

Setup Instructions

Overview

Please note that using VICIdial on AWS requires some basic Amazon Web Services, Linux, Asterisk, Networking, and SIP knowledge. The learning curve can be steep but well worth the effort.

Setup

1. Launch a new EC2 instance and choose your instance type. An instance with ephemeral storage is recommended. See the tips section on **Swap and Recording**.
2. Provision at least 20 GB of storage. The base AMI is 10 GB for advanced users.
3. It is a good idea to assign an elastic IP to the instance.
4. SSH into the server with **ec2-user** and your key.
5. Switch to root by typing '**sudo -i**'.
6. Complete the config.
7. To begin a single server setup run '**express-setup**'. Advanced users with specific needs should run '**standard-setup**'.
8. Follow the setup directions and reboot the server.
9. After the setup, you may need to open/close additional ports in the instance's security group depending on your configuration.
10. An SSL certificate is recommended and can be installed by pointing a domain to your IP and running '**vicibox-ssl**' as root.
11. Visit <https://your-instance-ip-or-domain>, click Administration and enter '**6666**' for the username and your **instance ID** for the password.
12. Enter a new password and proceed to users.
13. Click the default user and add all necessary permissions under the “**ADMIN INTERFACE OPTIONS**” section.

14. Do **NOT** change the “**Modify Same User Level**” section until you have created a new user with the appropriate permissions.
15. After setting all of the admin interface options you need to 1, you’ll be free to configure VICIdial.

VICIpnone Configuration

VICIpnone is a browser based WebRTC phone. While you're free to use any phone you like for the agent leg, having the phone right in the agent session without extensions or additional software is extremely convenient. You do not have to use VICIpnone .

VICIdial can be used with any software or hardware that supports SIP. You can even route the agent leg through your carrier to a cellphone if you're inclined to do so.

https://viciphone.com/?page_id=353

Example Carrier Configurations

While you can use any SIP provider(s) you like, these are some popular ones in our experience. You can combine them as needed in a single dialplan or setup multiple carriers to use as backups. Please spend some time familiarizing yourself with how Asterisk Dialplans work.

- [Flowroute](#)

Carrier ID:	Flowroute
Carrier Name:	Flowroute ?
Carrier Description:	?
Admin User Group:	--ALL-- ?
Registration String:	?
Template ID:	--NONE-- ?
Account Entry:	<pre>[Flowroute] disallow=all allow=ulaw type=friend host=flowroute-endpoint dtmfmode=rfc2833 context=trunkinbound</pre> ?
Protocol:	SIP ?
Globals String:	TECHPREFIX=your-prefix* ?
Dialplan Entry:	<pre>exten => _91NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log) exten => _91NXXNXXXXXX,2,Dial(SIP/\${TECHPREFIX}1\${EXTEN:2}@Flowroute,,To) exten => _91NXXNXXXXXX,3,Hangup</pre> ?
Server IP:	172.31.19.53 (0.0.0.0 is all servers) ?
Active:	☒ ?
SUBMIT	

The TECHPREFIX does not need to be a global string. This is just an example. You can include it in the dialplan. Don't forget to include the asterisk. For example.

```
Carrier ID: Flowroute
Carrier Name: Flowroute

Account Entry:
[Flowroute]
disallow=all
allow=ulaw
type=friend
host=us-east-va.sip.flowroute.com
dtmfmode=rfc2833
context=trunkinbound

Dialplan Entry:
exten => _94NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)
exten => _94NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)
```

```
exten => _94NXXNXXXXXX, 2, Set(CALLERID(num)=+1${CALLERID(num)})
exten => _94XNXXNXXXXXX, 2, Set(CALLERID(num)=+1${CALLERID(num)})

exten => _94NXXNXXXXXX, 3, Dial(SIP/12345678*1${EXTEN:2}@Flowroute,,To)
exten => _94XNXXNXXXXXX, 3, Dial(SIP/12345678*${EXTEN:2}@Flowroute,,To)

exten => _94NXXNXXXXXX, 4, Hangup
exten => _94XNXXNXXXXXX, 4, Hangup

Protocol: SIP

Server IP: 0.0.0.0
```

- **Twilio**

See <https://www.twilio.com/docs/sip-trunking/ip-addresses> for more host addresses. You will likely want them all to be safe.

Carrier ID:	Twilio
Carrier Name:	Twilio ?
Carrier Description:	?
Admin User Group:	---ALL--- ?
Registration String:	?
Template ID:	--NONE-- ?
Account Entry:	<pre> [OutTwilio] disallow=all allow=ulaw type=friend host=your-twilio-endpoint dtmfmode=rfc2833 context=trunkinbound [Twilio](!) disallow=all allow=ulaw type=peer dtmfmode=rfc2833 context=trunkinbound [twilio-va-1](Twilio) host=54.172.60.0 [twilio-va-2](Twilio) host=54.172.60.1 [twilio-va-3](Twilio) host=54.172.60.2 etc... </pre>
Protocol:	SIP ?
Globals String:	?
Dialplan Entry:	<pre> exten => _91NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log) exten => _91NXXNXXXXXX,2,Set(CALLERID(num)=+1\${CALLERID(num)}) exten => _91NXXNXXXXXX,3,Dial(SIP/+1\${EXTEN:2}@OutboundTwilio,,To) exten => _91NXXNXXXXXX,4,Hangup </pre>
Server IP:	172.31.19.53 ? (0.0.0.0 is all servers) ?
Active:	Y ?
SUBMIT	

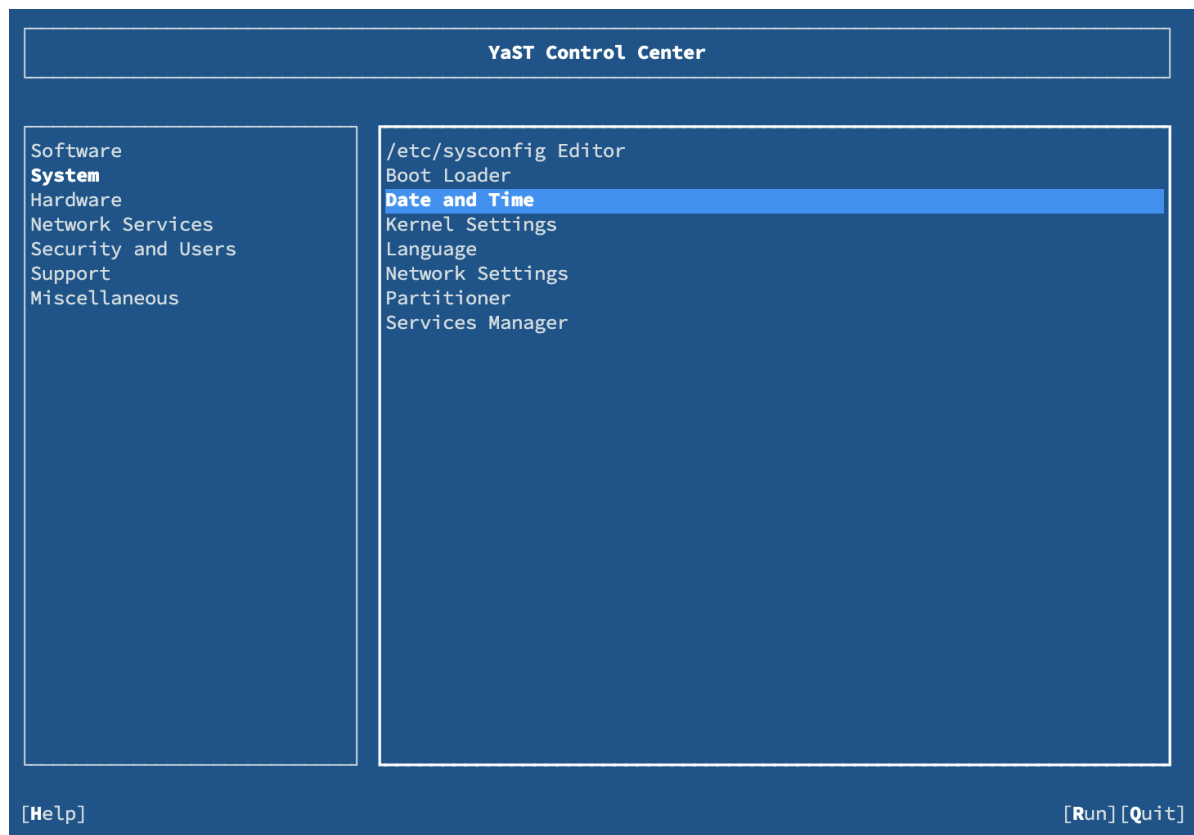
- **AWS Chime**

Carrier ID:	AWSChime
Carrier Name:	AWS Chime ⓘ
Carrier Description:	ⓘ
Admin User Group:	--ALL-- ⓘ
Registration String:	ⓘ
Template ID:	--NONE-- ⓘ
Account Entry:	<pre>[AWS] disallow=all allow=ulaw type=friend host=your-aws-chime-endpoint dtmfmode=rfc2833 context=trunkinbound</pre> ⓘ
Protocol:	SIP ⓘ
Globals String:	ⓘ
Dialplan Entry:	<pre>exten => _91NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log) exten => _91NXXNXXXXXX,2,Set(CALLERID(num)=+1\${CALLERID(num)}) exten => _91NXXNXXXXXX,3,Dial(SIP/AWS/+1\${EXTEN:2},,+To) exten => _91NXXNXXXXXX,4,Hangup</pre> ⓘ
Server IP:	172.31.19.53 ⓘ (0.0.0.0 is all servers) ⓘ
Active:	Y ⓘ
SUBMIT	

Tips

- **YaST**

In OpenSUSE Leap, YaST is your best friend. You can open the YaST Control Center by simply typing `yast` in your SSH session.



- **Timezones**

If you need to change your timezone for any reason. Be sure to do so in the following locations. Phones and servers may need to be modified as well depending on your needs.

- YaST > System > Date and Time

Note: If you wish to use the AWS Time Sync Service, you can configure that here too.

- /etc/php7/apache2/php.ini under the **date.timezone** option.
- /etc/php7/cli/php.ini under the **date.timezone** option.
- In VICIdial under Admin > System Settings > Default Local GMT.
- In VICIdial under Admin > System Settings > Default Voicemail Zone.

- **Swap and Recording**

A service called swap-monitor.service has been added to the 11.0.0 update. The script is located at /usr/local/bin/swap-monitor.sh.

- If you choose an instance without ephemeral storage it will provision a 4GB /swap file in the root directory.
- If you choose an instance with ephemeral storage, it will provision a 16GB swap partition on the ephemeral storage.

If you plan on a larger (greater than 10 agents) production environment, we recommend you choose an instance with ephemeral storage like c5d. The reason for this is that swap will drastically increase the load on your server while the EBS backed swap is in use. A very similar situation occurs with /var/spool/asterisk/monitor when the recording features are used.

If you don't intend to use a swap file, you're going to want at least 2GB of RAM + the size of your monitor folder in production when recording. You'll also want to disable the swap-monitor.service. If you need to modify the swap size simply edit the script listed above, then stop and start the instance.

If you intend to mount another volume for some reason, be sure it is mounted after the ephemeral device and after the initial setup. The script utilizes the first extra device found.

• Troubleshooting

If you start having problems, some of the best places to start looking are the following.

- VICIdial> Reports

Make sure your server(s) are green and the times are synced.

SERVER	DESCRIPTION	IP	ACT	LOAD	CHAN	AGNT	DISK	TIME	VER
ip-172-31-	Test install of Asterisk server	172.31.19.53	Y/Y/Y	39 - 6%	0	0	42%	2020-12-10 09:39:45	3336
		PHP Time						2020-12-10 09:39:45	
		DB Time						2020-12-10 09:39:45	

- If the asterisk time doesn't match PHP and DB, go to Admin > Servers > Asterisk Version and make sure this string matches your current Asterisk version. In your terminal you can confirm your current version with 'asterisk -V'

The 'V' is capitalized. You can also set the server to automatically restart asterisk or manually restart it with 'systemctl restart asterisk'.

- VICIdial> Reports > Admin Utilities

There are a number of reports here to assist your efforts.

- Run the 'htop' command.

This will give you a good idea of the state of your system resources and processes under the hood.

- In your terminal run 'asterisk -vr'

This can help a great deal in troubleshooting SIP issues. Add more v's to the command for more verbose output. See asterisk -h for more options.