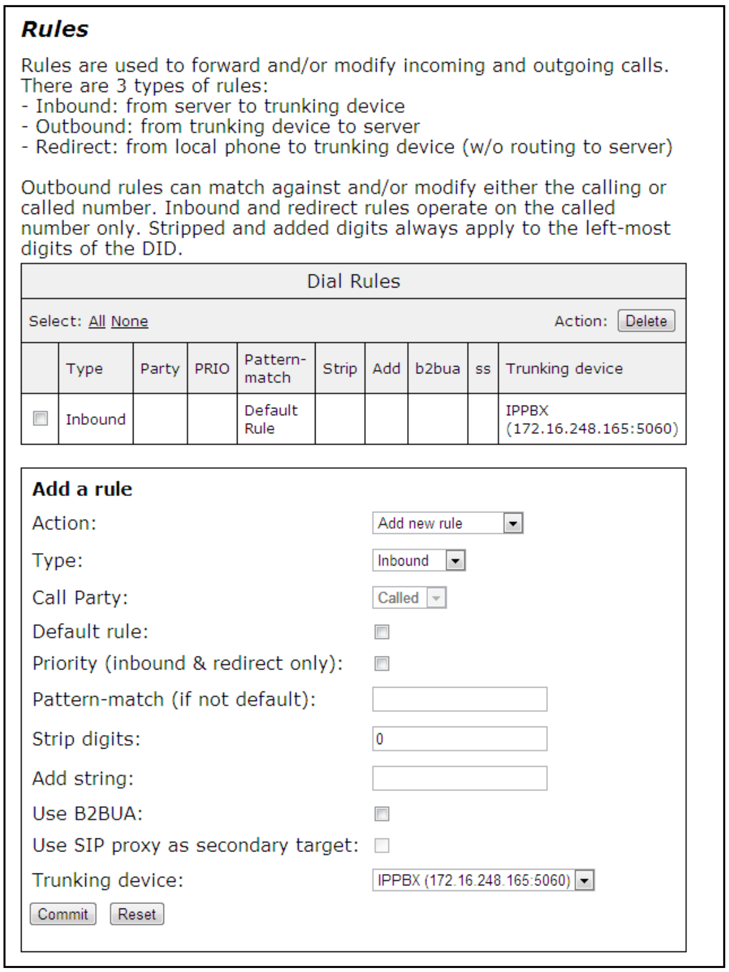

Figure 3.5.3 SIP Trunking, Rules, Add a rule.

3.6 Checking the SIP Trunk Registration

1. From the command-line of the FreeSWITCH server, issue the following command to access the FreeSWITCH CLI:

   /usr/local/freeswitch/bin/fs_cli

2. Issue the following command to see the SIP trunk registration status:

   sofia status

The following line will appear in the status output, indicating the SIP trunk registration status:

   external:<gateway name>   gateway sip:<username>@<hostname or host IP address>   REGED