VIPole Enterprise

SIP Server Integration Guide

VIPole
www.vipole.com

SECURE AND ENCRYPTED MESSENGER
for secure communications and encrypted data storage

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1. VIPole server settings

- VIPole provides interface for integration with SIP servers to make only outgoing calls.
- VIPole is compatible with such SIP server as Asterisk and 3CX.
- VIPole and Asterisk should be installed on different servers.

For correct SIP server performance enabling calling to short numbers within local SIP server:

- You need to configure VIPole server autoconfiguration.
- It is desirable that the VIPole server and SIP server were on the same subnet.

1.1 Edit server configuration file /etc/vipole/server.config

Do not use FQDN if the parameters specify that you need an IP address.

```
# Enable sip
sip-enable=true

# Enable SIP management in VIPole Administrator of desktop client
sip-enable-admin=true

# IP address of SIP server. If blank that SIP module is disabled.
sip-server-address = 192.168.1.20 #  Asterisk IP address

# Realm asterisk
sip-domain = asterisk.local
# UDP port of SIP server
sip-server-port=5060

# Self IP address used for interactions with SIP server (use real address here, don't use 0.0.0.0)
sip-local-address = local IP address of VIPole server

# SIP phone domain name.

# This domain name is appended to phone number for making phone calls (e.g. +12345678@sip-to-phone.example.com - sip-to-phone.example.com is appended to phone number)
sip-phone-domain = vipole.company.com
```

1.2 Edit Relay configuration file /etc/vipople/relay.config

```
# Self IP address used for interactions with SIP server

sip-listen-address = local IP address of VIPole server
```

1.3 Edit /etc/vipole/client_autoconf.config

```
# SIP module is enabled

sip-enable=1

# Check balance in client before making SIP calls If set then calls are available when balance is
more than 0

sip-client-check-balance=0

# Allow auto initialization of SIP account using web module

sip-allow-auto-init=0

# Use strict phone number in E.123 international notation. The number must start from plus followed
by country code and number (+123456789)

# When disabled then phone number can be either in plain digital or in international format

sip-strict-phone-number=0
```

After that, rewrite autoconfigutation

(root))# vipoleadm --operation save_clients_autoconf --clients-autoconf-file \ client_autoconf.config --domain vipole.domain --dbname vipole_server

And reload VIPole server

(root))# /etc/init.d/vipoleserver restart

2. SIP settings in Administrator dashboard

Log in to VIPole Administrator dashboard:

- Open user profile and enter SIP login and password of the user.
- Enter SIP login without SIP domain.

3. Asterisk SIP server settings

3.1. Install Asterisk on another computer or server.

3.2. Edit sip.conf

```
[general]
pedantic=no
localnet=192.168.1.0/255.255.255.0 ; your subnet
autodomain=no
domain= 192.168.1.20 ; (Asterisk local IP)

[authentication]
dtmfmode=rfc2833
type=friend
directmedia=no
host=dynamic
directmedia=yes
disallow=all
allow=ilbc
allow=g729
allow=gsm
allow=g723
allow=ulaw
disallow=all
allow=ulaw

; Enter SIP provider peer to make outgoing calls
; SIP provider connection settings you can get on official website of the provider.
[line1]
type=peer
secret=PASSWORD****
username=LOGIN
host=SIP_DOMAIN
fromuser=LOGIN
fromdomain=SIP_DOMAIN
insecure=invite
nat=force_rport
dtmfmode=inband
context=line1
directmedia=no
```

```
[defaults](!)
canreinvite=no
directmedia=no
disallow=all
allow=alaw
progressinband=yes
; allow=g729
; allow=g723

[peer](!,defaults)
type=friend
insecure=invite,port
context=local_user
;fromdomain=<domain name or IP>
host=dynamic
qualify=yes
rtpkeepalive=300
;dtmfmode=inband

[user-sip1](peer) ; your user login (sip-login)
nat=no
username=user-vp1 ; (sip-login)
secret=pass1 ; (sip-password)
context=local_user
callerid=101

; another user
[user-sip2](peer); your user login (sip-login)
nat=no
username=user-sip2 ; (sip-login)
secret=pass2 ; (sip-password)
context=local_user
callerid=102
```

3.3. Enter your dial plan for outgoing calls in extensions.conf

```
[general]

[globals]

[default]

[local_user]

exten => _8XXXXXXXXXX,1,Dial(SIP/line1/${EXTEN})

exten => _7XXXXXXXXXX,1,Dial(SIP/line1/${EXTEN})

exten => _+8XXXXXXXXXX,1,Dial(SIP/line1/${EXTEN:1})

exten => _+7XXXXXXXXXX,1,Dial(SIP/line1/${EXTEN:1})
```

4. SIP debugging

For debugging, you can use the parameters of the VIPole server by adding them at the end of the **server.config** configuration /

```
debug-modules=sip

debug-level = 8
```

For the Asterisk use command

```
(root)# asterisk -vvvvvcdr
```