

BULKVS

Description of this guide:

This FreePBX guide will show you how to configure a SIP TRUNK to BulkVS using IP Based Authentication

Key:

For this guide we used Freepbx version 14.0.4.5

Termination means outbound calling

Origination means inbound calling

Disclaimer:

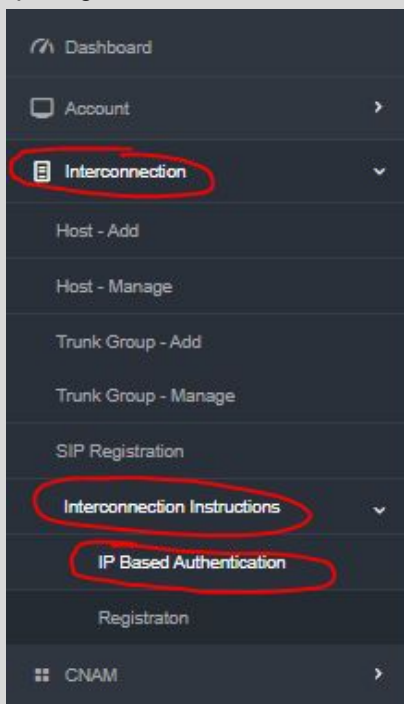
The below is a template to guide you. Please have your PBX vendor review all the settings in the event additional security or configuration is needed.

Prerequisites:

Login to portal.bulkvs.com to obtain your SIP TRUNK credentials

1) Login to bulkvs.com

2) Navigate to Interconnection -> Interconnection Instructions -> IP Based Authentication



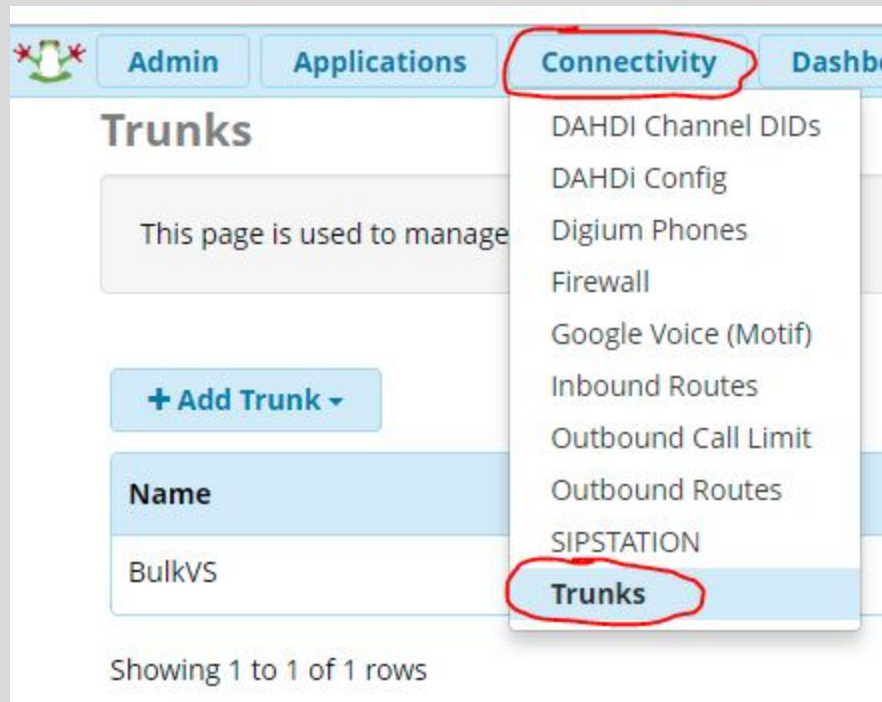
3) Make note of your **SIP Proxy** information:

SIP Proxy / SRV Record: sip.bulkvs.com
Supported Codecs: G.711u, G729a, T.38 (Pass Through where available)
DTMF Support: RFC2833 and Inband(G.711u)
SIP Port: 5060
IP Addresses:
199.255.157.198
162.249.171.198
69.12.88.198
76.8.29.198

Configuring Termination (outbound calling) and Origination (inbound calling) on FreePBX

1) Login to your FreePBX system

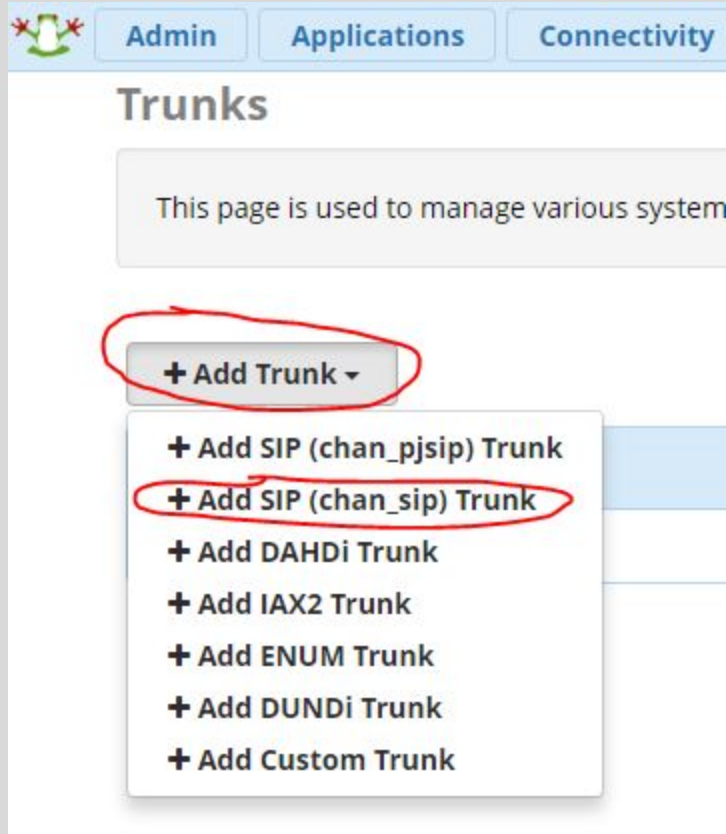
2) Click on Connectivity -> Trunks -> then + Add SIP (chan_sip) Trunk



The screenshot shows the FreePBX web interface. At the top, there are navigation tabs: Admin, Applications, Connectivity, and Dashboard. The 'Connectivity' tab is selected and circled in red. Below the tabs, the 'Trunks' page is displayed. The page title is 'Trunks' and it contains a description: 'This page is used to manage...'. There is a '+ Add Trunk' button. Below the button is a table with one row: 'BulkVS'. The 'Connectivity' dropdown menu is open, showing various options: DAHDI Channel DIDs, DAHDi Config, Digium Phones, Firewall, Google Voice (Motif), Inbound Routes, Outbound Call Limit, Outbound Routes, SIPSTATION, and Trunks. The 'Trunks' option is circled in red.

Name
BulkVS

Showing 1 to 1 of 1 rows



4) Create the BulkVS SIP TRUNK

A) Under **General** fill in your information as follows

Trunk Name: BulkVS-TRUNK

HideCallerID: No

Outbound CallerID: <your number>

**the number you would like to display when making outbound calls

CID Options: Allow AnyCID

Maximum Channels: 10 (change this to your desired limit)

Asterisk Trunk Dial Options: T and System

Continue if Busy: No

Disable Trunk: No

Monitor Trunk Failures: No

Edit Trunk

In use by 1 route

General

Dialed Number Manipulation Rules

sip Settings

Trunk Name	BulkVS-TRUNK
Hide CallerID	<input type="radio"/> Yes <input checked="" type="radio"/> No
Outbound CallerID	3109060901
CID Options	<input checked="" type="radio"/> Allow Any CID <input type="radio"/> Block Foreign CIDs <input type="radio"/> Remove CNAM <input type="radio"/> Force Trunk CID
Maximum Channels	10
Asterisk Trunk Dial Options	T <input type="radio"/> Override <input checked="" type="radio"/> System
Continue if Busy	<input type="radio"/> Yes <input checked="" type="radio"/> No
Disable Trunk	<input type="radio"/> Yes <input checked="" type="radio"/> No
Monitor Trunk Failures	<input type="radio"/> Yes <input checked="" type="radio"/> No

B) Under **Dialed Number Manipulation Rules**, leave this as default

Edit Trunk

In use by 1 route

General

Dialed Number Manipulation Rules

sip Settings

Dial Number Manipulation Rules

These rules can manipulate the dialed number before sending it out this trunk. If no rule applies, the number is not changed. The original dialed number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the **prefix** plus the **match pattern** boxes, the rule will be applied and all subsequent rules ignored.

Upon a match, the **prefix**, if defined, will be stripped. Next the **prepend** will be inserted in front of the **match pattern** and the resulting number will be sent to the trunk. All fields are optional.

Rules:

X matches any digit from 0-9

Z matches any digit from 1-9

N matches any digit from 2-9

[1237-9] matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)

. wildcard, matches one or more characters (not allowed before a | or +)

[Dial patterns wizards](#)

(prepend) prefix | match pattern

Outbound Dial Prefix

Submit Duplicate Reset Delete



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C) Under sip Settings and Outgoing

Trunk Name: BulkVS

PEER Details:

host=sip.bulkvs.com
disallow=all
allow=ulaw
canreinvite=no
dtmfmode=rfc2833
insecure=very
port=5060
qualify=yes
type=friend

The screenshot shows the Asterisk SIP Trunk configuration interface. At the top, there is a navigation bar with tabs for Admin, Applications, Connectivity, Dashboard, Reports, Settings, and UCP. Below this is the 'Edit Trunk' page for 'BulkVS', which is noted as being 'In use by 1 route'. The configuration is divided into three main sections: 'General', 'Dialed Number Manipulation Rules', and 'sip Settings'. Under 'sip Settings', there are two sub-sections: 'Outgoing' and 'Incoming'. The 'Outgoing' section is active, showing the 'Trunk Name' as 'BulkVS'. Below this, the 'PEER Details' section is expanded, displaying the following configuration parameters: host=sip.bulkvs.com, disallow=all, allow=ulaw, canreinvite=no, dtmfmode=rfc2833, insecure=very, port=5060, qualify=yes, and type=friend.

D) Under sip Settings and Incoming

User Context: <leave blank>

User Details: <leave blank>

Register String: <leave blank>

Admin Applications Connectivity Dashboard Reports Settings UCP

Edit Trunk

In use by 1 route

General Dialed Number Manipulation Rules sip Settings

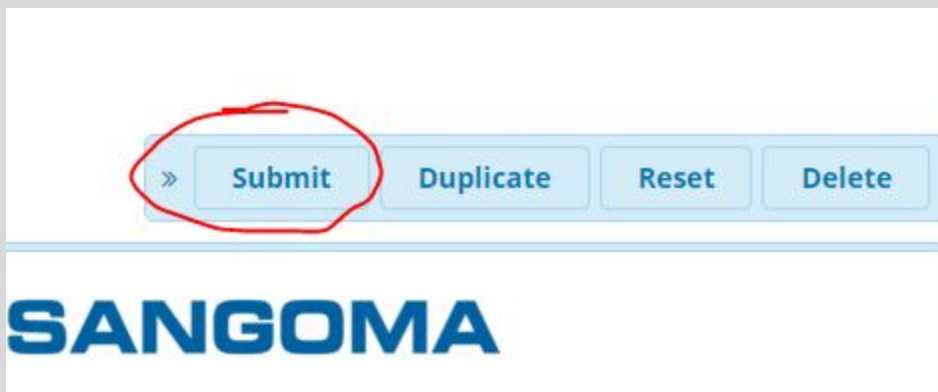
Outgoing Incoming

USER Context ?

USER Details ?

Register String ?

E) Now press **submit** (on the bottom right) to save the changes

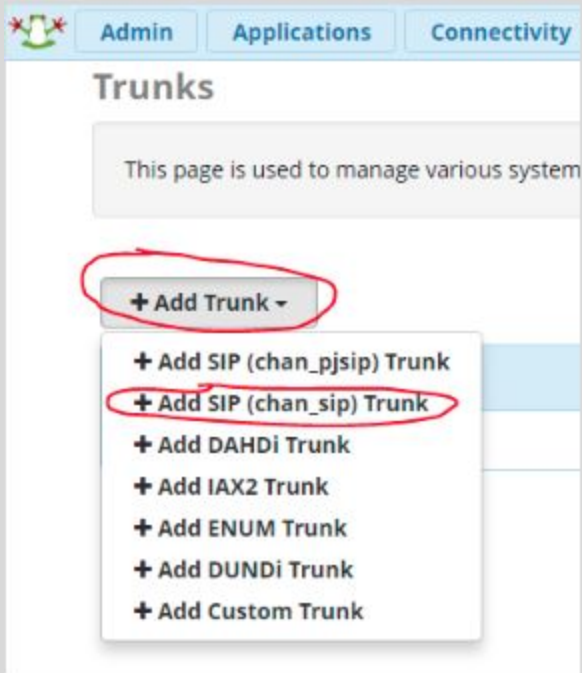


F) Configure BulkVS IPs in to your system.

We will repeat steps F to K 4 times, one of each Bulk Solutions SBC IP. In our example below we take care of 162.249.171.198. Repeat steps for the other IPs listed below as well

162.249.171.198
69.12.88.198
76.8.29.198
199.255.157.198

Click on Add Trunk Group again



G) Under General Fill out the information as follows
(update your outbound CallerID to your number)

Trunk Name: BulkVS-TRUNK-162.249.171.198

The screenshot shows the Asterisk web interface for configuring a trunk. The 'General' tab is selected, and the form is filled with the following values:

- Trunk Name: BulkVS-TRUNK-162.249.171.198
- Hide CallerID: No
- Outbound CallerID: 3109060901
- CID Options: Allow Any CID (selected), Block Foreign CIDs, Remove CNAM, Force Trunk CID
- Maximum Channels: 24
- Asterisk Trunk Dial Options: T (selected), Override, System
- Continue if Busy: No
- Disable Trunk: No
- Monitor Trunk Failures: No

At the bottom right, there are buttons for 'Submit', 'Duplicate', 'Reset', and 'Delete'.

H) Dialed Number Manipulation Rules will not be changed (use the default)

Edit Trunk

In use by 1 route +

General **Dialed Number Manipulation Rules** sip Settings

Dial Number Manipulation Rules

These rules can manipulate the dialed number before sending it out this trunk. If no rule applies, the number is not changed. The original dialed number is passed down from the route where some manipulation may have already occurred. This trunk has the option to further manipulate the number. If the number matches the combined values in the **prefix** plus the **match pattern** boxes, the rule will be applied and all subsequent rules ignored.

Upon a match, the **prefix**, if defined, will be stripped. Next the **prepend** will be inserted in front of the **match pattern** and the resulting number will be sent to the trunk. All fields are optional.

Rules:

- X** matches any digit from 0-9
- Z** matches any digit from 1-9
- N** matches any digit from 2-9
- [1237-9]** matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
- .** wildcard, matches one or more characters (not allowed before a | or +)

[Dial patterns wizards](#)

(prepend) prefix | [match pattern +

Outbound Dial Prefix ?

» **Submit** **Duplicate** **Reset** **Delete**

I) Under SIP Settings -> Outgoing use the following

Trunk Name: BulkVS-162.249.171.198

Peer Details:

host=sip.bulkvs.com

disallow=all

allow=ulaw,alaw

canreinvite=yes

dtmfmode=rfc2833

insecure=very

port=5060

qualify=yes

type=friend

nat=no

General Dialed Number Manipulation Rules sip Settings

Outgoing Incoming

Trunk Name ? BulkVS-162.249.171.198

PEER Details ?

```

host=sip.bulkvs.com
disallow=all
allow=ulaw,alaw
canreinvite=yes
dtmfmode=rfc2833
insecure=very
port=5060
qualify=yes
type=friend
nat=no

```

J) Under SIP Settings -> Incoming use the following

User Context: from-sip-external²

User Details:

host=**162.249.171.198**

type=friend

port=5060

insecure=port,invite

nat=no

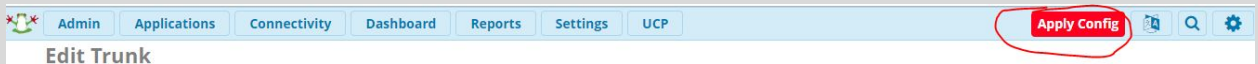
disallow=all

allow=ulaw,alaw

qualify=yes

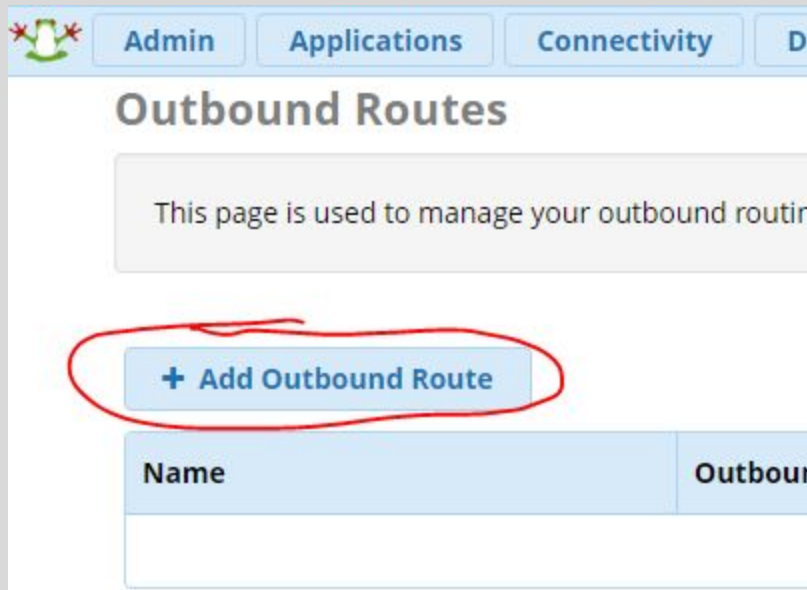
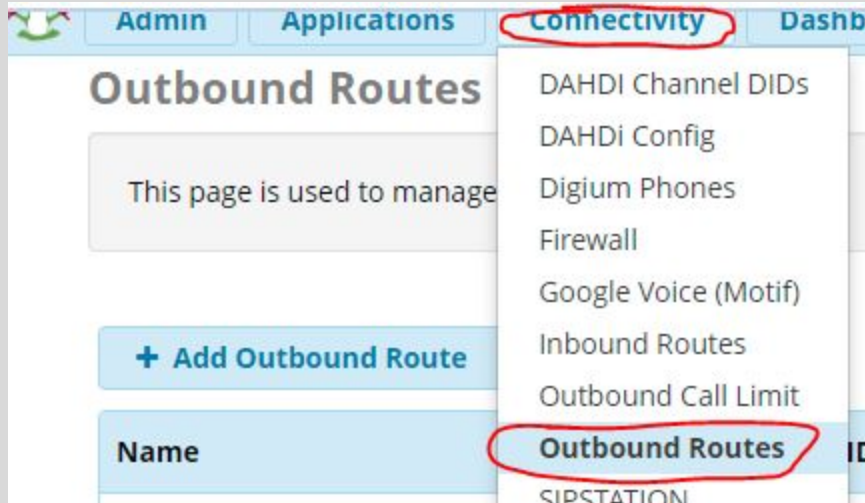
canreinvite=yes

K) Then press **Apply Config** on the top right to apply the changes



5) Create the BulkVS Outbound Routes

A) Click on **Connectivity, Outbound Routes** and **+ Add Outbound Route**



B) Under **Route Settings**

Route Name: BulkVS-Route

Route CID: <your number>

Override Extension: No

Route Password: <leave blank>

Route Type: <do not select any option>

Music On Hold: default

Time Match Time Zone: User System Time Zone

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

Route Position: -- No Change --

Trunk Sequence for Matched Routes: BulkVS-TRUNK

Optional Destination On Congestion: Normal Congestion

[Admin](#)
[Applications](#)
[Connectivity](#)
[Dashboard](#)
[Reports](#)
[Settings](#)
[UCP](#)

Outbound Routes

Edit Route: BulkVS-Term: BulkVS-Term

[Route Settings](#)
[Dial Patterns](#)
[Import/Export Patterns](#)
[Additional Settings](#)

Route Name [?](#) BulkVS-TERM

Route CID [?](#) 3109060901

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 [?](#)

- BulkVS-TRUNK
-

Optional Destination on Congestion [?](#) Normal Congestion

Note: Extension Routes is not registered

C) under **Dial Patterns** configure it as follow

1800NXXXXXX
 1833NXXXXXX
 1844NXXXXXX
 1855NXXXXXX
 1866NXXXXXX
 1877NXXXXXX
 1888NXXXXXX
 1NXXNXXXXXX

922
911

1 NXXNXXXXXX (10 digit calling)
1301 NXXXXXX (7 digit dialing)

* In this example my area code is 301 so NXXXXXX will add 1301 area code if I dial 7 digits
** In this example we expect to have your E911 service (911 and 922 dialing) via BulkVS.

!!! If not please remove the 911 and 922 routes !!!

Admin Applications Connectivity Dashboard Reports Settings UCP

Dial Patterns that will use this Route

Pattern Help +

Dial patterns wizards

(prepend)	prefix	[.]	/	CallerID	+ -
(prepend)	prefix	[1800NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1833NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1844NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1855NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1866NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1877NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1888NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[1NXXNXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[911]	/	CallerID	+ -
(prepend)	prefix	[922]	/	CallerID	+ -
(1)	prefix	[NXXNXXXXXX]	/	CallerID	+ -
(1301)	prefix	[NXXXXXX]	/	CallerID	+ -
(prepend)	prefix	[match pattern]	/	CallerID	+ -

» Submit Duplicate Reset Delete

D) under Import/Export Patterns

** No changes needed

Admin Applications Connectivity Dashboard Reports Settings UCP

Outbound Routes

Edit Route: BulkVS-TERM: BulkVS-TERM

Route Settings Dial Patterns Import/Export Patterns Additional Settings

Upload from CSV [?](#)

Export Dialplans as CSV [?](#)

E) under **Additional Settings**

** No changes needed

Admin Applications Connectivity Dashboard Reports Settings UCP

Outbound Routes

Edit Route: BulkVS-TERM: BulkVS-TERM

Route Settings Dial Patterns Import/Export Patterns Additional Settings

Note that the meaning of these options has changed. Please read the wiki for further information on these changes.

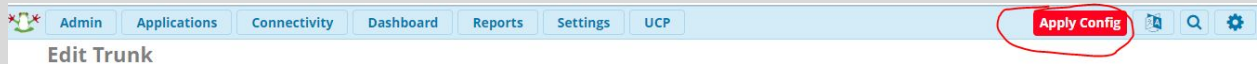
Call Recording [?](#)

PIN Set [?](#)

F) Now press **submit** (on the bottom right) to save the changes

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G) Then press **Apply Config** on the top right to apply the changes

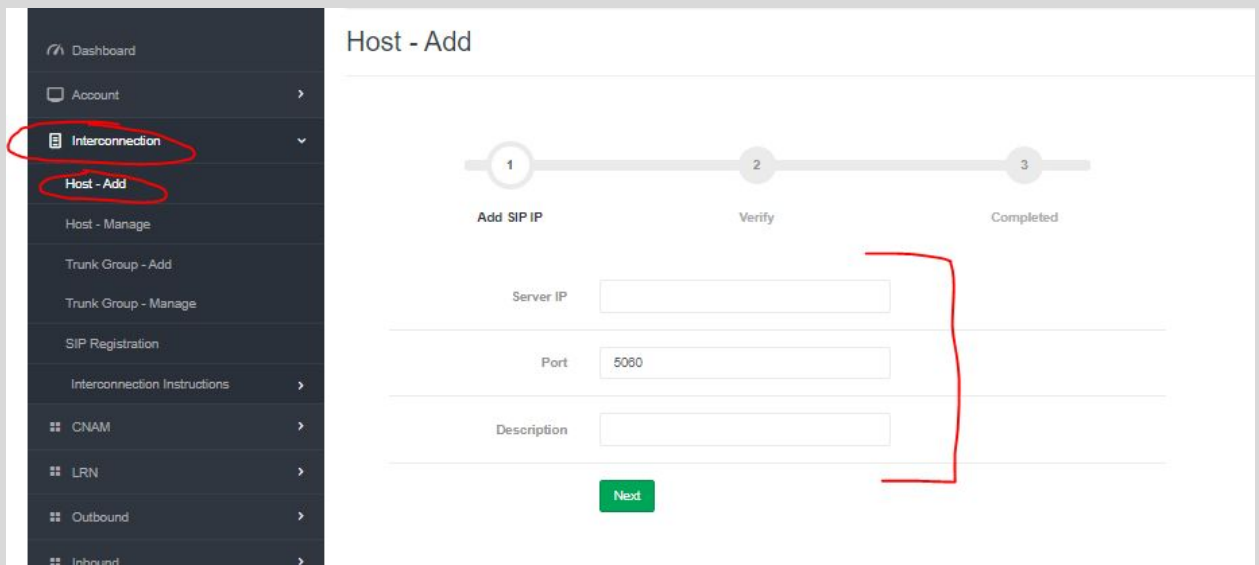


6) Add your SIP Server IP and Create a Trunk Group

A) Login to portal.bulkVS.com

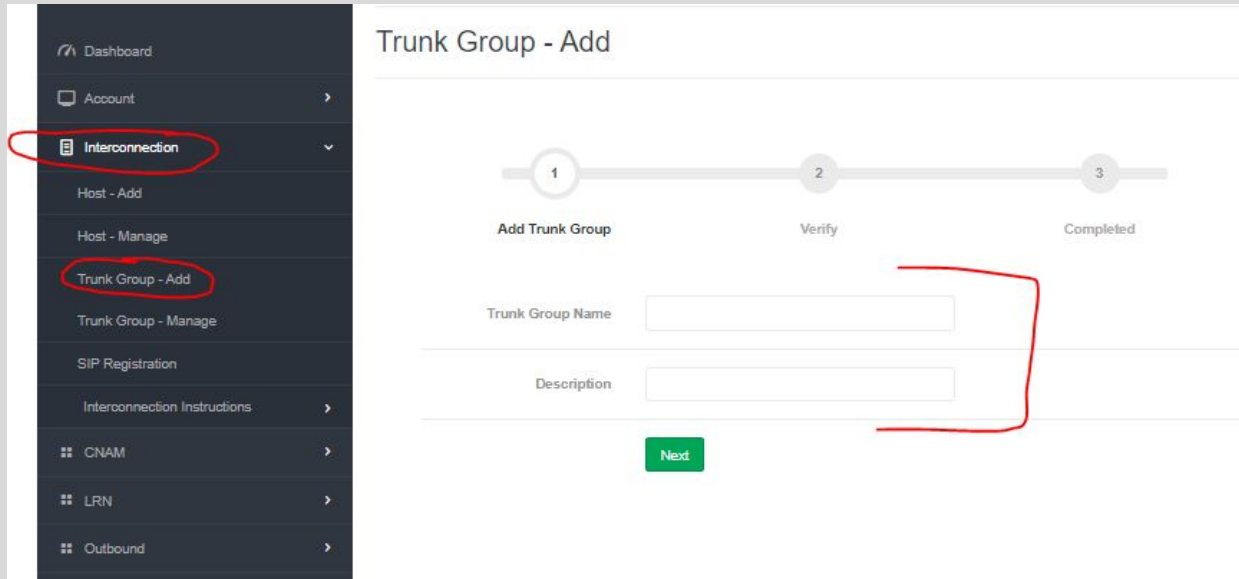
B) Interconnection -> Host - Add -> Enter your Public IP information for your SIP Server

C) Repeat steps A and B if you have multiple IPs



E) Create a Trunk Group

Click on Interconnection -> Trunk Group - Add



F) After the Trunk Group is created add a Primary Host IP and optionally a Secondary Host IP.
 Interconnection -> Trunk Group - Manage -> Select your trunk group

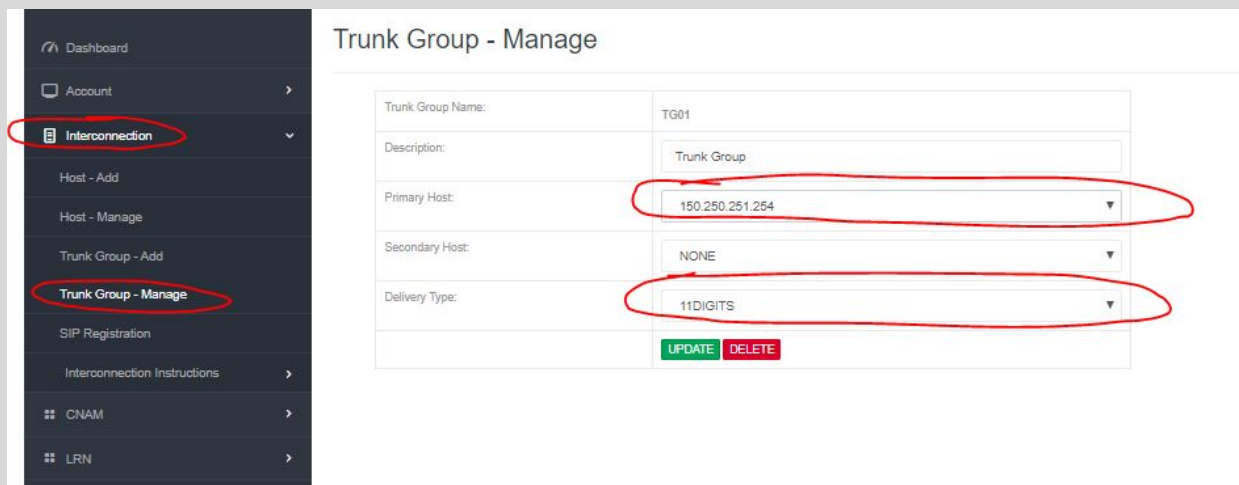
*Secondary Host is optional, it will be used if we can't reach you on the primary IP

** Delivery type is how you would like to receive the call our options are as follows

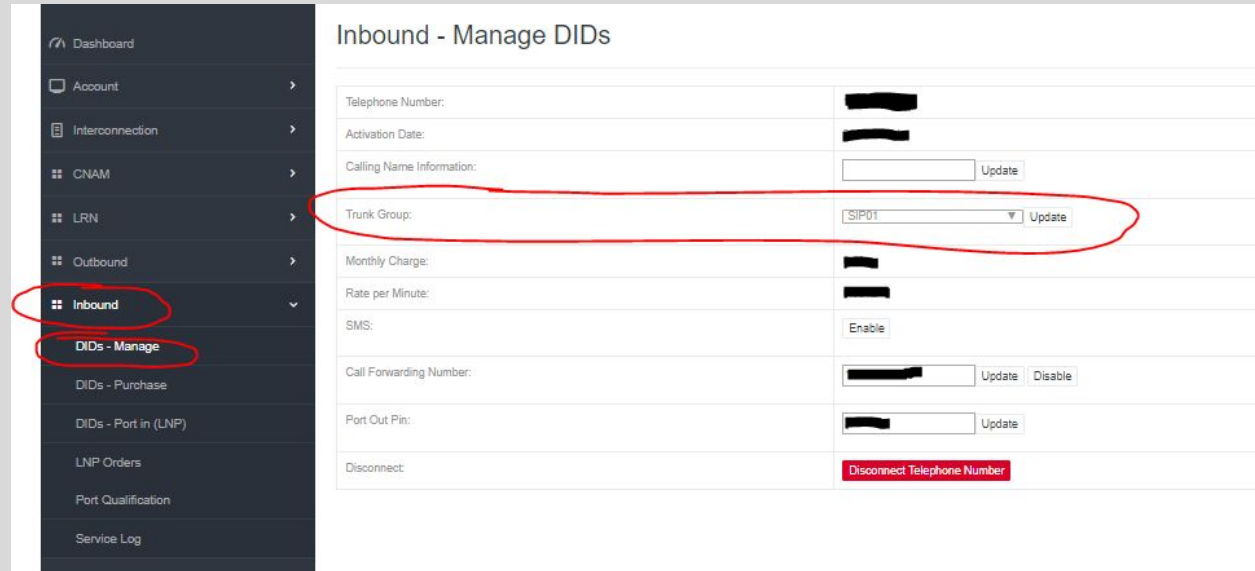
E164: +13109060901

10Digits: 3109060901

11Digits: 13109060901 (Default)



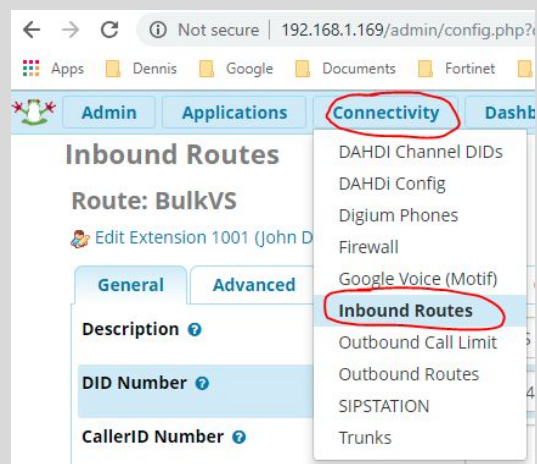
G) To receive inbound calls via a trunk group update your number
 Inbound -> DIDs - Manage -> Select your number ->




K) At this point inbound calls for this number will reach your FreePBX when dialed from the PSTN (the public switching network, for example, your cell phone)

L) Remember to create an inbound route on FreePBX. In this example I will route the number to our test extension 1001. If you call your BulkVS number it will reach ring x1001


Click on **Connectivity** -> **Inbound Routes**












 [Admin](#) [Applications](#) [Connectivity](#) [Dashboard](#) [Reports](#) [Settings](#) [UCP](#)

Inbound Routes

Route: BulkVS

 [Edit Extension 1001 \(John Doe\)](#)

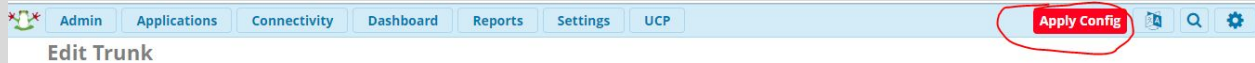
[General](#) [Advanced](#) [Privacy](#) [Fax](#) [Other](#)

Description 	<input type="text" value="BulkVS"/>
DID Number 	<input type="text" value="REDACTED"/>
CallerID Number 	<input type="text" value="ANY"/>
CID Priority Route 	<input type="radio"/> Yes <input checked="" type="radio"/> No
Alert Info 	<input type="text" value="None"/>
Ringer Volume Override 	<input type="text" value="None"/>
CID name prefix 	<input type="text"/>
Music On Hold 	<input type="text" value="Default"/>
Set Destination 	<input type="text" value="Extensions"/> <input type="text" value="1001 John Doe"/>

Now press **submit** (on the bottom right) to save the changes

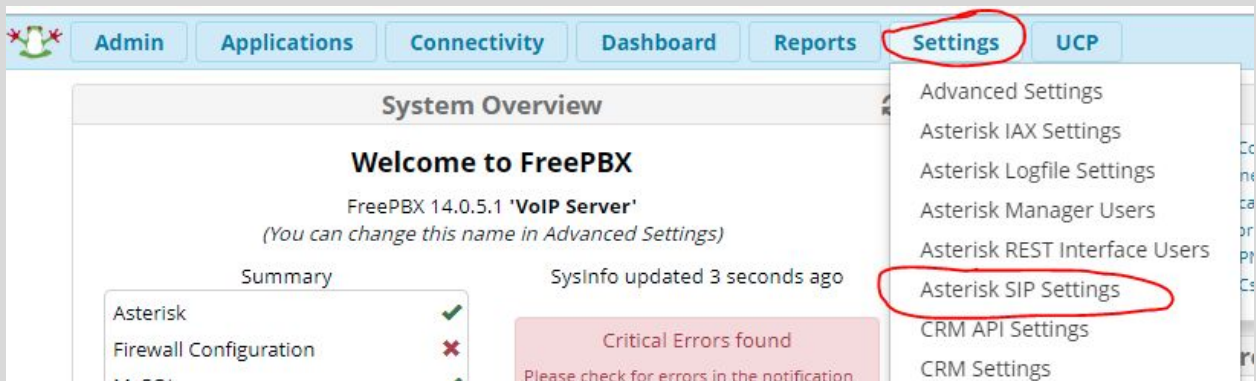
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Then press **Apply Config** on the top right to apply the changes

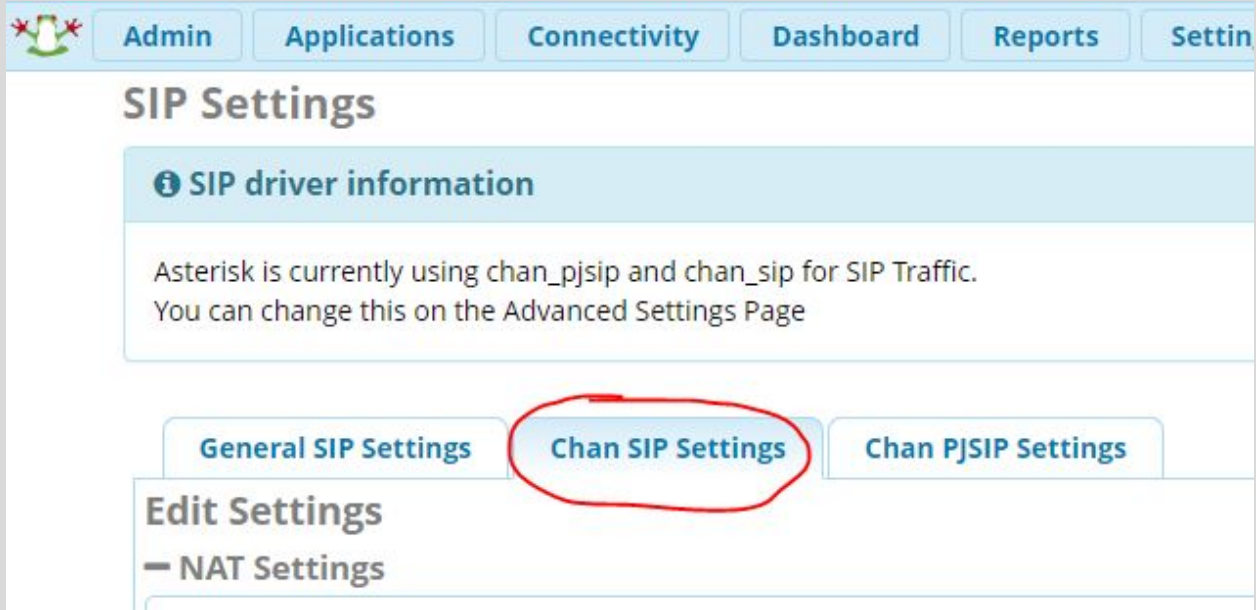


M)BulkVS only uses port 5060. You will need to update the default port from 5160 to 5060 for SIP.

Click on **Settings** and then **Asterisk SIP Settings**



Now click on **Chan SIP Settings**



Scroll down to **Bind Port** and **TLS Bind Port** and update them from **5160** to **5060**

Advanced General Settings




Default Context ?	from-sip-external
Bind Address ?	0.0.0.0
Bind Port ?	5060
TLS Bind Address ?	:::
TLS Bind Port ?	5060
Enable SRV Lookup ?	<input type="radio"/> Yes <input checked="" type="radio"/> No
Enable TCP ?	<input type="radio"/> Yes <input checked="" type="radio"/> No
Call Events ?	<input checked="" type="radio"/> Yes <input type="radio"/> No

Now press **submit** (on the bottom right) to save the changes

[»](#) **Submit** Duplicate Reset Delete

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Then press **Apply Config** on the top right to apply the changes

Admin Applications Connectivity Dashboard Reports Settings UCP **Apply Config**   

Edit Trunk

Conclusion:

That should complete your configuration of your FreePBX SIP TRUNK to Bulk Solutions.

If you have any questions you could always reach out to us at support@bulkvs.com