



SIP Trunking Configuration Guide: Configuring a SIP Trunk from Bandwidth to Amazon Chime SDK Voice Connector and a SIP Media Application

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1 Audience

This document is intended for technical staff and Value-Added Resellers (VARs) with installation and operational responsibilities. This configuration guide provides steps for configuring **SIP Trunks** between operator **Bandwidth** and an **Amazon Chime SDK Voice Connector**. An example is provided of the trunk then connecting to a **SIP Media Application (SMA)** that is used to bridge an inbound call to a meeting.

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1.1 Amazon Chime SDK Voice Connector

Amazon Chime SDK 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 SDK Voice Connector uses the industry-standard Session Initiation Protocol (SIP). Amazon Chime SDK 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 SDK Voice Connector offers cost-effective rates for inbound and outbound calls. Calls into Amazon Chime meetings, as well as calls to other Amazon Chime SDK Voice Connector customers are at no additional cost. With Amazon Chime SDK Voice Connector, companies can reduce their voice calling costs without having to replace their on-premises phone system.

SIP media applications make it easier and faster for you to create custom signaling and media instructions that you would normally build on your private branch telephone exchange (PBX).

SIP rules specify how a SIP media application can connect to an Amazon Chime SDK meeting. Calls can go to and from private phone numbers that you own or to and from a Request URI hostname, the name assigned to an Amazon Chime SDK Voice Connector. The Amazon Chime SDK runs the SIP rules when a user places or receives a call.

You must be an AWS Lambda user before you can create SIP media applications.

2 SIP Trunking Network Components

The network for SIP Trunk reference configuration is illustrated below and is representative of call routing through **Bandwidth** with **Amazon Chime SDK Voice Connector** and **SMA**.

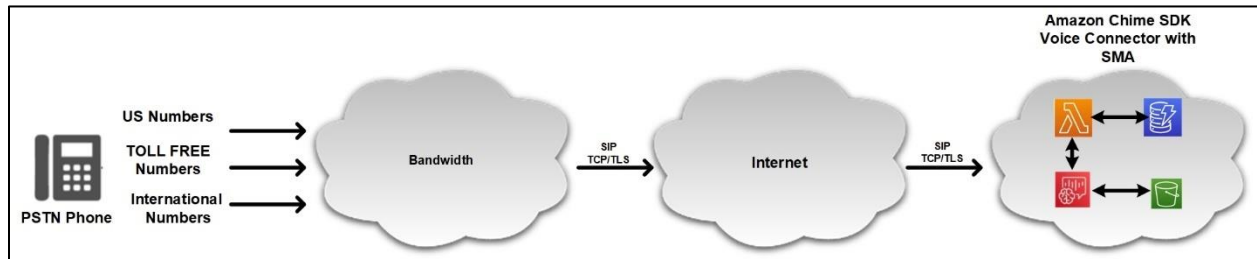


Figure 1 : Network Topology

2.1 Hardware Components

- None

2.2 Software Requirements

- None

3 Features

3.1 Features Supported

The below call scenarios are tested with TCP and TLS.

- Inbound calls to SMA using following numbers.
 - * US Toll free numbers
 - * US Toll numbers
 - * International Numbers (UK, AUSTRALIA, BELGIUM)
- Calling Party Number Presentation
- DTMF-RFC 2833
- Long duration calls

3.2 Features Not Supported

- None

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Amazon Chime SDK Voice Connector does not accept calls from source regions other than US.
- No ring back to the caller when the call is transferred to another DID from Amazon SMA, after pressing DTMF key number to transfer the call.

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

In this section we present an overview of the steps that are required for establishing a **Bandwidth** SIP Trunk to **Amazon Chime SDK Voice Connector** and then creating a **SIP media application** action to bridge an inbound SIP call to a meeting.

Table 1 – PBX Configuration Steps

Steps	Description	Reference
Step 1	Amazon Chime SDK Voice Connector and SIP Media Application Configuration	Section 4.2
Step 2	Bandwidth SIP Trunk Configuration	Section 4.3

4.2 Amazon Chime SDK Voice Connector and SIP Media Application Configuration

4.2.1 Create SIP Trunk in Amazon Chime SDK Voice Connector

To create an Amazon Chime SDK Voice Connector

1. Login to Amazon Chime console at <https://console.aws.amazon.com/chime-sdk/home>
2. For **SIP Trunking**, choose **Voice Connectors**.

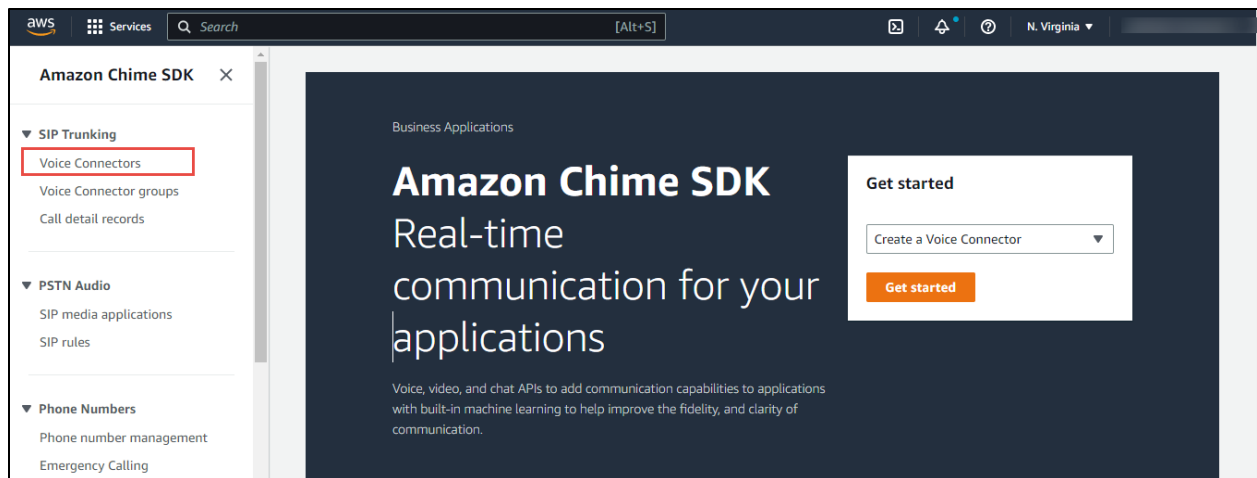


Figure 2 : Create Amazon Chime SDK Voice Connector

3. Choose **Create Voice Connector**.

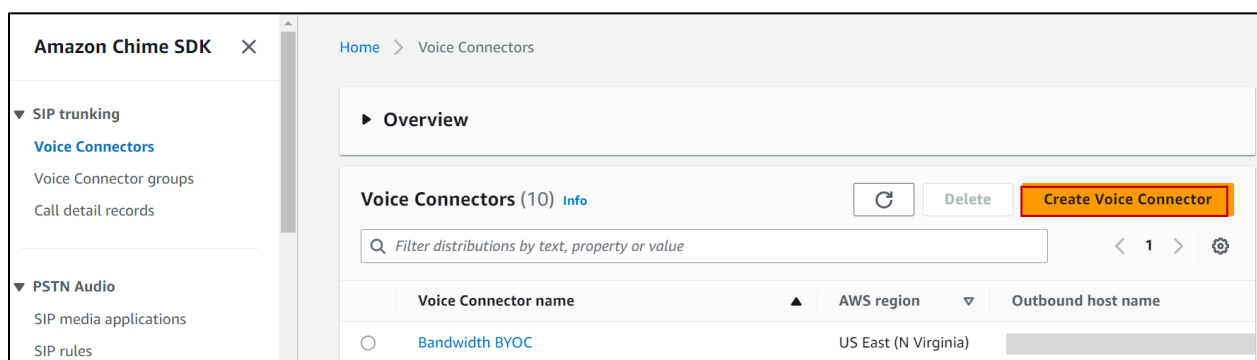


Figure 3 : Create Amazon Chime SDK Voice Connector (cont.)

4. Enter a **Voice Connector name**, (Bandwidth BYOC is used in this example) to create a trunk to Bandwidth.
5. For **Encryption (TLS)** select **Enabled** or **Disabled** for **TCP**
6. Choose **Create Voice Connector**.

Amazon Chime SDK > Voice Connectors > Create Voice Connector

Create Voice Connector [Info](#)

Create an Amazon Chime Voice Connector to make phone calls using your existing SIP infrastructure. [Find out more](#)

Set-up

Voice Connector name

Bandwidth BYOC

AWS region

US East (N Virginia)

Encryption

☐ Enabled – default

☒ Disabled

Tags - optional [Info](#)

A tag is a label that you assign to an AWS resource. Each tag consists of a key and an optional value. You can use tags to search and filter your resources or track your AWS costs.

No items associated with the resource.

[Add new tag](#)

You can add up to 50 tags.

Cancel [Create Voice Connector](#)

Figure 4 : Create Amazon Chime SDK Voice Connector (cont.)

Note

Enabling encryption configures your Amazon Chime SDK Voice Connector to use TLS transport for SIP signaling and Secure RTP (SRTP) for media. Inbound calls use TLS transport, and unencrypted outbound calls are blocked.

4.2.2 Access List in Amazon Chime SDK Voice Connector

1. Open the Amazon Chime console at <https://console.aws.amazon.com/chime-sdk/home>
2. For **Calling**, choose **Voice Connectors**.
3. Choose the name of the Amazon Chime SDK Voice Connector to edit.
4. Choose **Termination** and select Termination status **Enabled**. Termination describes calls routed from your SIP infrastructure and terminating on a Voice Connector
5. The **Outbound host name** is present in the Termination tab of Amazon Chime SDK Voice Connector. Bandwidth must send the SIP INVITE to Amazon Chime SDK Voice Connector containing this host name in the Request-URI.
6. For **Allowed hosts list**, choose **New**, enter the CIDR notations and values to allow list, and choose **Add**. The subnets of Bandwidth's IP address are given in the **Allowed hosts list**.

The screenshot shows the Amazon Chime SDK Voice Connector console. The breadcrumb navigation is 'Home > Voice Connectors > Bandwidth BYOC'. The main heading is 'Bandwidth BYOC'. Below it are tabs: 'General', 'Termination' (highlighted with a red box), 'Origination', 'Emergency calling', 'Phone numbers', 'Streaming', 'Logging', and 'Tags'. The 'Termination' tab content includes a message: 'Enable termination settings to control outbound calling from your SIP infrastructure. [Find out more](#)'. Below this is the 'Termination status' section with two radio buttons: 'Enabled' (selected and highlighted with a red box) and 'Disabled'. The 'Outbound host name' section has a text input field containing 'epitoh' followed by '.voiceconnector.chime.aws'. Below that is the 'Allowed hosts list*' section with a description: 'IP addresses allowed to make calls using your Voice Connector. You can create up to 10 entries.' At the bottom of this section are two buttons: 'New' (highlighted with a red box) and 'Actions' with a dropdown arrow.

Figure 5 : Create Amazon Chime SDK Voice Connector (cont.)

Note

Adding host addresses is not limited to inbound and outbound configuration. Multiple host addresses may be required due to SIP infrastructure dependency.

4.2.3 Create AWS Lambda function

In this example, a lambda function is used to bridge an inbound PSTN call from Bandwidth to an Amazon Chime SDK meeting. The lambda function executes the instruction, a SIP media application is used to call the function, and a SIP rule is used to trigger the SIP media application when the inbound phone number is detected by the SIP rule. See the [Administrator Guide](#) for an overview of SIP applications.

1. Open the Amazon console at <https://console.aws.amazon.com/console/>.
2. In the menu select Services > All services and choose **Lambda**.

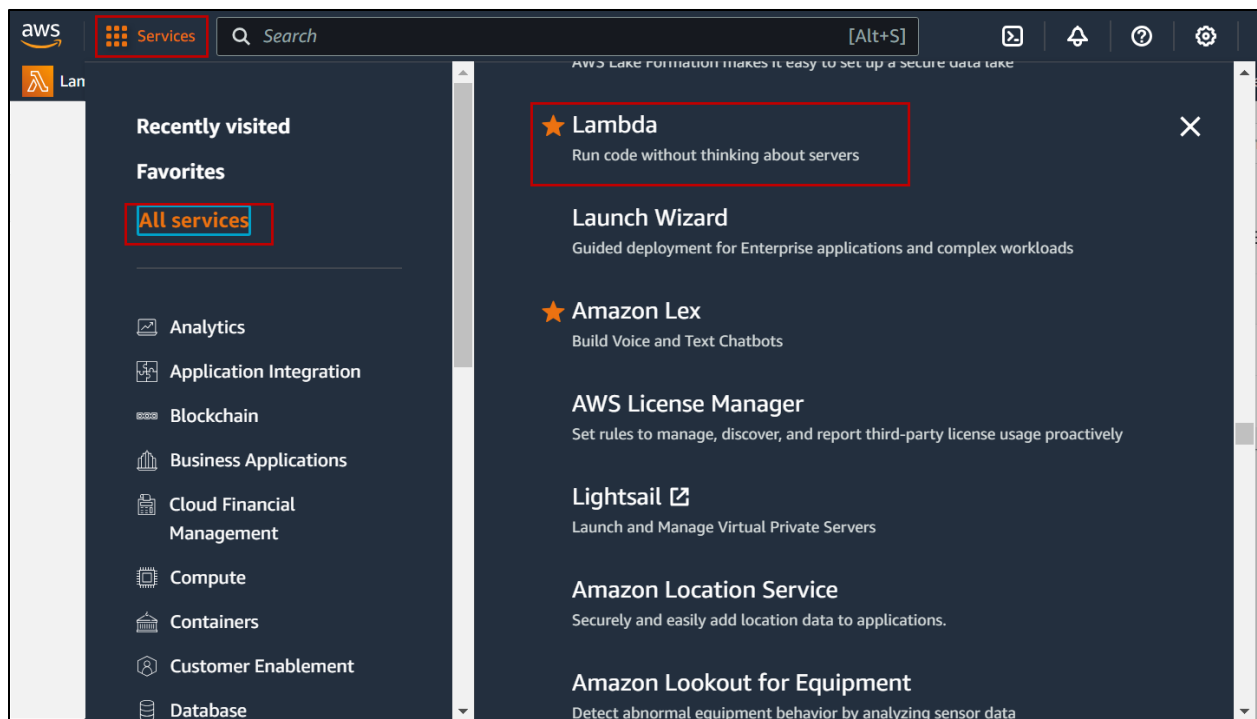


Figure 6 : Create Lambda

3. In the AWS Lambda menu, select the **Functions** and click **Create function** button to create a new lambda function.

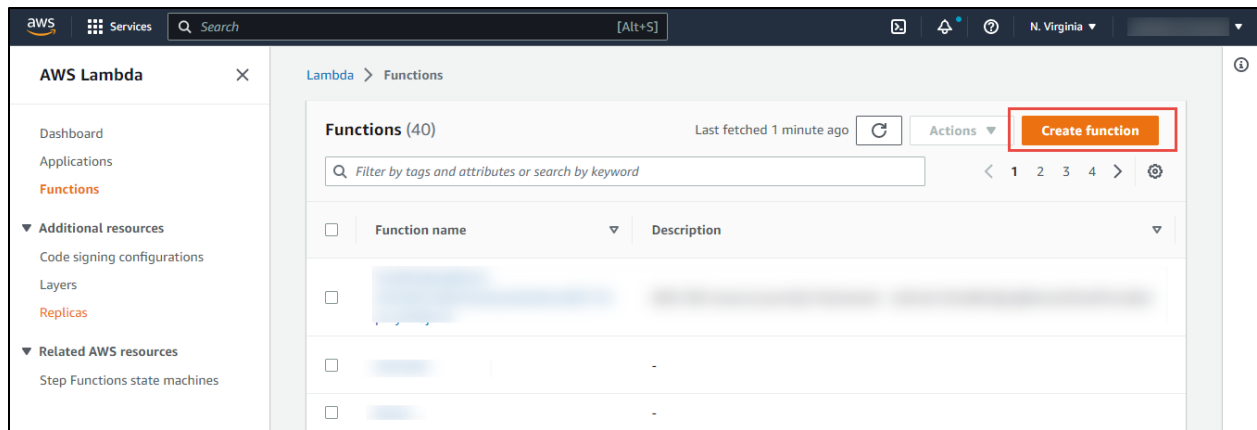


Figure 7 : Create Lambda (cont.)

Open the lambda **Function** and click **Copy ARN** button. This ARN will be used while creating SIP Media Applications.

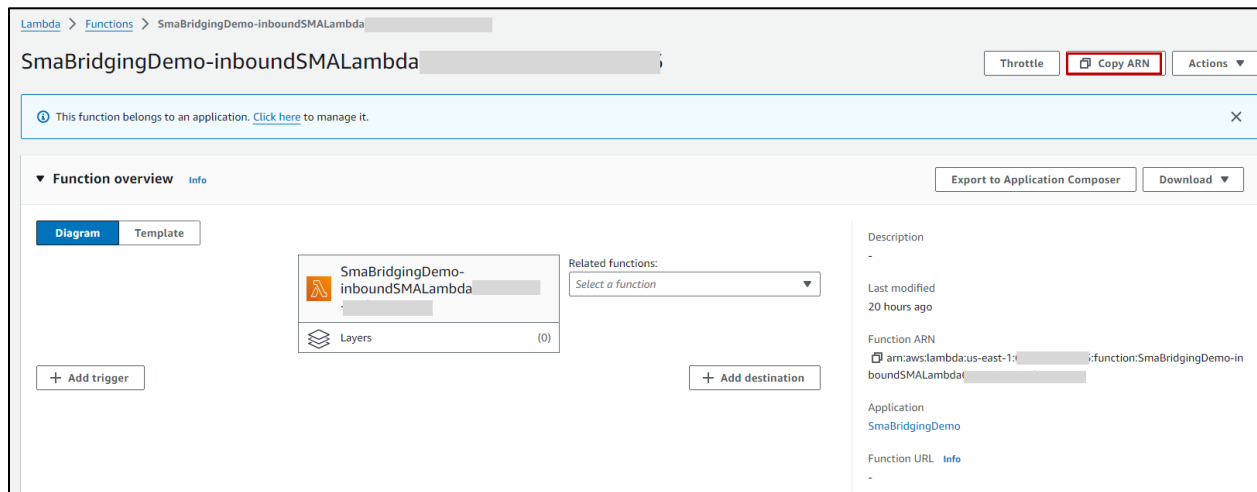


Figure 8 : Create Lambda (cont.)

4.2.4 Create SIP Media Application

SIP media applications are used to call lambda functions. The ARN for the lambda function is required, and therefore it is necessary to create the lambda function prior to the SIP media application.

To create a SIP media application:

1. Open the Amazon Chime console at <https://console.aws.amazon.com/chime-sdk/home>.
2. In the Amazon Chime SDK console, in the navigation pane, choose **SIP media applications**.
3. Choose **Create a SIP media application**. The **Create a SIP media application** page appears.

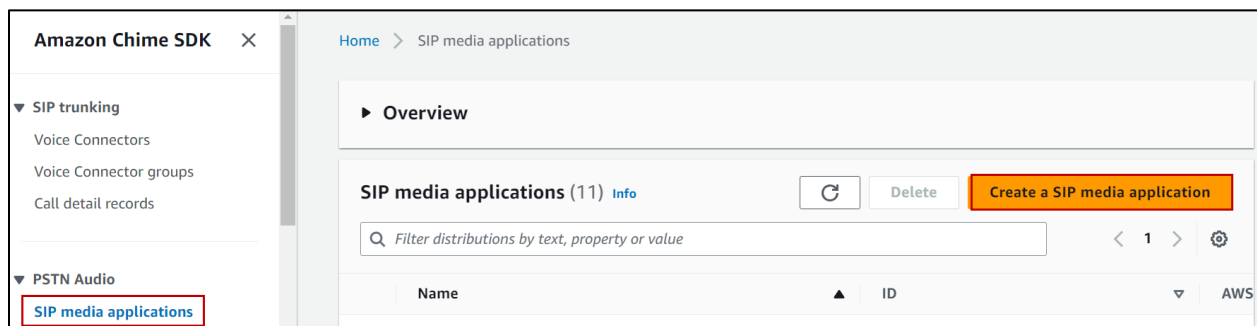


Figure 9 : Create SIP Media Application

4. For **Name**, enter a name for your application.
5. Copy your Lambda function's ARN and paste it into the **Lambda function ARN** box.
6. Choose **Create a SIP media application**.
7. A success message appears at the top of the **Create a SIP media application** page, and your media application appears in the list of applications.

Amazon Chime SDK > SIP media applications > Create a SIP media application

Create a SIP media application [Info](#)

Create an Amazon Chime SDK SIP media application using your existing Lambda function. [Find out more](#)

Set-up

Name

SipMediaApplication

Must have a length less than or equal to 256

AWS region

us-east-1

Lambda function ARN

arn:aws:lambda:us-east-1: [redacted] :function:SmaBridgingDemo-inboundSMALa

Must be a valid ARN

Tags - optional [Info](#)

A tag is a label that you assign to an AWS resource. Each tag consists of a key and an optional value. You can use tags to search and filter your resources or track your AWS costs.

No items associated with the resource.

Add new tag

You can add up to 50 tags.

[Cancel](#) [Create a SIP media application](#)

Figure 10 : Create SIP Media Application (cont.)

4.2.5 Create SIP Rules

SIP rules are triggers that invoke SIP media applications. There are two trigger types, either a phone number (used in this example) or a Voice Connector request URI.

To create a SIP rule:

1. Open the Amazon Chime console at <https://console.aws.amazon.com/chime-sdk/home>.
2. In the navigation pane, choose **SIP rules**. The **SIP rules** page appears.
3. Choose **Create SIP rule**. The **Create SIP rule** dialog box appears.

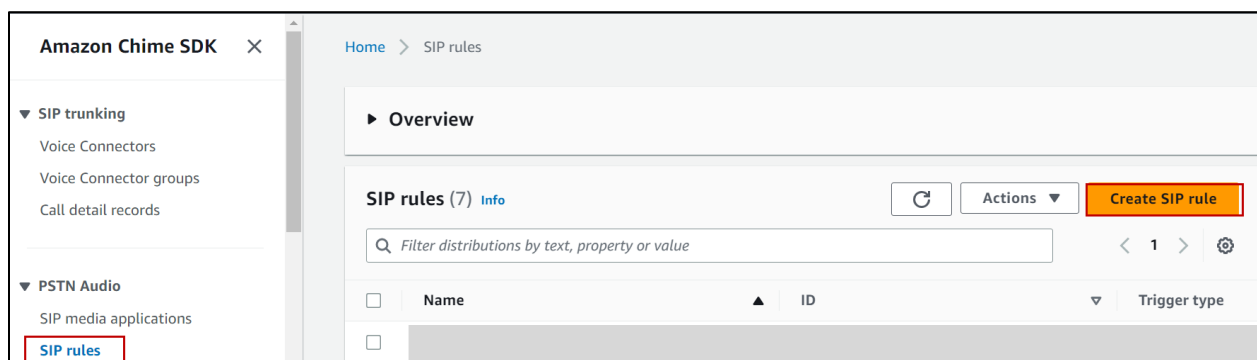


Figure 11 : Create SIP Rule

4. Enter a Name for the rule.
5. Set the Trigger type to **RequestURIHostname**.
6. From the drop down, select the Request URI hostname as the voice connector host created.
7. Click **Add a SIP media application**.
8. open the **SIP media application** list and select the SIP application that you want to use.
9. Choose **Create SIP rule**.

Amazon Chime SDK > SIP rules > Create SIP rule

Create SIP rule [Info](#)

Enter a name and select a target type and trigger type for your SIP rule. Choose the tick box to enable the rule. Add target SIP media applications with priority. [Find out more](#)

Set-up

Name

Bandwidth

Must have a length less than or equal to 256

Trigger type

RequestUriHostname

Request URI hostname

Currently, only Voice Connector hosts are supported

.voiceconnector.chime.aws

Status

☒ Enabled

SIP media applications

SIP media application	Priority	
SmaBridgingDemo-inbound-SMA ...	1	Remove
Add a SIP media application		

[Cancel](#) [Create SIP rule](#)

Figure 12 : Create SIP Rule (cont.)

4.2.6 Enable SIP Logs in Amazon Chime SDK Voice Connector

The Amazon Chime SDK disables logging for Voice Connectors by default. When you enable logging, the system sends the data to an Amazon CloudWatch log group.

To enable SIP logs for inbound calls:

1. Open the Amazon Chime console at <https://console.aws.amazon.com/chime-sdk/home>.
2. For **Collecting SIP logs**, Choose the name of the Amazon Chime SDK Voice Connector to edit.
3. Choose **Logging** and select SIP message logs **Enabled**.
4. Choose **Save**.

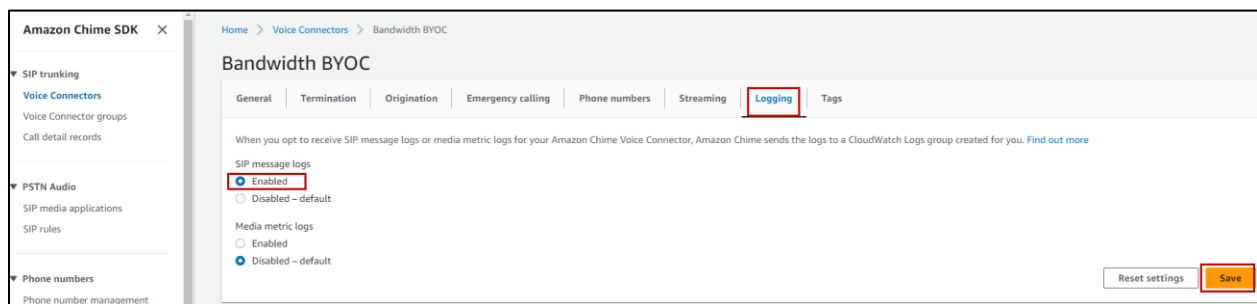


Figure 13 : Enable SIP message logs in Cloud Watch

4.2.7 Collect CloudWatch SIP Logs

To collect the SIP Logs

1. Open the Amazon console at <https://console.aws.amazon.com/console/>.
2. In the menu select services choose **CloudWatch**.
3. Select the **log groups** in AWS CloudWatch and filter the SIP log using the Voice Connector Outbound host name.
4. Select the **SIP messages** from Log group.

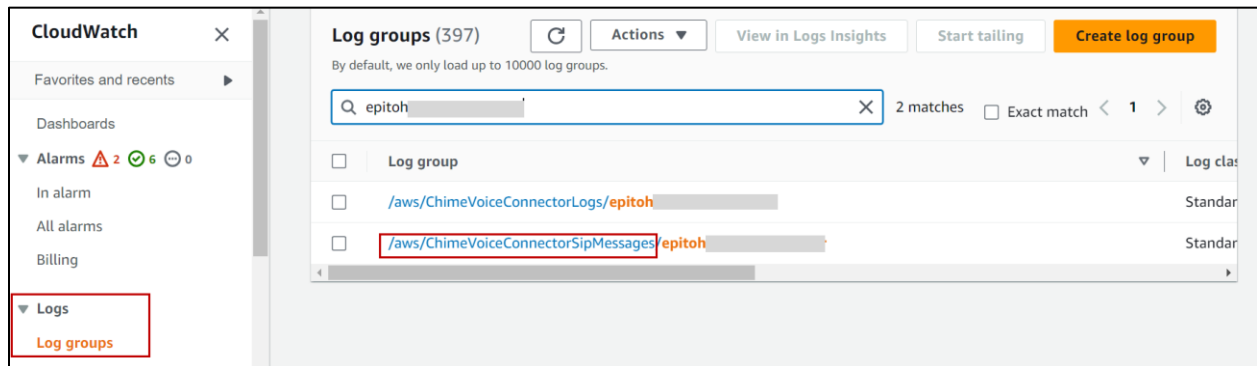


Figure 14 : Collect CloudWatch SIP Logs

4.3 Bandwidth SIP Trunk Configuration

Bringing Bandwidth to Amazon Chime SDK Voice Connector as your carrier of choice delivers numerous benefits to your business, including:

- Greater control over your PSTN calling;
- Instant PSTN voice access to 60+ countries representing +90% of global GDP;
- Fully compliant services for peace of mind over long-term stability and availability;
- High quality voice and low latency on your calls, consistently across every market;
- Better cross-application workflows by using Bandwidth provisioning APIs to automate advanced functions such as phone number procurement, configuration and regulatory submissions.

4.3.1 Prerequisites for the setup:

A registered Bandwidth account with Secure Comms (SRTP) enabled and access to Global Portal <https://app.voxbone.com>.

Telephone Numbers purchased and provisioned under 'Numbers' tab. Bringing Bandwidth to Amazon Chime SDK.

4.3.2 Create Voice URI

1. Login to the Bandwidth Portal at <https://login.voxbone.com/login> with appropriate credentials.
2. Navigate to **Settings > Voice URIs** and create a Voice URI to create a SIP trunk towards Amazon Chime SDK Voice Connector.
3. Select **PROTOCOL** as SIP.
4. Enter **URI** as +1323xxxxxxx@epitohxxxxxxxxx.voiceconnector.chime.aws;transport=tls (Configure transport = tcp for TCP). +1323xxxxxxx is the Bandwidth US DID number. epitohxxxxxxxxx is the Amazon Chime SDK Voice Connector Outbound host name.
5. Enter **DESCRIPTION** as Voice echo.
6. Click **SAVE**.

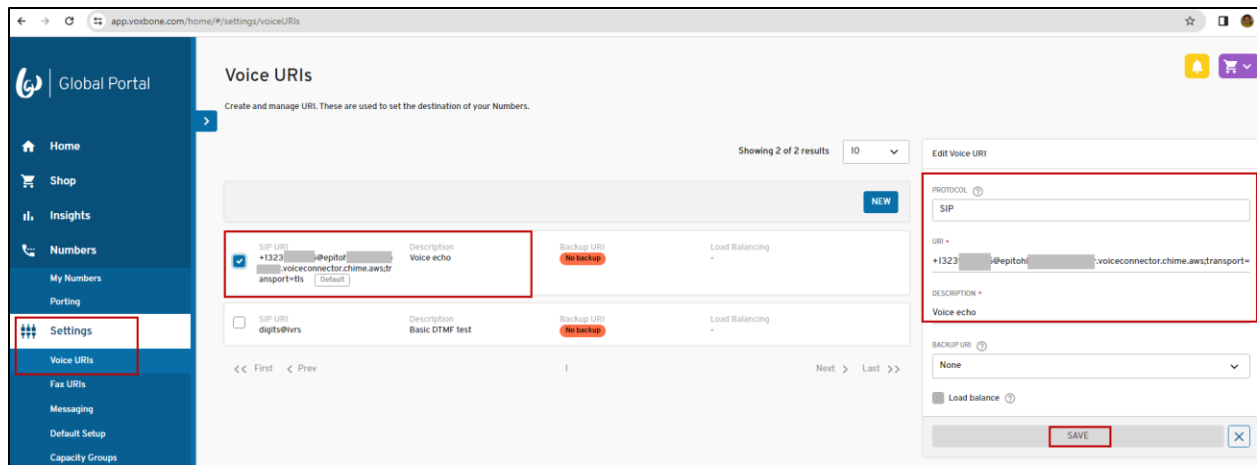


Figure 15 : Configure Voice URI

4.3.3 Configure Numbers

1. Navigate to **Numbers > My Numbers**.
2. Navigate to **Basic** tab and select **VOICE URI** as
+1323xxxxxxx@epitohxxxxxxxxx.voiceconnector.chime.aws;transport=tls.
3. Click **SAVE**.

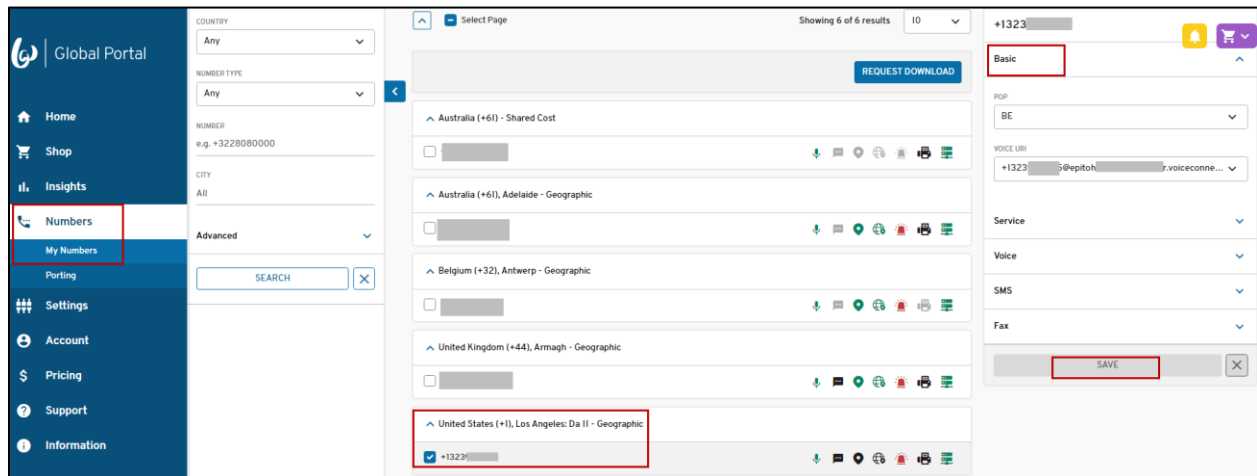
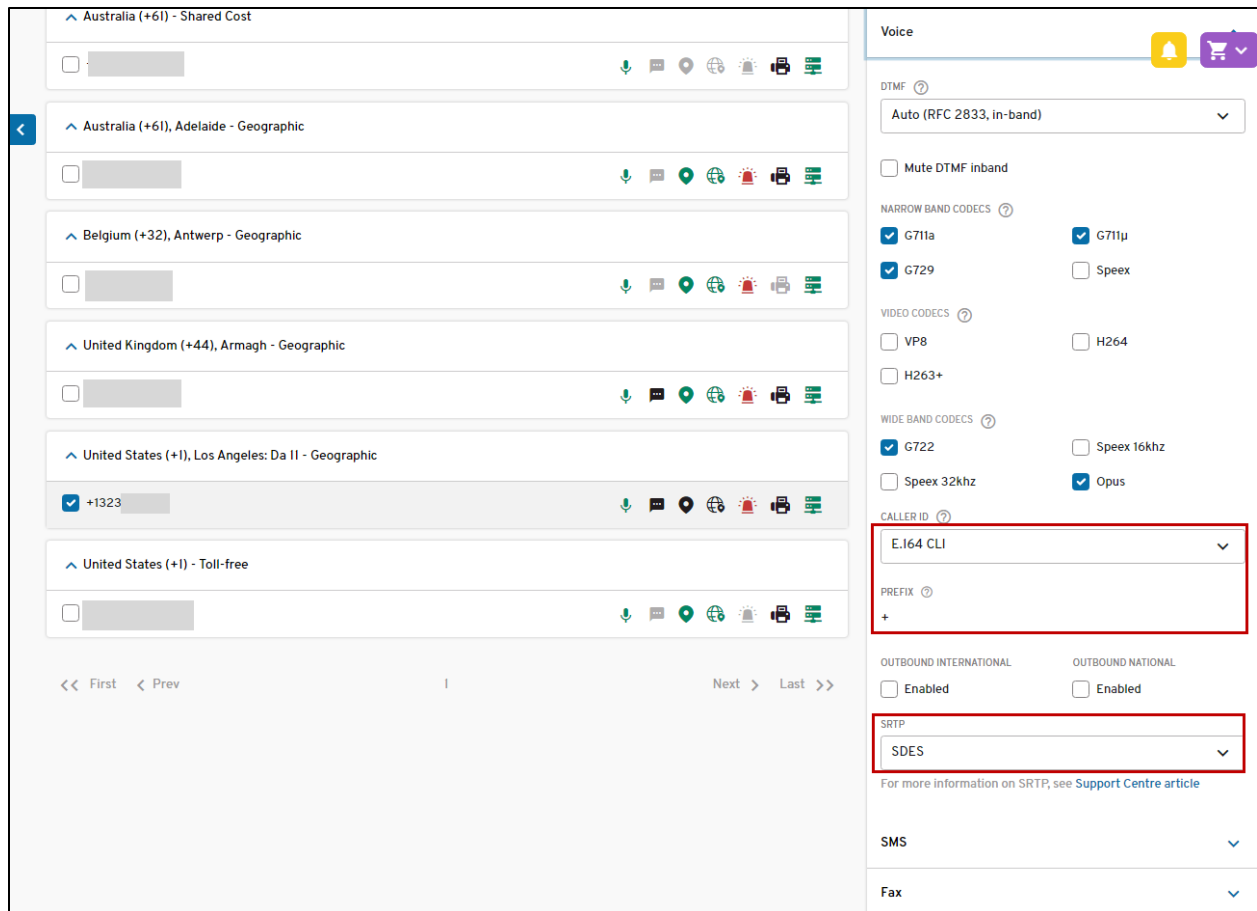


Figure 16 : Configure Number

4. Navigate to **Voice** tab and select **CALLER ID** as E.164 CLI
5. Enter **PREFIX** as +
6. Select **SRTP** as SDES
7. Click **SAVE**.
8. Repeat the same steps for the other DIDs.



The screenshot displays the AWS IAM console interface for configuring a DID number. The left pane shows a list of DIDs, and the right pane shows the configuration options for the selected DID.

Left Pane (DID List):

- Australia (+61) - Shared Cost
- Australia (+61), Adelaide - Geographic
- Belgium (+32), Antwerp - Geographic
- United Kingdom (+44), Armagh - Geographic
- United States (+1), Los Angeles: Da II - Geographic
- United States (+1) - Toll-free

Right Pane (Voice Configuration):

- DTMF:** Auto (RFC 2833, in-band)
- Mute DTMF inband:** ☐
- NARROW BAND CODECS:**
 - ☒ G711a
 - ☒ G711u
 - ☒ G729
 - ☐ Speex
- VIDEO CODECS:**
 - ☐ VP8
 - ☐ H264
 - ☐ H263+
- WIDE BAND CODECS:**
 - ☒ G722
 - ☐ Speex 16khz
 - ☐ Speex 32khz
 - ☒ Opus
- CALLER ID:** E.164 CLI
- PREFIX:** +
- OUTBOUND INTERNATIONAL:** ☐ Enabled
- OUTBOUND NATIONAL:** ☐ Enabled
- SRTP:** SDES
- SMS:** ☐ Enabled
- Fax:** ☐ Enabled

Figure 17 : Configure Number (cont.)