miniSQL Documentation

DevOpSec

Mar 31, 2023

Contents

CHAPTER 1

Intro to dSIPRouter

dSIPRouter allows you to quickly turn Kamailio into an easy to use SIP Service Provider platform, which enables the following two basic use cases:

- **SIP Trunking services:** Provide services to customers that have an on-premise PBX such as FreePBX, FusionPBX, Avaya, etc. We have support for IP and credential based authentication.
- **Hosted PBX services:** Proxy SIP Endpoint requests to a multi-tenant PBX such as FusionPBX or single-tenant such as FreePBX. We have an integration with FusionPBX that make is really easy and scalable!

1.1 Demo System

You can checkout our demo system by clicking the link below and enter the listed username and password:

http://demo.dsiprouter.org:5000

username: admin

password: ZmIwMTdmY2I5NjE4

API Token: 9lyrny3HOtwgjR6JIMwRaMej9LijIS835zhVbD8ywHDzXT07Xm6vem1sgfvWkFz3

1.2 Credits

I'd like to say thank you to Nicole D., John O. and Courtney G. for their time in fulfilling this document. I'd also like to give a hardy thank you to dOpensource for their monetary support in funding this document.

1.3 Support

Free support is available via our group

Paid support is available here

CHAPTER 2

Installing dSIPRouter

2.1 Installing dSIPRouter

Install dSIPRouter takes approximately 9-12 minutes to install. The following video shows you the install process:

2.1.1 Prerequisites:

- Must run this as the root user (you can use sudo)
- git needs to be installed
- Hostname needs to be set to a FQDN (for certbot to get LetsEncrypt certificate)
- The installer will handle all other dependencies

2.1.2 Install Options

- Proxy SIP Traffic Only (Don't Proxy audio (RTP) traffic)
- Proxy SIP Traffic and Audio when it detects a SIP Agent is behind NAT
- Proxy SIP Traffic, Audio and it configures the system to work properly when the PBX's and dSIPRouter are behind a NAT.

2.1.3 OS Support

OS / Distro	Current Support
Debian 11 (bullseye)	STABLE
Debian 10 (buster)	STABLE
Debian 9 (stretch)	STABLE
RedHat Linux 8	ALPHA
Alma Linux 8	ALPHA
Rocky Linux 8	ALPHA
Amazon Linux 2	STABLE
Ubuntu 22.04 (jammy)	ALPHA
Ubuntu 20.04 (focal)	ALPHA

Kamailio will be automatically installed along with dSIPRouter. Must be installed on a fresh install of Debian Stretch, Debian Buster or CentOS. You will not be prompted for any information. It will take anywhere from 9-12 minutes to install - depending on the processing power of the machine. You can secure the Kamailio database after the installation.

We maintain installation documentation for the following operating systems. Please open a pull request if you want to add and maintain addtional documentation:

- debian_install
- rhel_install

2.1.4 Amazon AMI's

We now provide Amazon AMI's (pre-built images) which allows you to get up and going even faster. You can find a list of the images here. The images are a nominal fee, which goes toward supporting the project.

CHAPTER 3

Command Line Options

3.1 Command Line Options

Execute "./dsiprouter.sh" followed by one of the listed commands. **NOTE** Once installed the command will be available globally as *dsiprouter* with tab-completion.

Command	What does it do?
install	Installs dSIPRouter and related services
uninstall	Uninstall dSIPRouter and related services
clusterinstall	Install dSIPRouter (via SSH) on a cluster of nodes
upgrade	Upgrade dSIPRouter platform (requires license)
start	Starts dSIPRouter
stop	Stops dSIPRouter
restart	Restarts dSIPRouter
configurekam	Reconfigures the Kamailio configuration file based on dSIPRouter settings
configuresslcert	Reconfigures SSL certificate used by Kamailio and dSIPRouter
renewsslcert	Renew configured letsencrypt SSL certificate
installmodules	Install / uninstall dDSIProuter modules
resetpassword	Generate new random dSIPRouter admin account password
setcredentials	Set various credentials manually
chown	Update file permissions for dSIPRouter and related services
version	Show dSIPRouter version
help	List all of the options

Refer to *Installing dSIPRouter* to get the complete one line version of the command.

To start dSIPRouter:

```
./dsiprouter.sh start
```

To stop dSIPRouter:

```
./dsiprouter.sh stop
```

To restart dSIPRouter:

```
./dsiprouter.sh restart
```

To uninstall dSIPRouter:

```
./dsiprouter.sh unistall
```

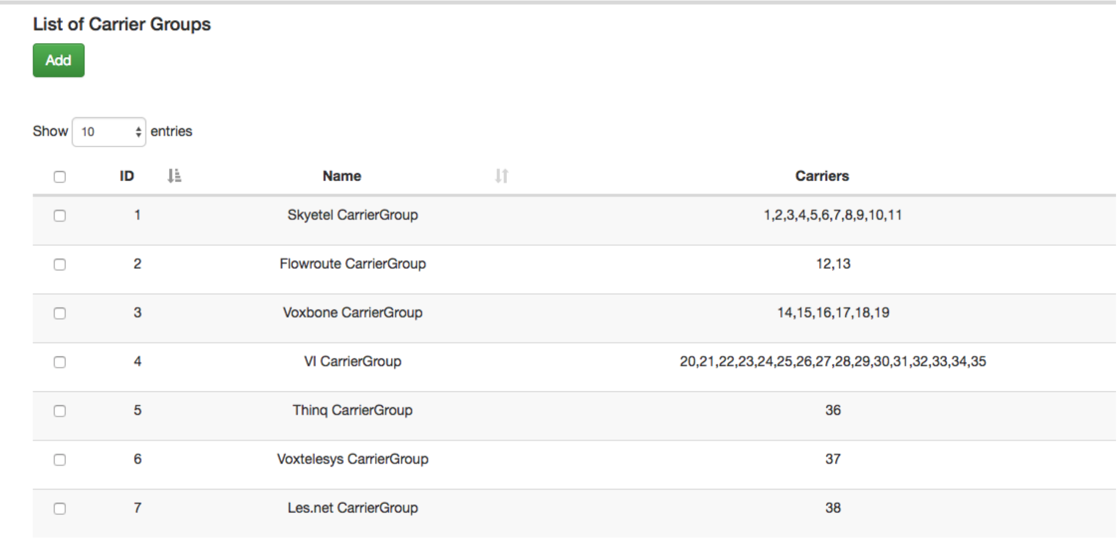
CHAPTER 4

Configuring dSIPRouter

4.1 Configuring dSIPRouter

4.1.1 Carrier Groups

The Carrier Group section of dSIPRouter allows you to define which carriers will be used to provide Internet service (aka ISP) for your VOIP (Voice Over IP) services. Carrier groups support IP Authentication and Username/Password authentication. Below is an example of a carrier groups list.



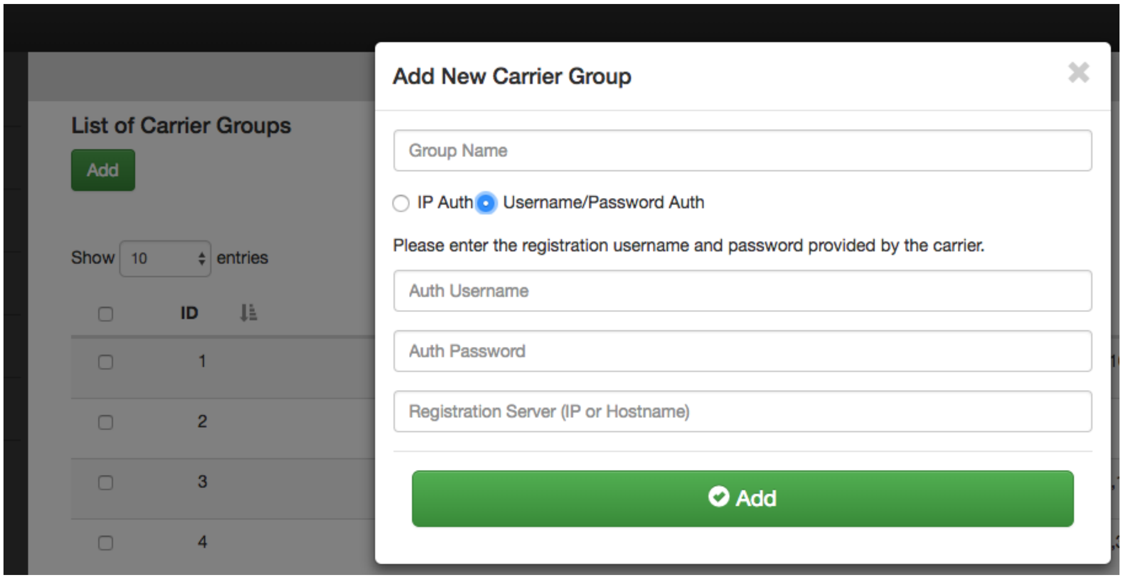
List of Carrier Groups

□	ID	Name	Carriers
□	1	Skyetel CarrierGroup	1,2,3,4,5,6,7,8,9,10,11
□	2	Flowroute CarrierGroup	12,13
□	3	Voxbone CarrierGroup	14,15,16,17,18,19
□	4	VI CarrierGroup	20,21,22,23,24,25,26,27,28,29,30,31,32,33,34,35
□	5	Thinq CarrierGroup	36
□	6	Voxtelesys CarrierGroup	37
□	7	Les.net CarrierGroup	38

4.1.2 Adding a Carrier

- Log into dSIPRouter using proper username and password.

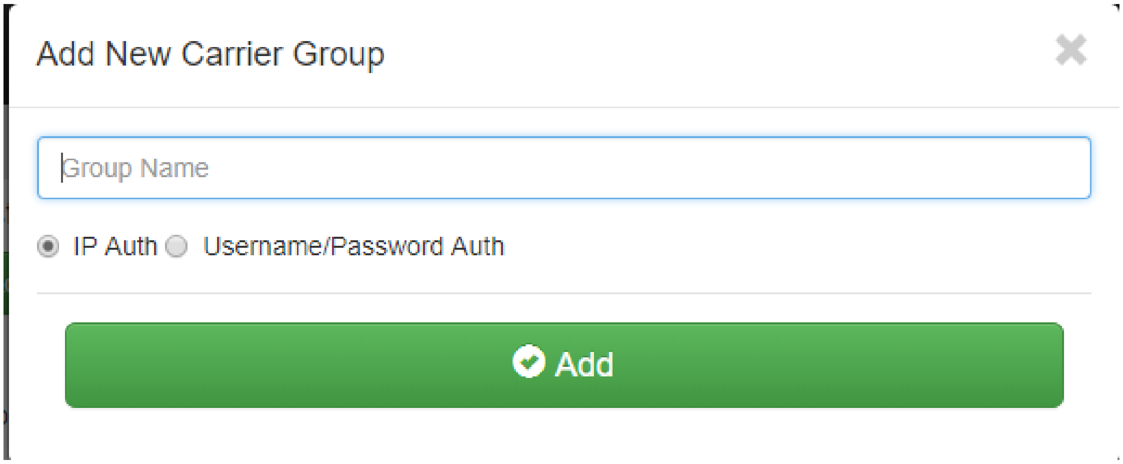
- Click "Add" to create a Carrier Group. A carrier group can contain 1 or more SIP endpoints provided by the carrier. A SIP Endpoint represents a device that makes or receives calls via your Gateway. This could be a physical IP phone, a softphone app such as Skype, on a PC or smartphone, an Analog Telephone Adapter (ATA) such as for fax machines, or even a PBX system.
- Select Username/Password Auth, fill in the username, password of your registration server and the registration server name. Then click ADD.



○ IP Auth ● Username/Password Auth

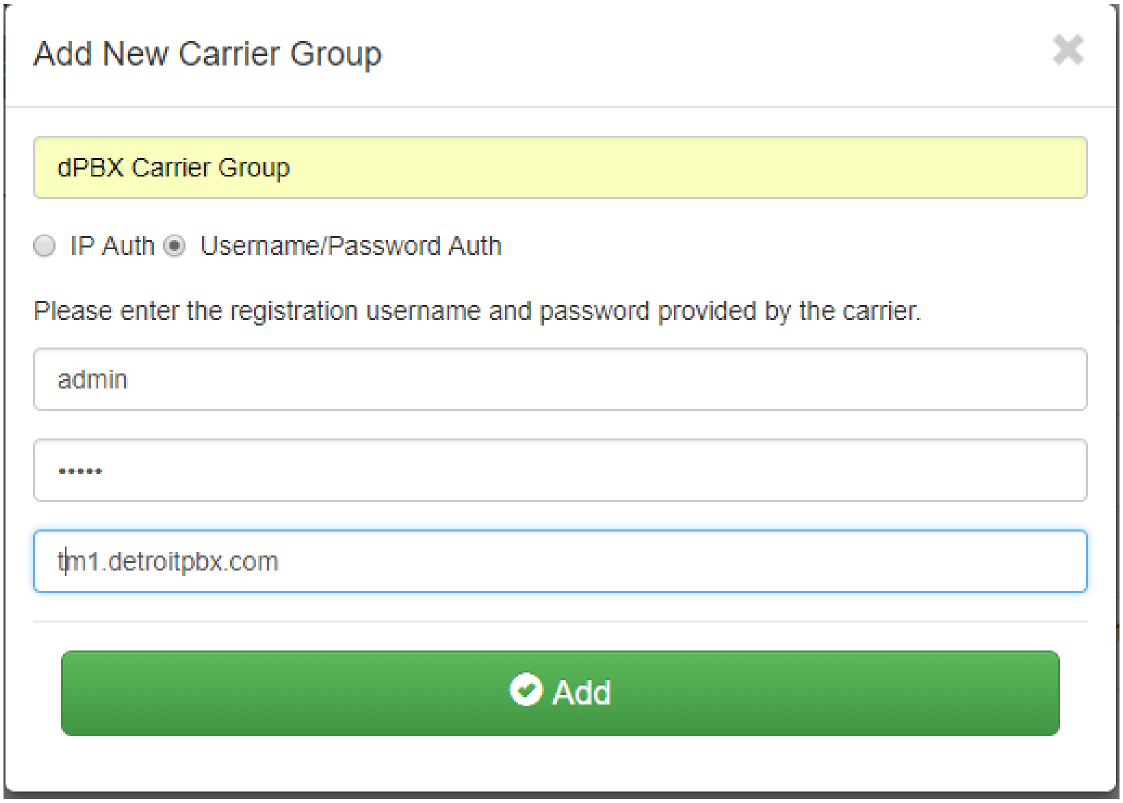
Please enter the registration username and password provided by the carrier.

NOTE: Click IP authenication to use only the IP address of your PBX/endpoint.



● IP Auth ○ Username/Password Auth

For example:



Add New Carrier Group

dPBX Carrier Group

○ IP Auth ● Username/Password Auth

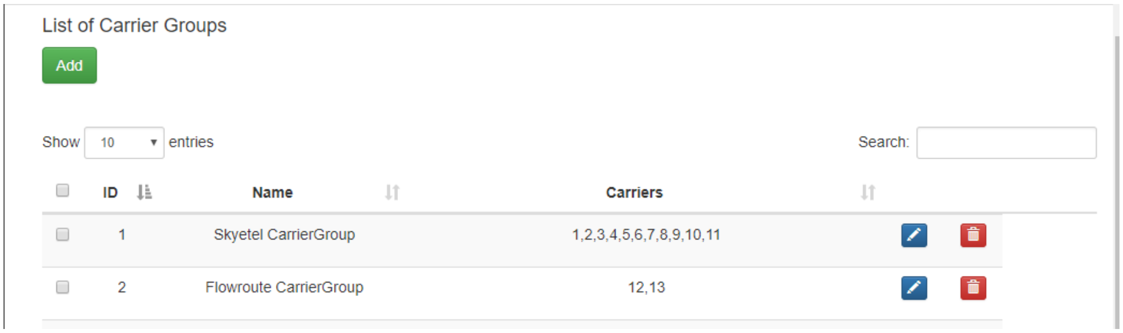
Please enter the registration username and password provided by the carrier.

admin

•••••

tm1.detroitpbx.com

After you have added the new group, the screen will return back to the List of Carriers Group page. Select the pencil in the blue box to the right to allow editing the Config and Endpoints.



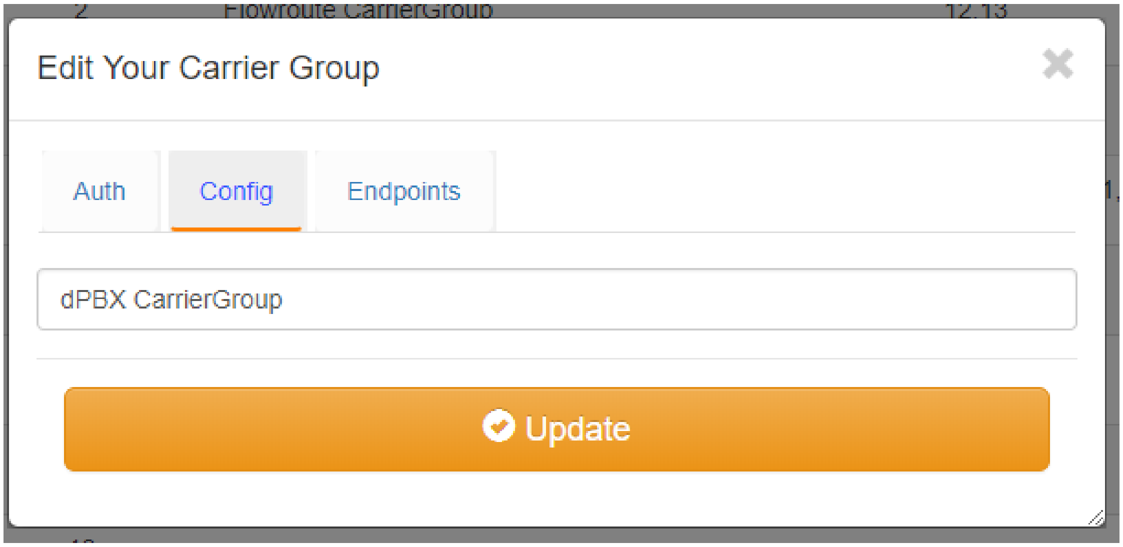
List of Carrier Groups

Show 10 entries

Search:

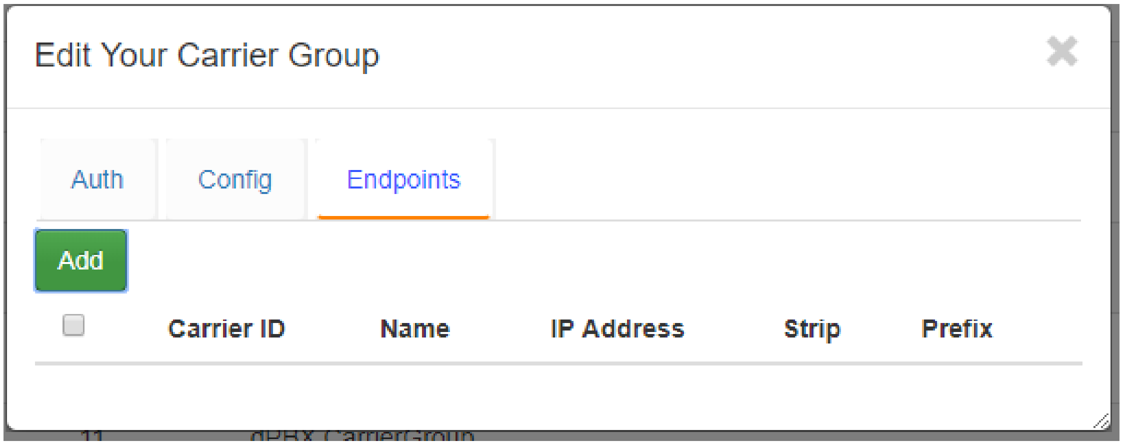
□	ID	Name	Carriers		
□	1	Skyetel CarrierGroup	1,2,3,4,5,6,7,8,9,10,11	✎	🗑
□	2	Flowroute CarrierGroup	12,13	✎	🗑

Select the Config tab. The Config tab allows you to edit/change the Carrier group name. Then click Update.

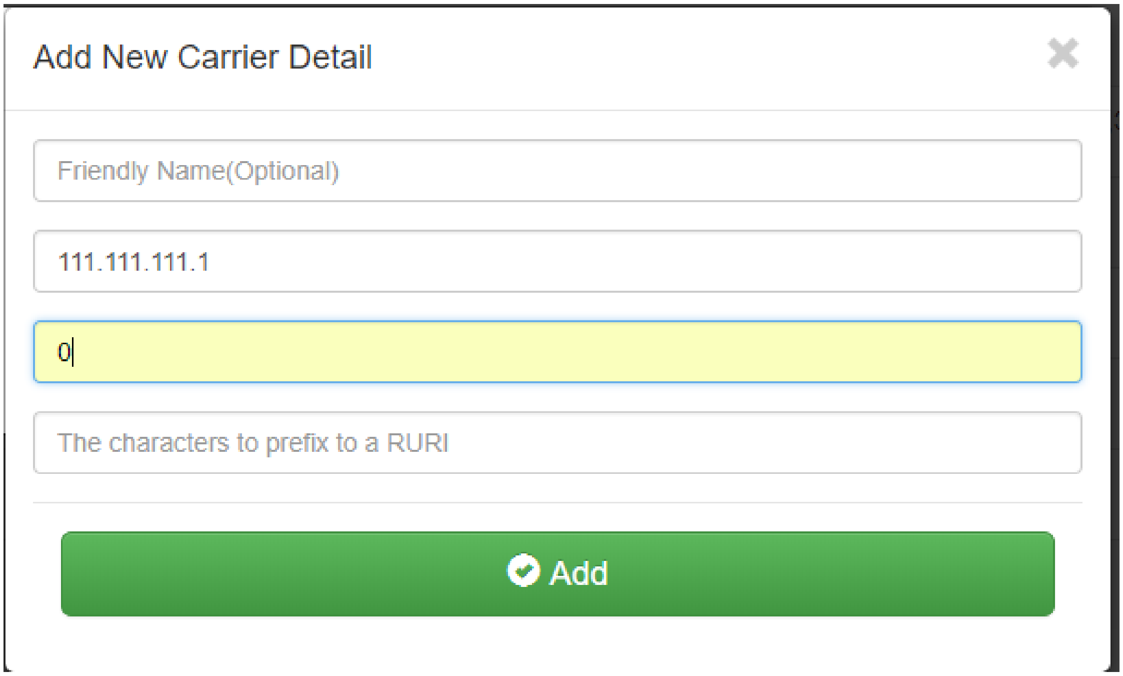


dPBX CarrierGroup

To add an Endpoint, click the Endpoint tab.



Click ADD, enter the Friendly name (optional), the IP address of the endpoint/device, # of characters to strip from RURI, the character to prefix to a RURI then click ADD again. RURI is when a number is dialed such as 9 plus the 1 and the number, a carrier may only want to see the phone number so the RURI would strip the "9" from in front of the telephone number. For example, if a PBX sends a number over as 914443332222 but the carrier wants the number to be sent as 14443332222 then the field should have a 1, which would strip off the 9. Some carriers request added digits, aka prefixes, in front of the phone number for validation that the call is from that carrier. In this example, the # of characters to strip from RURI is 0 as in none.



Add New Carrier Detail

Friendly Name(Optional)

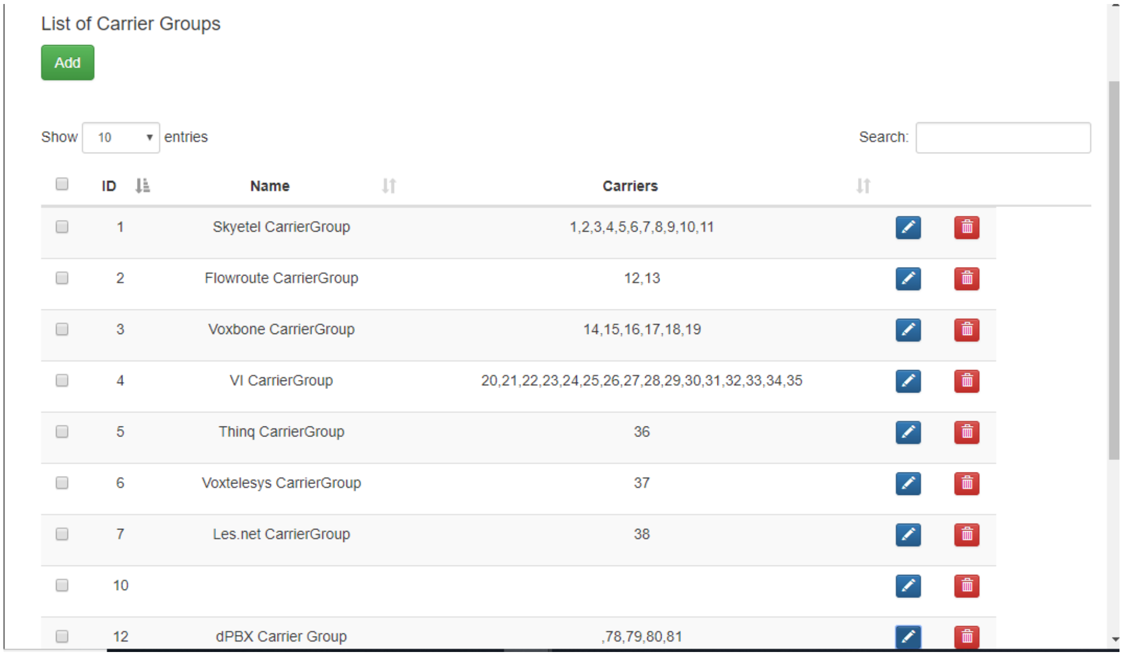
111.111.111.1

0

The characters to prefix to a RURI

Edit and click ADD again to add addtional endpoints. Click the gray X in that box to save window and close.

You should now see your added carrier with endpoints in the Carrier Group List.

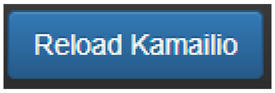


List of Carrier Groups

Show 10 entries Search:

☐	ID	Name	Carriers
☐	1	Skyetel CarrierGroup	1,2,3,4,5,6,7,8,9,10,11
☐	2	Flowroute CarrierGroup	12,13
☐	3	Voxbone CarrierGroup	14,15,16,17,18,19
☐	4	VI CarrierGroup	20,21,22,23,24,25,26,27,28,29,30,31,32,33,34,35
☐	5	Thinq CarrierGroup	36
☐	6	Voxtelesys CarrierGroup	37
☐	7	Les.net CarrierGroup	38
☐	10		
☐	12	dPBX Carrier Group	,78,79,80,81

Be sure to click the Reload Kamailio button to apply changes.

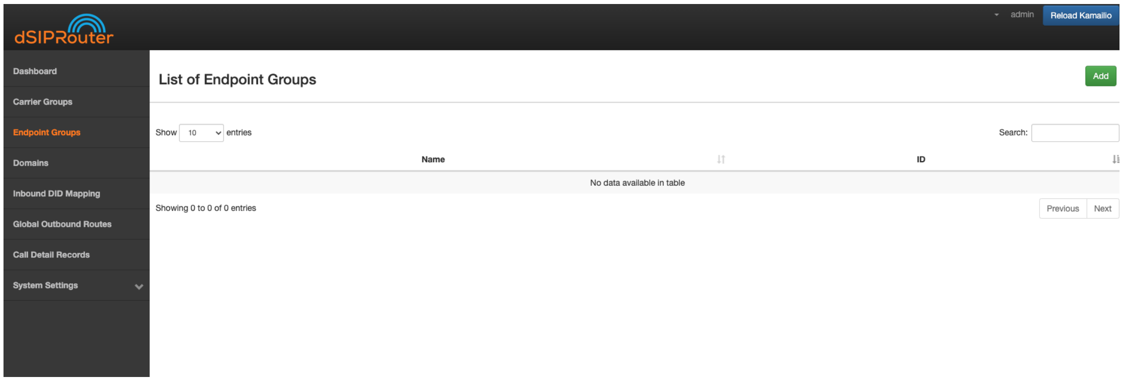


4.1.3 PBX(s) and Endpoints

Allows you to define a PBX or Endpoint that will send or receive calls from dSIPRouter. The PBX or Endpoint can use IP authentication or a username/password can be defined.

To add an Endpoint Group:

1) Click on Endpoints Groups.
2) Click on the green Add button.



List of Endpoint Groups

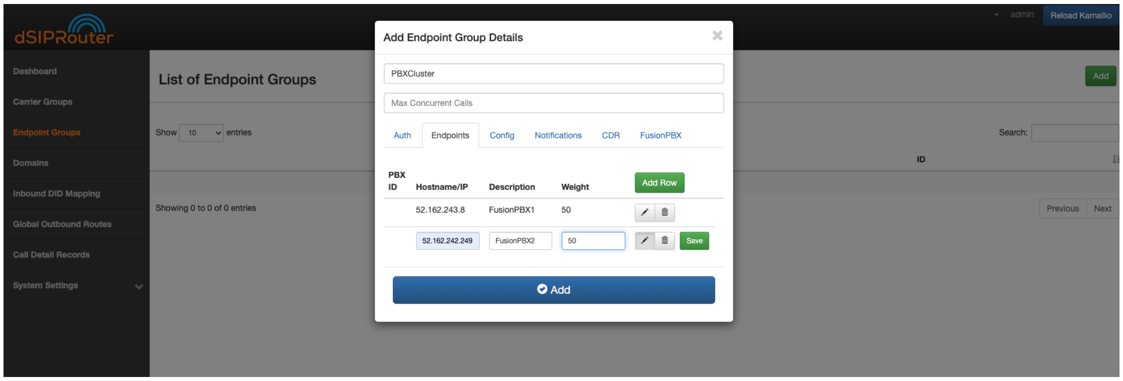
Name	ID
No data available in table	

Showing 0 to 0 of 0 entries

3) Configure the Endpoint Group

   The Endpoint Tab is where you specify the endpoints that will be signaling with dSIPRouter. The weight field allows you to define how much SIP traffic is distributed to a particular endpoint. If you don't specify a weight for an endpoint the system will automatically generate a weight. If you are using FusionPBX Domain Auth then Register and INVITE requests will be distributed to the endpoints based upon the weights. You will also have the option to route Inbound calls to the endpoints based on the weights by selecting the name of the Endpoint Group with an LB concatenated to the name. For example, if the name of the Endpoint Group is **PBXCluster** then you would select **PBXCluster LB** from the Inbound Mapping Endpoint Group drop down.

b) Click the green Add button.



Add Endpoint Group Details

PBXCluster

Max Concurrent Calls

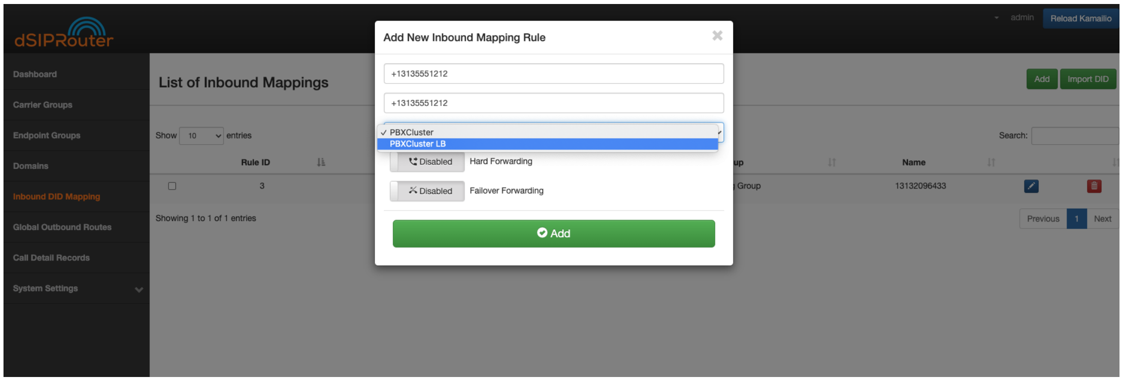
PBX ID	Hostname/IP	Description	Weight
	52.162.243.8	FusionPBX1	50
	52.162.242.249	FusionPBX2	50

4) Click on the Reload Kamailio button in order for the changes to be updated.

4.1.4 Inbound DID Mapping

To Import a DID from a CSV file:

1) Click on Inbound DID Mapping.



Add New Inbound Mapping Rule

+13135551212

+13135551212

✓ PBXCluster
PBXCluster LB

Disabled Hard Forwarding

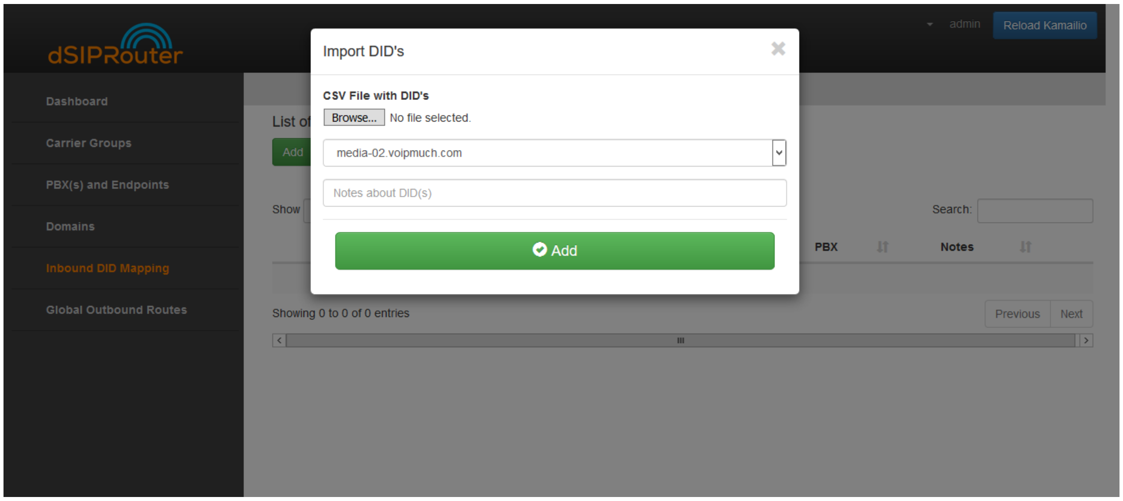
Disabled Failover Forwarding

Add

	Rule ID		Name
□	3	Group	13132096433

Showing 1 to 1 of 1 entries

2) Click on the green Import DID button underneath List on Inbound Mappings.



Import DID's

CSV File with DID's

Browse... No file selected.

media-02.voipmuch.com

Notes about DID(s)

Add

Showing 0 to 0 of 0 entries

3) Click the Browse button and select the file that contains the DID numbers that you wish to use.

4) Click the green Add button.

   Click CSV Example to view a sample of the .CSV file

5) Click on the Reload Kamailio button in order for the changes to be updated.

To Manually import a DID:

1) Click on Inbound DID Mapping

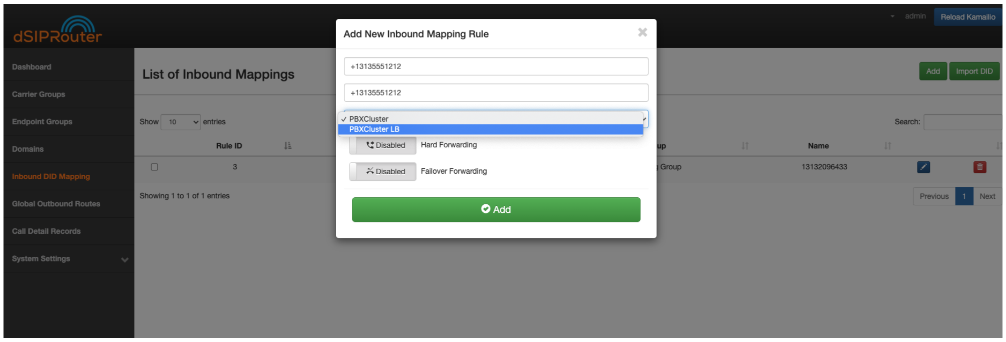
2) Click on the green ADD button.

   - Enter the name of the Inbound mapping
   - Enter the DID number in the DID field.
   - Select the Endpoint Group from the drop-down list

     Note: Each endpoint will contain at least two entries. One that leverages load balancing weights and another that randomly selects an endpoint. The one denoted with a LB is the one that uses the load balancing algorithm. If FusionPBX

Domain Support is enabled you will see an additional entry for routing to the external interface of the FusionPBX server.

- Click the green Add button.



Add New Inbound Mapping Rule

+13135551212

+13135551212

✓ PBXCluster
PBXCluster LB

Disabled Hard Forwarding

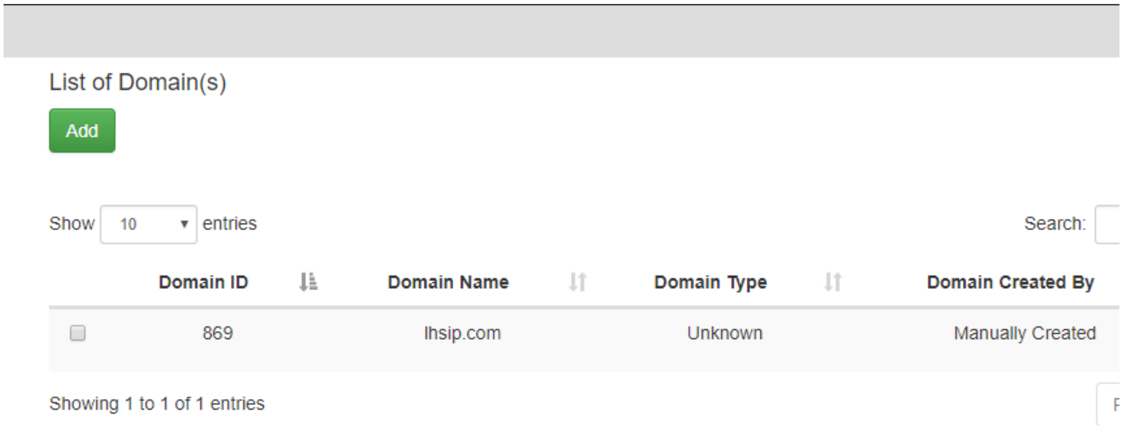
Disabled Failover Forwarding

Rule ID		up	Name
3		Group	13132096433

3) Click on the Reload Kamailio button in order for the changes to be updated.

4.1.5 Adding a Domain

To add a domain click on Domains then click the green add button.



List of Domain(s)

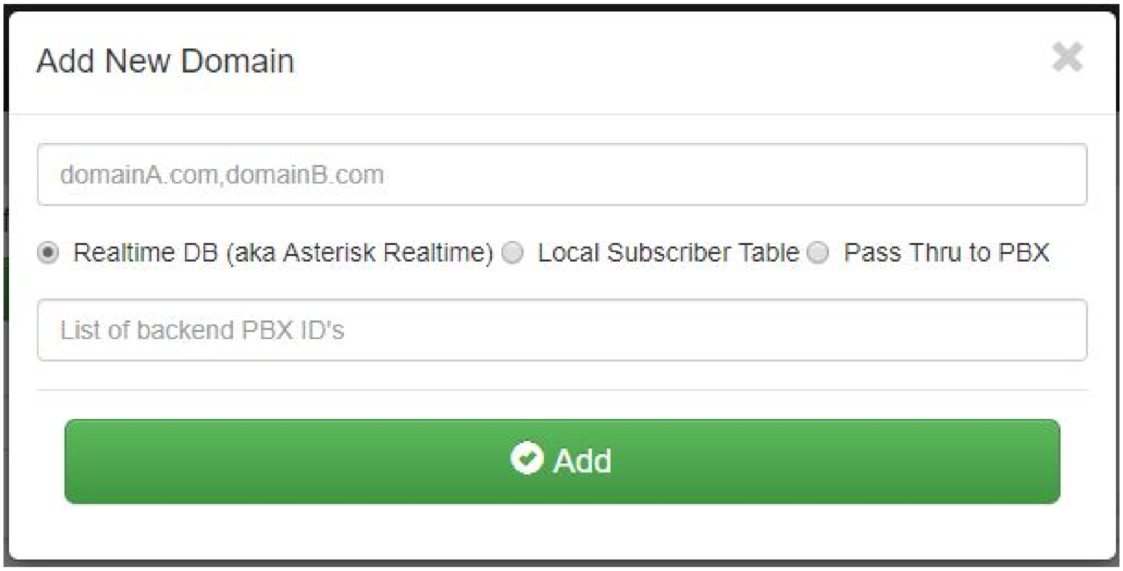
	Domain ID	Domain Name	Domain Type	Domain Created By
□	869	lhsip.com	Unknown	Manually Created

Showing 1 to 1 of 1 entries

Fill in the domain name. (Note: You can create 1 or more domains by separating them with commas).

- Select Realtime DB or Local Subscriber table (for multiple domains)
- Select Pass Thru to PBX (single domain).

Note: Details can be found in Realtime DB if you want to ensure that the Kamailio configuration file is setup to point to the Asterisk Realtime database configuration. Details on how to populate the table can be found in the Local Suscriber table if you want to use the built in subscriber table that's part of Kamailio. Use the pass thru to register info to the FreePBX server so that you don't have to change how authentication is done.



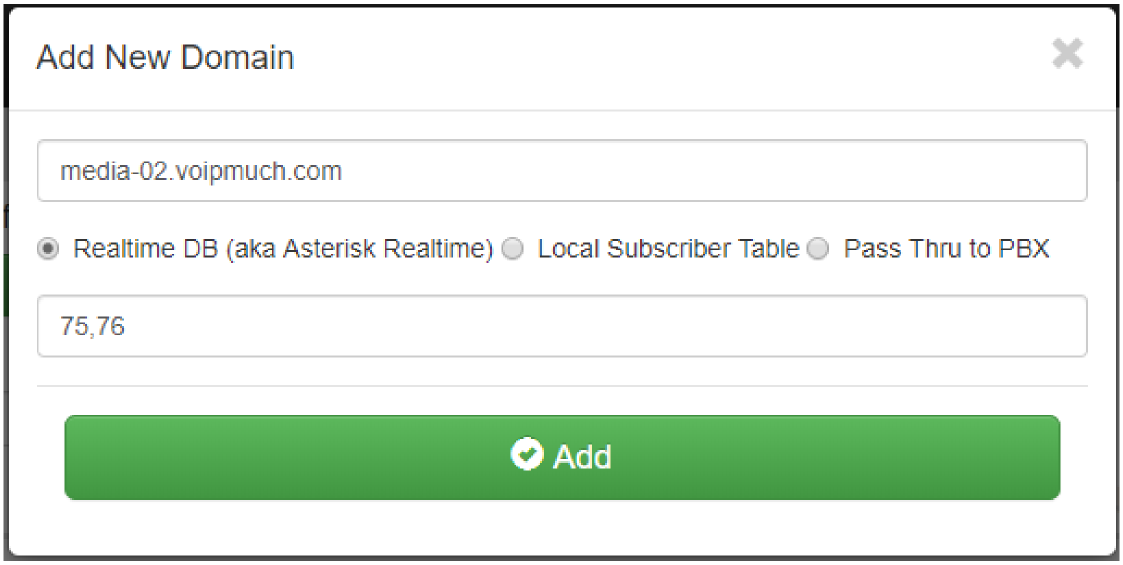
Add New Domain

domainA.com,domainB.com

● Realtime DB (aka Asterisk Realtime) ○ Local Subscriber Table ○ Pass Thru to PBX

List of backend PBX ID's

- For the List of backend PBX ID's you should use the ID assigned to each PBX that you want to be part of that domain. Such as naming the ID number thats assigned to media-02.voipmuch.com for example in *PBX(s) and Enpoints*.



Add New Domain

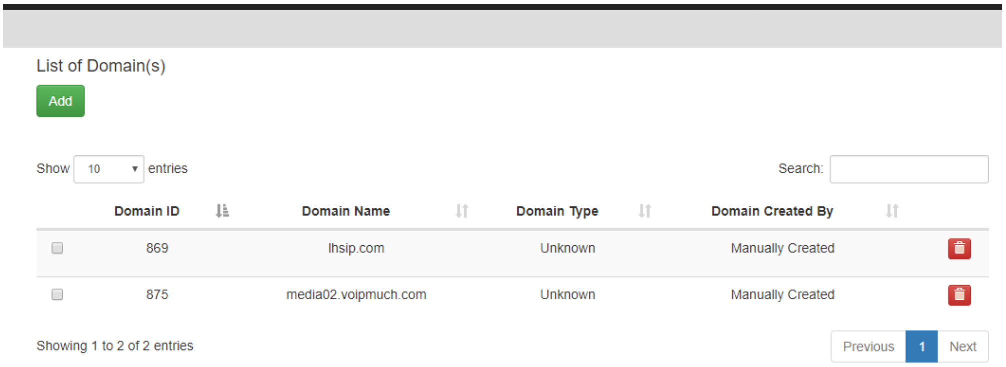
media-02.voipmuch.com

● Realtime DB (aka Asterisk Realtime) ○ Local Subscriber Table ○ Pass Thru to PBX

75,76

- Click ADD

You will then be returned back to the List of domains page and you should see your new domain added. You can delete this domain by clicking the red trash can to the right of the page.

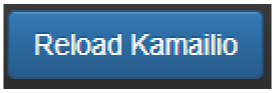


List of Domain(s)

	Domain ID	Domain Name	Domain Type	Domain Created By	
□	869	lhsip.com	Unknown	Manually Created	🗑
□	875	media02.voipmuch.com	Unknown	Manually Created	🗑

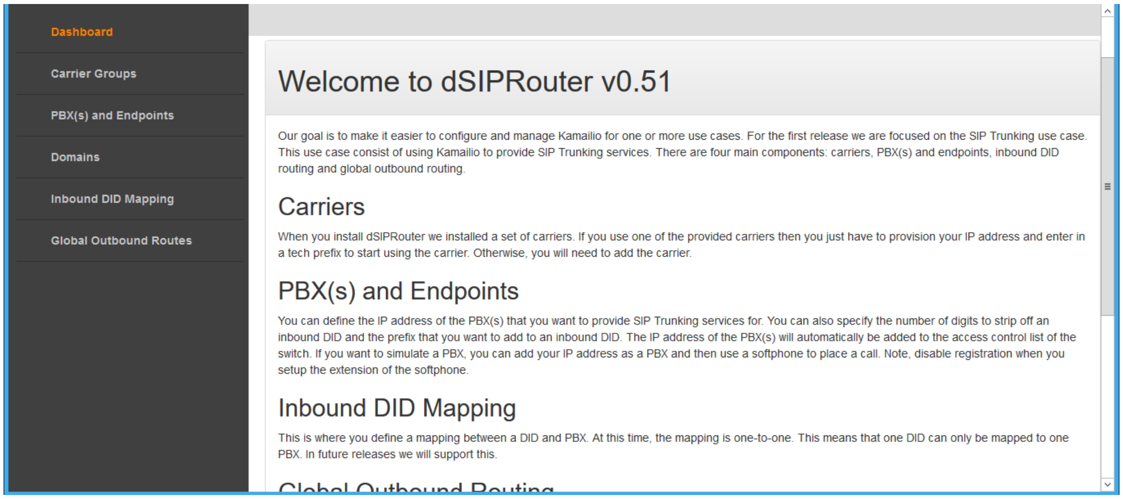
Showing 1 to 2 of 2 entries

Be sure to click the Reload Kamailio button to apply changes.



4.1.6 Global Outbound Routes

1) Go to the Dashboard screen.



Welcome to dSIPRouter v0.51

Our goal is to make it easier to configure and manage Kamailio for one or more use cases. For the first release we are focused on the SIP Trunking use case. This use case consist of using Kamailio to provide SIP Trunking services. There are four main components: carriers, PBX(s) and endpoints, inbound DID routing and global outbound routing.

Carriers

When you install dSIPRouter we installed a set of carriers. If you use one of the provided carriers then you just have to provision your IP address and enter in a tech prefix to start using the carrier. Otherwise, you will need to add the carrier.

PBX(s) and Endpoints

You can define the IP address of the PBX(s) that you want to provide SIP Trunking services for. You can also specify the number of digits to strip off an inbound DID and the prefix that you want to add to an inbound DID. The IP address of the PBX(s) will automatically be added to the access control list of the switch. If you want to simulate a PBX, you can add your IP address as a PBX and then use a softphone to place a call. Note, disable registration when you setup the extension of the softphone.

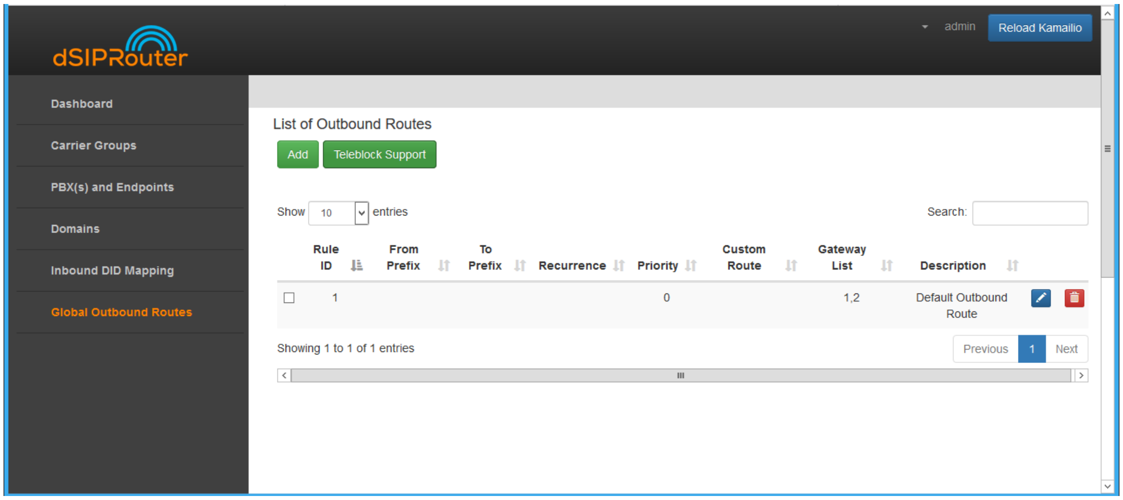
Inbound DID Mapping

This is where you define a mapping between a DID and PBX. At this time, the mapping is one-to-one. This means that one DID can only be mapped to one PBX. In future releases we will support this.

Global Outbound Routing

2) Click on Global Outbound Routes.

3) Click on the green Add button.

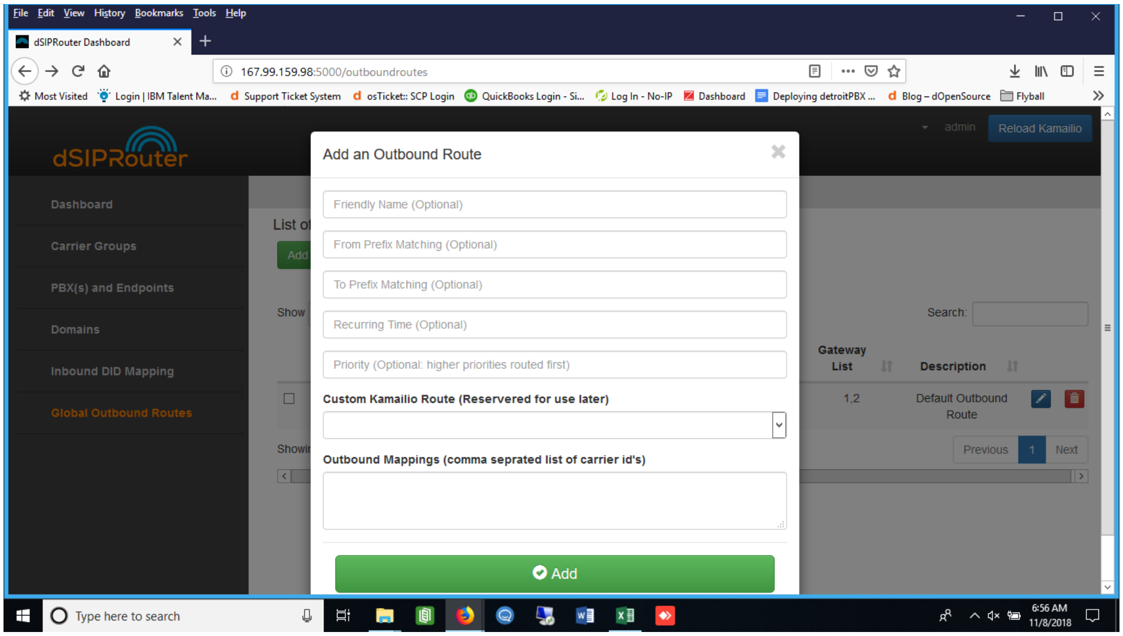


Rule ID	From Prefix	To Prefix	Recurrence	Priority	Custom Route	Gateway List	Description
1				0		1,2	Default Outbound Route

Showing 1 to 1 of 1 entries

4) a) Enter in the Outbound Route information.

b) Click on the green Add button.



5) Click on the blue Reload Kamailio button in order for the changes to be updated.

CHAPTER 5

API

5.1 API

KAMAILIO API		
PUT	Update existing information at endpoint	N/A
GET	Get Information from Endpoint	• /api/v1/kamailio/stats/ • /api/v1/kamailio/reload/
POST	Create new information at endpoint	N/A
DELETE	Delete information at endpoint	N/A

ENDPOINT API		
PUT	Update existing information at endpoint	• /api/v1/endpoint/lease/<int:leaseid>/revoke
GET	Get Information from Endpoint	• /api/v1/endpoint/lease/
POST	Create new information at endpoint	• /api/v1/endpoint/<int:id>
DELETE	Delete information at endpoint	N/A

INBOUND MAPPING API		
PUT	Update existing information at endpoint	• /api/v1/inboundmapping
GET	Get Information from Endpoint	• /api/v1/inboundmapping
POST	Create new information at endpoint	• /api/v1/inboundmapping
DELETE	Delete information at endpoint	• /api/v1/inboundmapping

The steps to obtain the API Token key and using the different curl commands are listen below.

Note: Make sure to to login to your instance via ssh.

5.1.1 Getting Your Token

```
DSIP_TOKEN=$(grep 'API_TOKEN' /etc/dsiprouter/gui/settings.py | cut -d "'" -
↪f 2)
```

5.1.2 Executing Kamailio stats API

```
curl -H "Authorization: Bearer $DSIP_TOKEN"
-X GET http://demo.dsiprouter.org:5000/api/v1/kamailio/stats
```

One Line Version:

```
curl -H "Authorization: Bearer $DSIP_TOKEN" -X GET http://
↪<addressOfYourInstance>:5000/api/v1/kamailio/stats
```

5.1.3 Executing Lease Point API

Getting the endlease

```
curl -H "Authorization: Bearer $DSIP_TOKEN" -H "Content-Type: application/
↪json"
-X GET "http://demo.dsiprouter.org:5000/api/v1/endpoint/lease?ttl=15&
↪email=mack@dsiprouter.org"
```

One Line Version:

```
curl -H "Authorization: Bearer $DSIP_TOKEN" -H "Content-Type: application/
↪json" -X GET "http://demo.dsiprouter.org:5000/api/v1/endpoint/lease?ttl=15&
↪email=mack@dsiprouter.org"
```

Revoking and replacing with your own lease ID

```
curl -H "Authorization: Bearer $DSIP_TOKEN" -H "Content-Type: application/
↪json"
-X PUT "http://demo.dsiprouter.org:5000/api/v1/endpoint/lease/1/revoke"
```

One Line Version:

```
curl -H "Authorization: Bearer $DSIP_TOKEN" -H "Content-Type: application/
↪json" -X PUT "http://demo.dsiprouter.org:5000/api/v1/endpoint/lease/1/
↪revoke"
```

5.1.4 Inbound Mapping Valid commands

GET /api/v1/inboundmapping

```
curl -X GET -H "Authorization: Bearer ${token}" "http://demo.dsiprouter.
↪org:5000/api/v1/inboundmapping"
curl -X GET -H "Authorization: Bearer ${token}" "http://demo.dsiprouter.
↪org:5000/api/v1/inboundmapping?ruleid=3"
curl -X GET -H "Authorization: Bearer ${token}" "http://demo.dsiprouter.
↪org:5000/api/v1/inboundmapping?did=1313"
```

POST /api/v1/inboundmapping

```
curl -X POST -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping" -d '{"did": "1313", "servers": ["66","67"], "notes": "1313␣
↪DID Mapping"}'
curl -X POST -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping" -d '{"did": "1313","servers": ["66","67"]}'
curl -X POST -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping" -d '{"did": "", "servers": ["66"], "notes": "Default DID␣
↪Mapping"}'
```

PUT /api/v1/inboundmapping

```
curl -X PUT -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping?ruleid=3" -d '{"did": "01234", "notes": "01234 DID Mapping"}
↪'
curl -X PUT -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping?did=1313" -d '{"servers": ["67"]}'
curl -X PUT -H "Authorization: Bearer ${token}" --connect-timeout 3 -H
↪"Content-Type: application/json" "http://demo.dsiprouter.org:5000/api/v1/
↪inboundmapping?did=1313" -d '{"did": "01234", "notes": "01234 DID Mapping"}
↪'
```

DELETE /api/v1/inboundmapping

```
curl -X DELETE -H "Authorization: Bearer ${token}" "http://demo.dsiprouter.
↪org:5000/api/v1/inboundmapping?ruleid=3"
curl -X DELETE -H "Authorization: Bearer ${token}" "http://demo.dsiprouter.
↪org:5000/api/v1/inboundmapping?did=1313"
```

CHAPTER 6

Implementing Use Cases

6.1 Use Cases

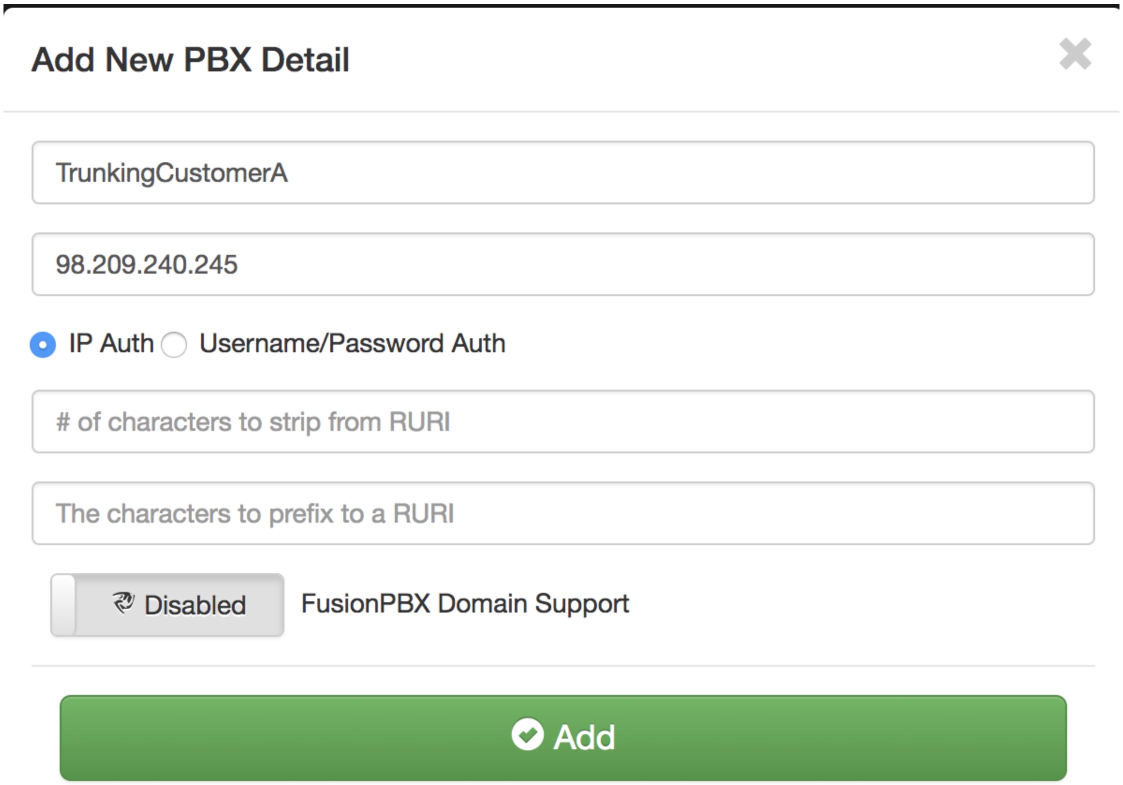
This section contains a list of the common use cases that are implemented using dSIPRouter

6.1.1 SIP Trunking Using IP Authentication

dSIPRouter enables an organization to start supporting SIP Trunking within minutes. Here are the steps to set it up using IP Authentication:

1. Login to dSIPRouter
2. Validate that your carrier is defined and specified in the Global Outbound Routes. If not, please follow the steps in **:ref:‘carrier_groups‘_** and/or **:ref:‘global_outbound_routes‘_** documentation.
3. Click on PBX’s and Endpoints
4. Click “Add”
5. Select **IP Authentication** and fill in the fields specified below:

- Friendly Name
- IP Address of the PBX or Endpoint Device



Add New PBX Detail

TrunkingCustomerA

98.209.240.245

● IP Auth ○ Username/Password Auth

# of characters to strip from RURI
The characters to prefix to a RURI
Disabled
Add

FusionPBX Domain Support

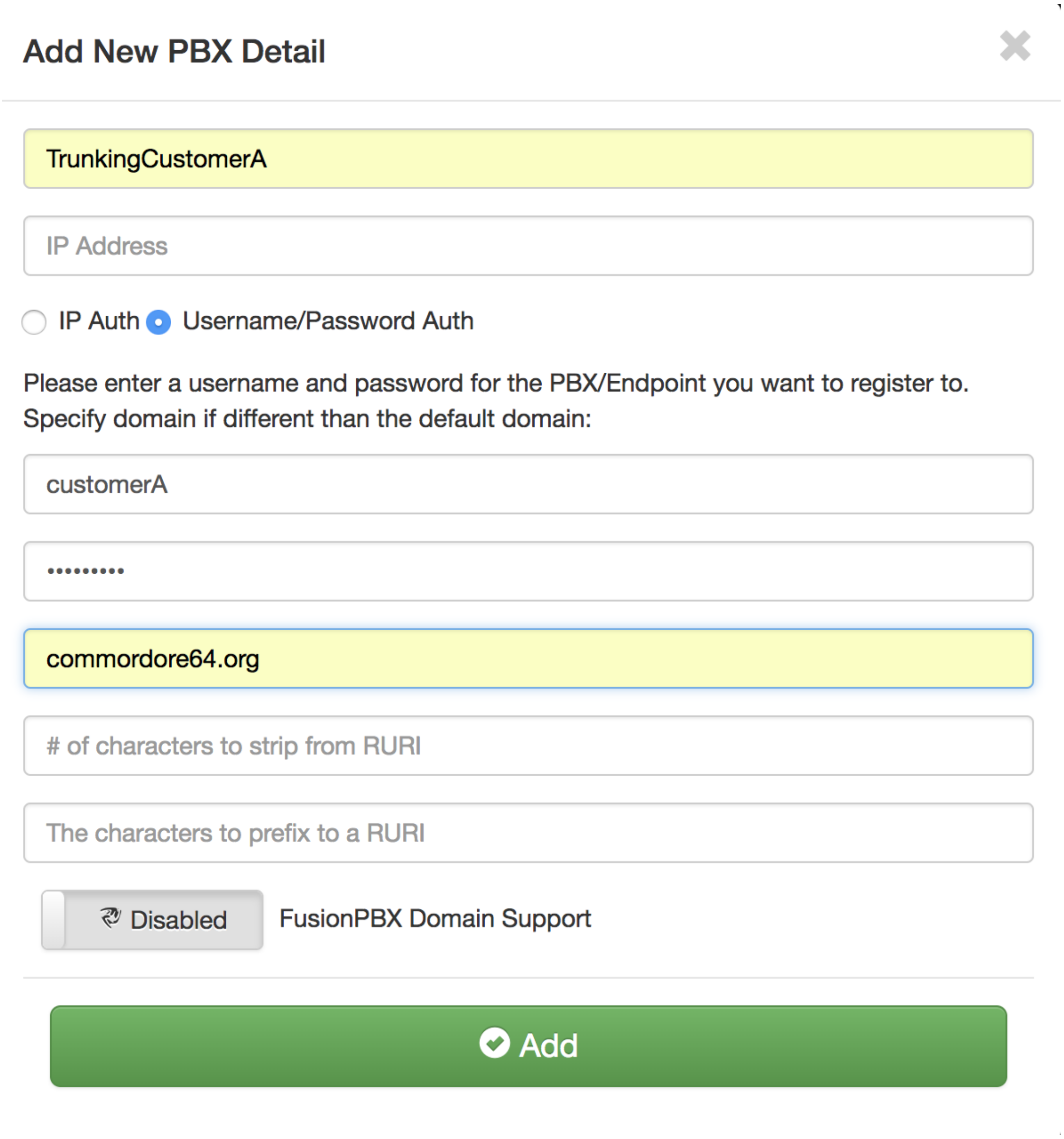
6. Click "Add"
7. Click "Reload" to make the change active.

6.1.2 SIP Trunking Using Username/Password Authentication

Here are the steps to set it up using Username/Password Authentication:

1. Login to dSIPRouter
2. Valiate that your carrier is defined and specified in the Global Outbound Routes. If not, please follow the steps in carrier_groups.rst and/or global_outbound_routes documentation.
3. Click on PBX's and Endpoints
4. Click "Add"
5. Select **Username/Password Authentication** and fill in the fields specified below:

- Friendly Name
- Click the "Username/Password Auth" radio button
- Enter a username
- Enter a domain. Note, you can make up the domain name. If you don't specify one then the default domain will be used, which is sip.dsiprouter.org by default.
- Enter a password



Add New PBX Detail

TrunkingCustomerA

IP Address

○ IP Auth ● Username/Password Auth

Please enter a username and password for the PBX/Endpoint you want to register to. Specify domain if different than the default domain:

customerA

•••••••••

commordore64.org

# of characters to strip from RURI

The characters to prefix to a RURI

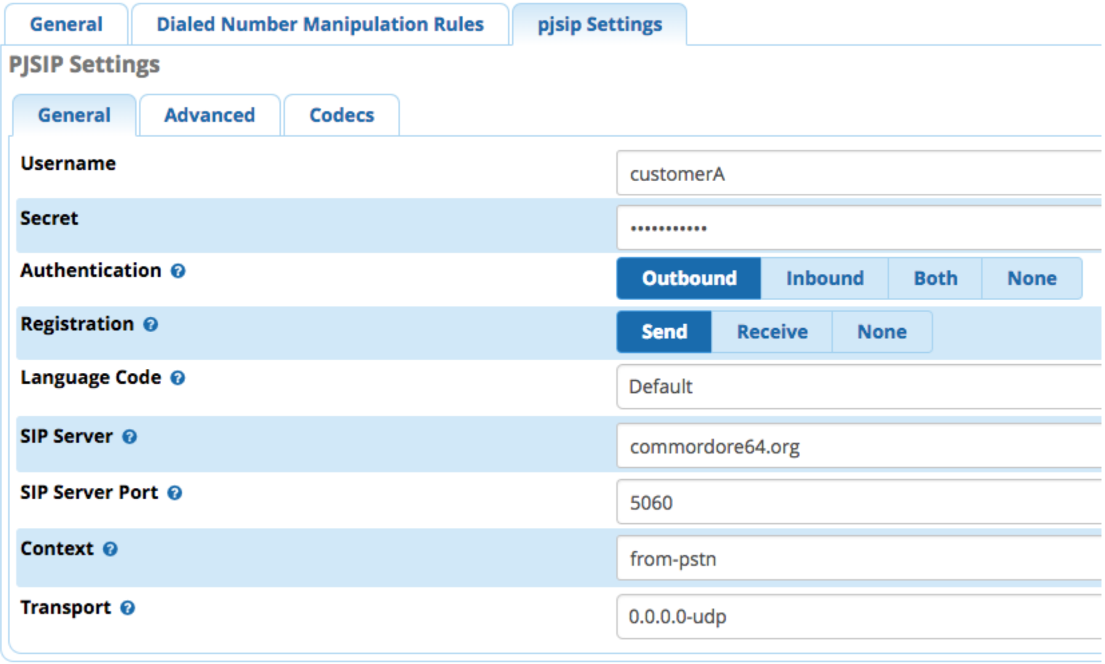
Disabled FusionPBX Domain Support

6. Click "Add"
7. Click "Reload" to make the change active.

6.1.3 Using PJSIP Trunking - FreePBX Example

The following screenshot(s) shows how to configure a PJSIP trunk within FreePBX for Username/Password Authentication.

The first screenshot shows the General tab of the "pjsip settings" page:



Username: customerA

Secret: ••••••••••

Authentication: [✓ Outbound] Inbound Both None

Registration: [✓ Send] Receive None

Language Code: Default

SIP Server: commordore64.org

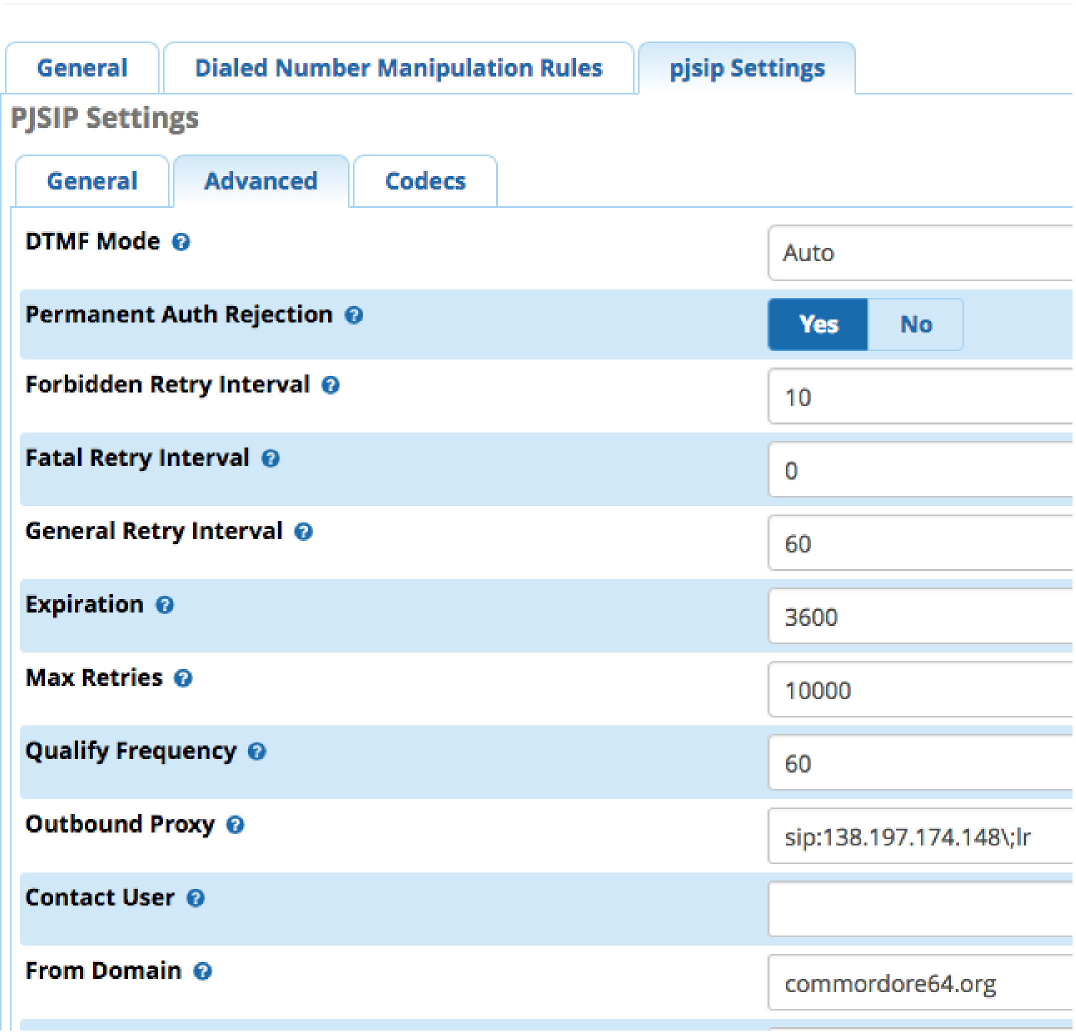
SIP Server Port: 5060

Context: from-pstn

Transport: 0.0.0.0-udp

The following fields needs to be entered

Field	Value
Username	Username from dSIPRouter PBX Setup
Secret	Password from dSIPRouter PBX Setup
Authentication	Outbound
Registration	Send
SIP Server	Domain name defined in the dSIPRouter PBX Setup
SIP Server	SIP port, which is 5060 in dSIPRouter



DTMF Mode: Auto

Permanent Auth Rejection: [✓ Yes] No

Forbidden Retry Interval: 10

Fatal Retry Interval: 0

General Retry Interval: 60

Expiration: 3600

Max Retries: 10000

Qualify Frequency: 60

Outbound Proxy: sip:138.197.174.148\;lr

Contact User:

From Domain: commordore64.org

The following fields needs to be entered

Field	Value
Outbound Proxy	IP address of dSIPRouter - must include the ";lr" at the end
From Domain	The name of the domain defined in the dSIPRouter PBX Setup

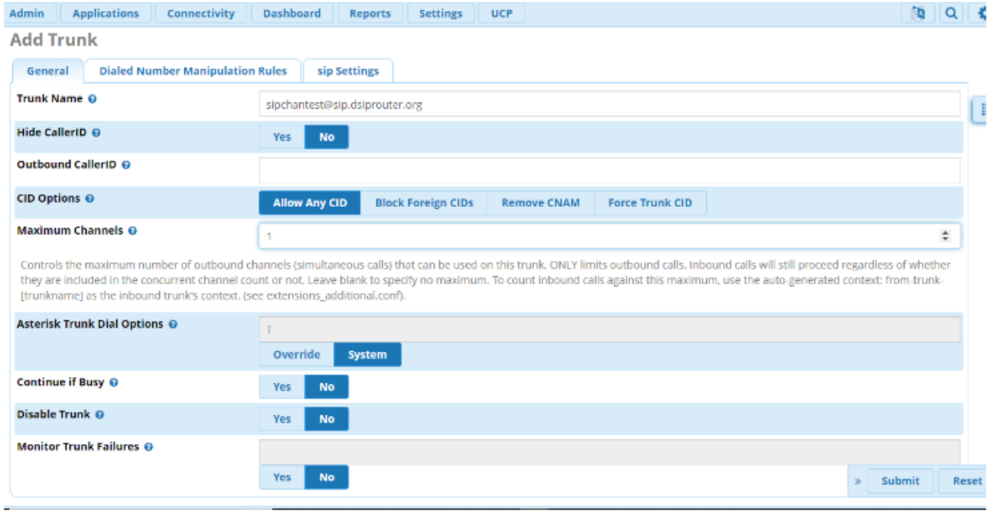
6.1.4 Using chanSIP Trunking - FreePBX Example

The following screenshot(s) shows how to configure a chanSIP trunk within FreePBX for Username/Password Authentication.

1. Log into FreePBX server
2. Click Connectivity→Trunks
3. Select Add SIP (chan_sip) Trunk
4. Under General tab enter

   The following fields needs to be entered

Field	Value
Trunk Name	Labeled in dsiprouter
Outbound Caller ID	Phone# that you want to appear during a outbound call (if applicable)



Add Trunk

Trunk Name: sipchantest@sip.dsiprouter.org

Hide CallerID: Yes [✓ No]

Outbound CallerID:

CID Options: [✓ Allow Any CID] Block Foreign CIDs Remove CNAM Force Trunk CID

Maximum Channels: 1

Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. ONLY limits outbound calls. Inbound calls will still proceed regardless of whether they are included in the concurrent channel count or not. Leave blank to specify no maximum. To count inbound calls against this maximum, use the auto generated context: from-trunk-[trunkname] as the inbound trunk's context. (see extensions_additional.conf).

Asterisk Trunk Dial Options: T — Override [✓ System]

Continue if Busy: Yes [✓ No]

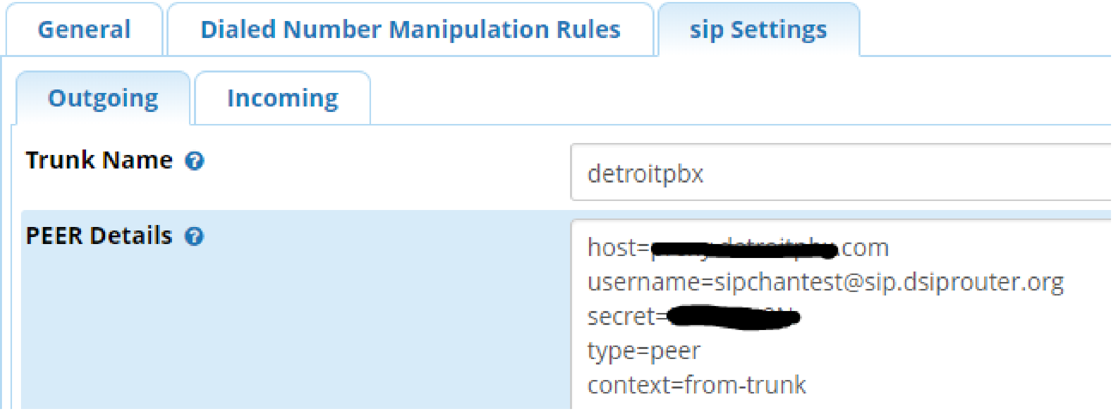
Disable Trunk: Yes [✓ No]

Monitor Trunk Failures: Yes [✓ No]

5. Next you will enter the configurations under the SIP Settings. Here you will enter the SIP settings for outgoing calls by selecting the **Outbound** tab. You will need the following information: The following fields needs to be entered

Field	Value
Host	<host name or IP address of dsiprouter>
Username	<Specified in dsiprouter@domainname>
Secret	<Specified in dsiprouter>
Type	peer
Context	from-trunk

The domain name has to be included and correct.



Trunk Name: detroitpbx

PEER Details:

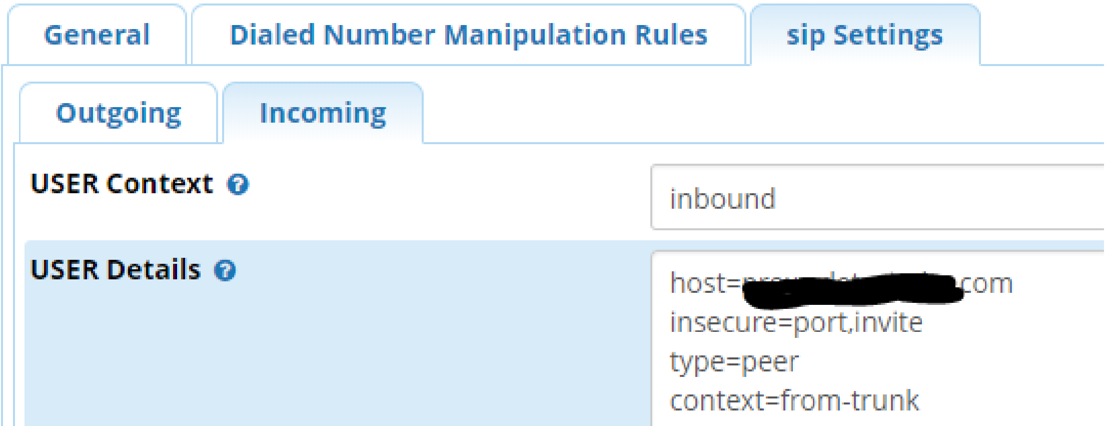
```
host=[redacted].com
username=sipchantest@sip.dsiprouter.org
secret=[redacted]
type=peer
context=from-trunk
```

NOTE:** Type <context=from-trunk> underneath the <type=peer> in the Peer Details box if it does not appear.

6. Next you will enter the configurations for incoming by selecting the **Incoming** tab in the SIP Settings. Here you will enter the SIP settings for inbound calls. You will need:

User Context: This is most often the account name or number your provider expects. In this example we named it “inbound”. The following User Details needs to be entered:

Field	Value
Host	<host name or IP address of dsiprouter>
Insecure	port,invite
Type	peer
Context	from-trunk

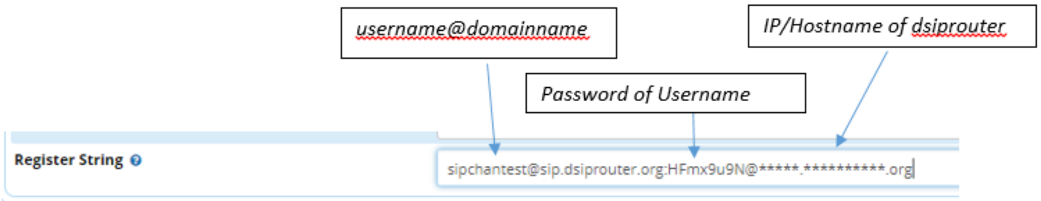


USER Context: inbound

USER Details:

```
host=[redacted]com
insecure=port,invite
type=peer
context=from-trunk
```

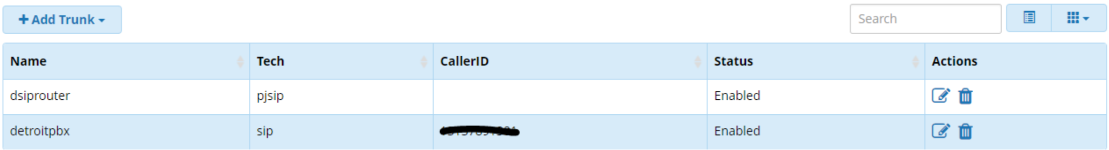
In the **Register String** enter: <username@domainname>:<password>@<ip address **or** hostname>. In this example it would be sipchantest@sip.dsiprouter.org:HFmx9u9N@demo.dsiprouter.org. **The domain name has to be included and correct.**



Register String: sipchantest@sip.dsiprouter.org:HFmx9u9N@*****.**********.org

7. Click Submit
8. Be sure to click the **Apply Config** button after submitting to confirm.

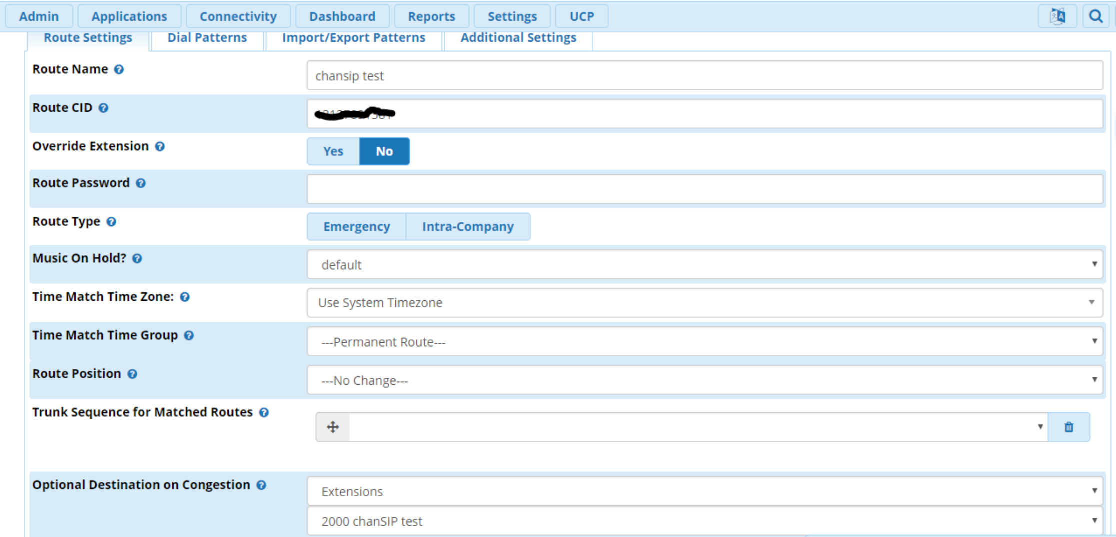
You will now be able to see the new chanSIP added in the truck.



Name	Tech	CallerID	Status	Actions
dsiprouter	pjsip		Enabled	
detroitpbx	sip	[redacted]	Enabled	

9. Next you will need to setup an outbound route. Select Connectivity→ Outbound Routes. Click the “+” sign to add a outbound route. In this tab you will need to enter:

Field	Value
Route Name	Type desired name
Route CID	Number you want to appear on caller ID
Trunk Sequence for Matched Routes	Trunk name (select from drop down box)



Route Name: chansip test

Route CID: [redacted]

Override Extension: Yes **No**

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?: default

Time Match Time Zone: Use System Timezone

Time Match Time Group: ---Permanent Route---

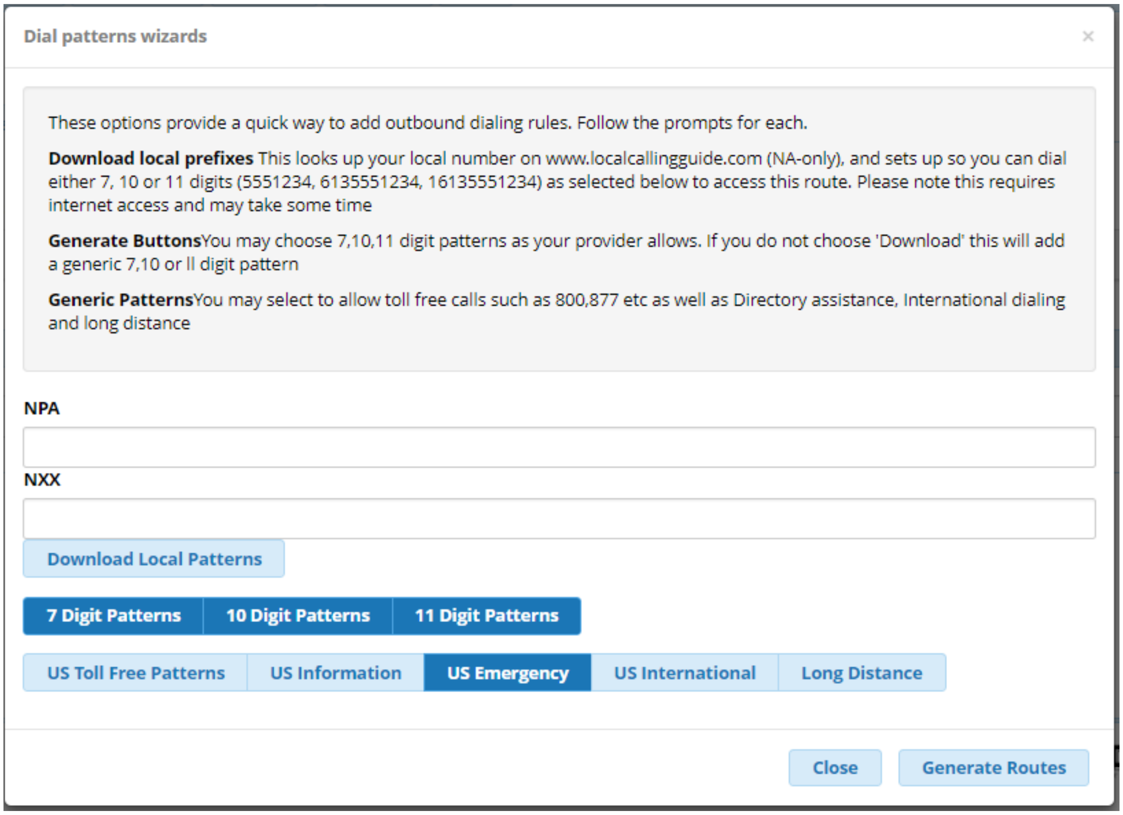
Route Position: ---No Change---

Trunk Sequence for Matched Routes:

Optional Destination on Congestion: Extensions

2000 chanSIP test

10. Click the Dial Patterns tab to set the dial patterns. If you are familiar with dial patterns, you can enter the dial patterns manually or you can click the Dial Patterans Wizard to auto create dial patterns if you like. You can choose 7, 10 or 11 digit patterns. Click Generate Routes.



Dial patterns wizards

These options provide a quick way to add outbound dialing rules. Follow the prompts for each.

Download local prefixes This looks up your local number on www.localcallingguide.com (NA-only), and sets up so you can dial either 7, 10 or 11 digits (5551234, 6135551234, 16135551234) as selected below to access this route. Please note this requires internet access and may take some time

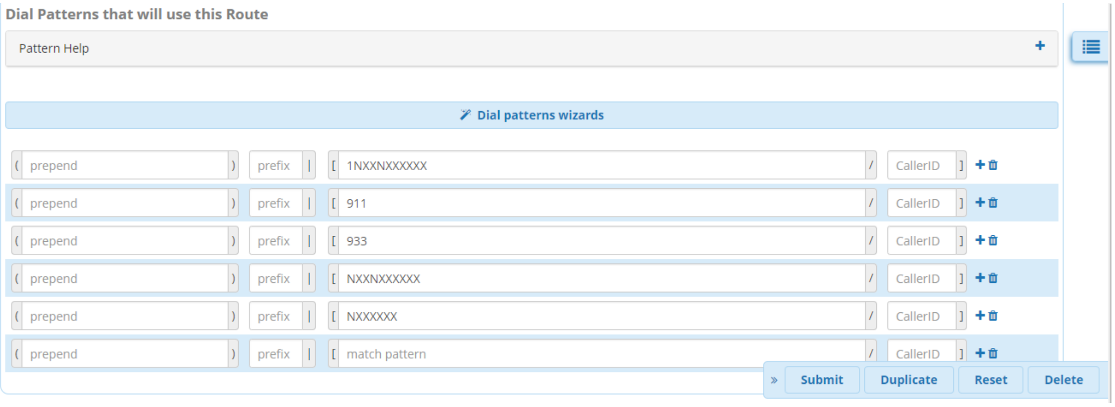
Generate Buttons You may choose 7,10,11 digit patterns as your provider allows. If you do not choose 'Download' this will add a generic 7,10 or ll digit pattern

Generic Patterns You may select to allow toll free calls such as 800,877 etc as well as Directory assistance, International dialing and long distance

NPA

NXX

Dial pattern is set to your preference. Prefixes are optional, not required.



Dial Patterns that will use this Route

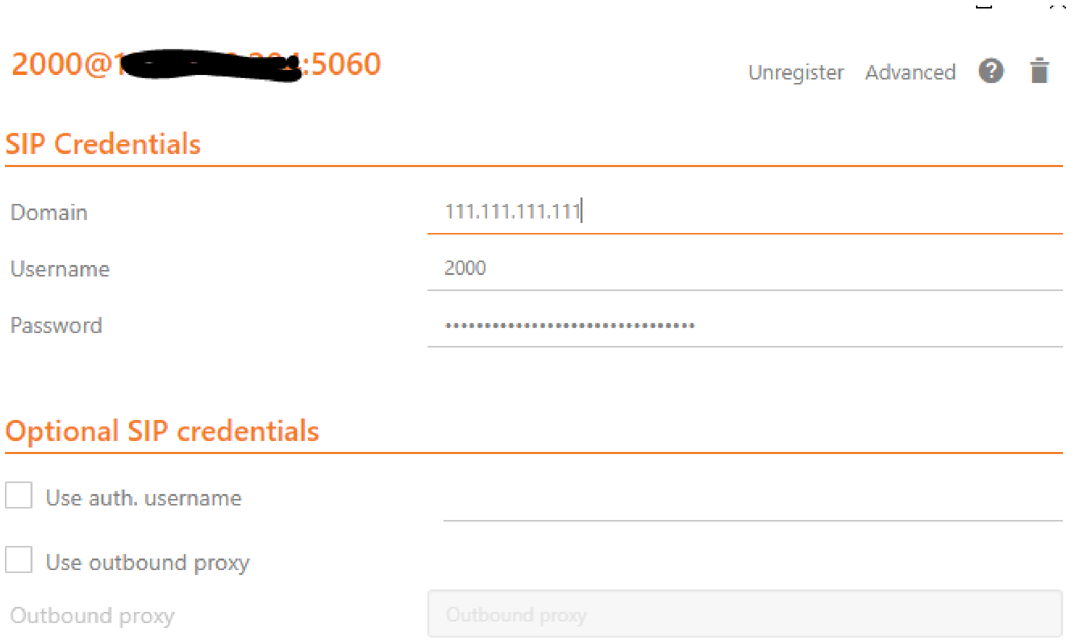
prepend	prefix	match pattern	CallerID
(prepend)	prefix |	[1NXXNXXXXXX /	CallerID]
(prepend)	prefix |	[911 /	CallerID]
(prepend)	prefix |	[933 /	CallerID]
(prepend)	prefix |	[NXXNXXXXXX /	CallerID]
(prepend)	prefix |	[NXXXXXX /	CallerID]
(prepend)	prefix |	[match pattern /	CallerID]

11. Click Submit and Apply Config button.

Assuming you already have an extention created in your FreePBX, you can validate incoming/outgoing calls by configuring a softphone or a hard phone. Below is an example of the information you would enter if you use a softphone: In this example we are using Zoiper. Once you've downloaded Zoiper application on your PC or smart device you would enter the following to configure the soft phone:

Field	Value
Username	<extension>@<siptrunkipaddress>
secret	<Password of that extension>
Hostname	<IP address of your FreePBX> (should autofill)

Note Skip Authenication and Outbound Proxy



2000@1[redacted]:5060

SIP Credentials

Domain: 111.111.111.111

Username: 2000

Password: ••••••••••••••••••••••••••••••••••

Optional SIP credentials

□ Use auth. username

□ Use outbound proxy

Outbound proxy: Outbound proxy

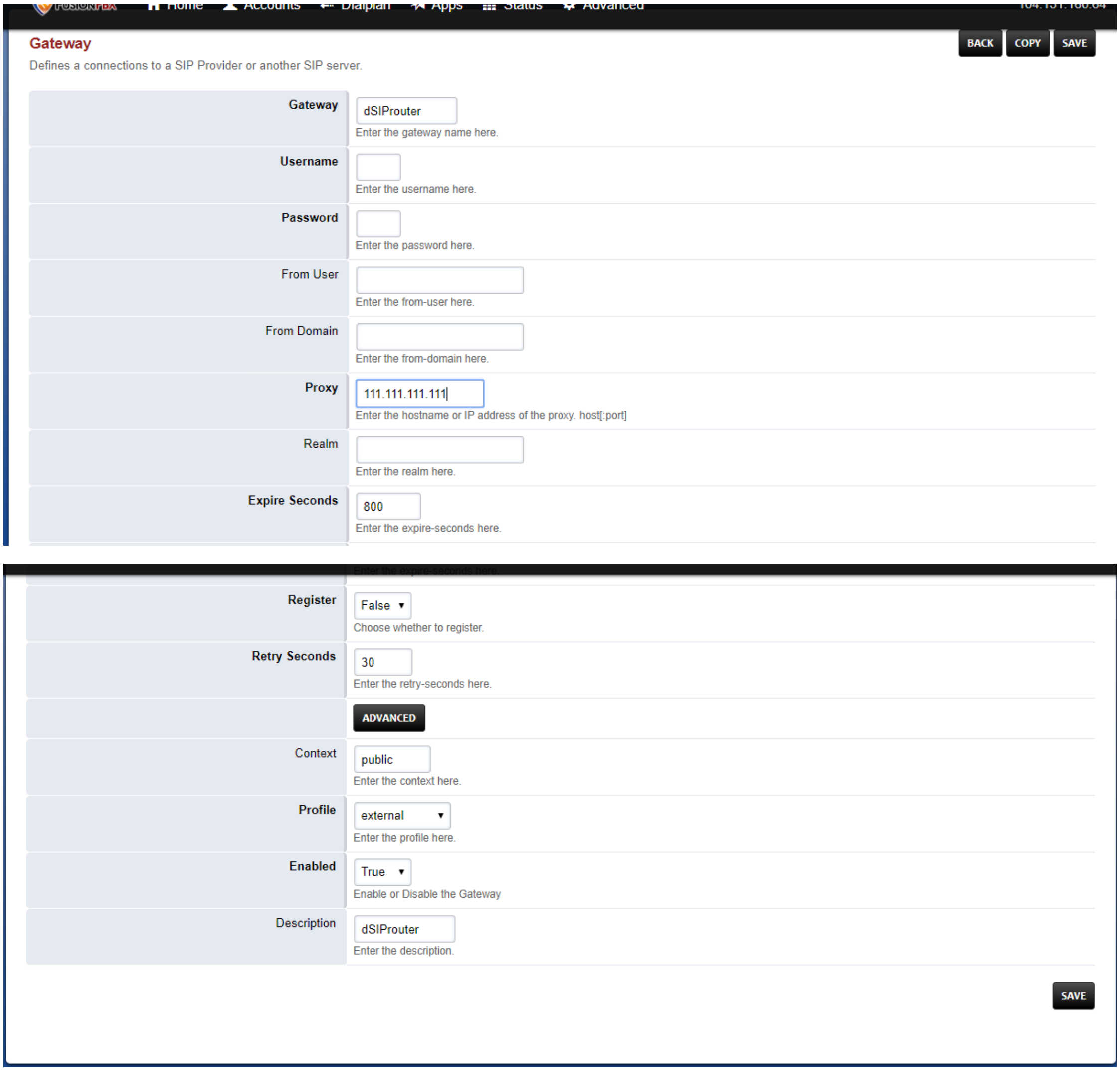
You should now be able to make a inbound and outbound call successfully!

6.1.5 Using SIP Trunking - FusionPBX IP Authenication

The following screenshot(s) shows how to configure a SIP trunk within FusionPBX for IP Authenication.

1. Log into your FusionPBX.
2. Click Accounts –> Gateways–>Click the + sign to add a gateway/SIP Trunk. The only fields you will need to fill here are:

- Gateway= Name of the SIP Trunk
- Proxy= IP address of the SIP trunk
- Register= Change to False because you are using IP authenication.



Gateway

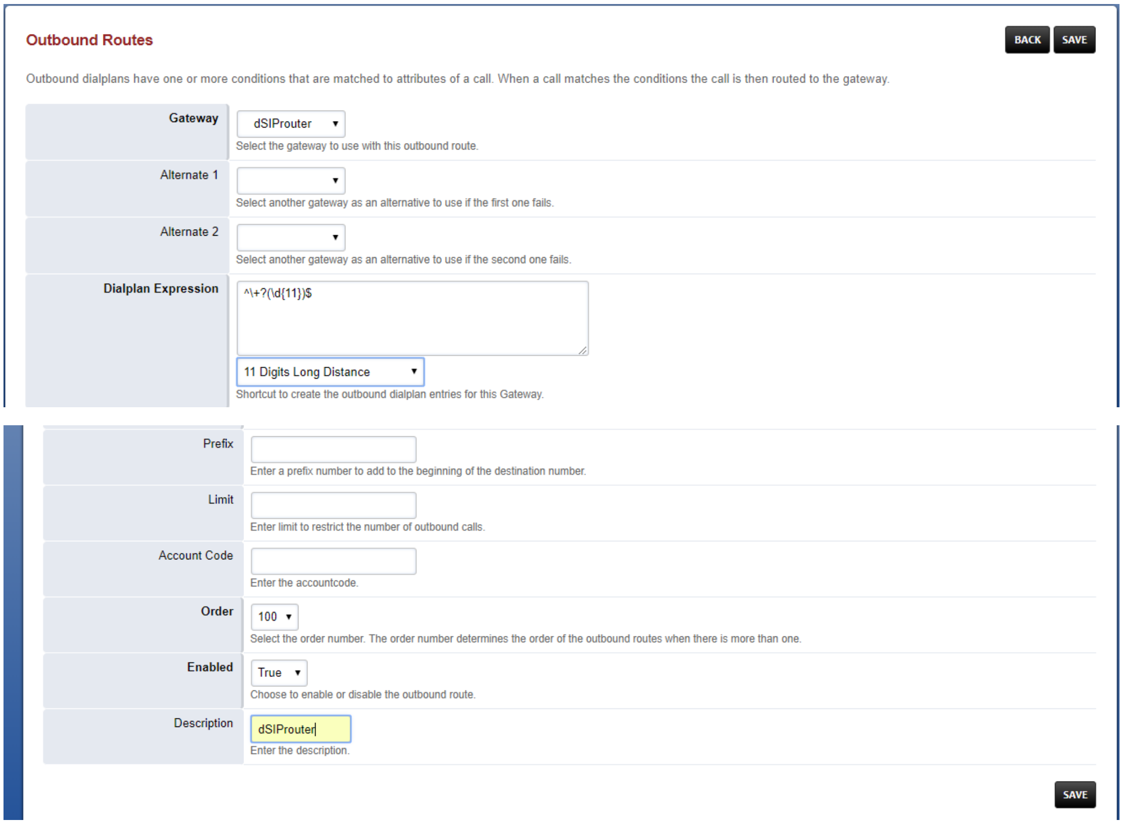
Defines a connections to a SIP Provider or another SIP server.

Gateway	dSIProuter Enter the gateway name here.
Username	Enter the username here.
Password	Enter the password here.
From User	Enter the from-user here.
From Domain	Enter the from-domain here.
Proxy	111.111.111.111 Enter the hostname or IP address of the proxy. host[:port]
Realm	Enter the realm here.
Expire Seconds	800 Enter the expire-seconds here.
Register	False ▾ Choose whether to register.
Retry Seconds	30 Enter the retry-seconds here.
Context	public Enter the context here.
Profile	external ▾ Enter the profile here.
Enabled	True ▾ Enable or Disable the Gateway
Description	dSIProuter Enter the description.

3. Click Save
4. Click DialPlan–>Outboung Routes–>Click the + sign to add a outbound route. Here you will enter in the following fields:

- Gateway= Name of the SIP Trunk
- Alternate gateways (if applicable)
- DialPlan Expression= 11d (standard setup in FusionPBX). To change the dialplan expression click on the dropdown box where it says “Shortcut to create the outbound dialplan entries for this Gateway.”
- Description= (if desired)

5. Click Save

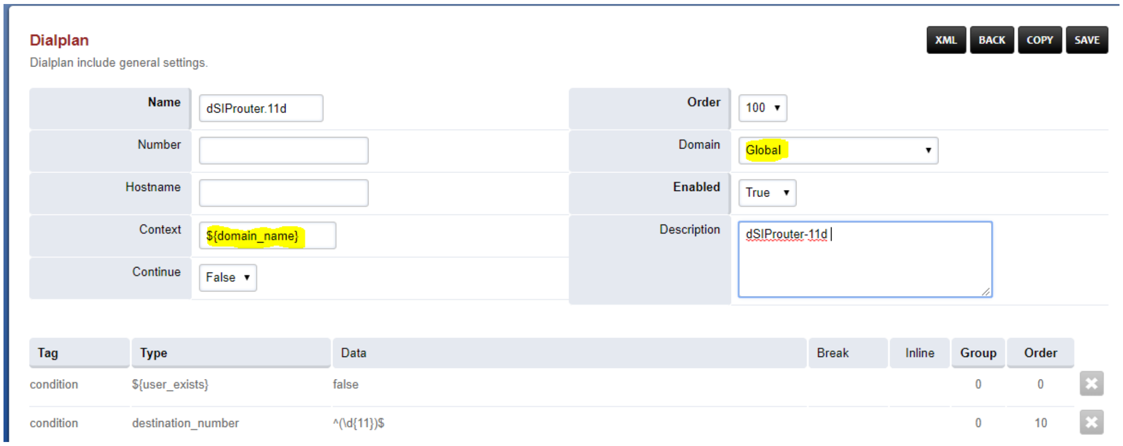


Outbound Routes

Outbound dialplans have one or more conditions that are matched to attributes of a call. When a call matches the conditions the call is then routed to the gateway.

Gateway	dSIProuter ▾ Select the gateway to use with this outbound route.
Alternate 1	▾ Select another gateway as an alternative to use if the first one fails.
Alternate 2	▾ Select another gateway as an alternative to use if the second one fails.
Dialplan Expression	^\+?(\d{11})$ 11 Digits Long Distance ▾ Shortcut to create the outbound dialplan entries for this Gateway.
Prefix	Enter a prefix number to add to the beginning of the destination number.
Limit	Enter limit to restrict the number of outbound calls.
Account Code	Enter the accountcode.
Order	100 ▾ Select the order number. The order number determines the order of the outbound routes when there is more than one.
Enabled	True ▾ Choose to enable or disable the outbound route.
Description	dSIProuter Enter the description.

NOTE To make these changes global for ALL domains for this SIP Trunk: reopen outbound routes and change the Domain to Global and the Context to ${domain_name} as shown below.



Dialplan

Dialplan include general settings.

Name	dSIProuter.11d	**Order**	100 ▾
Number		Domain	Global ▾
Hostname		**Enabled**	True ▾
Context	${domain_name}	Description	dSIProuter-11d
Continue	False ▾		

Tag	Type	Data	Break	Inline	Group	Order
condition	${user_exists}	false			0	0
condition	destination_number	^(\d{11})$			0	10

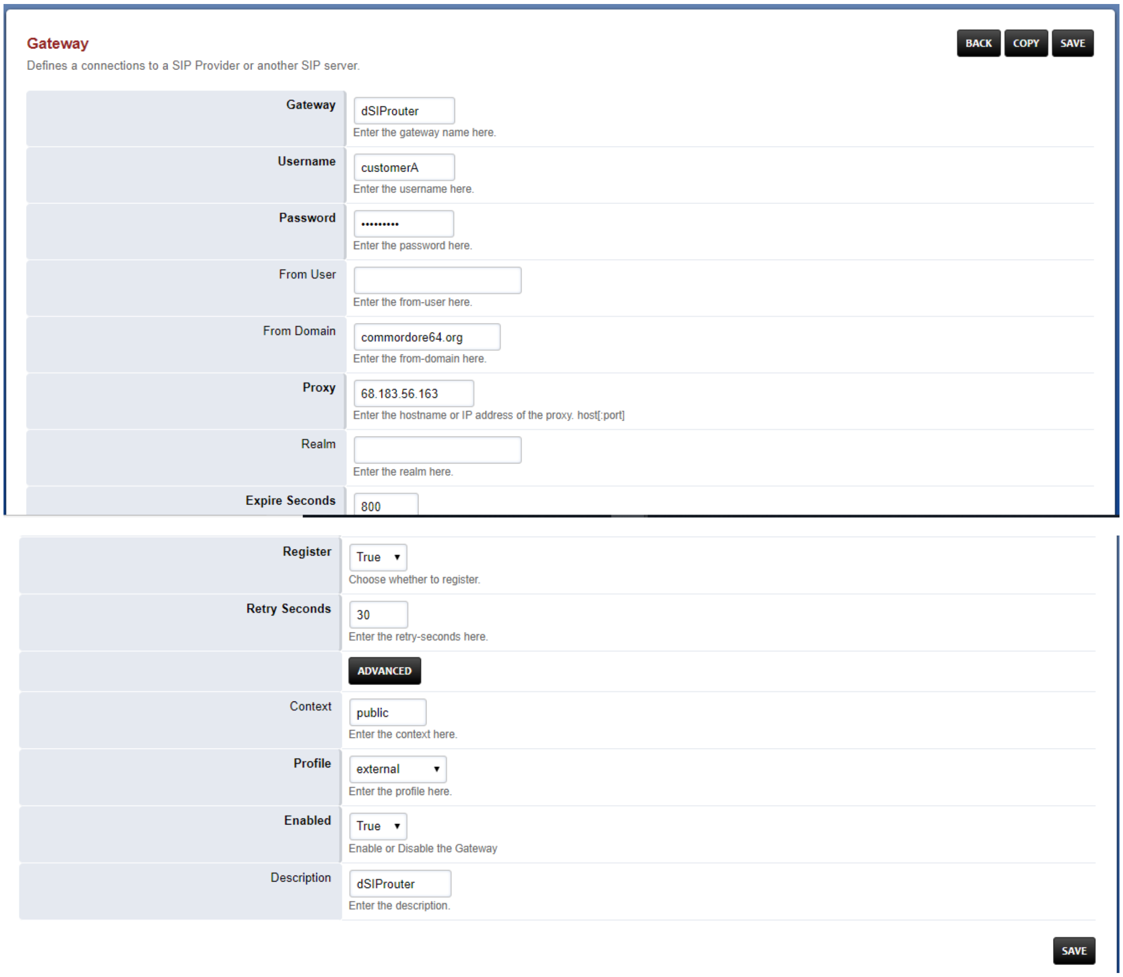
6.1.6 Using SIP Trunking - FusionPBX Username/Password Authenication

The following screenshot(s) shows how to configure a SIP trunk within FusionPBX for Username/Password Authenication with IP Authenication off.

1. Log into your FusionPBX.

2. Click Accounts –> Gateways–>Click the + sign to add a gateway/SIP Trunk. The following fields you will need to fill here are:

- Gateway= Name of the SIP Trunk
- Username= specified by dSIPRouter provider
- Password= specified by dSIPRouter provider
- From Domain= Specified or set by default
- Proxy= IP address of the SIP trunk
- Register= set to True because you are using Username/Password authenication.



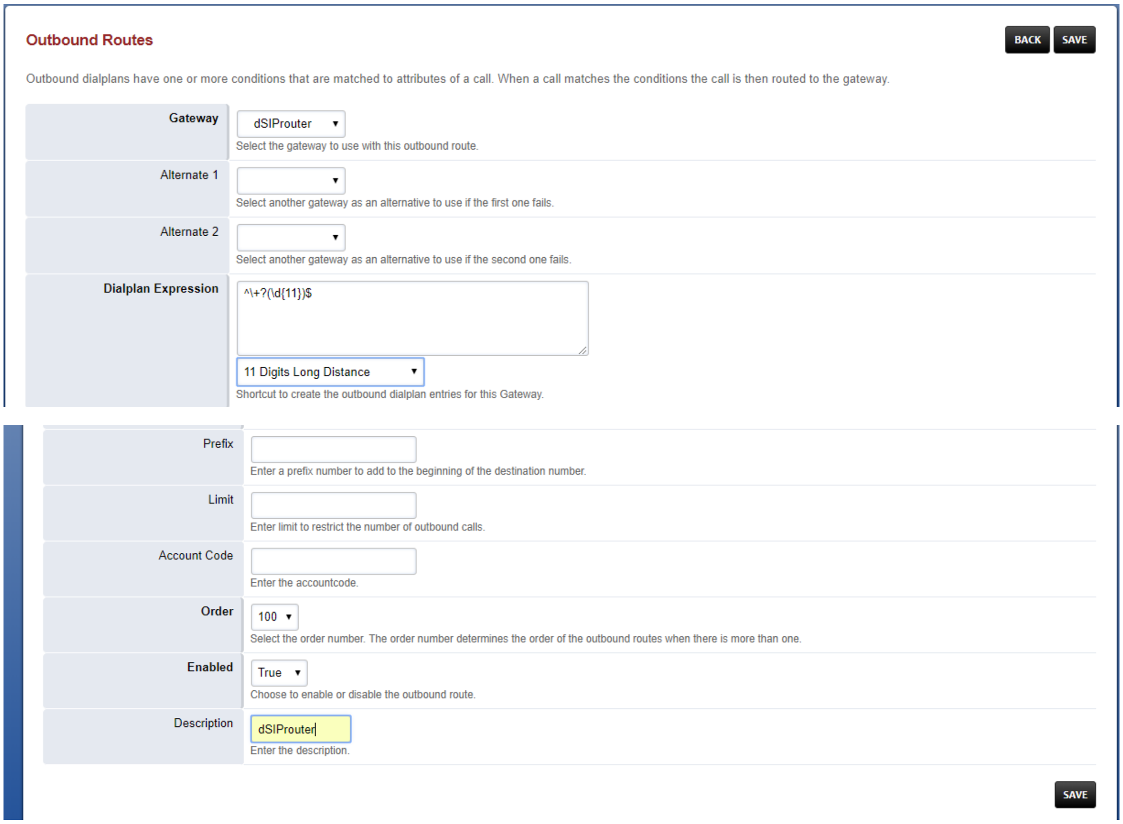
Gateway

Defines a connections to a SIP Provider or another SIP server.

Gateway	dSIProuter Enter the gateway name here.
Username	customerA Enter the username here.
Password	•••••••• Enter the password here.
From User	Enter the from-user here.
From Domain	commordore64.org Enter the from-domain here.
Proxy	68.183.56.163 Enter the hostname or IP address of the proxy. host[:port]
Realm	Enter the realm here.
Expire Seconds	800
Register	True ▾ Choose whether to register.
Retry Seconds	30 Enter the retry-seconds here.
	ADVANCED
Context	public Enter the context here.
Profile	external ▾ Enter the profile here.
Enabled	True ▾ Enable or Disable the Gateway
Description	dSIProuter Enter the description.

3. Click Save.
4. Click DialPlan–>Outboung Routes–>Click the + sign to add a outbound route. Here you will enter in the following fields:

- Gateway= Name of the SIP Trunk
- Alternate gateways (if applicable)
- DialPlan Expression= 11d (standard setup in FusionPBX). To change the dialplan expression click on the dropdown box where it says “Shortcut to create the outbound dialplan entries for this Gateway.”
- Description= (if desired)



Outbound Routes

Outbound dialplans have one or more conditions that are matched to attributes of a call. When a call matches the conditions the call is then routed to the gateway.

Gateway	dSIProuter ▾ Select the gateway to use with this outbound route.
Alternate 1	▾ Select another gateway as an alternative to use if the first one fails.
Alternate 2	▾ Select another gateway as an alternative to use if the second one fails.
Dialplan Expression	^\+?(\d{11})$ 11 Digits Long Distance ▾ Shortcut to create the outbound dialplan entries for this Gateway.
Prefix	Enter a prefix number to add to the beginning of the destination number.
Limit	Enter limit to restrict the number of outbound calls.
Account Code	Enter the accountcode.
Order	100 ▾ Select the order number. The order number determines the order of the outbound routes when there is more than one.
Enabled	True ▾ Choose to enable or disable the outbound route.
Description	dSIProuter Enter the description.

5. Click Save

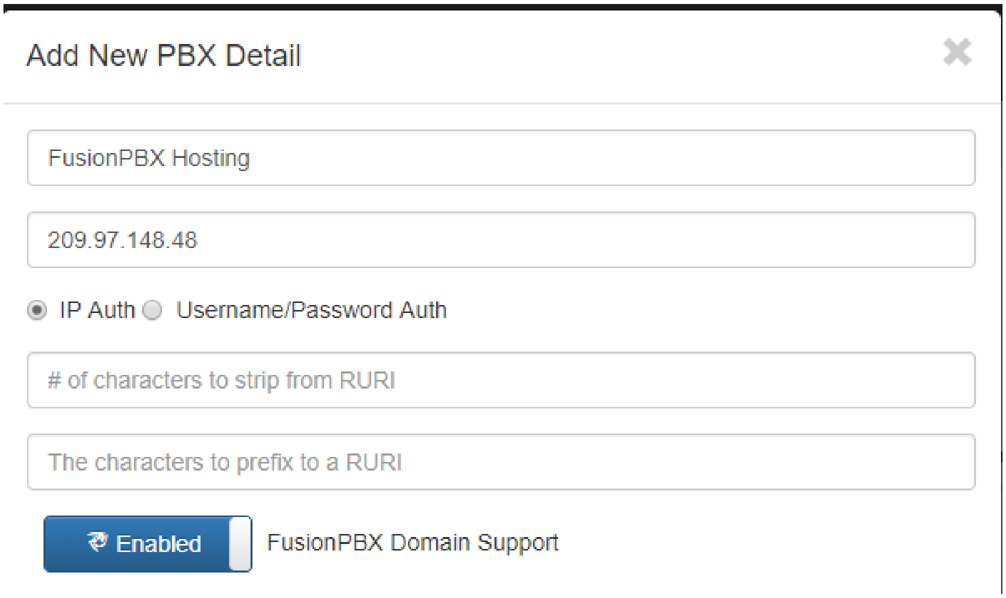
6.1.7 FusionPBX Hosting

Here we will demostrate how to setup dSIPRouter to enable hosting FusionPBX. We have built-in support for FusionPBX that allows domains to be dynamically pulled from FusionPBX.

1. Login to dSIPRouter
2. Click PBX(s) and EndPoints
3. Click ADD; enter the following fields

- Friendly Name (opional)
- IP address
- IP Auth
- Click to enable FusionPBX Domain Support
- FusionPBX Database IP or Hostname

4. Click ADD



Add New PBX Detail

FusionPBX Hosting

209.97.148.48

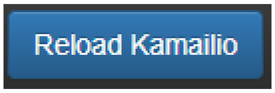
● IP Auth ○ Username/Password Auth

# of characters to strip from RURI

The characters to prefix to a RURI

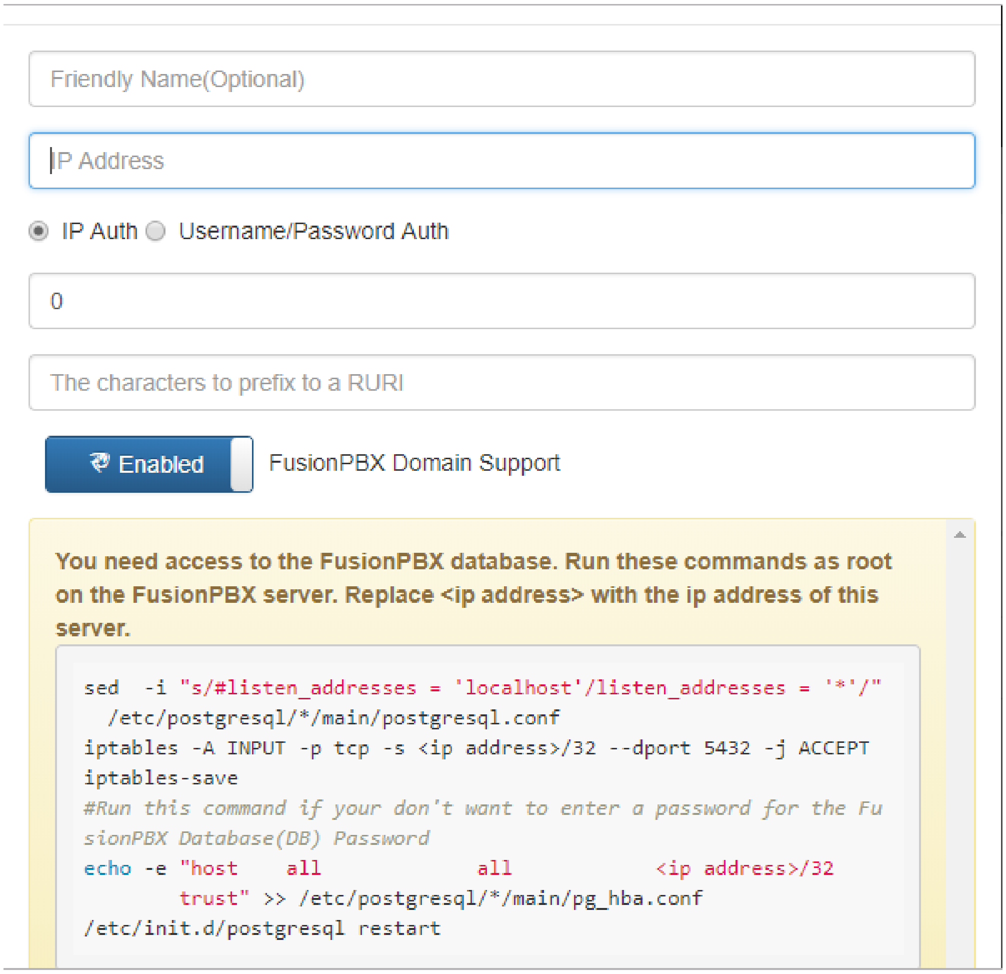
Enabled FusionPBX Domain Support

5. Click Reload Kamailio. (when changes are made reload button will change to orange)



6. Access your FusionPBX database via ssh.

7.Run the command as illustrated in the "Edit your PBX Detail" window as root on the FusionPBX server. Replace <ip address> (not including the brackets) with the IP address of the dSIPRouter server you're adding. Command line will look simulair to the following picture. **NOTE** After you have entered the first two lines of commands you will not see a form of reply. If command is entered correctly it will return back to your root line. If the command line is incorrect you will receive a "command not found" error message. Recheck the command line and IP address.



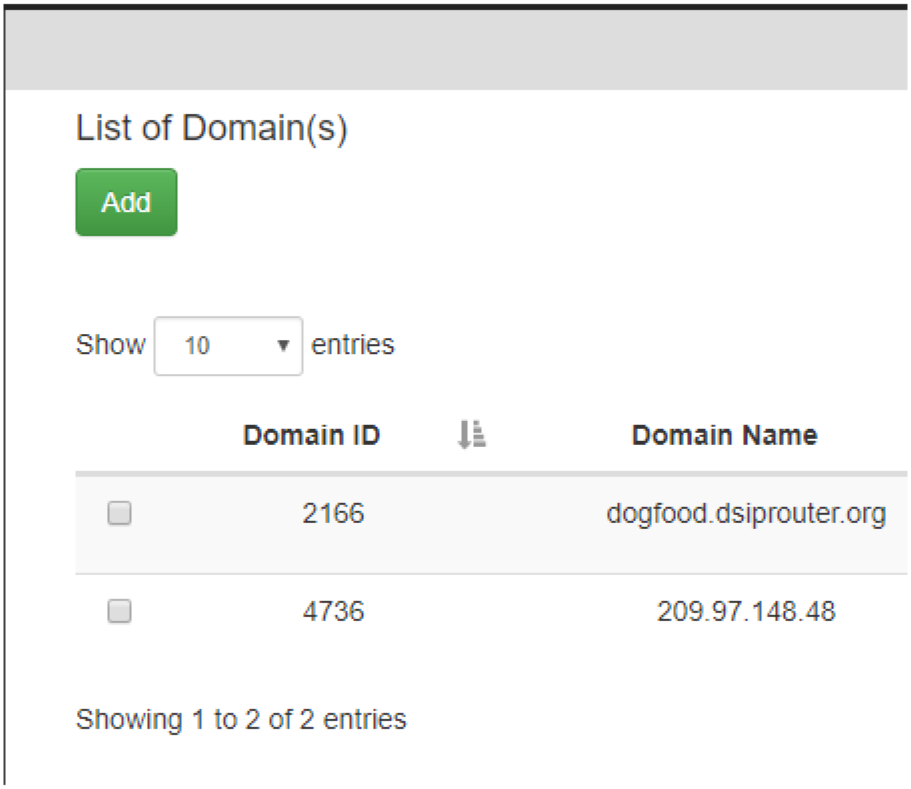
● IP Auth ○ Username/Password Auth

0

You need access to the FusionPBX database. Run these commands as root on the FusionPBX server. Replace <ip address> with the ip address of this server.

```
sed  -i "s/#listen_addresses = 'localhost'/listen_addresses = '*'/"
  /etc/postgresql/*/main/postgresql.conf
iptables -A INPUT -p tcp -s <ip address>/32 --dport 5432 -j ACCEPT
iptables-save
#Run this command if your don't want to enter a password for the Fu
sionPBX Database(DB) Password
echo -e "host    all             all             <ip address>/32
        trust" >> /etc/postgresql/*/main/pg_hba.conf
/etc/init.d/postgresql restart
```

After the command is run you should now be able to see the domains of that PBX in dSIPRouter.



List of Domain(s)

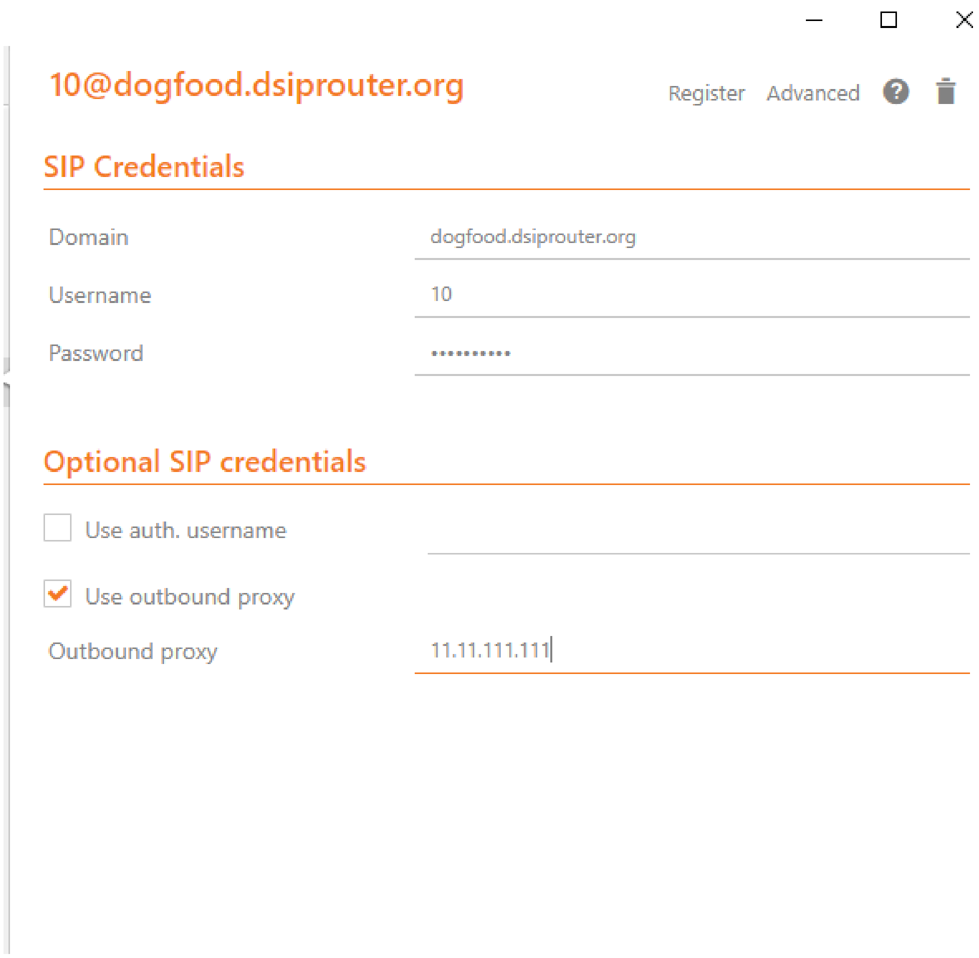
	Domain ID	Domain Name
□	2166	dogfood.dsiprouter.org
□	4736	209.97.148.48

Showing 1 to 2 of 2 entries

You can test PBX Hosting is valid by configuring a softphone or a hard phone. Below is an example using a softphone:

Now that domains have been synced in dSIPRouter you are able to register a softphone. In this example we are using Zoiper. Once you've downloaded Zopier appliaction on your PC or smart device you would add:

- username (extension@domainname)
- password (password of that extension)
- outbound proxy (IP address of the dSIPRouter)



10@dogfood.dsiprouter.org

SIP Credentials

Domain dogfood.dsiprouter.org

Username 10

Password ••••••••••

Optional SIP credentials

□ Use auth. username

☑ Use outbound proxy

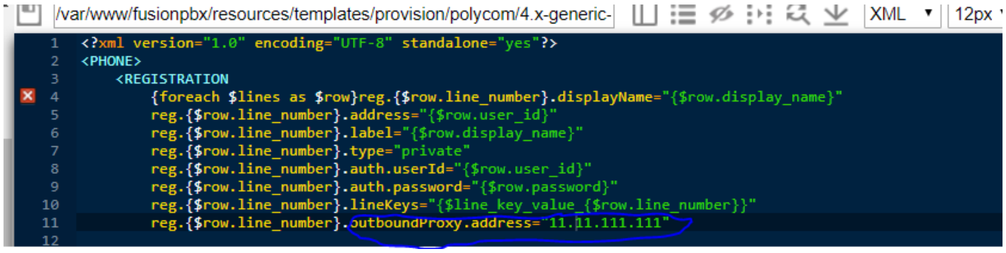
Outbound proxy 11.11.111.111

6.1.8 Provisioning and Registering a Polycom VVX Phone

Now that domains have been synced in dSIPRouter you are able to register a endpoint/hardphone. In this example we are using a Polycom VVX410 desk phone.

1. Log into your FusionPBX box

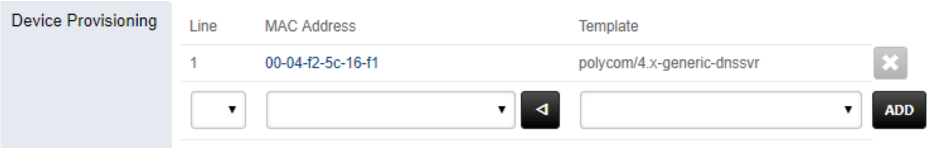
   a) Update the "outboundProxy.address" of the template with the IP address or hostname of the dSIPRouter in the provisioning editor.



/var/www/fusionpbx/resources/templates/provision/polycom/4.x-generic-

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<PHONE>
    <REGISTRATION
        {foreach $lines as $row}reg.{$row.line_number}.displayName="{$row.display_name}"
        reg.{$row.line_number}.address="{$row.user_id}"
        reg.{$row.line_number}.label="{$row.display_name}"
        reg.{$row.line_number}.type="private"
        reg.{$row.line_number}.auth.userId="{$row.user_id}"
        reg.{$row.line_number}.auth.password="{$row.password}"
        reg.{$row.line_number}.lineKeys="{$line_key_value_{$row.line_number}}"
        reg.{$row.line_number}.outboundProxy.address="11.11.111.111"
```

2. Assign the phone to a template.



Device Provisioning

Line	MAC Address	Template
1	00-04-f2-5c-16-f1	polycom/4.x-generic-dnssvr

3. Configuring the Provisioning Server section of the phone. Enter the appropriate information into the fields.

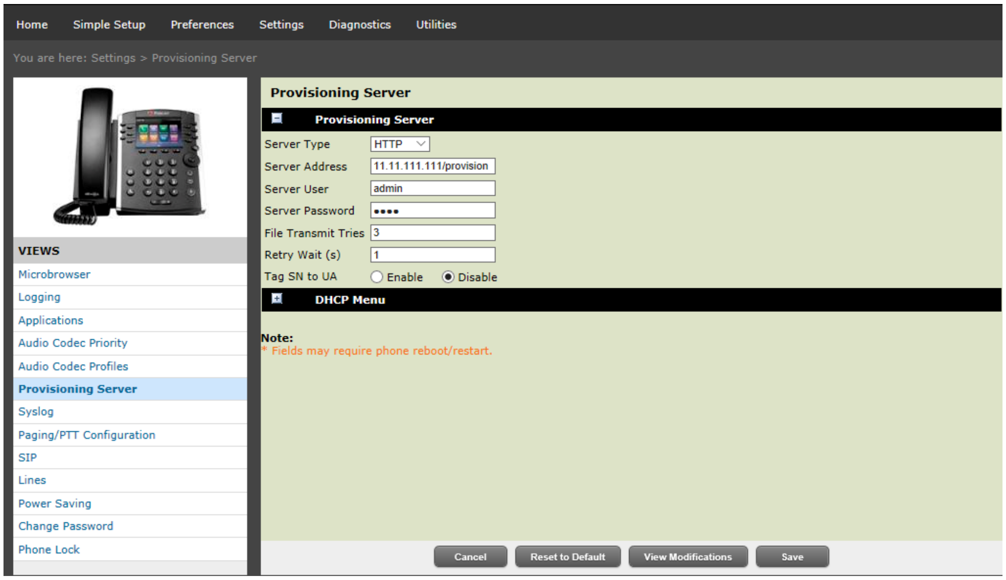
a) Server Type (dSIPRouter uses HTTP by default)

b) Server Address

c) Server Username (device provisioning server name)

d) Server Password

4. Click Save



Provisioning Server

Provisioning Server

Server Type: HTTP

Server Address: 11.11.111.111/provision

Server User: admin

Server Password: ••••

File Transmit Tries: 3

Retry Wait (s): 1

Tag SN to UA: ○ Enable ● Disable

DHCP Menu

Note:
* Fields may require phone reboot/restart.

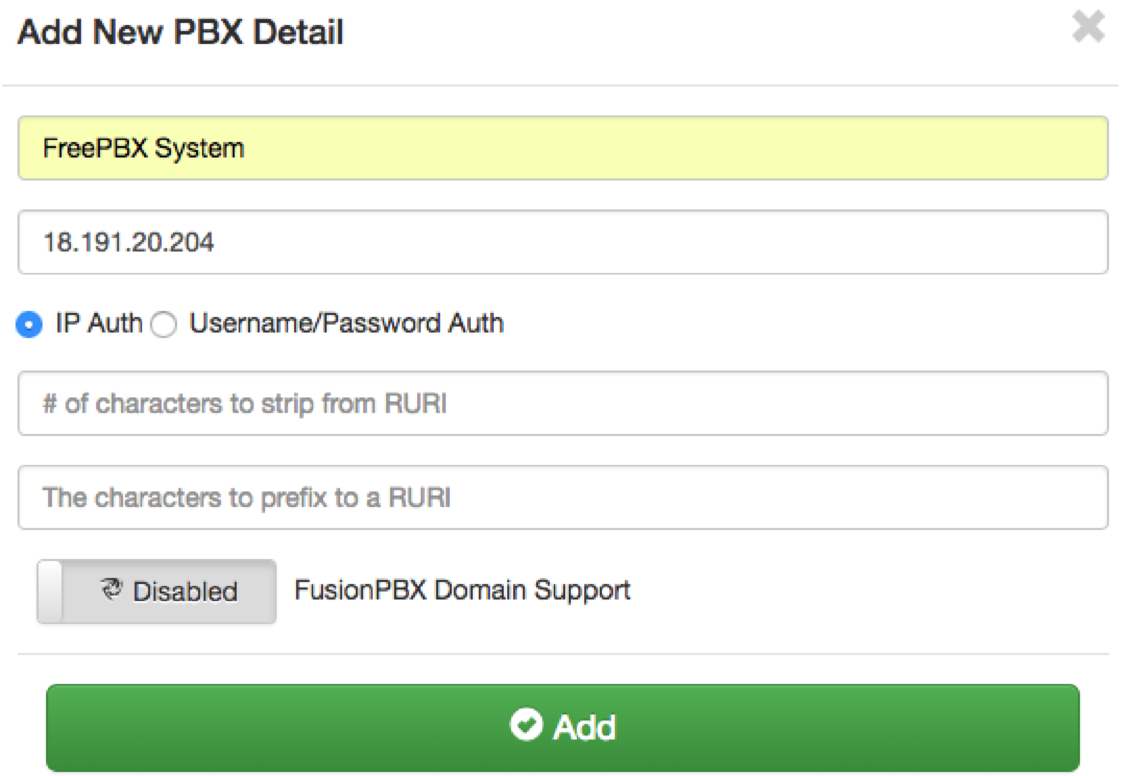
5. Reboot the phone

6.1.9 FreePBX Hosting - Pass Thru Authentication

Here we will demostrate how to setup dSIPRouter to enable hosting FreePBX using Pass Thru Authentication. FreePBX is designed to be a single tenant system or in other words, it was built to handle one SIP Domain. So, we use dSIPRouter to define a SIP Domain and we pass thru Registration info to the FreePBX server so that you don't have to change how authentication is done. However, this will only work for one FreePBX server. If you have a cluster of FreePBX servers then use "Local Subscriber Table" authentication. The value of having dSIPRouter in front of FreePBX is to provide you with flexibility. After setting this up you will have the ability upgrade or migrate users from one FreePBX instance to another without having to take an outage. The following video shows how to configure this. The steps to implement this is below the video.

Steps to Implement

1. Click PBX and Endpoints
2. Click Add



Add New PBX Detail

FreePBX System

18.191.20.204

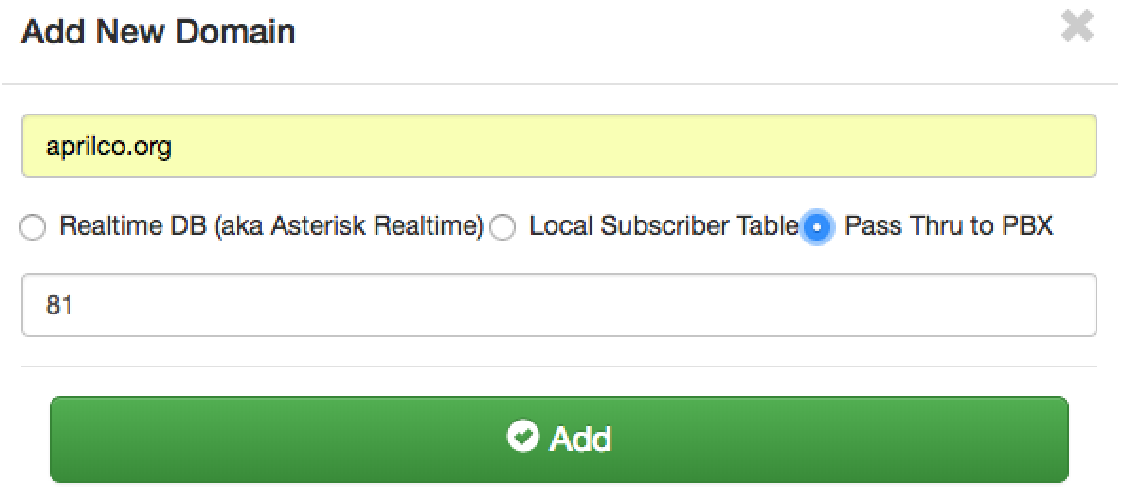
● IP Auth ○ Username/Password Auth

# of characters to strip from RURI

The characters to prefix to a RURI

Disabled FusionPBX Domain Support

3. Reload Kamailio
4. Click Domains
5. Click Add



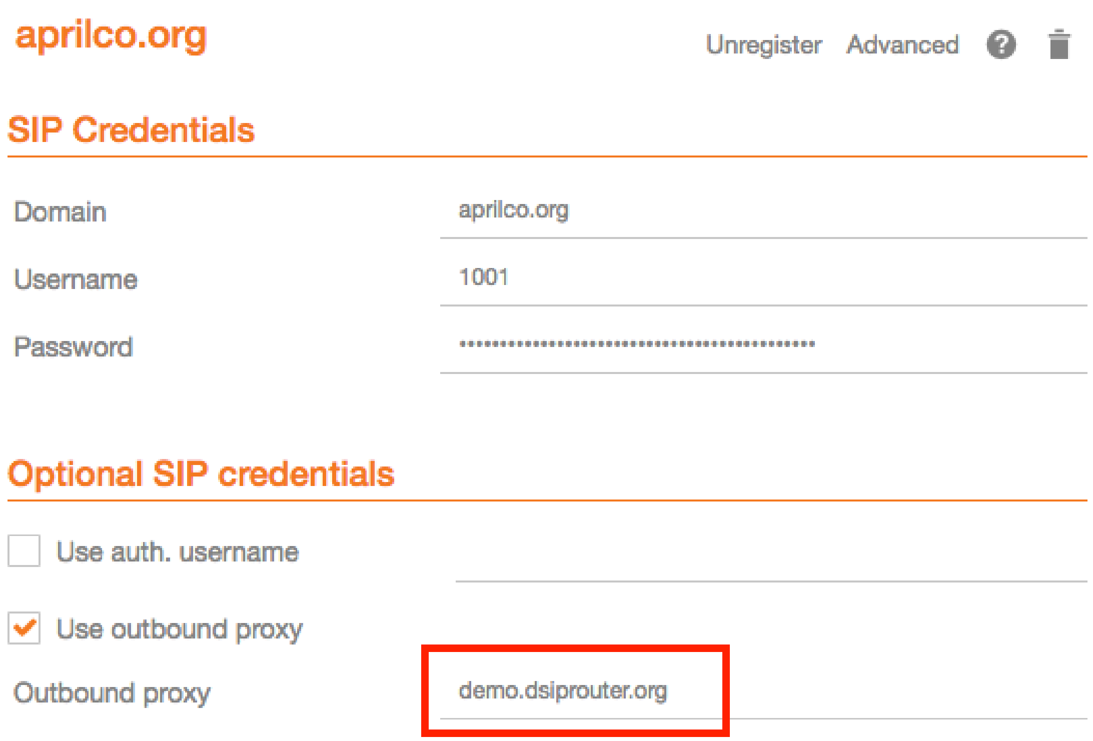
Add New Domain

aprilco.org

○ Realtime DB (aka Asterisk Realtime) ○ Local Subscriber Table ● Pass Thru to PBX

81

6. Reload Kamailio
7. Register a phone via dSIPRouter - notice that we used the hostname of dSIPRouter as the Outbound Proxy. This forces the registration thru the proxy.



aprilco.org

SIP Credentials

Domain	aprilco.org
Username	1001
Password	••

Optional SIP credentials

□ Use auth. username

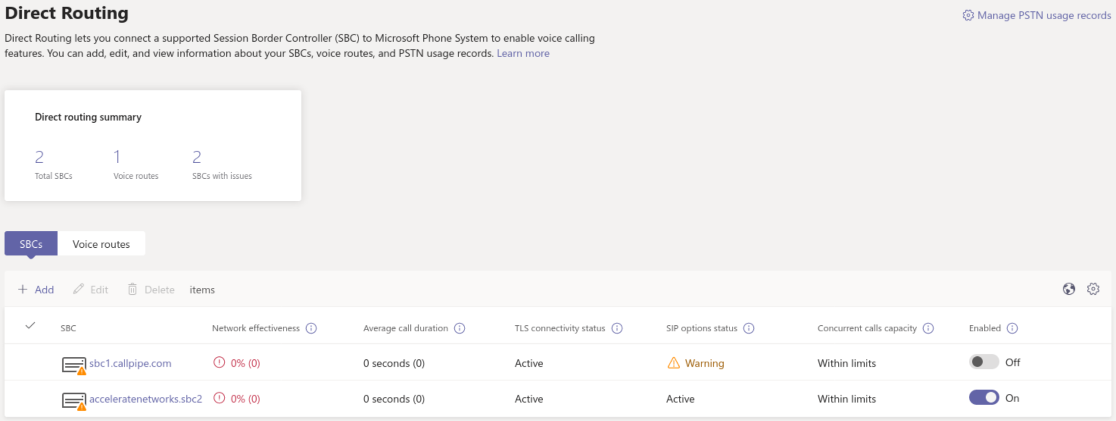
☑ Use outbound proxy

Outbound proxy	demo.dsiprouter.org

6.1.10 Microsoft Teams Direct Routing

dSIPRouter can act as an intermediary Session Border Controller between Microsoft Teams Direct Routing and your SIP provider or SIP servers.

An instance of dSIPRouter can either be a single tenant configuration (like sbc.example.com) or multi-tenant under a single wildcard subdomain (like *.sbc.example.com where * is the tenant's name).



Direct Routing

Direct Routing lets you connect a supported Session Border Controller (SBC) to Microsoft Phone System to enable voice calling features. You can add, edit, and view information about your SBCs, voice routes, and PSTN usage records. Learn more

Direct routing summary

2	1	2
Total SBCs	Voice routes	SBCs with issues

✓	SBC	Network effectiveness	Average call duration	TLS connectivity status	SIP options status	Concurrent calls capacity	Enabled
	sbc1.callpipe.com	0% (0)	0 seconds (0)	Active	⚠ Warning	Within limits	Off
	acceleratenetworks.sbc2	0% (0)	0 seconds (0)	Active	Active	Within limits	On

Steps to Implement

1. Buy a license and follow the license installation instructions that are emailed to you.

2. Add any carriers you need for inbound and outbound routing, define appropriate routes.

3. Authorize your SBC's domain with Microsoft 365 by adding a TXT record starting with ms= per Microsoft's documentation. Note: For multi-tenant use, authorizing the root subdomain or domain (if you use *.sbc.example.com, you would authorize sbc.example.com) should avoid the need to authorize each subdomain below this (like clientname.example.com)

4. Create a global admin user with proper Teams licensing associated with the domain (or for multi-tenant both the root subdomain (eg: sbc.example.com) and client's domain (eg: client.sbc.example.com))

5. Add the Teams session border controller in Teams Admin Center. Ensure the SIP port is correct (usually 5061) and the SBC is enabled!

6. Install PowerShell type pwsh then:

Install-Module -Name MicrosoftTeams Import-Module MicrosoftTeams $userCredential = Get-Credential Connect-MicrosoftTeams -Credential $userCredential

code

Login Note: If your using multi-factor authentication (MFA/2FA), log in by typing Connect-MicrosoftTeams Debian 10 Note: If you run into this OpenSSL issue , here is a workaround! **Replace sbc.example.com, user@example.com and +13137175555** with your SBC's FQDN, the user's email address and their phone number (with + then country code, use +1 if you are in the North American Numbering Plan)

Set-CsOnlinePstnUsage -Identity Global -Usage @{Add="US and Canada"} Set-CsOnlineVoiceRoute -Identity "LocalRoute" -NumberPattern ".*" -OnlinePstnGatewayList sbc.example.com New-CsOnlineVoiceRoutingPolicy "US Only" -OnlinePstnUsages "US and Canada"

# This is suppose to stop MSTeams from using the Microsoft Dialing Plan and using the routing policies that was defined above Set-CsTenantHybridConfiguration -UseOnPremDialPlan $False

# Apply and the US Only Voice Routing Policy to the user Grant-CsOnlineVoiceRoutingPolicy -Identity "user@example.com" -PolicyName "US Only"

# If it doesn't return a value of US Only, then wait 15 minutes and try it again. It sometime takes a while for the policy to be ready. Get-CsOnlineUser "user@example.com" | select OnlineVoiceRoutingPolicy

# Define a outgoing phone number (aka DID) and set Enterprise Voice and Voicemail Set-CsUser -Identity "user@example.com" -OnPremLineURI tel:+13137175555 -EnterpriseVoiceEnabled $true -HostedVoiceMail $true

code

Note: Log out by typing Disconnect-MicrosoftTeams

Credits to Mack at dSIPRouter for the SkypeForBusiness script and this blog post for helping me update these commands for the new MicrosoftTeams PowerShell module.

Add a single Teams User

If you have an existing dSIPRouter SBC configured in Teams and have added a DID as an inbound route already, then run the commands below in PowerShell to add an additional user.

Replace user@example.com and +13137175555 with your SBC's FQDN, the user's email address and their phone number (with + then country code, use +1 if you are in the North American Numbering Plan)

# Get Credentials, if using MFA/2FA just run Connect-MicrosoftTeams $userCredential = Get-Credential Connect-MicrosoftTeams -Credential $userCredential

# Apply and the US Only Voice Routing Policy to the user Grant-CsOnlineVoiceRoutingPolicy -Identity “user@example.com“ -PolicyName “US Only”

# Define a outgoing phone number (aka DID) and set Enterprise Voice and Voicemail Set-CsUser -Identity “user@example.com” -OnPremLineURI tel:+13137175555 -EnterpriseVoiceEnabled $true -HostedVoiceMail $true

code

Note: Log out by typing Disconnect-MicrosoftTeams

CHAPTER 7

Resources

7.1 Resources

CSV Example

7.1.1 Proxy FusionPBX UI

Add the following stanza before “location /images/” stanza to proxy the FusionPBX UI thru dSIPRouter. Once the following text is added to /opt/dsiprouter/gui/modules/fusionpbx/dsiprouter.nginx.tpl you will be able to access the FusionPBX GUI via: https://dSIPRouter_IP/ or https://dSIPRouter_IP:

```
location / {

        proxy_pass https://fusionpbx;
        proxy_redirect off;
        proxy_next_upstream error timeout http_404 http_403 http_500 http_
↪502 http_503 http_504 non_idempotent;

}
```

CHAPTER 8

Supported Configurations

8.1 Supported Configurations

8.1.1 Pass Thru to PBX Authentication Supported Configurations

PBX Distribution	PBX Version	Driver Type	Registration Test	Ext to Ext Test	Notes
FreePBX	Asterisk 13.22.0	chan_sip	Pass	Pass	see *Enabling the Path Header for Asterisk chan_sip*
FreePBX	Asterisk 13.22.0	chan_pjsip	Pass	Not Tested	suppport_path needs to be enabled
FusionPBX	FreeSWITCH 1.6	Sofia	Pass	Pass	

8.1.2 Enabling the Path Header for Asterisk chan_sip

1. Login into the FreePBX Admin GUI
2. Click Settings -> Asterisk SIP Settings
3. Click Chan SIP Settings
4. Find the "Other SIP Settings" field
5. Add the following field and click "Add Field"

   supportpath = yes
6. Click Submit
7. Click the red "Apply" settings button at the very top of the page

CHAPTER 9

Troubleshooting

9.1 Troubleshooting

Here you can troubleshoot logs for dSIPRouter, Kamailio and rtpengine:

All of our services are using syslog. For more information on syslog click here.

Default log facilities:

Log Facility	Service
local0	kamailio
local1	rtpengine
local2	dsiprouter

9.1.1 Kamailio Logging

1. How to turn logging on

Edit /etc/rsyslog.d/kamailio.conf and ensure the line beginning with local0 is not commented out:

```
vi /etc/rsyslog.d/kamailio.conf
```

Then restart syslog:

```
systemctl restart rsyslog
```

2. How to turn logging off

Edit /etc/rsyslog.d/kamailio.conf and ensure the line beginning with local0 is commented out:

```
vi /etc/rsyslog.d/kamailio.conf

```

(continues on next page)

(continued from previous page)

```

Then restart syslog:
```

```
systemctl restart rsyslog
```

3. Location of the log files

The default location is found here: /var/log/kamailio.log

4. How to configure it

Edit /etc/kamailio/kamailio.conf and change the variable ‘debug’ to the syslog logging verbosity of your choice.

```
vi /etc/kamailio/kamailio.conf
```

5. For more information see the documentation below:

https://www.kamailio.org/wiki/tutorials/3.2.x/syslog

9.1.2 rtpengine Logging

1. How to turn logging on

Edit /etc/rsyslog.d/rtpengine.conf and ensure the line beginning with local1 is not commented out:

```
vi /etc/rsyslog.d/rtpengine.conf
```

Then restart syslog:

```
systemctl restart rsyslog
```

2. How to turn logging off

Edit /etc/rsyslog.d/rtpengine.conf and ensure the line beginning with local1 is commented out:

```
 vi/etc/rsyslog.d/rtpengine.conf


Then restart syslog:
```

```
systemctl restart rsyslog
```

3. Location of the log files

The default location is found here: /var/log/rtpengine.log

4. How to configure it

Edit /etc/rtpengine/rtpengine.conf and change the variable ‘debug’ to the syslog logging verbosity of your choice.

```
vi /etc/rtpengine/rtpengine.conf
```

5. For more information see the documentation below:

https://github.com/sipwise/rtpengine

9.1.3 dSIPRouter Logging

1. How to turn logging on

Edit /etc/rsyslog.d/dsiprouter.conf and ensure the line beginning with local2 is not commented out:

```
vi /etc/rsyslog.d/dsiprouter.conf
```

Then restart syslog:

```
systemctl restart rsyslog
```

2. How to turn logging off

Edit /etc/rsyslog.d/dsiprouter.conf and ensure the line beginning with local2 is commented out:

```
vi /etc/rsyslog.d/dsiprouter.conf
```

Then restart syslog:

```
systemctl restart rsyslog
```

3. Location of the log files

The default location is found here: /var/log/dsiprouter.log

4. How to configure it Edit /etc/dsiprouter/gui/settings.py and change the variable 'DSIP_LOG_LEVEL' to the syslog logging verbosity of your choice.

```
vi /etc/dsiprouter/gui/settings.py
```

5. For more infornation see the documentation below:

https://success.trendmicro.com/solution/TP000086250-What-are-Syslog-Facilities-and-Levels

CHAPTER 10

Upgrade

10.1 Upgrading dSIPRouter

10.1.1 Upgrade 0.522 to 0.523

Upgrade dSIPRouter

In this section we will show you how to upgrade from 0.522 to 0.523.

Before starting the upgrade process you will need to backup your kamailio database using the following command:

```
cd /opt/
mysqldump kamailio > kamailio-bk.sql
```

After you've backed up your database you can now uninstall dsiprouter v0.50 by running the following commands:

```
cd /opt/dsiprouter
./dsiprouter.sh uninstall
```

Once the uninstall is complete you will need to either move or delete the /dsiprouter directory using the following command.

```
mv /dsiprouter /usr/local/src (moving directory)
```

Alternatively:

```
rm -r /dsiprouter (removing directory)
```

Installing dsiprouter v0.523

```
cd /opt/
apt-get update
apt-get install -y git curl
cd /opt
git clone -b v0.523 https://github.com/dOpensource/dsiprouter.git
cd dsiprouter
./dsiprouter.sh install
```

Note: please take note of the credentials given after the script has completed.

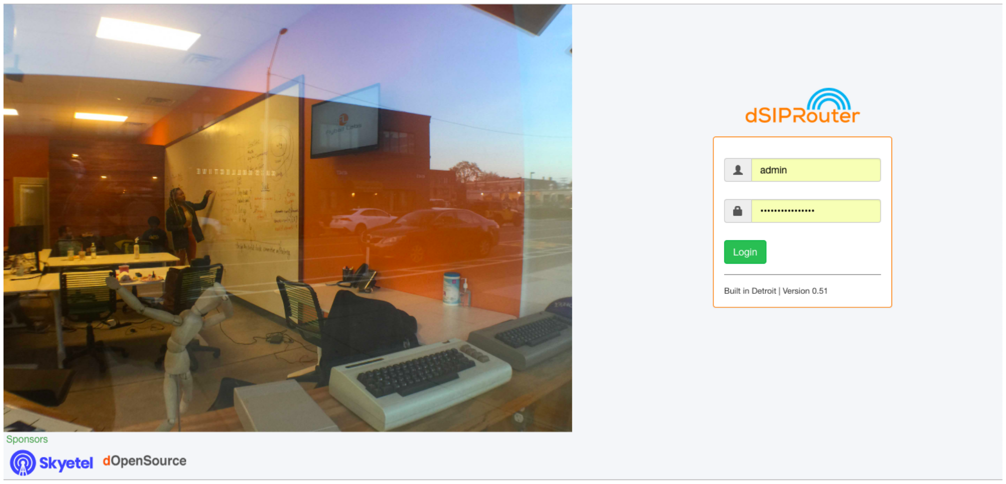
After the install is completed you can now restore your kamailio database using the following command:

```
cd /opt/
mysql  kamailio < kamailio-bk.sql
mysql kamailio -e "alter table dsip_multidomain_mapping add column domain_list_hash varchar(255) after domain_list;"
```

Now please restart dsiprouter using the following commands:

```
cd /opt/disprouter/
./dsiprouter.sh restart
```

After the install is complete and the dsiprouter service has been restarted, the login screen should now reflect v0.51 and you should be able to login with the dsiprouter credentials provided after the install completed.



admin

Built in Detroit | Version 0.51

10.1.2 Upgrade 0.50 to 0.51

Upgrade dSIPRouter

In this section we will show you how to upgrade from 0.50 to 0.51.

Before starting the upgrade process you will need to backup your kamailio database using the following command:

```
cd /opt/
mysqldump kamailio > kamailio-bk.sql
```

After you've backed up your database you can now uninstall dsiprouter v0.50 by running the following commands:

```
cd /opt/dsiprouter
./dsiprouter.sh uninstall
```

Once the uninstall is complete you will need to either move or delete the /dsiprouter directory using the following command.

```
mv /dsiprouter /usr/local/src (moving directory)
```

Alternatively:

```
rm -r /dsiprouter (removing directory)
```

Installing dsiprouter v0.51

```
cd /opt/
apt-get update
apt-get install -y git curl
cd /opt
git clone -b v0.51 https://github.com/dOpensource/dsiprouter.git
cd dsiprouter
./dsiprouter.sh install
```

Note: please take note of the credentials given after the script has completed.

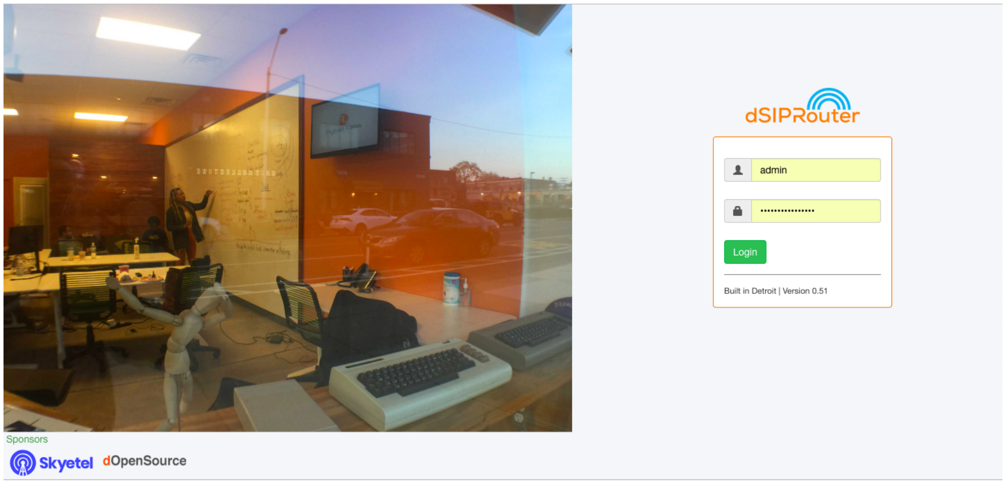
After the install is completed you can now restore your kamailio database using the following command:

```
cd /opt/
mysql  kamailio < kamailio-bk.sql
```

After the kamailio database is restored you need to restart dsiprouter using the following commands:

```
cd /opt/disprouter/
./dsiprouter.sh restart
```

After the install is complete and the dsiprouter service has been restarted, the login screen should now reflect v0.51 and you should be able to login with the dsiprouter credentials provided after the install completed.



admin

••••••••••••••••

10.1.3 Upgrade 0.621 to 0.63

Upgrade dSIPRouter

In this section we will show you how to upgrade from 0.621 to 0.63. This is the first release to contain our new upgrade approach.

The following steps will upgrade your Kamailio configuration from 0.621 to 0.63.

```
cd /opt/dsiprouter
git stash
git checkout v0.63
dsiprouter upgrade -rel 0.63
```

You should now be able to login to dSIPRouter and see that the new release has been applied.