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. **c5d** is recommended for production. 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. Run '**zypper ref**' and '**zypper up**' to update the system.
8. To begin a single server setup run '**express-setup**'. Advanced users with specific needs should run '**standard-setup**'.
9. Follow the setup directions and reboot the server.
10. After the setup, you may need to open/close additional ports in the instance's security group depending on your configuration.
11. An SSL certificate is recommended and can be installed by pointing a domain to your IP and running '**certbot-setup**' as root.
12. Visit https://your-instance-ip-or-domain, click Administration and enter '**6666**' for the username and your **instance ID** for the password.
13. Enter a new password and proceed to users.

14. Click the default user and add all necessary permissions under the "**ADMIN INTERFACE OPTIONS**" section.
15. Do **NOT** change the "**Modify Same User Level**" section until you have created a new user with the appropriate permissions.
16. After setting all of the admin interface options you need to 1, you'll be free to configure Vicidial.
17. Be sure to go to Admin > Servers > Modify and set the Asterisk Version to 13.34.0-vici or whatever your current version is. In SSH you can confirm your current version with 'asterisk -V' The 'V' is capitalized.

Viciphone Configuration

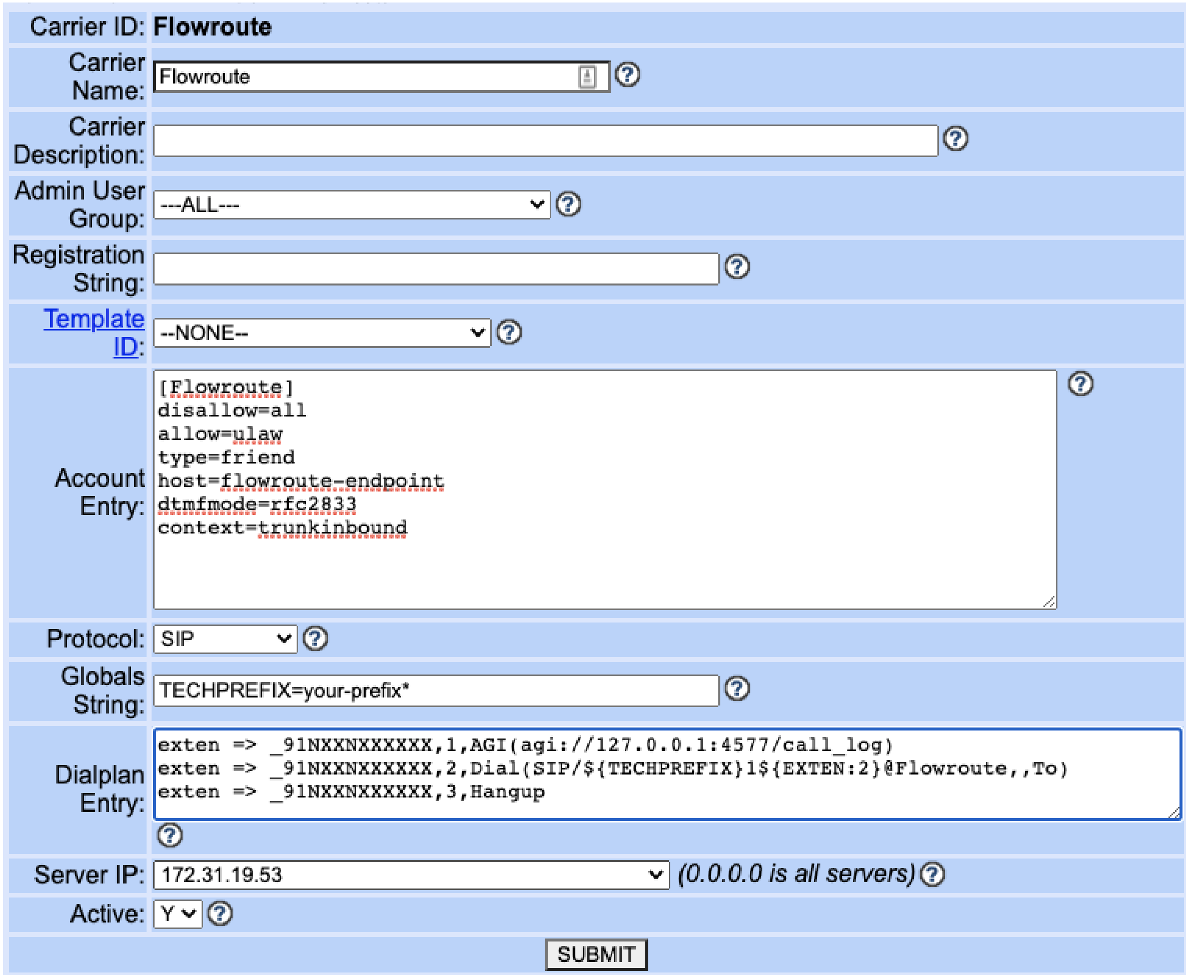
Viciphone is a browser based WebRTC phone. While you're free to use any phone you like for the agent leg, having the phone right in the agent session without extensions or additional software is extremely convenient. You do not have to use Viciphone. Vicidial can be used with any software or hardware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

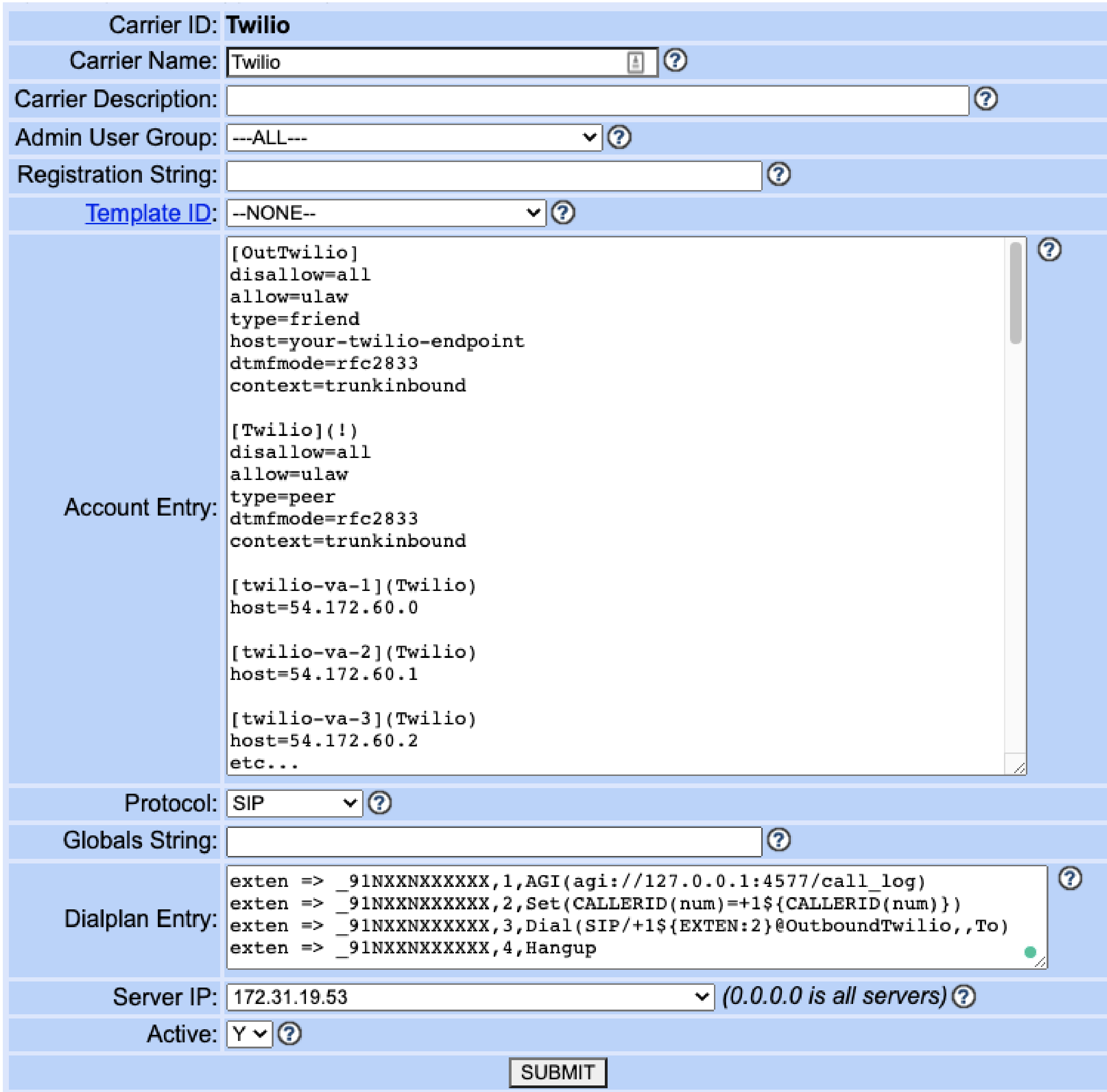
While you can use any SIP provider(s) you like, these are some popular ones in our experience. You can combine them as needed in a single dialplan or setup multiple carriers to use as backups. Please spend some time familiarizing yourself with how Asterisk Dialplans work.

- **Flowroute**



- **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:	(see below)
Protocol:	SIP
Globals String:	
Dialplan Entry:	(see below)
Server IP:	172.31.19.53 *(0.0.0.0 is all servers)*
Active:	Y

Account Entry:

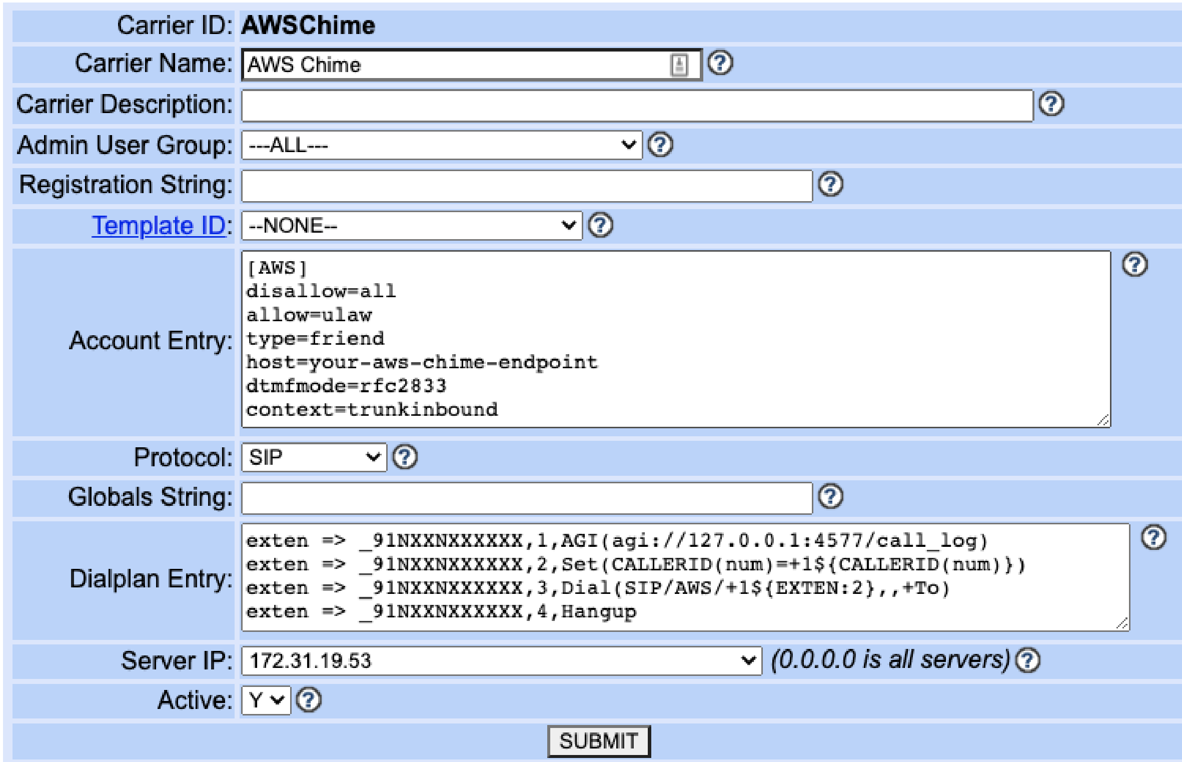
```
[OutTwilio]
disallow=all
allow=ulaw
type=friend
host=your-twilio-endpoint
dtmfmode=rfc2833
context=trunkinbound

[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...
```

Dialplan Entry:

```
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
```

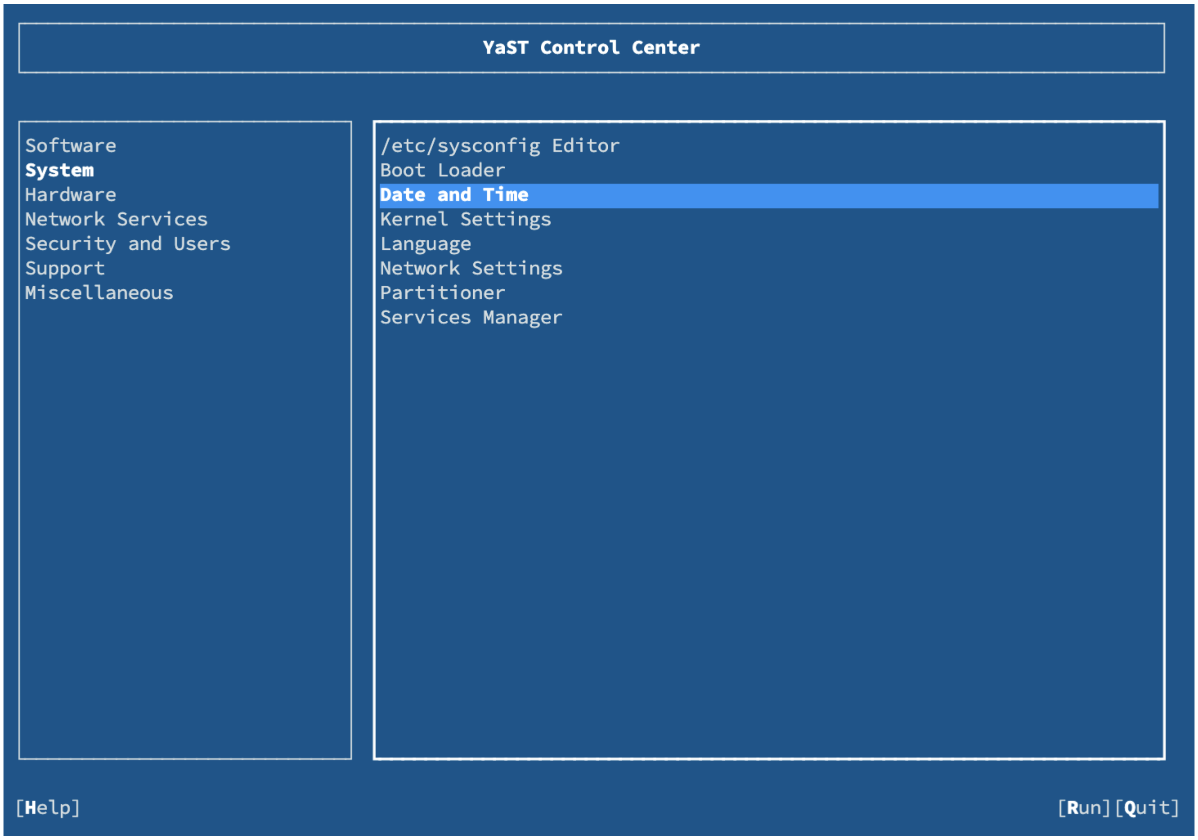
SUBMIT

- **AWS Chime**



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**

The following mostly applies to larger production environments with recording requirements.

You'll find that '/var/spool/asterisk/monitor' is mounted to tmpfs. See '/etc/fstab'. This is where recording processing takes place. You don't want this directory to exist anywhere but in memory or a swap file in ephemeral (locally attached) storage. **c5d** is an example of an instance with this type of storage.

You can set up a swap file on your EBS storage. You may even want to in a test environment on a smaller t3 instance for example. However, this type of setup will increase the load very quickly when you start adding agents and recordings. The problem with this configuration is that you'll have started storing virtual memory on what is essentially a network drive. No matter how good that network is, it will never be as fast as a locally attached volume or RAM. Also, AWS limits input/output operations per second (IOPS) on EBS volumes. You can overcome this limitation somewhat by provisioning more IOPS but this will become much more expensive and less effective than simply choosing an appropriate instance type.

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 can use YaST to configure your swap by navigating to System > Partitioner > nvme1n1 (or whatever device) > Partitions > Add Partition...

So your general options are...

1. Swap on ephemeral storage (recommended).
2. Swap on EBS storage (very small or test environments only).
3. No swap and lots of RAM.

- **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	

- Vicidial > Reports > Admin Utilities
  There are a number of reports here to assist your efforts.
- In SSH run the 'htop' command.
  This will give you a good idea of the state of your system resources and processes under the hood.
- In SSH 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.