



How to Use the Asterisk Marketplace Instance – Asterisk Version 17.0.0 – With AWS Chime

Thank you for your interest in open-source telephony from Asterisk. Below, you will find instructions on how to use this instance.

Quick Setup:

1. Login for this instance is centos.
2. To work with and run Asterisk, you will need to sudo -i.
3. Asterisk should start automatically. If does not, then issue this command:
 - a. asterisk
4. To view the Asterisk CLI, type asterisk -r
5. Default directory for configuration files:
 - a. /etc/asterisk
6. Modify your security group to restrict UDP 5060-5061 to only known sources. Otherwise, you may get hacked.
7. PHPMyAdmin URL – http://YOUR_INSTANCES_IP/phpmyadmin.
 - a. Username is root.
 - b. Password is your instance ID. Here's where to find it - <http://drops.tiltx.com/ZZO111>
 - c. For security reasons, please update your password. Also, please restrict your security group on port TCP 80 to only sources you know.
8. MySQL root password – It is your instance ID. Here's where to find it - <http://drops.tiltx.com/ZZO111>

AWS Chime Voice Connector Quick Setup:

1. Example of extensions.conf to place an outbound call.
 - a. Get the Outbound host name from your Voice Connector Group. Here's a screenshot of where to find it - <http://drops.tiltx.com/J78yGc>. Also, make sure your instance's public IP is in the 'Allowed Hosts List'.
 - b. In extensions, to place an outbound call, your dialplan should look something like this - <http://drops.tiltx.com/l6scAP>. Please note the +1 in front of the caller ID and the number sent to Amazon. In our case, +1 is our (US) country code (E.164 format). Please replace this as necessary.
2. Example of extensions.conf to receive a call:
 - a. The Chime hosts will look for the context 'DID-INBOUND' in your extensions.conf. Therefore, your extensions.conf should look something like this - <http://drops.tiltx.com/IDSE9w>. Please note that numbers are in E.164 format.

Assumptions:



- You have basic knowledge of Amazon Web Services, Linux, Asterisk, and an SSH client such as Putty (<https://www.chiark.greenend.org.uk/~sgtatham/putty/latest.html>) or SecureCRT.
- Very important note – If you assign an Elastic IP to your instance, you must reboot it in order for your SIP settings to take effect.
- If you need a SIP provider to provide inbound and outbound calling, we use TBI Telecom. Feel free to contact them at sales@tbitelecom.com.

If you have any questions, or need help, please feel free to email us:

- asterisk-awsmarketplace@tiltx.com.

Steps to Using and Configuring your Asterisk Instance

1. Once you've launched your instance, please copy/write down your instance's public IP; you will need this to connect your SIP phones and your SIP provider(s). Screenshot below and available here - <http://drops.tiltx.com/Pbm4FP>

The screenshot shows the AWS Management Console interface for an Amazon EC2 instance named 'Asterisk 17.0.0'. The instance is in the 'running' state and is located in the 'us-east-1a' availability zone. The 'IPv4 Public IP' is listed as '18.234.31.157', which is highlighted with a red box. Below the instance details, the 'Description' tab is selected, showing various instance attributes such as Instance ID, Instance state, Instance type, Elastic IPs, and Availability zone. The 'Public DNS (IPv4)' is also listed as 'ec2-18-234-31-157.compute-1.amazonaws.com'.

2. Log in via SSH to the IP address in from the 'IPv4 Public IP' with the certificate created during instance setup:
 - a. You will need the certificate you created when you launched the instance.
3. If you do not have an SSH client, you may download one here:
 - a. Putty: <https://www.chiark.greenend.org.uk/~sgtatham/putty/latest.html>
4. The login username is centos
5. Once logged in, you will need to issue the following command. Otherwise, you will not be able to access Asterisk:
 - a. `sudo -i`
6. **This step is very important.** Once you have are in as `sudo -s`, you will need to modify the `sip.conf` file to suit your needs. Most importantly, you will need to modify the entry(ies) at the very bottom of the file. This is where you specify your SIP providers and users:
 - a. Issue the following command – `vi /etc/asterisk/sip.conf` and press enter.



- b. Go all the way to the bottom of the file and update SIP profile to match what you need. Usually, you will only need to modify the 12.34.56.78 values to match yours:
 - i. Screenshot: <http://drops.tiltx.com/xcV7ID>
 - c. Save your settings (:wq!).
 7. Before sending calls, you will need to update your EC2 instance security group to allow SIP and RTP traffic. Here are the settings you need:
 - a. SIP: UDP 5060 - 5061; b. RTP: UDP 10000 – 20000
 - b. Remember to restrict your security group to only known SIP sources.
 8. Very important note – If you assign an Elastic IP to your instance, you must reboot it in order to your SIP settings to take effect.

If you need custom Asterisk development, or something more advanced, please feel free to let us know. We have an Asterisk Certified professional (since 2005) on staff that has seeing just about everything. You may email us at asterisk-aws-marketplace@tiltx.com. We want to see you succeed. Feel free to let us know how we are doing and what we may do to improve this template.