Amazon Chime Voice Connector

SIP Trunking Configuration Guide:

FreeSWITCH

December 2020

Document History

Rev. No.	Date	Description
1.0	Dec-15-2020	Draft SIP Trunk Configuration Guide
1.1	Feb-03-2021	Updated the document based on feedback

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1 Audience

This document is intended for technical staff and Value Added Resellers (VAR) with installation and operational responsibilities. This configuration guide provides steps for configuring SIP trunk using **FreeSWITCH** to connect to **Amazon Chime Voice Connector** for inbound and/or outbound telephony capabilities.

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1.1 Amazon Chime Voice Connector

Amazon Chime Voice Connector is a pay-as-you-go service that enables companies to make or receive secure phone calls over the internet or AWS Direct Connect using their existing telephone system or session border controller (SBC). The service has no upfront fees, elastically scales based on demand, supports calling both landline and mobile phone numbers in over 100 countries, and gives customers the option to enable inbound calling, outbound calling, or both.

Amazon Chime Voice Connector uses the industry-standard Session Initiation Protocol (SIP). Amazon Chime Voice Connector does not require dedicated data circuits. A company can use their existing Internet connection or AWS Direct Connect public virtual interface for SIP connectivity to AWS. Voice connectors can be configured in minutes using the AWS Management Console or Amazon Chime API. Amazon Chime Voice Connector offers cost-effective rates for inbound and outbound calls. Calls into Amazon Chime meetings, as well as calls to other Amazon Chime Voice Connector customers are at no additional cost. With Amazon Chime Voice Connector, companies can reduce their voice calling costs without having to replace their on-premises phone system.

2 SIP Trunking Network Components

The network for SIP Trunk reference configuration is illustrated below and is representative of **FreeSWITCH** with **Amazon Chime Voice Connector**

IP PBX is used as a secondary PBX in the topology to perform call failover and call distribution

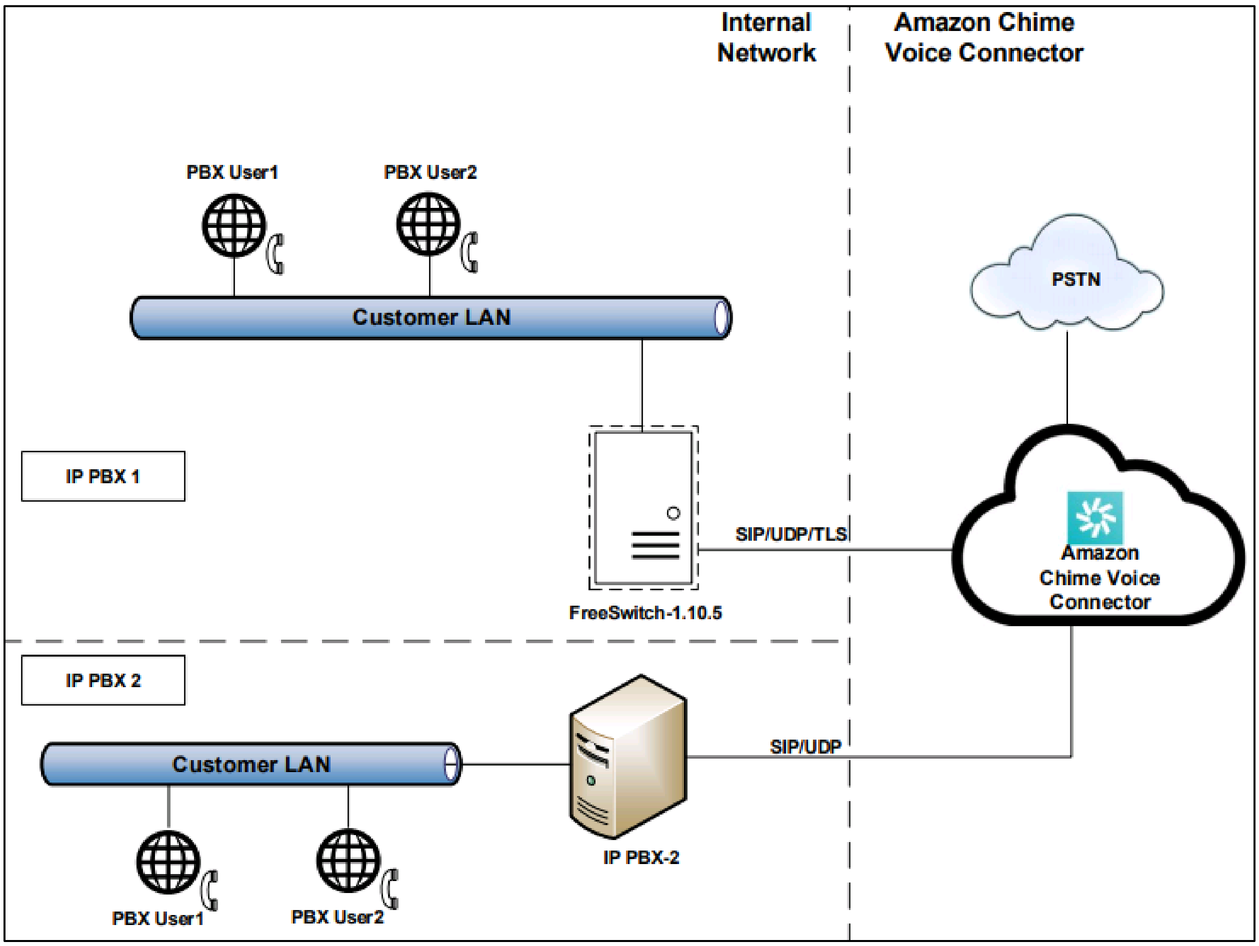

Figure 1 Network Topology

2.1 Hardware Components

- FreeSWITCH hosted on UCS-C240 VMWare running on ESXi-5.5.0

2.2 Software Requirements

- FreeSWITCH – 1.10.5-release-17-25569c1631~64bit

3 Features

3.1 Features Supported

- Calls to and from non Toll Free number
- Calls to Toll Free number
- Calls to Premium Telephone number
- Calling Party Number Presentation
- Calling Party Number Restriction
- Inbound Calls to an IVR
- International Calls
- Anonymous call
- DTMF-RFC 2833
- Long duration calls
- Calls to conference scheduled by Amazon Chime user
- Call Distribution
- Call Failover

3.2 Features Not Supported

- Amazon Chime Voice Connector responds to OPTIONS and TCP Keep Alive messages received from customer equipment, but does not send OPTIONS or TCP Keep Alive messages to customer equipment

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Amazon Chime Voice Connector,
    - does not support SIP NOTIFY or SIP INFO for DTMF
    - does not send SIP session refresher for long duration calls
- When the WAN link is down and a call is in progress, the PSTN call leg is not disconnected automatically after a period of inactivity. The call has to be cleared manually

4 Configuration

The specific values listed in this guide are used in the lab configuration described in this document and are for illustrative purposes only. You must obtain and use the appropriate values for your deployment. Encryption is always recommended if supported.

4.1 Configuration Checklist

This section presents an overview of the steps that are required to configure **FreeSWITCH** for SIP Trunking with **Amazon Chime Voice Connector**

Table 1 – PBX Configuration Steps

Steps	Description	Reference
Step 1	FreeSWITCH Configuration	Section 4.2
Step 2	Amazon Chime Voice Connector Configuration	Amazon Chime Voice Connector

4.2 FreeSWITCH Configuration

This section with screen shots taken from FreeSWITCH used for the interoperability testing gives a general overview of the FreeSWITCH configuration.

4.2.1 FreeSWITCH Login and Version

1. Using an application that supports **ssh**, access FreeSWITCH, enter the IP address of FreeSWITCH and enter the credentials to login

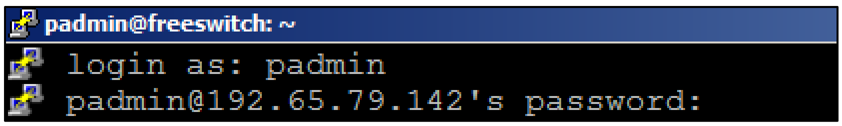

Figure 2: FreeSWITCH Login

2. To verify the system version of **FreeSWITCH** being tested, Login as **Root** user, enter the command **fs_cli** to enter the Command Line Interface(CLI) of FreeSWITCH and enter the command **version**

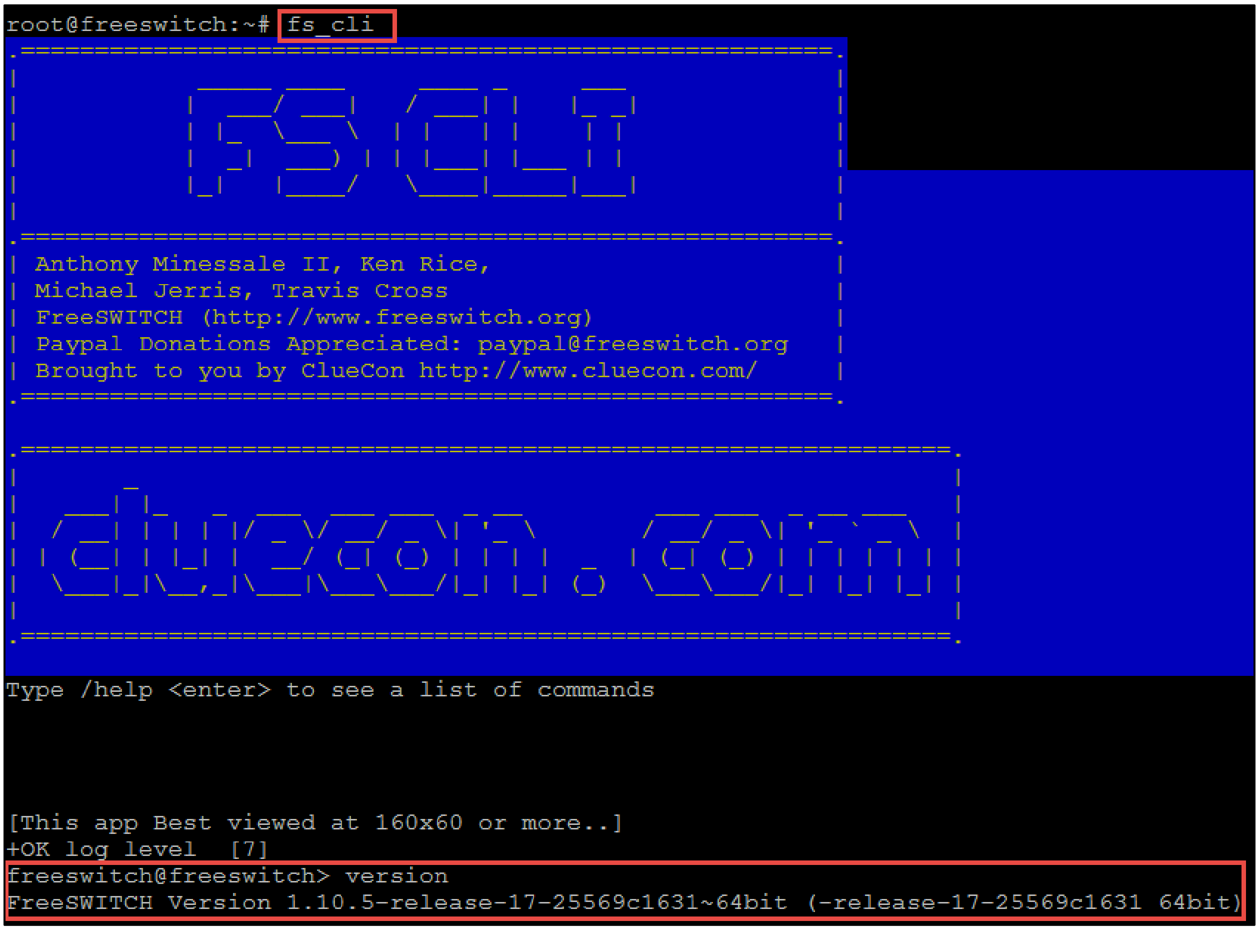

Figure 3: FreeSWITCH Version

4.2.2 Directory Configuration

1. Navigate to **/etc/freeswitch/directory/default**
2. Create an XML file for new directory
3. Enter the **user id, password, effective caller id name** and **number** and **Outbound caller id name** and **number** as shown below

```
root@freeswitch:/etc/freeswitch/directory/default# ls
[redacted]3711.xml [redacted]
root@freeswitch:/etc/freeswitch/directory/default# vi [redacted]3711.xml
```

Figure 4 Directory Configuration

```
<include>
  <user id="[redacted]3711">
    <params>
      <param name="password" value="$${default_password}"/>
      <param name="vm-password" value="[redacted]3711"/>
    </params>
    <variables>
      <variable name="toll_allow" value="domestic,international,local"/>
      <variable name="accountcode" value="[redacted]3711"/>
      <variable name="user_context" value="default"/>
      <variable name="effective_caller_id_name" value="[redacted]3711"/>
      <variable name="effective_caller_id_number" value="[redacted]3711"/>
      <variable name="outbound_caller_id_name" value="$${outbound_caller_name}"/>
      <variable name="outbound_caller_id_number" value="$${outbound_caller_id}"/>
      <variable name="callgroup" value="techsupport"/>
    </variables>
  </user>
</include>
```

Figure 5 Directory Configuration Contd.,

4.2.3 SIP Trunk using UDP

4.2.3.1 Gateway

1. Navigate to **/etc/freeswitch/sip_profiles/external**
2. Create a XML file (**aws.xml**) for SIP Gateway pointing to Amazon Chime Voice Connector
3. Enter the details of **gateway name**, **proxy** (FQDN of Amazon Chime Voice Connector) , **register transport**, **from-domain** (FQDN of Amazon Chime Voice Connector), **caller-id-in-from, enable-timer, session-timeout, minimum-session-expires, ping, ping-max, ping-min** as show below

```
root@freeswitch:/etc/freeswitch/sip_profiles/external# ls
aws.xml [redacted]
```

Figure 6 SIP Gateway

```
<include>
  <gateway name="awsvc">
  <param name="proxy" value="[redacted].voiceconnector.chime.aws"/>
  <param name="register-transport" value="udp"/>
  <param name="register" value="false"/>
  <param name="from-domain" value="[redacted].voiceconnector.chime.aws"/>
  <param name="caller-id-in-from" value="true"/>
  <param name="enable-timer" value="true"/>
  <param name="session-timeout" value="900"/>
  <param name="minimum-session-expires" value="120"/>
  <param name="ping" value="60"/>
  <param name="ping-max" value="10"/>
  <param name="ping-min" value="1"/>
  </gateway>
</include>
```

Figure 7 SIP Gateway Contd.,

4.2.3.2 Vars.xml

1. Navigate to **/etc/freeswitch/**
2. In the **vars.xml** file, in **External SIP Profile** section enter the fields **external_sip_port, external_tls_port, external_ssl_enable** as shown below and leave the rest to default

```
root@freeswitch:/etc/freeswitch# ls vars.xml
vars.xml
```

Figure 8 vars.xml configuration

```
<!-- External SIP Profile -->
<X-PRE-PROCESS cmd="set" data="external_auth_calls=false"/>
<X-PRE-PROCESS cmd="set" data="external_sip_port=5060"/>
<X-PRE-PROCESS cmd="set" data="external_tls_port=5061"/>
<X-PRE-PROCESS cmd="set" data="external_ssl_enable=false"/>
```

Figure 9 vars.xml configuration contd.,

4.2.3.3 Outbound Dial Plan

1. Navigate to **/etc/freeswitch/dialplan/default**
2. Create XML file (**awsdpob.xml**) for an outbound dial plan to Amazon Chime Voice Connector
3. Enter the details of **destination_number, sip_cid_type, dtmf_type, nolocal:absolute_codec_string, bridge=sofia/gateway/awsvc/+1$1** (Provide the Gateway name created in the Gateway section) as shown below for Local and International outbound calls

```
root@freeswitch:/etc/freeswitch/dialplan/default# ls awsdpob.xml
awsdpob.xml
```

Figure 10 Outbound Dial Plan configuration

```
<extension name="local - aws">
    <condition field="${toll_allow}" expression="local"/>
    <condition field="destination_number" expression="^9(\d{10})$">
      <action application="export" data="sip_cid_type=none"/>
      <action application="set" data="sip_cid_type=pid"/>
      <action application="export" data="dtmf_type=rfc2833"/>
      <action application="export" data="nolocal:absolute_codec_string=PCMU"/>
      <action application="bridge" data="sofia/gateway/awsvc/+1$1"/>
    </condition>
</extension>
<extension name="International - aws">
    <condition field="${toll_allow}" expression="international"/>
    <condition field="destination_number" expression="^9(\d+)$">
      <action application="export" data="sip_cid_type=none"/>
      <action application="set" data="sip_cid_type=pid"/>
      <action application="export" data="dtmf_type=rfc2833"/>
      <action application="export" data="nolocal:absolute_codec_string=PCMU"/>
      <action application="bridge" data="sofia/gateway/awsvc/+$1"/>
    </condition>
</extension>
```

Figure 11 Outbound Dial Plan configuration contd.,

4.2.3.4 Inbound Dial Plan

1. Navigate to **/etc/freeswitch/dialplan/public**
2. Create a XML file (**awsdpib.xml**) for inbound dial plan from Amazon Chime Voice Connector to FreeSWITCH to route the call to a directory number
3. Enter the details of **destination_number, transfer** (Directory number created in the previous steps)

```
padmin@freeswitch:/etc/freeswitch/dialplan/public$ ls awsdpib.xml
awsdpib.xml
```

Figure 12 Inbound Dial Plan configuration

```
<extension name="public_did_aws-[redacted]3711">
    <condition field="destination_number" expression="^\+[redacted]3711$">
      <action application="transfer" data="[redacted]3711 XML default"/>
    </condition>
</extension>
```

Figure 13 Inbound Dial Plan configuration contd.,

4.2.4 SIP Trunk using TLS

4.2.4.1 Gateway

1. Navigate to **/etc/freeswitch/sip_profiles/external**
2. Create a XML file (**aws.xml**) for SIP Gateway pointing to Amazon Chime Voice Connector
3. Configure **tls-version** and **register-transport** as show below and leave the rest of the fields to values as configured earlier

```
<include>
  <gateway name="awsvc">
  <param name="proxy" value="[redacted].voiceconnector.chime.aws"/>
  <param name="register-transport" value="tls"/>
  <param name="tls-version" value="tlsv1"/>
  <param name="register" value="false"/>
  <param name="from-domain" value="[redacted].voiceconnector.chime.aws"/>
  <param name="caller-id-in-from" value="true"/>
  <param name="enable-timer" value="true"/>
  <param name="session-timeout" value="900"/>
  <param name="minimum-session-expires" value="120"/>
  <param name="ping" value="60"/>
  <param name="ping-max" value="10"/>
  <param name="ping-min" value="1"/>
  </gateway>
</include>
```

Figure 14 SIP Configuration-TLS

4.2.4.2 Vars.xml

1. Navigate to **/etc/freeswitch/**
2. In the **vars.xml** file, in **External SIP Profile** section, configure the field **external_ssl_enable** as shown below and leave the rest of the fields to values as configured earlier

```
<!-- External SIP Profile -->
<X-PRE-PROCESS cmd="set" data="external_auth_calls=false"/>
<X-PRE-PROCESS cmd="set" data="external_sip_port=5060"/>
<X-PRE-PROCESS cmd="set" data="external_tls_port=5061"/>
<X-PRE-PROCESS cmd="set" data="external_ssl_enable=true"/>
```

Figure 15 SIP Configuration-TLS

4.2.4.3 Outbound Dial Plan

1. Navigate to **/etc/freeswitch/dialplan/default**
2. Create a XML file (**awsdpob.xml**) for an outbound dial plan to Amazon Chime Voice Connector
3. Configure SRTP Crypto attribute **nolocal:rtp_secure_media=true:AES_CM_128_HMAC_SHA1_80** as shown below for Local and International outbound calls and leave the rest of the fields to values as configured earlier

```
root@freeswitch:/etc/freeswitch/dialplan/default# ls awsdpob.xml
awsdpob.xml
```

Figure 16 Outbound Dial Plan –TLS configuration

```
<extension name="local - aws">
    <condition field="${toll_allow}" expression="local"/>
    <condition field="destination_number" expression="^9(\d{10})$">
      <action application="export" data="sip_cid_type=none"/>
      <action application="set" data="sip_cid_type=pid"/>
      <action application="export" data="dtmf_type=rfc2833"/>
      <action application="export" data="nolocal:absolute_codec_string=PCMU"/>
      <action application="export" data="nolocal:rtp_secure_media=true:AES_CM_128_HMAC_SHA1_80"/>
      <action application="bridge" data="sofia/gateway/awsvc/+1$1"/>
    </condition>
</extension>
<extension name="International - aws">
    <condition field="${toll_allow}" expression="international"/>
    <condition field="destination_number" expression="^9(\d+)$">
      <action application="export" data="sip_cid_type=none"/>
      <action application="set" data="sip_cid_type=pid"/>
      <action application="export" data="dtmf_type=rfc2833"/>
      <action application="export" data="nolocal:absolute_codec_string=PCMU"/>
      <action application="export" data="nolocal:rtp_secure_media=true:AES_CM_128_HMAC_SHA1_80"/>
      <action application="bridge" data="sofia/gateway/awsvc/+$1"/>
    </condition>
</extension>
```

Figure 17 Outbound Dial Plan –TLS configuration contd.,