



**Amazon Chime SDK
Voice Connector
SIP Trunking Configuration
Guide:
FreePBX (Asterisk)**

March 2023

Document History

Rev. No.	Date	Description
1.0	Nov-17-2020	Draft SIP Trunk Configuration Guide
1.1	Dec-09-2020	Initial release
1.2	Feb-03-2021	Minor updates
2.0	March 28, 2023	Updated for FreePBX v16.0.21.9 and Asterisk v19.5.0

Table of Contents

1	Audience.....	5
1.1	Amazon Chime SDK Voice Connector.....	5
2	SIP Trunking Network Components.....	5
2.1	Hardware Components	6
2.2	Software Requirements	6
3	Features.....	7
3.1	Features Supported	7
3.2	Features Not Supported	7
3.3	Features Not Tested	7
3.4	Caveats and Limitations	7
4	Configuration.....	8
4.1	Configuration Checklist.....	8
4.2	Free PBX Configuration.....	8
4.2.1	FreePBX Login and Version	8
4.2.2	Extension Configuration.....	9
4.2.3	SIP Trunk using UDP.....	10
4.2.4	SIP Trunk using TLS	13
4.2.5	Outbound Route.....	14
4.2.6	Inbound route	15

Table of Figures

Figure 1: Network Topology	6
Figure 2: Free PBX login.....	9
Figure 3: Free PBX Version.....	9
Figure 4: Asterisk Version	9
Figure 5: User-Extension Configuration.....	10
Figure 6: User-Extension Configuration cont.....	10
Figure 7: SIP Configuration-UDP-chan_pjsip	11
Figure 8: SIP Configuration-UDP-chan_pjsip cont.....	11
Figure 9: SIP Configuration-General	11
Figure 10: SIP Configuration-General	12
Figure 11: SIP Configuration- Advanced	12
Figure 11: SIP Configuration- Advanced	13
Figure 13: SIP Configuration - Codecs.....	13
Figure 14: SIP Configuration – TLS	14
Figure 15: SIP Configuration – TLS	14
Figure 16: Outbound Route – Route Settings.....	15
Figure 17: Outbound Route – Dial Patterns	15
Figure 18: Inbound Route.....	16

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 **FreePBX (Asterisk)** to connect to **Amazon Chime SDK Voice Connector** for inbound and/or outbound telephony capabilities.

The information in this document is for informational purposes only. AWS does not guarantee the accuracy of this document and AWS has no responsibility or liability for errors or omissions related to this document. The document is subject to change without notice, and should not be construed as a commitment by AWS.

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.

2 SIP Trunking Network Components

The network for the SIP Trunk reference configuration is illustrated below and is representative of FreePBX (Asterisk) with Amazon Chime SDK Voice Connector.

IP PBX 2 is used as a secondary PBX in the topology to perform call failover and call distribution.

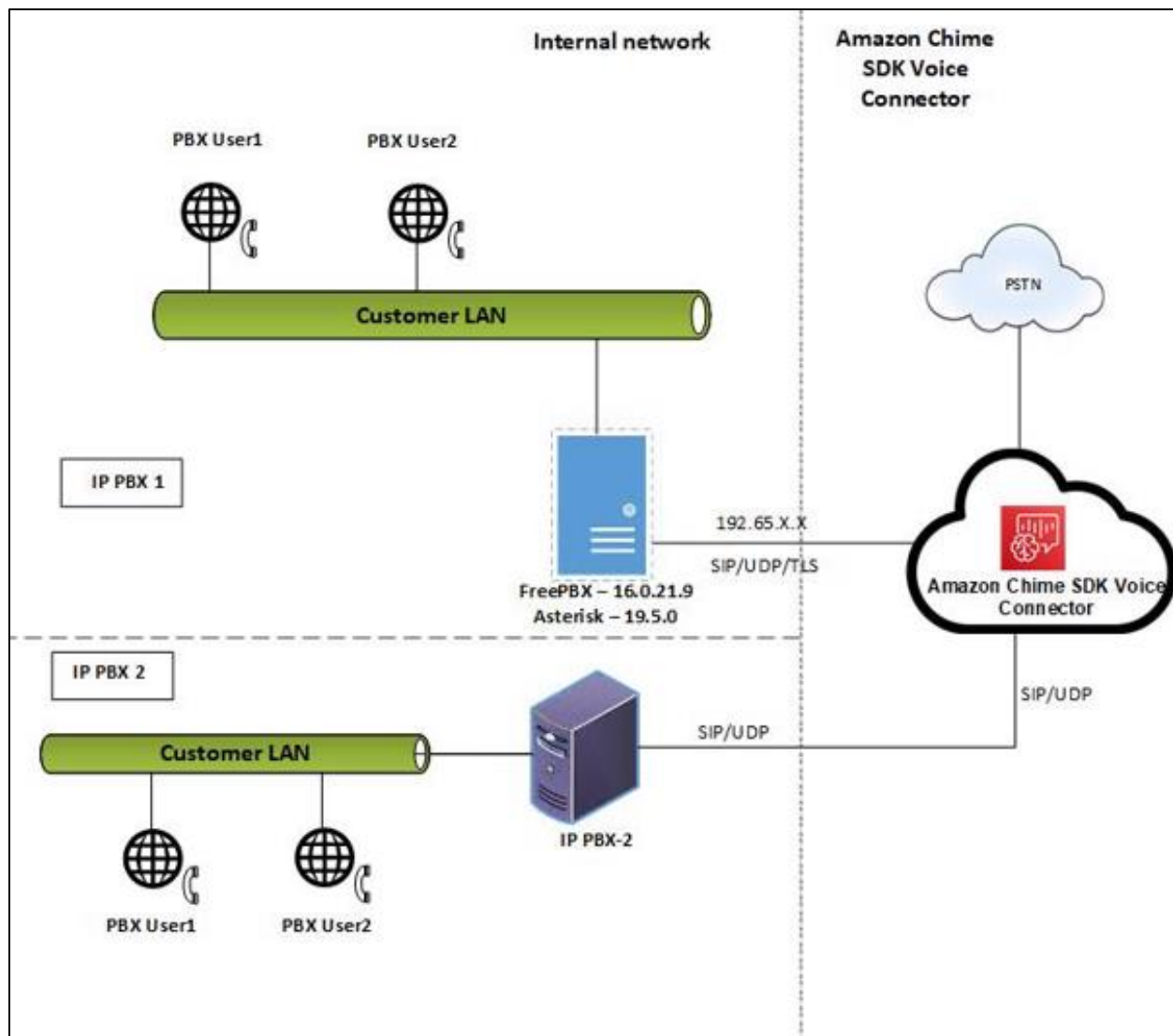


Figure 1: Network Topology

2.1 Hardware Components

- FreePBX (Asterisk) hosted on UCS-C240 VMWare running on ESXi

2.2 Software Requirements

- FreePBX – 16.0.21.9 & Asterisk – 19.5.0

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
- Calls to Business number
- 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 SDK Voice Connector responds to OPTIONS messages received from customer equipment, but does not send OPTIONS messages to customer equipment.
- Keep Alive – Double CRLF are not supported by Amazon Chime SDK Voice Connector and FreePBX (Asterisk).

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Amazon Chime SDK Voice Connector:
 - Does not support SIP NOTIFY or SIP INFO for DTMF.
 - Does not send SIP session refresher for long duration calls.
 - Does not acknowledge for SIP OPTIONS from FreePBX (Asterisk) when the REQ_URI format is in "user@fqdn" and it causes outbound calls using call authentication to fail.
- FreePBX (Asterisk) does not support wild card certificate sent by Amazon Chime SDK

Voice Connector hence disabled the Server/Peer certificate verification in FreePBX(Asterisk) for a successful secured inbound and outbound calling.

- FreePBX(Asterisk) does not send Session Refresh for long duration calls. Kept the call active for one hour and verified the call stays connected.

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. Customer 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 **FreePBX (Asterisk)** for SIP Trunking with **Amazon Chime SDK Voice Connector**.

Steps	Description	Reference
Step 1	FreePBX(Asterisk) Configuration	Section 4.2

Table 1 – PBX Configuration Steps

4.2 Free PBX Configuration

This section, with screen shots taken from the FreePBX (Asterisk) used for the interoperability testing, gives a general overview of the FreePBX (Asterisk) configuration.

4.2.1 FreePBX Login and Version

1. Open the browser and enter the IP address of FreePBX (Asterisk) and click on **FreePBX Administration** option to enter the credentials and click on **Continue** to login.

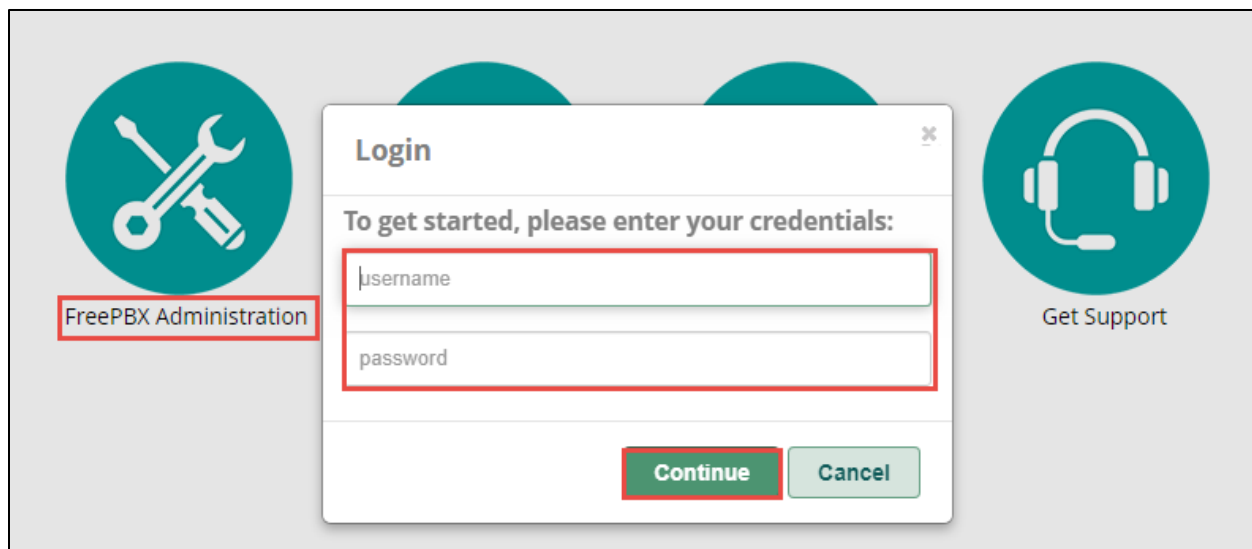


Figure 2: Free PBX login

2. To verify the system version of **FreePBX** being tested, click on **Dashboard** to find the version of **FreePBX**.

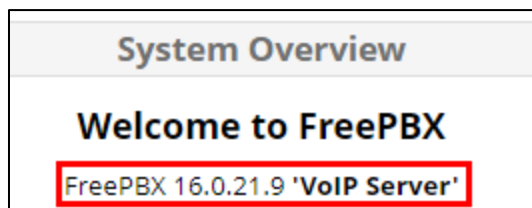


Figure 3: Free PBX Version

3. To verify the system version of **Asterisk** being tested, navigate to **Reports → Asterisk Info** to find the version of **Asterisk**



Figure 4: Asterisk Version

4.2.2 Extension Configuration

1. Navigate to **Applications → Extensions**
2. Choose **Add New SIP [chan_pjsip] Extension**
3. The following are the values that are configured in **Display Name, Outbound CID, Secret,**

Username and **Password for new user** in **General** Tab and leave the rest of the fields to default value.

The screenshot shows the Asterisk SIP Settings [chan_pjsip] tab, General section. The 'Edit Extension' section is highlighted with a red box, showing fields for Display Name (919), Outbound CID (+1919), Emergency CID, and Secret (9192893711). The 'Language' section shows Language Code set to Default. The 'User Manager Settings' section shows 'Linked to User 9192893711' and 'Select User Directory' set to 'PBX Internal Directory'. The 'Submit' button is highlighted with a red box.

Figure 5: User-Extension Configuration

The screenshot shows the Asterisk SIP Settings [chan_pjsip] tab, General section, User Manager Settings. The 'Username' and 'Password For New User' fields are highlighted with a red box. The 'Username' field is empty, and the 'Password For New User' field is empty. The 'Submit' button is highlighted with a red box.

Figure 6: User-Extension Configuration cont.

4.2.3 SIP Trunk using UDP

1. Navigate to **Settings → Asterisk SIP Settings**
2. The following are the values that are configured in **SIP Settings [chan_pjsip]** tab,
 - a. **udp-0.0.0.0-All** is set to **Yes** in **Transports** section
 - b. **Port to listen** on for **UDP** is **5060**
3. Leave the rest of the fields to default values.

Transports

Note that the interface is only displayed for your information, and is not referenced by asterisk. After you enable/disable a transport, asterisk needs to be **restarted**, not just reloaded.

udp

udp - 0.0.0.0 - All ☒ Yes ☐ No

tcp

tcp - 0.0.0.0 - All ☐ Yes ☒ No

tls

tls - 0.0.0.0 - All ☐ Yes ☒ No

ws

ws - 0.0.0.0 - All ☐ Yes ☒ No

Figure 7: SIP Configuration-UDP-chan_pjsip

0.0.0.0 (udp)

Port to Listen On

Domain the transport comes from

External IP Address

Local network

Figure 8: SIP Configuration-UDP-chan_pjsip cont.

4. Navigate to **Connectivity** → **Trunks** → click on **Add Trunk** and choose **Add SIP(chan_pjsip) Trunk**.
5. The following are the values that are configured in **Trunk Name**, **CID Options**, **Maximum Channels** in **General** Tab and leave the rest of the fields to default values.

General | Dialed Number Manipulation Rules | pjsip Settings

Trunk Name

Hide CallerID ☒ Yes ☐ No

Outbound CallerID

CID Options

Maximum Channels

Asterisk Trunk Dial Options

Continue if Busy ☒ Yes ☐ No

Disable Trunk ☒ Yes ☐ No

Monitor Trunk Failures

Figure 9: SIP Configuration-General

6. The following are the values that are configured in **SIP Server** (Outbound host name from Amazon Chime SDK Voice Connector), **SIP Server Port** (5060), **Context**(from-pstn), **Transport** (UDP) in **General** Tab in **pjsip Settings**. Leave the rest of the fields to default values.

General | Dialed Number Manipulation Rules | **pjsip Settings**

PJSIP Settings

General | Advanced | Codecs

Username: Authentication Disabled

Auth username: Authentication Disabled

Secret: Authentication Disabled

Authentication: Outbound | Inbound | Both | **None**

Registration: Send | Receive | **None**

Language Code: Default

SIP Server: gdr

SIP Server Port: 5060

Context: from-pstn

Transport: 0.0.0.0-udp

Submit | Duplicate | Reset | Delete

Figure 10: SIP Configuration-General

7. The following are the values that are configured in **Qualify Frequency** (60) for sending SIP OPTIONS, **From Domain** (Outbound host name from Amazon Chime SDK Voice Connector) and **Send RPID/PAI** is set to **Send P-Asserted-Identity header** in **Advanced** tab in **pjsip Settings**. Leave the rest of the fields to default values.

General | **Advanced** | Codecs

DTMF Mode: Auto

Permanent Auth Rejection: Yes | **No**

Forbidden Retry Interval: 30 Seconds

Fatal Retry Interval: 30 Seconds

General Retry Interval: 60 Seconds

Expiration: 3600 Seconds

Max Retries: 10000

Qualify Frequency: 60 Seconds

Outbound Proxy:

User = Phone: Yes | **No**

Contact User:

From Domain: gdr

Submit | Duplicate | Reset | Delete

Figure 11: SIP Configuration- Advanced

Trust RPID/PAI ⓘ	<input type="button" value="Yes"/> <input type="button" value="No"/>		
Send RPID/PAI ⓘ	<input type="button" value="No"/>	<input type="button" value="Send Remote-Party-ID header"/>	<input type="button" value="Send P-Asserted-Identity header"/>
	<input type="button" value="Both"/>		
Send Private CallerID Information ⓘ	<input type="button" value="Yes"/> <input type="button" value="No"/>		

Figure 12: SIP Configuration- Advanced

8. The following Codec (**G711 ulaw**) is selected in **Codec** tab in **pjsip Settings**

General	Dialed Number Manipulation Rules	pjsip Settings
PJSIP Settings		
General	Advanced	Codecs
Check the desired codecs, all others will be disabled. Drag to re-order.		
<div> <input checked="" type="checkbox"/> ulaw </div>		

Figure 13: SIP Configuration - Codecs

4.2.4 SIP Trunk using TLS

The following are the configuration settings that need to be entered to configure a SIP trunk using TLS in FreePBX (Asterisk).

1. Navigate to **Settings → Asterisk SIP Settings**
2. The following are the values that are configured in **SIP Settings [chan_pjsip]** tab,
 - a. **Certificate Manager** (Default), **SSL Method** (tlsv1_2), **Verify Client** (Yes), **Verify Server** (No) in **TLS/SSL/SRTP Settings** section
 - b. **tcp-0.0.0.0-All** and **tls-0.0.0.0-All** are set to **Yes** in **Transports** section
 - c. **Port to listen on** for **TCP** is **5062**
 - d. **Port to listen on** for **TLS** is **5067**
3. Leave the rest of the fields to default values.

Note: **Verify Server** is set to **NO** so that FreePBX (Asterisk) does not validate the Wild card certificate received from Amazon Chime SDK Voice Connector and the TLS handshake is successful.

TLS/SSL/SRTP Settings

Certificate Manager ⓘ default

SSL Method ⓘ Default

Verify Client ⓘ ☒ Yes ☐ No

Verify Server ⓘ ☒ Yes ☐ No

Transports

Note that the interface is only displayed for your information, and is not referenced by asterisk. After you enable/disable a transport, asterisk needs to be **restarted**, not just reloaded.

udp

udp - 0.0.0.0 - All ⓘ ☐ Yes ☒ No

tcp

tcp - 0.0.0.0 - All ⓘ ☒ Yes ☐ No

tls

tls - 0.0.0.0 - All ⓘ ☒ Yes ☐ No

»

Figure 14: SIP Configuration – TLS

4. Navigate to **Connectivity → Trunks →** Choose the trunk created towards Amazon Chime SDK Voice Connector
5. The following values are changed for TLS in **SIP Server** (Outbound host name from Amazon Chime SDK Voice Connector), **SIP Server Port** (5061), **Context** (from-pstn) and **Transport** (0.0.0.0-tls) in **General** Tab in **PJSIP Settings**. Leave the rest of the fields to default values.

SIP Server ⓘ gdn

SIP Server Port ⓘ 5061

Context ⓘ from-pstn

Transport ⓘ 0.0.0.0-tls

Figure 15: SIP Configuration – TLS

4.2.5 Outbound Route

1. Navigate to **Connectivity → Outbound Routes →** click on **Add Outbound Route**
2. The following are the values that are configured in **Route Name**, **Trunk Sequence for Matched Routes** (Trunk name created as per the Section 4.2.3) in **Route Settings** Tab and leave the rest of the fields to default values.

Route Settings | Dial Patterns | Import/Export Patterns | Notifications | Additional Settings

Route Name: Route_Amazon

Route CID:

Override Extension: Yes No

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?: default

Time Match Time Zone: Use System Timezone

Time Match Time Group: ---Permanent Route---

Trunk Sequence for Matched Routes: Trunk_Amazon

Submit Duplicate Reset Delete

Figure 16: Outbound Route – Route Settings

- The following are the values that are configured in **Prepend** (+1 or + for e.164 dialing), **Prefix** (Access code for route), **Match Pattern** (Dialed number to be matched) in **Dial Patterns** Tab and leave the rest of the fields to default values.

Route Settings | Dial Patterns | Import/Export Patterns | Notifications | Additional Settings

Dial Patterns that will use this Route

Pattern Help +

Dial patterns wizards

Prepend	Prefix	Match Pattern	CallerID
(prepend)	prefix	[NXXNXXXXXX]	/ CallerID] +
(+44)	7	[6XXXXXXX]	/ CallerID] +
(+1)	8	[5XXXXXXX]	/ CallerID] +
(+91)	9	[95XXXXXXX]	/ CallerID] +
(prepend)	prefix	[match pattern]	/ CallerID] +

Submit Duplicate Reset Delete

Figure 17: Outbound Route – Dial Patterns

4.2.6 Inbound route

- Navigate to **Connectivity** → **Inbound Routes** → click on **Add Inbound Route**
- The following are the values that are configured in **Description**, **DID Number**, **Set Destination** (Extensions / User created in Section 4.2.2) in **General** Tab and leave the rest of the fields to default values.

GeneralAdvancedPrivacyFaxOther

Description ⓘ

Inbound_Route_for Amazon

DID Number ⓘ

+191

CallerID Number ⓘ

ANY

CID Priority Route ⓘ

YesNo

Alert Info ⓘ

None

Ringer Volume Override ⓘ

None

CID name prefix ⓘ

Music On Hold ⓘ

Default

Set Destination ⓘ

Extensions

91

» SubmitResetDelete

Figure 18: Inbound Route