



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

November 2020

Document History

Rev. No.	Date	Description
1.0	Nov-17-2020	Draft SIP Trunk Configuration Guide
1.1	Dec-09-2020	Updated the document based on feedback
1.2	Feb-03-2021	Updated the document based on feedback

Table of Contents

1	Audience	5
1.1	Amazon Chime Voice Connector	5
2	SIP Trunking Network Components	6
2.1	Hardware Components	7
2.2	Software Requirements	7
3	Features	7
3.1	Features Supported	7
3.2	Features Not Supported	7
3.3	Features Not Tested	8
3.4	Caveats and Limitations	8
4	Configuration	9
4.1	Configuration Checklist	9
4.2	FreePBX Configuration	10
4.2.1	FreePBX Login and Version	10
4.2.2	Extension Configuration	12
4.2.3	SIP Trunk using UDP	14
4.2.4	SIP Trunk using TLS	19
4.2.5	Outbound Route	21
4.2.6	Inbound Route	23

Table of Figures

Figure 1 Network Topology.....	6
Figure 2: FreePBX Login	10
Figure 3: FreePBX Version.....	11
Figure 4: Asterisk Version	11
Figure 5 User-Extension Configuration.....	12
Figure 6 User – Extension Configuration contd.,	13
Figure 7 SIP Configuration-UDP- chan_pjsip.....	14
Figure 8 SIP Configuration-General	15
Figure 9 SIP Configuration-General	16
Figure 10 SIP Configuration- General.....	17
Figure 11 SIP Configuration-Advanced	17
Figure 12 SIP Configuration-Advanced	17
Figure 13 SIP Configuration-Codexs	18
Figure 14 SIP Configuration-TLS.....	19
Figure 15 SIP Configuration-TLS.....	20
Figure 16 Outbound Route- Route Settings	21
Figure 17 Outbound Route- Dial Patterns.....	22
Figure 18 Inbound Route	23

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 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 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 **FreePBX (Asterisk)** with **Amazon Chime Voice Connector**

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

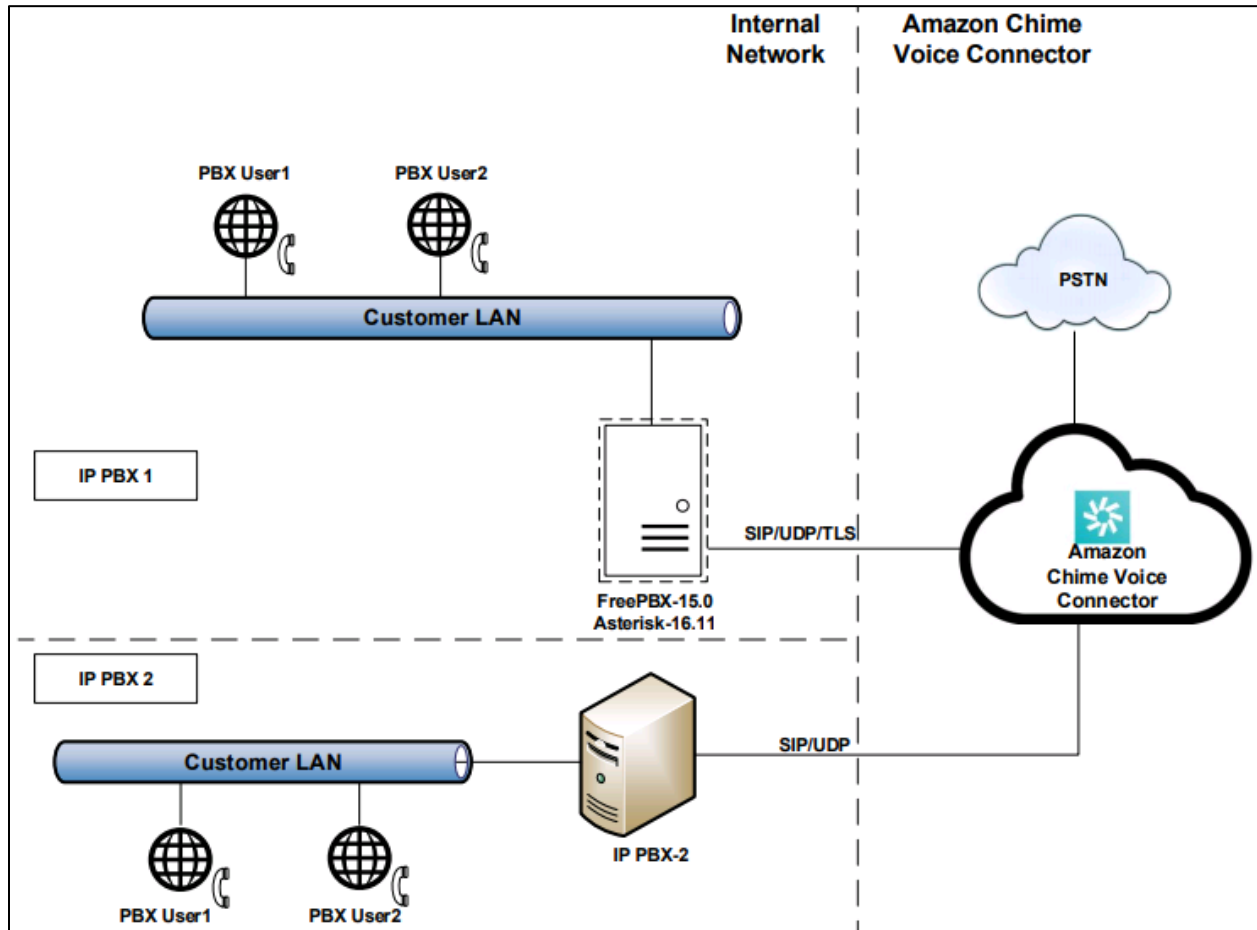


Figure 1 Network Topology

2.1 Hardware Components

- FreePBX (Asterisk) hosted on UCS-C240 VMWare running on ESXi-6.7.0

2.2 Software Requirements

- FreePBX – 15.0.16.72 & Asterisk – 16.11.1

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 messages received from customer equipment, but does not send OPTIONS messages to customer equipment
- Keep Alive – Double CRLF are not supported by Amazon Chime Voice Connector and FreePBX

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
 - does not acknowledge for SIP OPTIONS from FreePBX when the REQ_URI format is in "user@fqdn" and it causes outbound calls using call authentication to fail
- FreePBX does not support wild card certificate sent by Amazon Chime Voice Connector so disabled the Server/Peer certificate verification in FreePBX for a successful secured inbound and outbound calling
- FreePBX does not send Session Refresh for long duration calls. Kept the call active for one hour and verified the call stays connected
- Inbound call is rejected when FreePBX is configured with the IP address of Amazon Chime Voice Connector instead of FQDN

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 **FreePBX** for SIP Trunking with **Amazon Chime Voice Connector**

Table 1 – PBX Configuration Steps

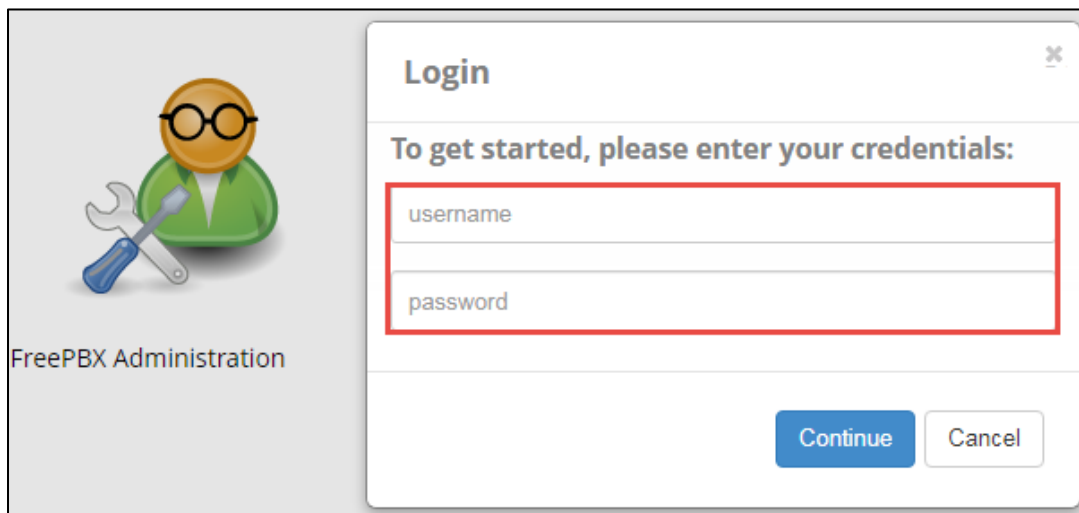
Steps	Description	Reference
Step 1	FreePBX Configuration	Section 4.2
Step 2	Amazon Chime Voice Connector Configuration	Amazon Chime Voice Connector

4.2 FreePBX Configuration

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

4.2.1 FreePBX Login and Version

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



The screenshot shows the FreePBX Administration login interface. On the left, there is a logo featuring a green figure with glasses and a wrench and screwdriver, with the text "FreePBX Administration" below it. On the right, a white login dialog box is displayed. The dialog box has a title "Login" and a close button (X) in the top right corner. Below the title, it says "To get started, please enter your credentials:". There are two input fields: "username" and "password", both highlighted with a red border. At the bottom right of the dialog box, there are two buttons: "Continue" (blue) and "Cancel" (white).

Figure 2: FreePBX Login

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

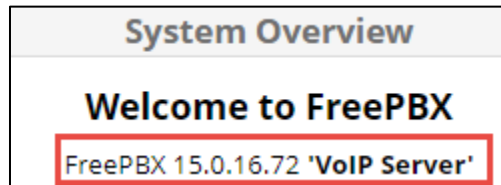


Figure 3: FreePBX Version

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

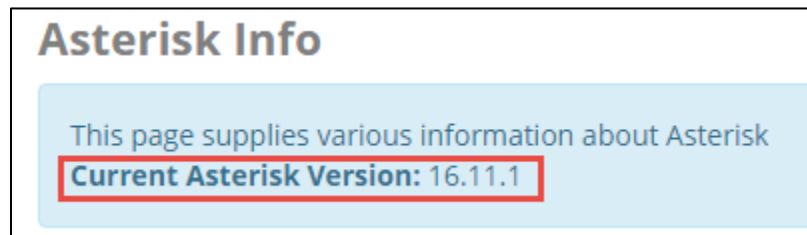


Figure 4: Asterisk Version

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 values

General Voicemail Find Me/Follow Me Advanced Pin Sets Other

— Edit Extension

This device uses **PJSIP** technology listening on Port 5060 (UDP)

Display Name ⓘ

919

Outbound CID ⓘ

+1919

Emergency CID ⓘ

Secret ⓘ

Figure 5 User-Extension Configuration

— User Manager Settings

Linked to User 919

Select User Directory: ⓘ

PBX Internal Directory

Link to a Different Default User: ⓘ

919 (Linked)

Username ⓘ

Use Custom Username

Password For New User ⓘ

Groups ⓘ

All Users

Figure 6 User – Extension Configuration contd.,

4.2.3 SIP Trunk using UDP

1. Navigate to **Settings > Asterisk SIP Settings Routes**
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

The screenshot displays the 'Transports' configuration page for Asterisk SIP Settings [chan_pjsip]. It features a list of transport protocols with toggle switches for enabling or disabling them. The 'udp' section is highlighted with a red box, showing 'udp - 0.0.0.0 - All' set to 'Yes'. Below it, 'tcp', 'tls', 'ws', and 'wss' are all set to 'No'. At the bottom, the '0.0.0.0 (udp)' section is also highlighted with a red box, showing the 'Port to Listen On' set to '5060'. A note at the top states: 'Note that the interface is only displayed for your information, and is not referenced by asterisk. just reloaded.'

Transport	Enabled
udp - 0.0.0.0 - All	Yes
tcp - 0.0.0.0 - All	No
tls - 0.0.0.0 - All	No
ws - 0.0.0.0 - All	No
wss - 0.0.0.0 - All	No
0.0.0.0 (udp)	Port to Listen On: 5060

Figure 7 SIP Configuration-UDP- chan_pjsip

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

The screenshot displays the 'General' configuration tab for a SIP trunk. The 'Trunk Name' field is highlighted with a red box and contains the text 'Trunk_to_Amazon_VC'. Below it, the 'Hide CallerID' field is set to 'No'. The 'Outbound CallerID' field is empty. The 'CID Options' section is highlighted with a red box and shows four buttons: 'Allow Any CID' (selected), 'Block Foreign CIDs', 'Remove CNAM', and 'Force Trunk CID'. The 'Maximum Channels' field is also highlighted with a red box and contains the value '10'.

Figure 8 SIP Configuration-General

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

The screenshot displays the AWS IAM console interface for configuring a user's pjsip settings. At the top, there are three tabs: 'General', 'Dialed Number Manipulation Rules', and 'pjsip Settings', with the latter being the active tab. Below the tabs, the 'PJSIP Settings' section is visible, containing sub-tabs for 'General', 'Advanced', and 'Codecs'. The 'General' sub-tab is selected. The configuration fields include: 'Username' (Authentication Disabled), 'Secret' (Authentication Disabled), 'Authentication' (Outbound, Inbound, Both, None - 'None' is selected), 'Registration' (Send, Receive, None - 'None' is selected), 'Language Code' (Default), and 'SIP Server' (cr7c). The 'SIP Server' field is highlighted with a red box.

Figure 9 SIP Configuration-General

SIP Server Port ⓘ

5060

Context ⓘ

from-pstn

Transport ⓘ

0.0.0.0-udp

Figure 10 SIP Configuration- General

- The following are the values that are configured in **Qualify Frequency** (60) for sending SIP OPTIONS, **From Domain** (Outbound host name from Amazon Chime 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

Qualify Frequency ⓘ

60 Seconds

Outbound Proxy ⓘ

User = Phone ⓘ

Contact User ⓘ

From Domain ⓘ

cr7c

Figure 11 SIP Configuration-Advanced

Send RPID/PAI ⓘ

Figure 12 SIP Configuration-Advanced

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

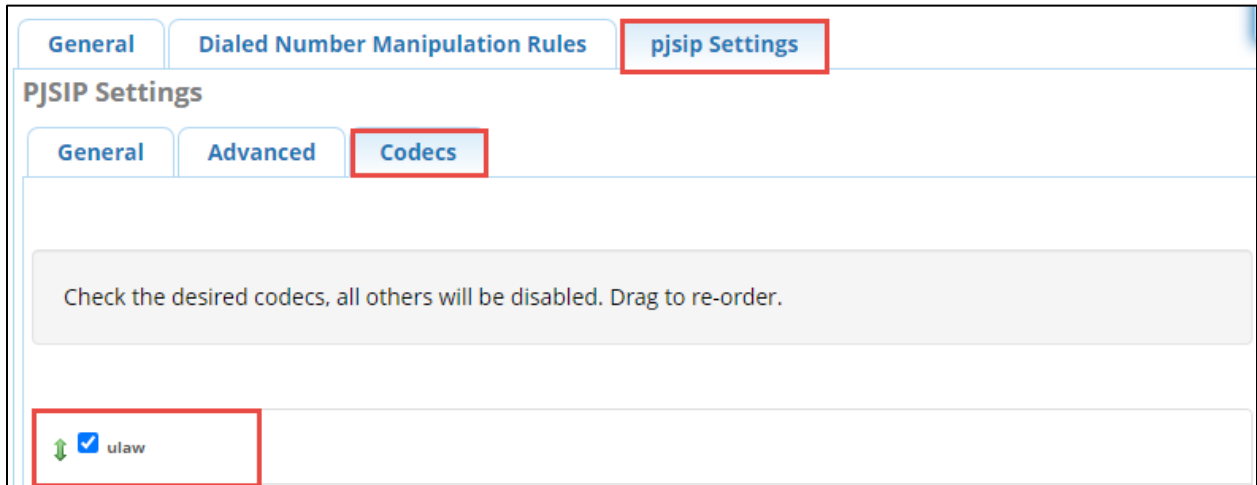


Figure 13 SIP Configuration-Codecs

4.2.4 SIP Trunk using TLS

The following are the configuration that needs to be performed to configure SIP trunk using TLS in FreePBX

1. Navigate to **Settings > Asterisk SIP Settings Routes**
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

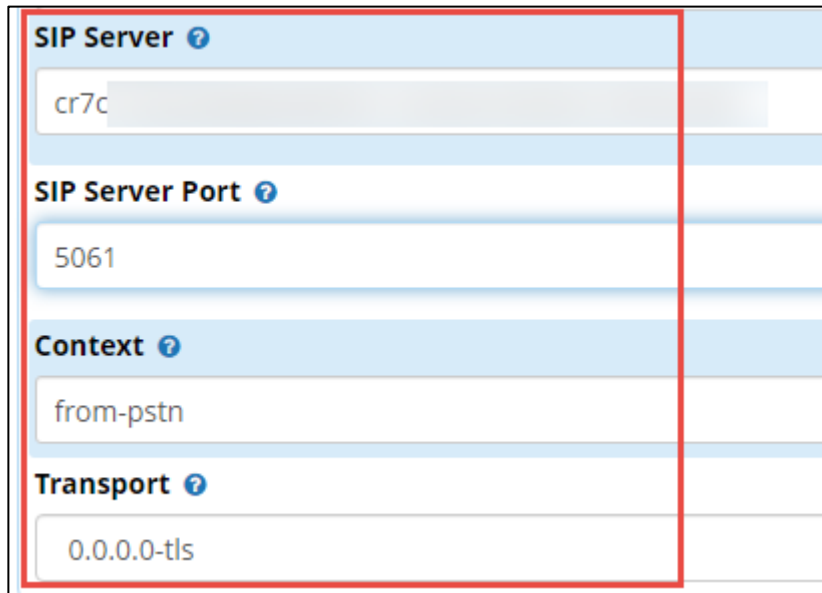
Note: **Verify Server** is set to **NO** so that FreePBX does not validate the Wild card certificate received from Amazon Chime Voice Connector and the TLS handshake is successful.

The screenshot displays the configuration interface for SIP Settings [chan_pjsip]. It is divided into two main sections: 'TLS/SSL/SRTP Settings' and 'Transports'. The 'TLS/SSL/SRTP Settings' section includes fields for 'Certificate Manager' (set to 'default'), 'SSL Method' (set to 'tlsv1_2'), 'Verify Client' (set to 'Yes'), and 'Verify Server' (set to 'No'). The 'Transports' section contains a note and three transport settings: 'udp - 0.0.0.0 - All' (set to 'No'), 'tcp - 0.0.0.0 - All' (set to 'Yes'), and 'tls - 0.0.0.0 - All' (set to 'Yes'). Red boxes highlight the 'TLS/SSL/SRTP Settings' section, the 'Transports' section, and the 'tcp' and 'tls' transport settings.

Figure 14 SIP Configuration-TLS

4. Navigate to **Connectivity > Trunks** > Choose the trunk created towards Amazon Chime Voice Connector

5. The following values are changed for TLS in **SIP Server** (Outbound host name from Amazon Chime Voice Connector), **SIP Server Port** (5061), **Context**, **Transport** (0.0.0.0-tls) in **General** Tab in **pjsip Settings**. Leave the rest of the fields to default values

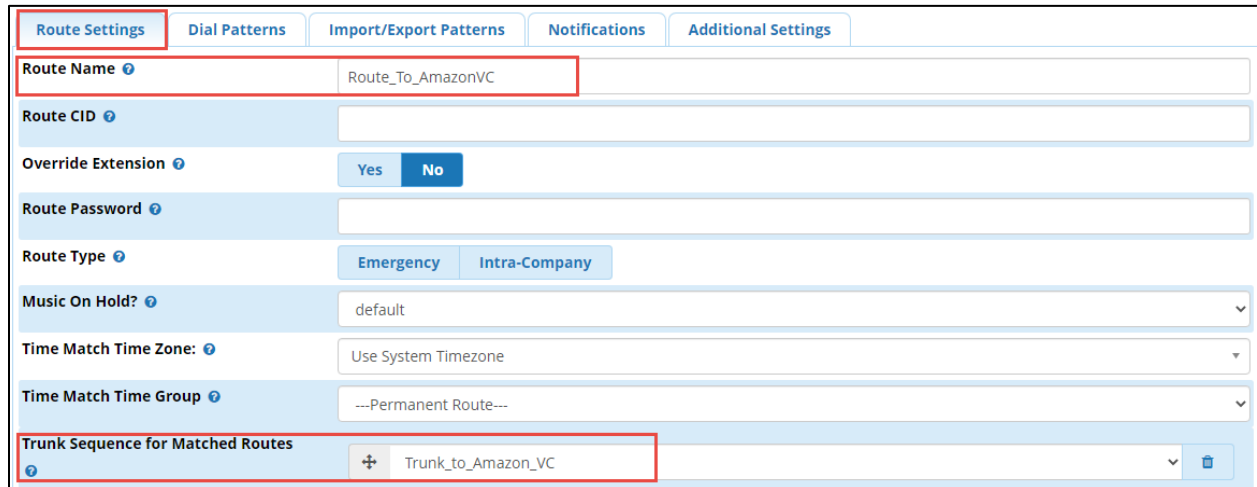


SIP Server ⓘ	cr7c
SIP Server Port ⓘ	5061
Context ⓘ	from-pstn
Transport ⓘ	0.0.0.0-tls

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 previous steps) in **Route Settings** Tab and leave the rest of the fields to default values



The screenshot shows the 'Route Settings' tab for an Outbound Route. The 'Route Name' field is highlighted with a red box and contains the text 'Route_To_AmazonVC'. The 'Trunk Sequence for Matched Routes' field is also highlighted with a red box and contains the text 'Trunk_to_Amazon_VC'. Other fields are at their default values: 'Route CID' is empty, 'Override Extension' is set to 'No', 'Route Password' is empty, 'Route Type' is set to 'Emergency', 'Music On Hold?' is set to 'default', 'Time Match Time Zone:' is set to 'Use System Timezone', and 'Time Match Time Group' is set to '---Permanent Route---'.

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

The screenshot shows the 'Dial Patterns' configuration page for an outbound route. The 'Dial Patterns' tab is selected and highlighted with a red box. Below the tabs, there is a section titled 'Dial Patterns that will use this Route'. A 'Pattern Help' link is visible. A blue bar labeled 'Dial patterns wizards' is present. Below this, three dial patterns are listed, each with a red box around its fields:

Prepend	Prefix	Match Pattern	Actions
(+)	9	[31XXXXXXXXXX /	+ 🗑️
(+1)	9	[900XXXXXXXX /	+ 🗑️
(+1)	9	[214XXXXXXXX /	+ 🗑️

Figure 17 Outbound Route- Dial Patterns

4.2.6 Inbound Route

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

General	Advanced	Privacy	Fax	Other
Description ?	InbountRouteforAmazonVC			
DID Number ?	+1919			
CallerID Number ?	ANY			
CID Priority Route ?	<input type="radio"/> Yes <input checked="" type="radio"/> No			
Alert Info ?	None			
Ringer Volume Override ?	None			
CID name prefix ?				
Music On Hold ?	Default			
Set Destination ?	Extensions			
	919			

Figure 18 Inbound Route