



Amazon Chime Voice Connector

SIP Trunking Configuration Guide:

Microsoft Skype for Business 2015 and AudioCodes Mediant Cloud Edition

October 2019

Document History

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Table of Contents

1	Audience.....	5
1.1	Amazon Chime Voice Connector.....	6
2	SIP Trunking Network Components.....	7
2.1	Hardware Components.....	8
2.2	Software Requirements.....	8
3	Features	8
3.1	Features Supported	8
3.2	Features Not Supported.....	9
3.3	Features Not Tested.....	9
3.4	Caveats and Limitations.....	9
4	Configuration	10
4.1	Configuration Checklist	10
4.2	Skype for Business Configuration.....	10
4.2.1	PSTN gateway configuration	10
4.2.2	Voice Routing Configuration.....	14
4.2.3	Trunk Configuration	19
4.2.4	User Configuration	23
4.3	AudioCodes CE Configuration.....	24
4.3.1	Network IP Interface configuration	24
4.3.2	Media Realm configuration	24
4.3.3	SRD configuration	26
4.3.4	SIP Interface configuration.....	26
4.3.5	Proxy Sets configuration.....	29
4.3.6	IP Group Table configuration	32
4.3.7	Coder Groups configuration.....	35
4.3.8	IP Profile configuration	36
4.3.9	IP-to-IP Routing	42
4.3.10	TLS Configuration.....	44
4.3.11	Message Manipulation configuration.....	47

Table of Figures

Figure 1:	Network Topology.....	7
-----------	-----------------------	---

Figure 2: Add new IP/PSTN Gateway	11
Figure 3: Enter the PSTN Gateway FQDN.....	11
Figure 4: Define the IP address.....	12
Figure 5: Define the root trunk.....	12
Figure 6: PSTN Gateways List.....	13
Figure 7: Publish the Topology	13
Figure 8: Successfully Published Topology	14
Figure 9: Add New User Voice Policy.....	14
Figure 10: New Voice Policy.....	15
Figure 11: New PSTN Usage Record.....	16
Figure 12: Creating New Voice Route.....	16
Figure 13: Create Route continuation.....	16
Figure 14: Trunk Association in Route.....	17
Figure 15: Commit all save the changes	17
Figure 16: Uncommitted Voice Configuration Settings	18
Figure 17: Successfully published voice routing changes.....	18
Figure 18: New Pool Trunk.....	19
Figure 19: Select a Service	19
Figure 20: Trunk Configuration	20
Figure 21: Trunk Configuration Continuation.....	21
Figure 22: Trunk Configuration Continuation.....	21
Figure 23: Uncommitted Trunk Configuration settings.....	22
Figure 24: Trunk Configuration after committed changes.....	22
Figure 25: User Configuration.....	23
Figure 26: IP Interfaces.....	24
Figure 27: Media Realms Table	24
Figure 28: Media Realm for Skype for Business	25
Figure 29: Media Realm for Amazon Chime Voice Connector	25
Figure 30: Default SRD.....	26
Figure 31: SRD Table Details	26
Figure 32: SIP Interfaces.....	27
Figure 33: SIP Interface for Skype for Business.....	27
Figure 34: SIP Interface for Skype for Business Continuation.....	28
Figure 35: SIP Interface for Amazon Chime Voice Connector.....	28
Figure 36: SIP Interface for Amazon Chime Voice Connector Continuation.....	29
Figure 37: Proxy Sets table.....	29
Figure 38: Proxy Set table for Skype for Business	30
Figure 39: Proxy Address for Skype for Business	31
Figure 40: Proxy Set table for Amazon Chime Voice Connector	31
Figure 41: Proxy Address for Amazon Chime Voice Connector	32
Figure 42: IP Group Table	32
Figure 43: IP Group Table for Skype for Business.....	32
Figure 44: IP Group table for Skype for Business Continuation	33

Figure 45: IP Group table for Skype for Business Continuation	33
Figure 46: IP Group table for Amazon Chime Voice Connector	34
Figure 47: IP Group table for Amazon Chime Voice Connector Continuation	34
Figure 48: IP Group table for Amazon Chime Voice Connector Continuation	35
Figure 49: Coder Groups.....	35
Figure 50: Allowed Audio Coders	36
Figure 51: IP Profiles	36
Figure 52: IP Profile for Skype for Business	37
Figure 53: IP Profile for Skype for Business Continuation	37
Figure 54: IP Profile for Skype for Business Continuation	38
Figure 55: IP Profile for Skype for Business Continuation	38
Figure 56: IP Profile for Skype for Business Continuation	39
Figure 57: IP Profile for Skype for Business Continuation	39
Figure 58: IP Profile for Amazon Chime Voice Connector	40
Figure 59: IP Profile for Amazon Chime Voice Connector Continuation	40
Figure 60: IP Profile for Amazon Chime Voice Connector Continuation	41
Figure 61: IP Profile for Amazon Chime Voice Connector Continuation	41
Figure 62: IP Profile for Amazon Chime Voice Connector Continuation	42
Figure 63: IP-to-IP Routing.....	42
Figure 64: IP-to-IP Routing for OPTIONS.....	43
Figure 65: IP-to-IP Routing from Skype for Business to Amazon Chime Voice Connector.....	43
Figure 66: IP-to-IP Routing from Amazon Chime Voice Connector to Skype for Business.....	44
Figure 67: TLS Context list.....	44
Figure 68: TLS Context for Amazon Chime Voice Connector.....	45
Figure 69: Trusted Root Certificate Import option.....	45
Figure 70: Media Security	46
Figure 71: SRTP option in IP Profile.....	46
Figure 72: From header modification Skype for Business	47
Figure 73: From header Modification Amazon Chime Voice Connector	47
Figure 74: OPTIONS RURI modification	48
Figure 75: OPTIONS To header modification	48
Figure 76: PAI header modification	48
Figure 77: Add Allow header-UPDATE	49
Figure 78: Session Expires Timer modification	49

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 trunks using **Microsoft Skype for Business 2015 (Skype for Business)** and **AudioCodes Mediant Cloud Edition (AudioCodes CE)** to connect

to **Amazon Chime Voice Connector** for inbound and/or outbound telephony capabilities.

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 the SIP trunk reference configuration is illustrated below and is representative of Microsoft Skype for Business 2015 and AudioCodes CE configuration.

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

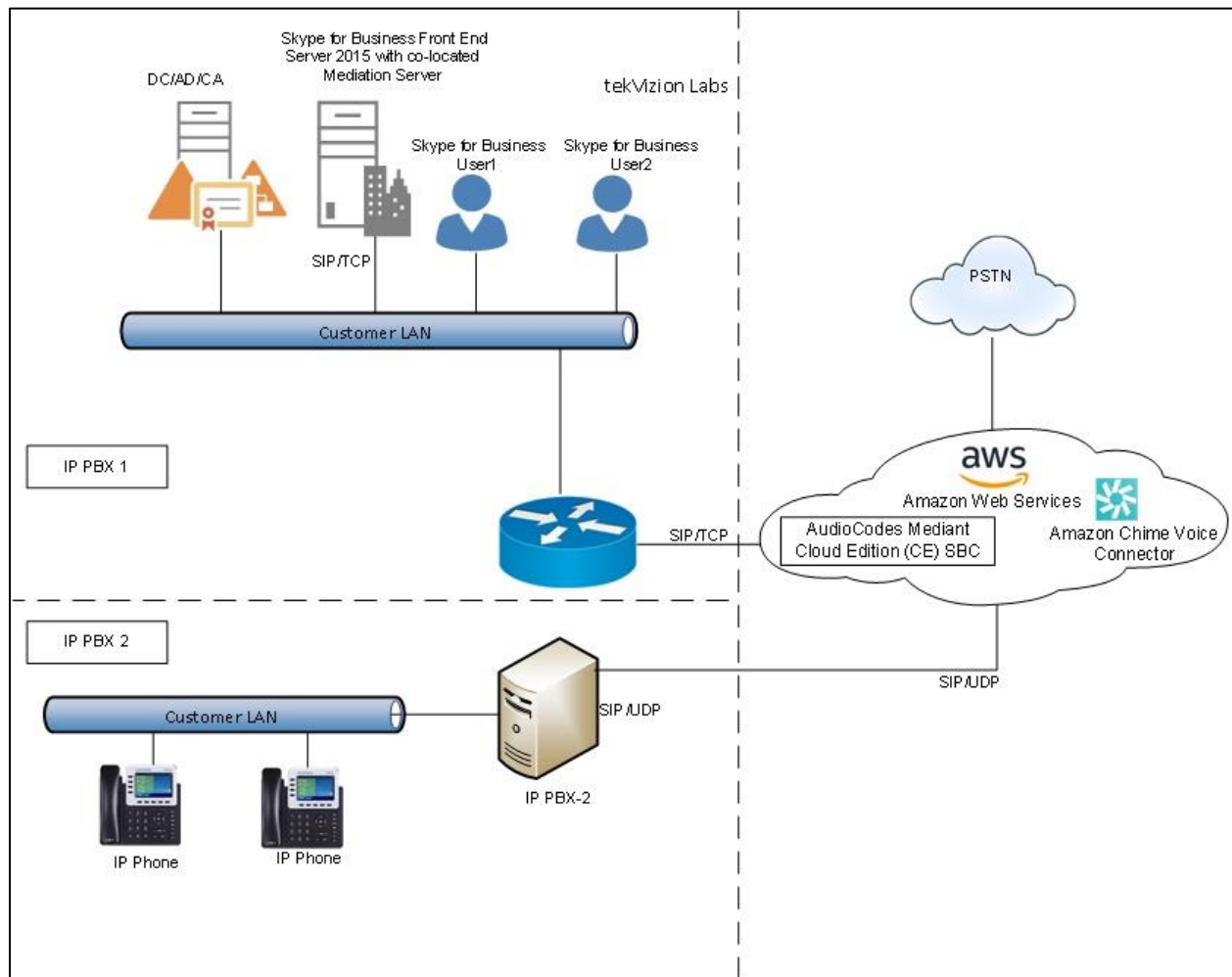


Figure 1: Network Topology

2.1 Hardware Components

- Windows 10 hosting Microsoft Skype for Business Server 2015
- AudioCodes CE running on Amazon Web Service

2.2 Software Requirements

- Microsoft Skype for Business Server 2015: 6.0.9319.562
- AudioCodes Mediant Cloud Edition: 7.20A.252.274
- Skype for Business Client: 15.0.5159.1000

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
- Call Authentication
- Anonymous call
- DTMF-RFC 2833
- Long duration calls
- Calls to conference scheduled by Amazon Chime user
- Calls to Amazon Chime Business number
- Call Distribution
- Call Failover

3.2 Features Not Supported

Amazon Chime Voice Connector does not support following features

- Keep Alive SIP OPTIONS
- Keep Alive – Double CRLF

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Amazon Chime Voice Connector does not send 'Allow' header for any request or response. Skype for Business requires 'UPDATE' method in 'Allow' header to send session refresh. SIP message manipulation is created in AudioCodes CE to add 'Allow' header with 'UPDATE' method for all request and response from AudioCodes CE towards Skype for Business.
- 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

4.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure **Skype for Business** and **AudioCodes CE** for SIP Trunking with **Amazon Chime Voice Connector**.

Table 1 – PBX Configuration Steps

Steps	Description	Reference
Step 1	Skype for Business Configuration	Section 4.2
Step 2	AudioCodes CE Configuration	Section 4.3

4.2 Skype for Business Configuration

This section with screen shots taken from Skype for Business used for the interoperability testing, gives a general overview of the PBX configuration.

4.2.1 PSTN gateway configuration

Open Skype for Business Server 2015 Topology Builder to create a trunk from Skype for Business to AudioCodes CE. Navigate to '*Shared Components*' and select '*PSTN gateways*' in Topology Builder window. Right click on '*PSTN gateways*' and select '*New IP/PSTN Gateway*'.

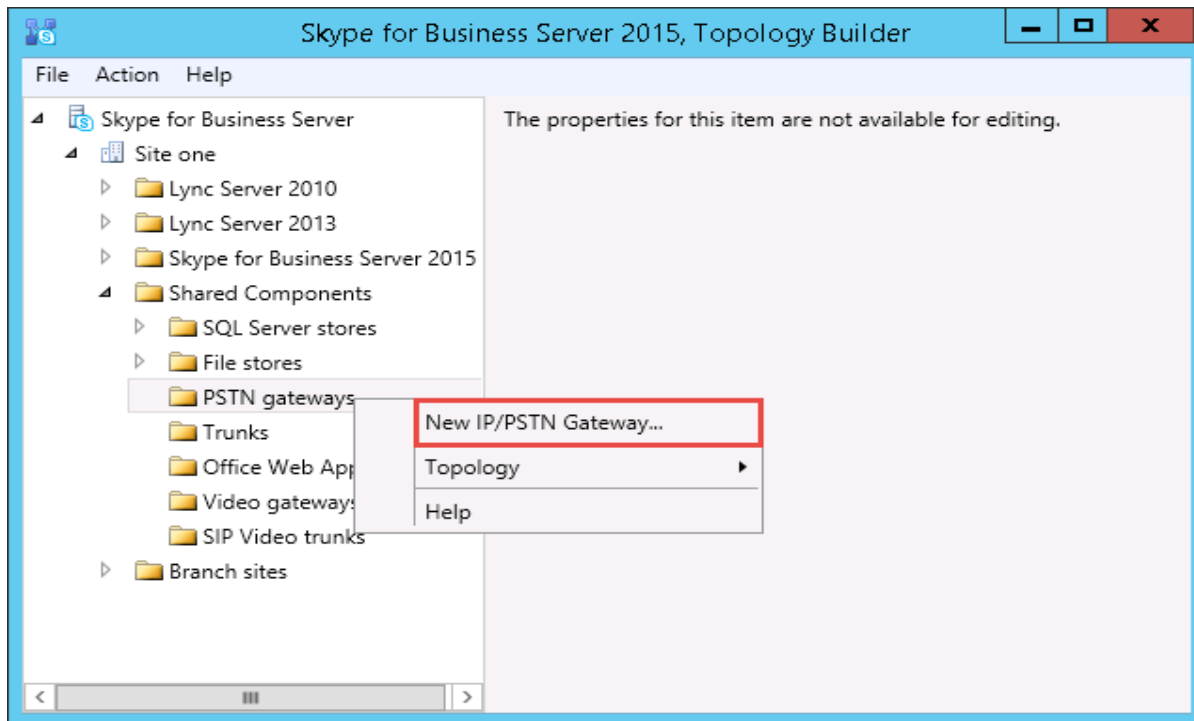


Figure 2: Add new IP/PSTN Gateway

The IP address or FQDN of AudioCodes Mediant CE is configured in 'Define New IP/PSTN Gateway' window.

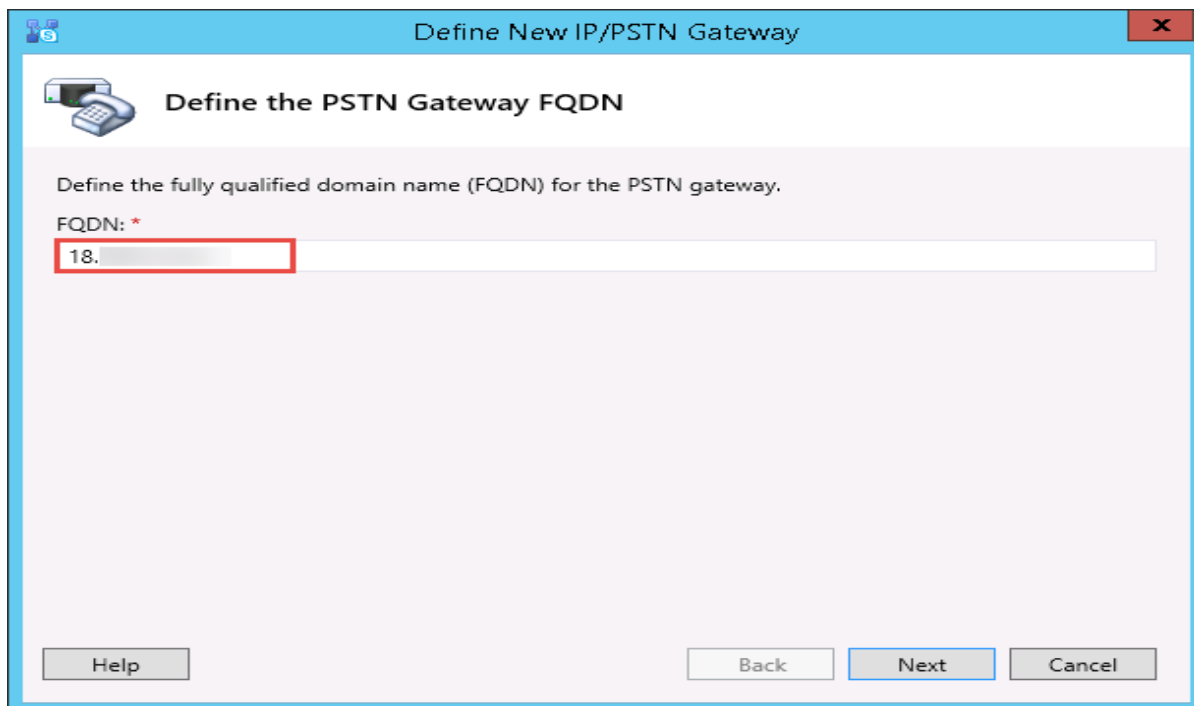


Figure 3: Enter the PSTN Gateway FQDN

Click on 'Next' and select 'Enable IPv4' and 'Use all configured IP addresses'.

The screenshot shows a window titled "Define New IP/PSTN Gateway" with a sub-header "Define the IP address". It contains two main sections for IPv4 and IPv6. In the IPv4 section, the "Enable IPv4" radio button is selected, and the "Use all configured IP addresses" option is chosen. The "PSTN IP address" field is empty. The IPv6 section is unselected. At the bottom, there are "Help", "Back", "Next", and "Cancel" buttons. The "Next" button is highlighted in blue.

Define New IP/PSTN Gateway

Define the IP address

☒ Enable IPv4

☒ Use all configured IP addresses.

☐ Limit service usage to selected IP addresses.

PSTN IP address:

☐ Enable IPv6

☒ Use all configured IP addresses.

☐ Limit service usage to selected IP addresses.

PSTN IP address:

Help Back Next Cancel

Figure 4: Define the IP address

Click on 'Next'. Configure the port number, transport protocol and associated Mediation Server.

The screenshot shows the same window, now at the "Define the root trunk" step. It contains several fields: "Trunk name:" with the value "18.", "Listening port for IP/PSTN gateway:" with the value "5060", "SIP Transport Protocol:" with a dropdown menu set to "TCP", "Associated Mediation Server:" with a dropdown menu set to "FE01.sfbblabre.local Site one", and "Associated Mediation Server port:" with the value "5060". At the bottom, there are "Help", "Back", "Finish", and "Cancel" buttons.

Define New IP/PSTN Gateway

Define the root trunk

Trunk name: *

18.

Listening port for IP/PSTN gateway: *

5060

SIP Transport Protocol:

TCP

Associated Mediation Server:

FE01.sfbblabre.local Site one

Associated Mediation Server port: *

5060

Help Back Finish Cancel

Figure 5: Define the root trunk

Click on 'Finish'. The newly created trunk will appear under the PSTN gateways with the associated Mediation Server.

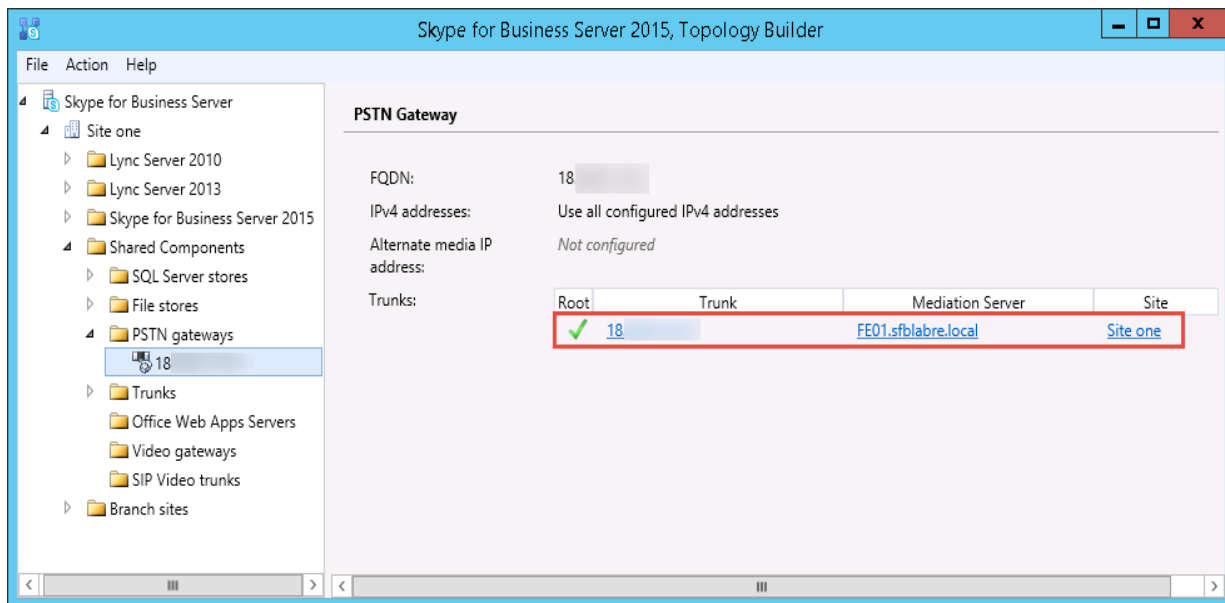


Figure 6: PSTN Gateways List

Navigate to menu 'Action' and select 'Topology', and in the submenu select 'Publish' to publish the topology.

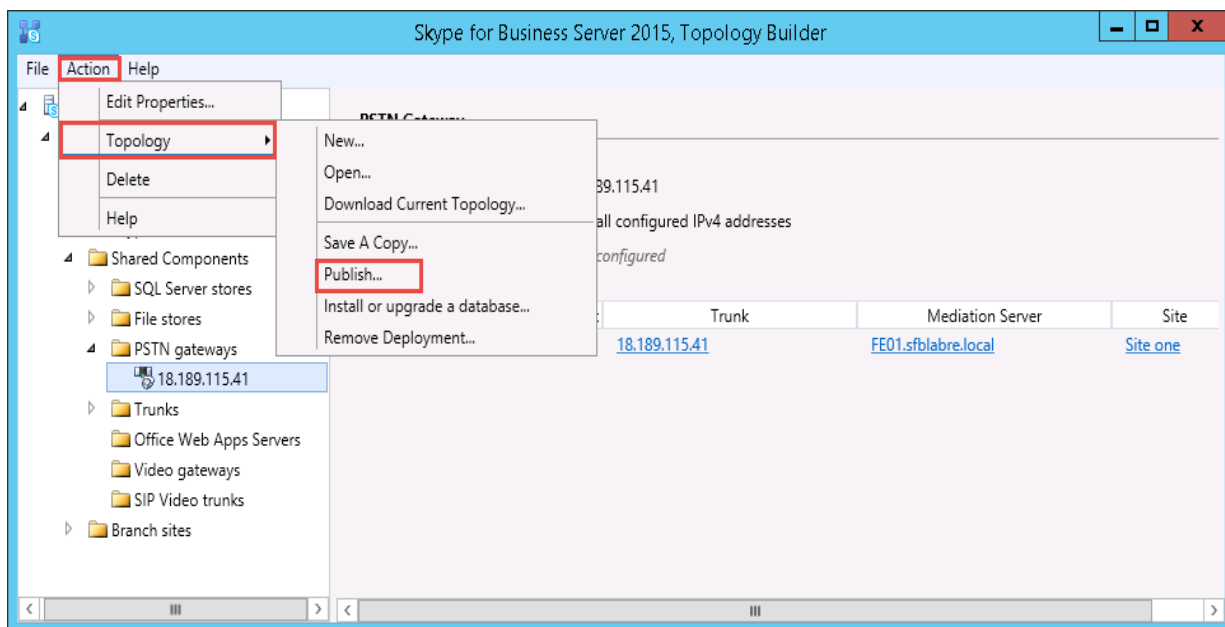


Figure 7: Publish the Topology

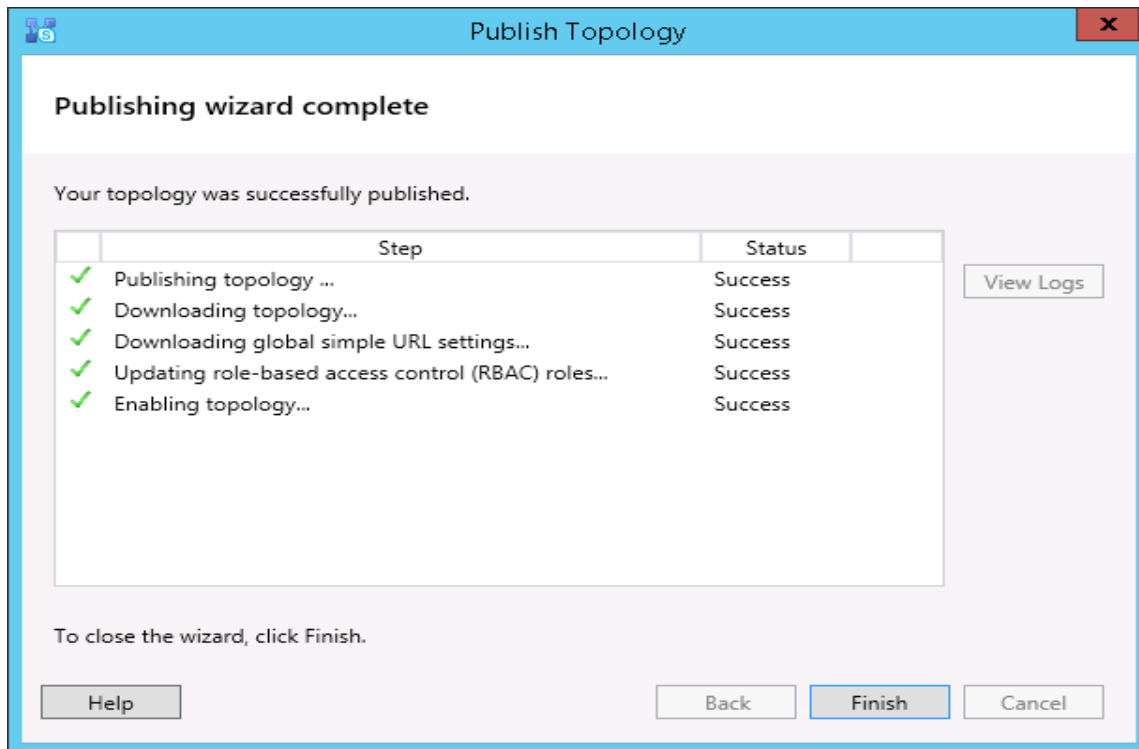


Figure 8: Successfully Published Topology

4.2.2 Voice Routing Configuration

Open Skype for Business Server 2015 Control Panel and click on 'Voice Routing' in left pane. Now navigate to 'Voice Policy' and click on 'New' and select 'User Policy' to add the new Voice Policy as shown below

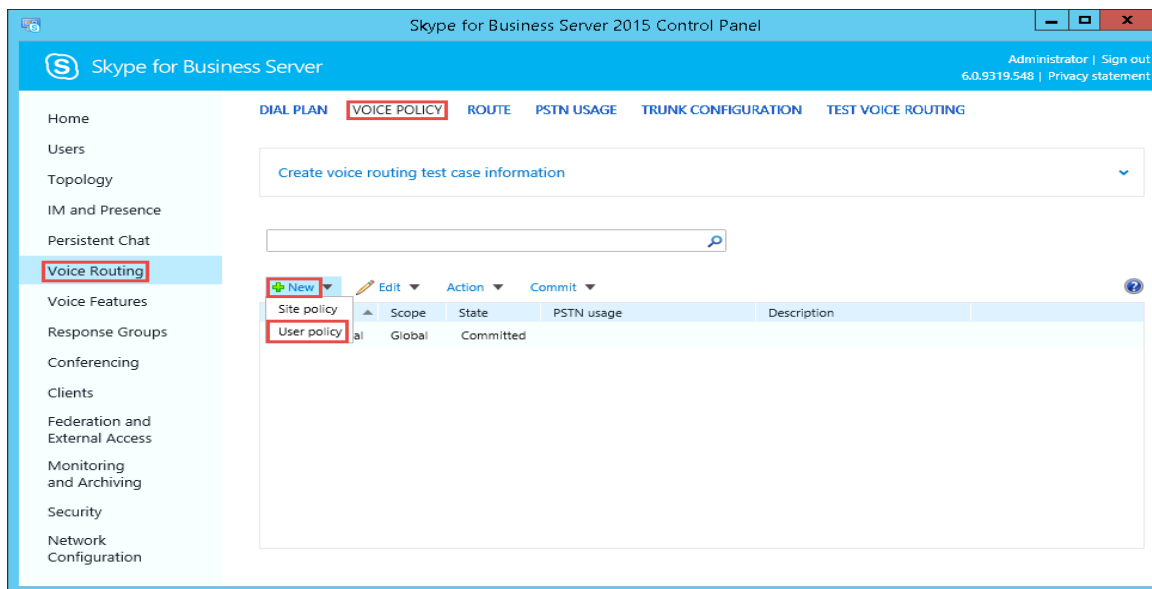


Figure 9: Add New User Voice Policy

Enter the name and description for the New Voice Policy and select the Calling Features. Click on 'New' under 'Associated PSTN Usages' to create a new PSTN Usage.

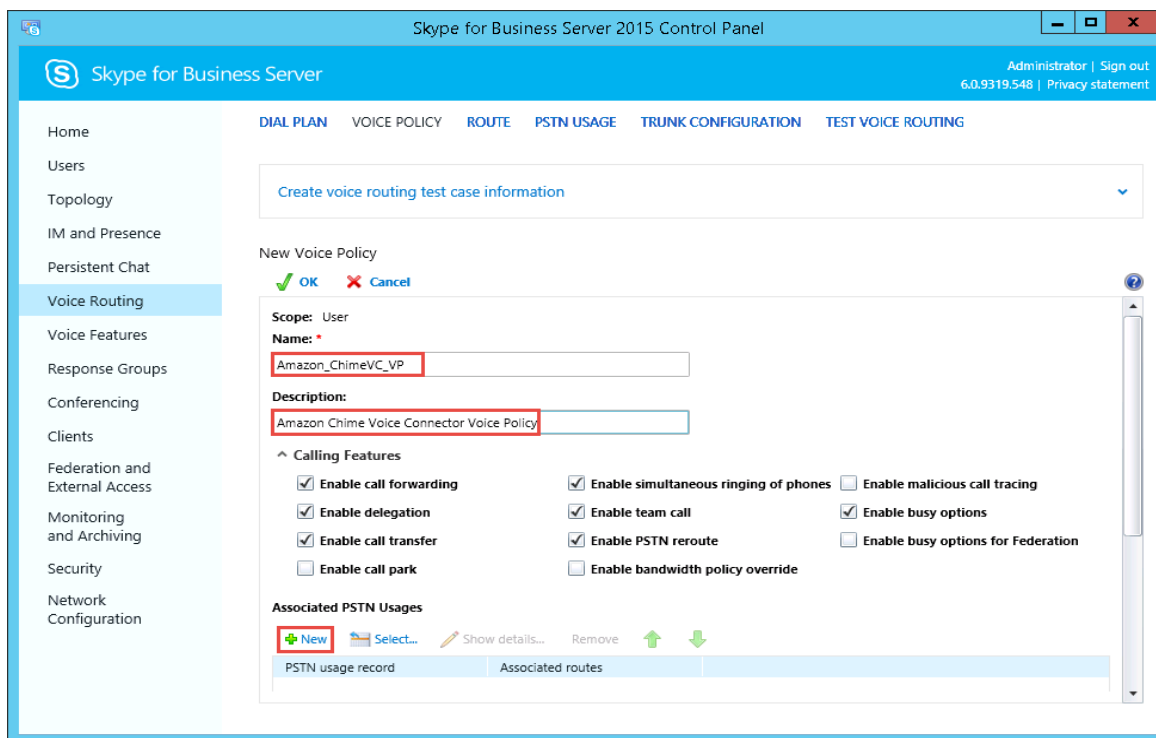


Figure 10: New Voice Policy

Enter the name of the PSTN Usage and click on 'New' under 'Associated Routes' to create a new Route.

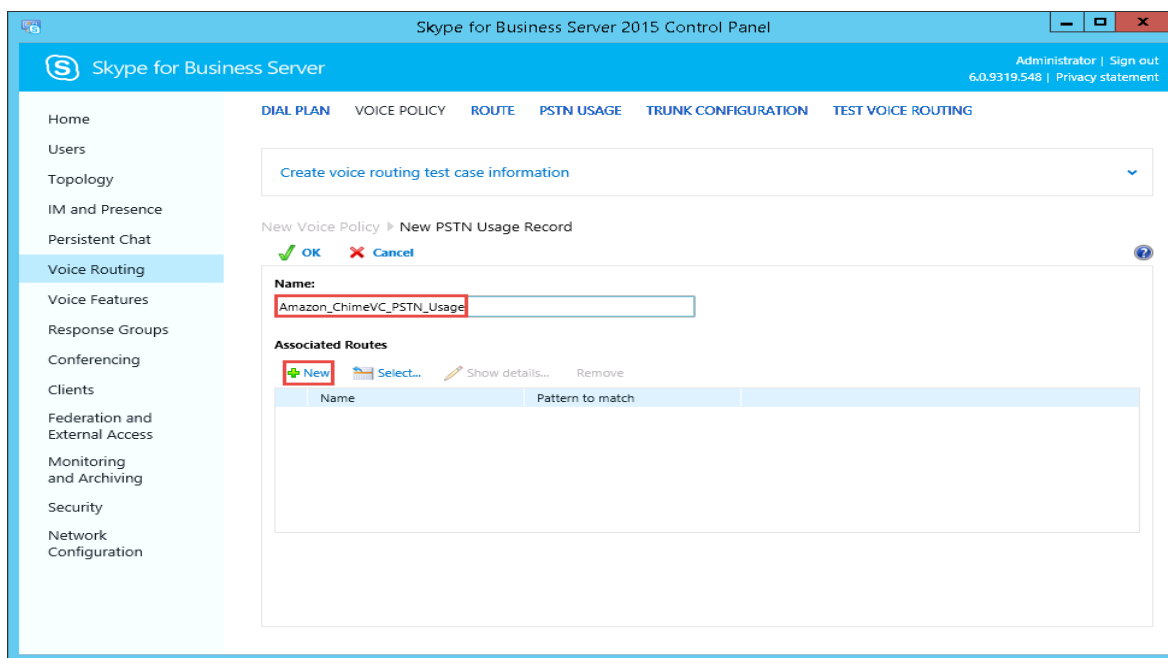
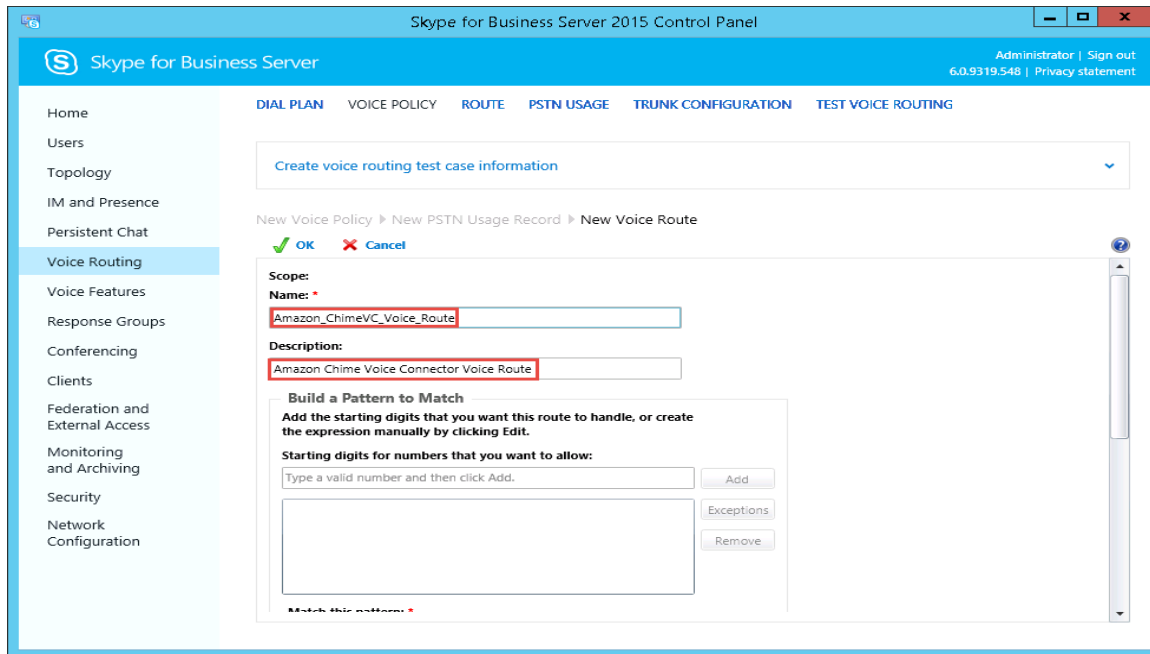


Figure 11: New PSTN Usage Record

Enter the name and the description for the new voice route.



Skype for Business Server 2015 Control Panel

Administrator | Sign out
6.0.9319.548 | Privacy statement

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

New Voice Policy > New PSTN Usage Record > New Voice Route

OK Cancel

Scope:

Name: *
Amazon_ChimeVC_Voice_Route

Description:
Amazon Chime Voice Connector Voice Route

Build a Pattern to Match
Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit.

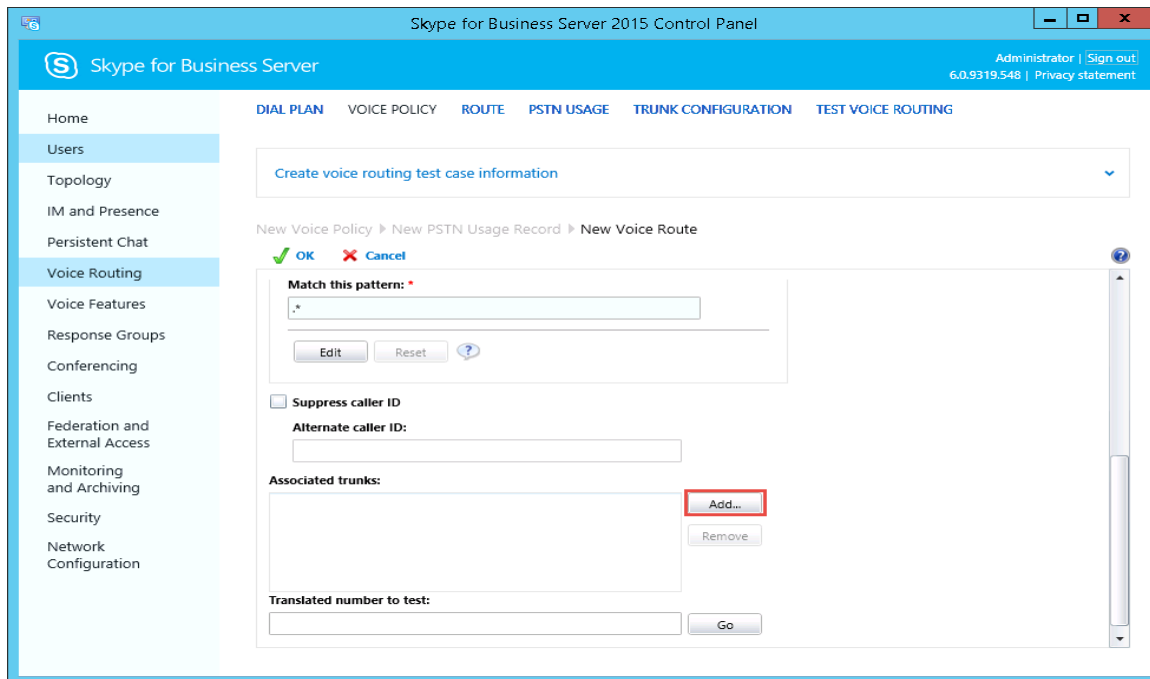
Starting digits for numbers that you want to allow:
Type a valid number and then click Add.

Add Exceptions Remove

Match this pattern: *

Figure 12: Creating New Voice Route

The pattern to match is set as `.*` which matches any dialed number from this voice policy. In the Associated Trunks section, select `Add` to choose the trunk towards AudioCodes CE.



Skype for Business Server 2015 Control Panel

Administrator | Sign out
6.0.9319.548 | Privacy statement

DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING

Create voice routing test case information

New Voice Policy > New PSTN Usage Record > New Voice Route

OK Cancel

Match this pattern: *
.*

Edit Reset ?

☐ Suppress caller ID

Alternate caller ID:

Associated trunks:

Add... Remove

Translated number to test:

Go

Figure 13: Create Route continuation

In the list of Trunks, select the appropriate trunk and click OK. Click OK in 'New Voice Route' window and click OK in 'New PSTN Usage Record' window. Click OK in 'New Voice Policy' window.

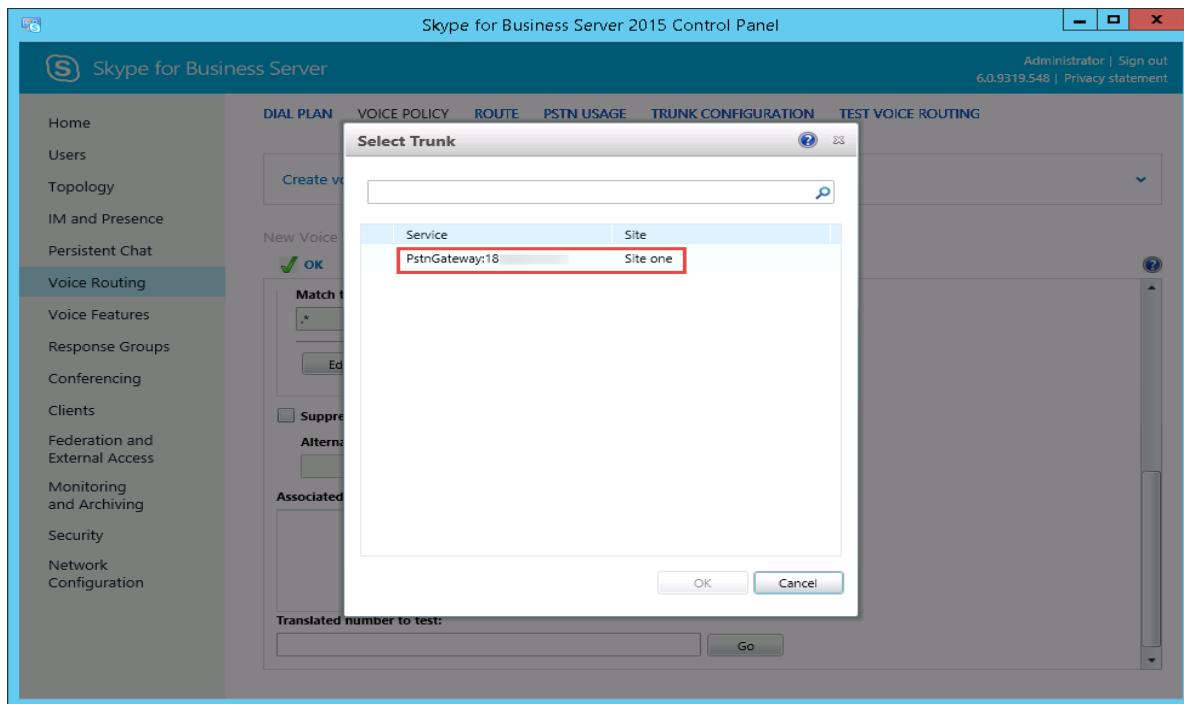


Figure 14: Trunk Association in Route

Click on 'Commit' drop down and select 'Commit all' to save the changes made.

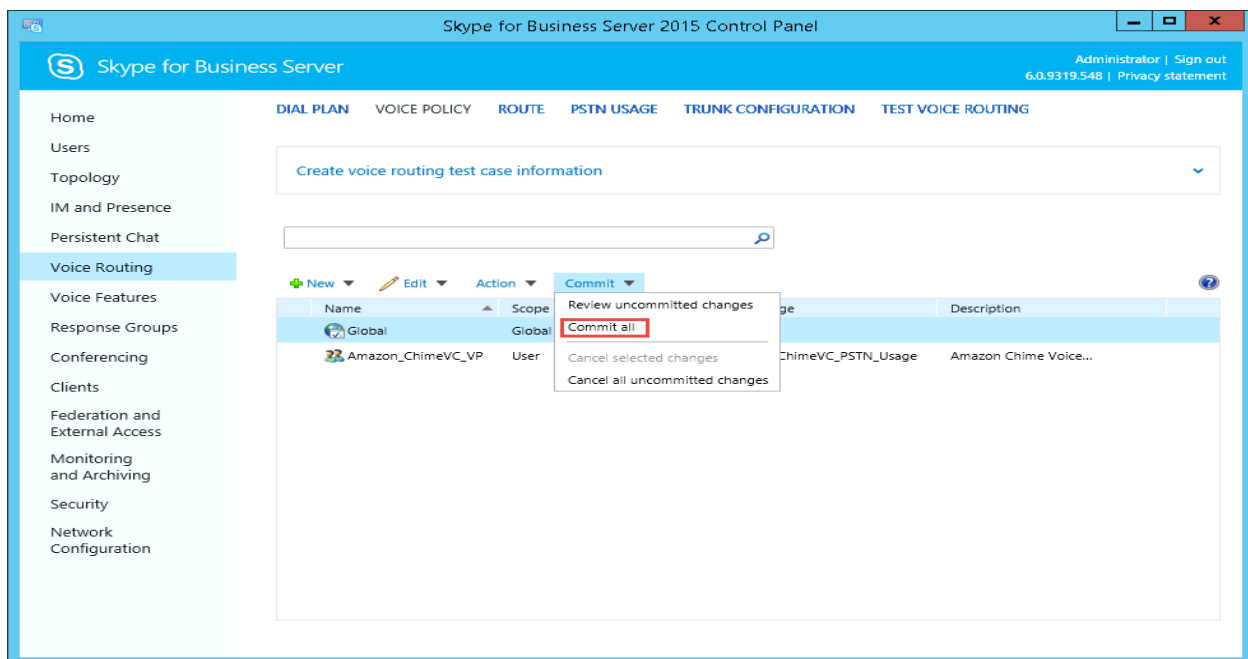


Figure 15: Commit all save the changes

All the changes made is displayed in the Uncommitted Voice Configuration Settings.

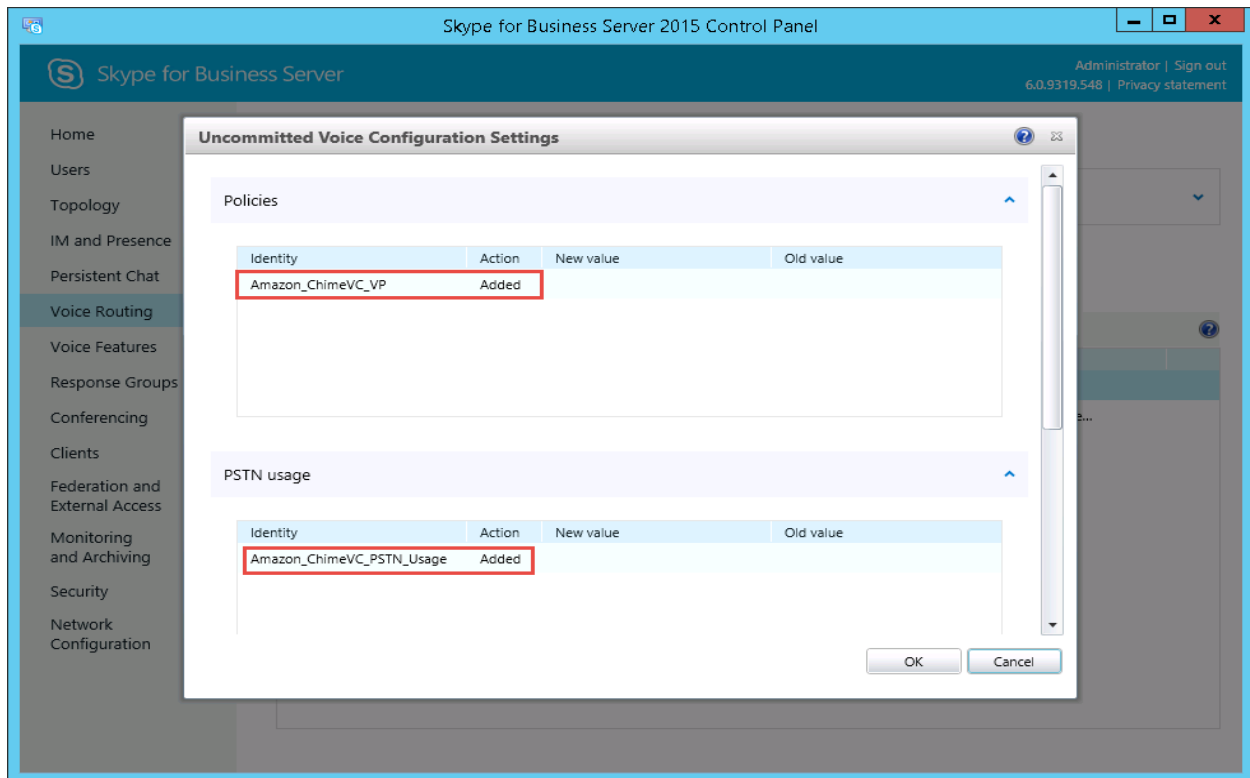


Figure 16: Uncommitted Voice Configuration Settings

The successfully published voice routing configuration window is displayed after the changes are committed.

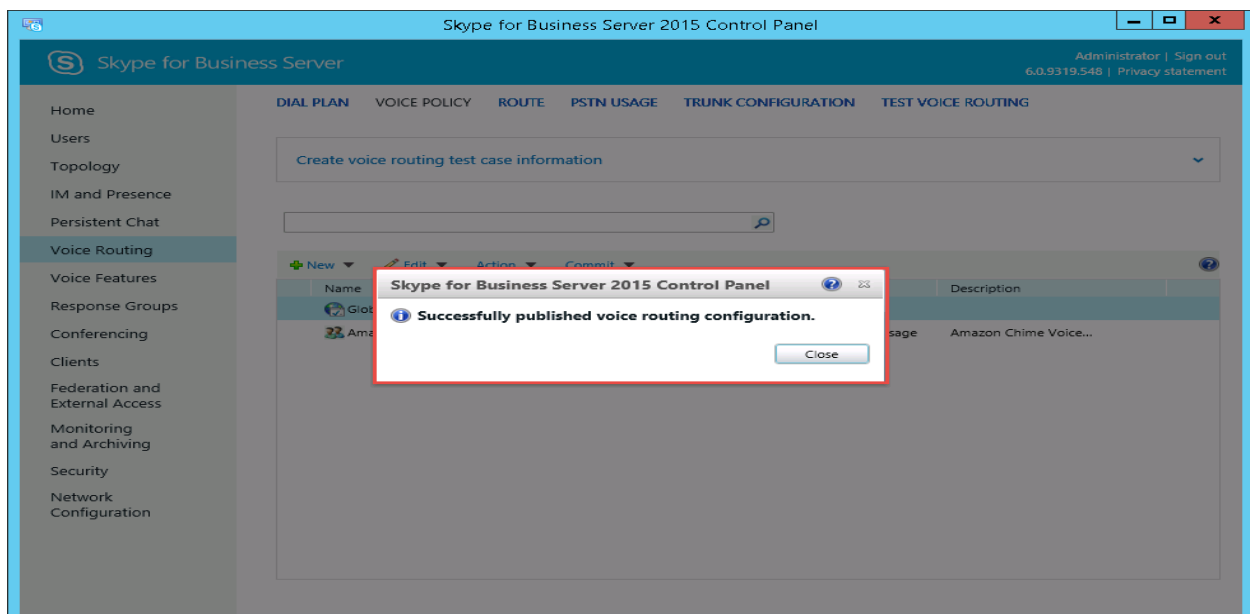


Figure 17: Successfully published voice routing changes

4.2.3 Trunk Configuration

In Skype for Business Control Panel, navigate to 'Trunk Configuration' under Voice Routing. Click on 'New' and select 'Pool trunk'.

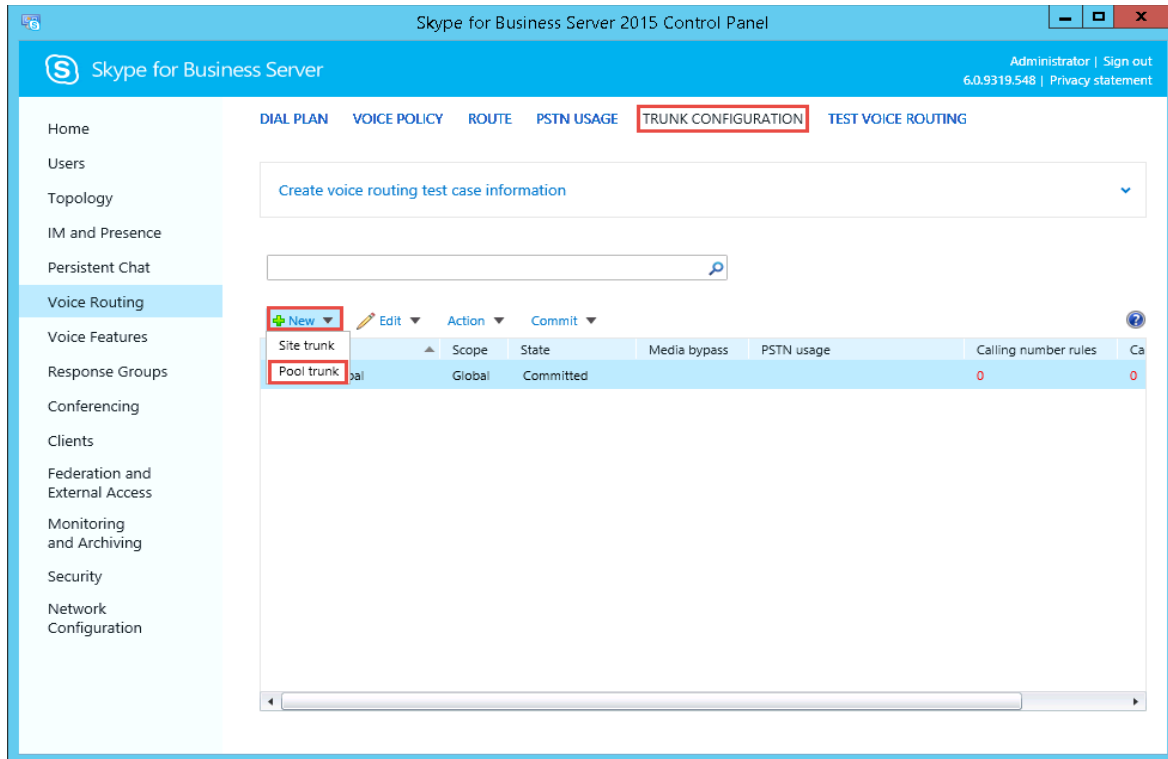


Figure 18: New Pool Trunk

Select the appropriate service created.

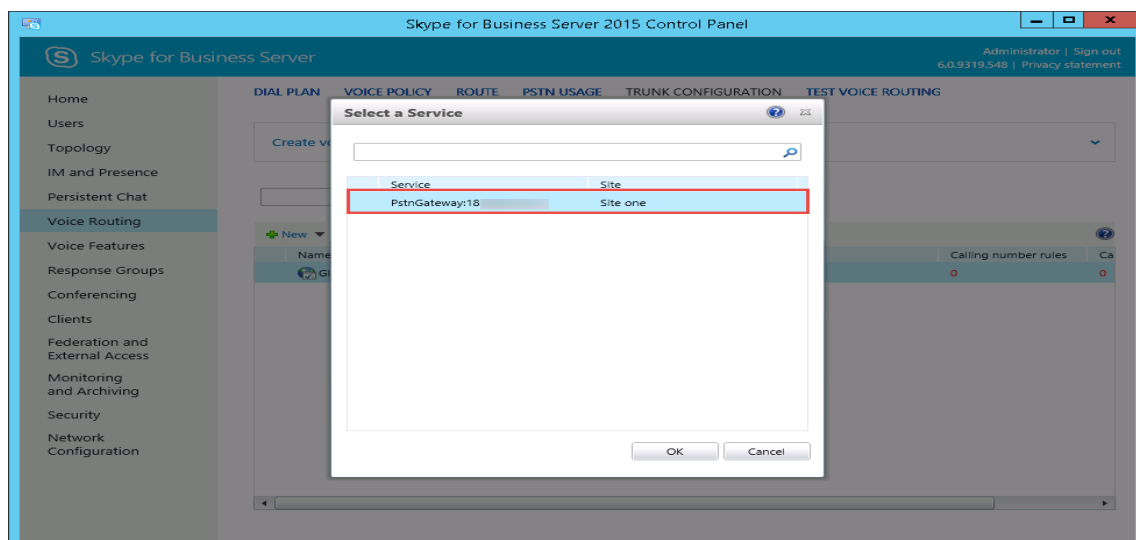


Figure 19: Select a Service

Ensure the parameters *'Enable media bypass'*, *'Centralized media processing'*, *'Enable forward call history'* and *'Enable outbound routing failover timer'* is enabled and the remaining parameters are set to default.

The screenshot displays the 'Skype for Business Server 2015 Control Panel' interface. The left-hand navigation pane lists various administrative sections, with 'Voice Routing' currently selected. The main content area is titled 'DIAL PLAN VOICE POLICY ROUTE PSTN USAGE TRUNK CONFIGURATION TEST VOICE ROUTING'. Below this, there is a section for 'New Trunk Configuration' with fields for 'Scope' (set to 'Pool'), 'Name' (set to 'PstnGateway:18'), 'Description', 'Maximum early dialogs supported' (set to '20'), 'Encryption support level' (set to 'Not supported'), and 'Refer support' (set to 'None'). At the bottom of this configuration form, three checkboxes are visible: 'Enable media bypass' (checked), 'Centralized media processing' (checked), and 'Enable RTP latching' (unchecked). A red rectangular box highlights the first two checked options.

Figure 20: Trunk Configuration

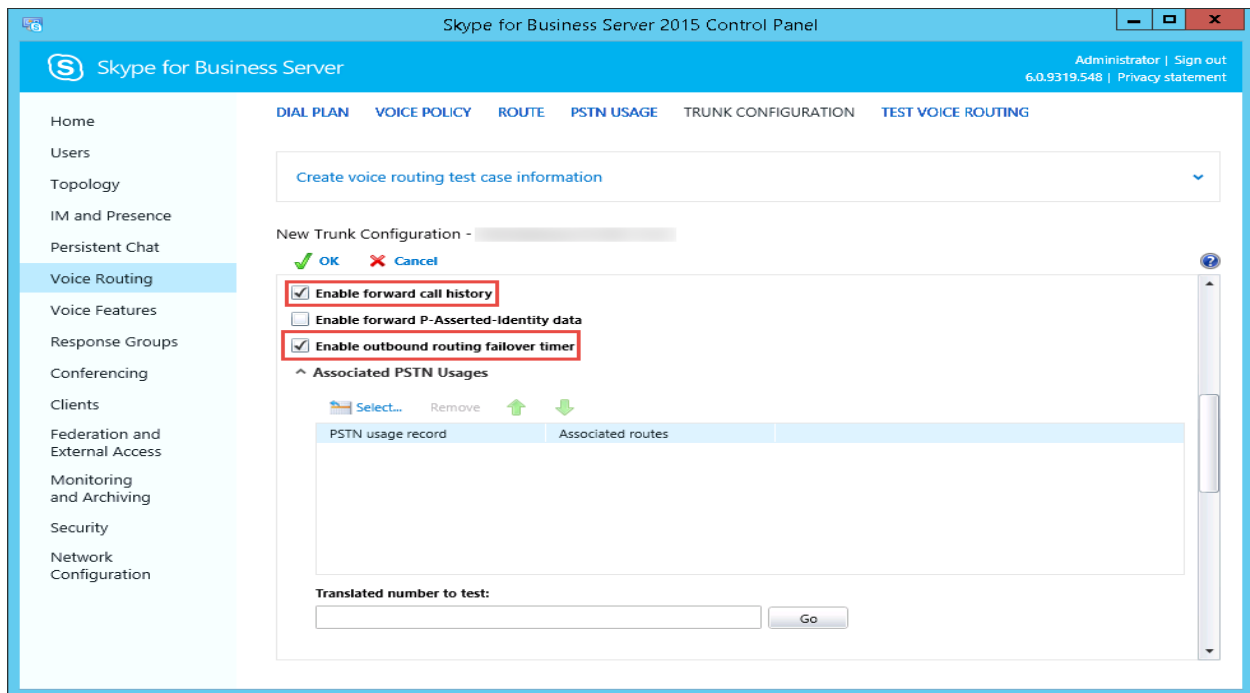


Figure 21: Trunk Configuration Continuation

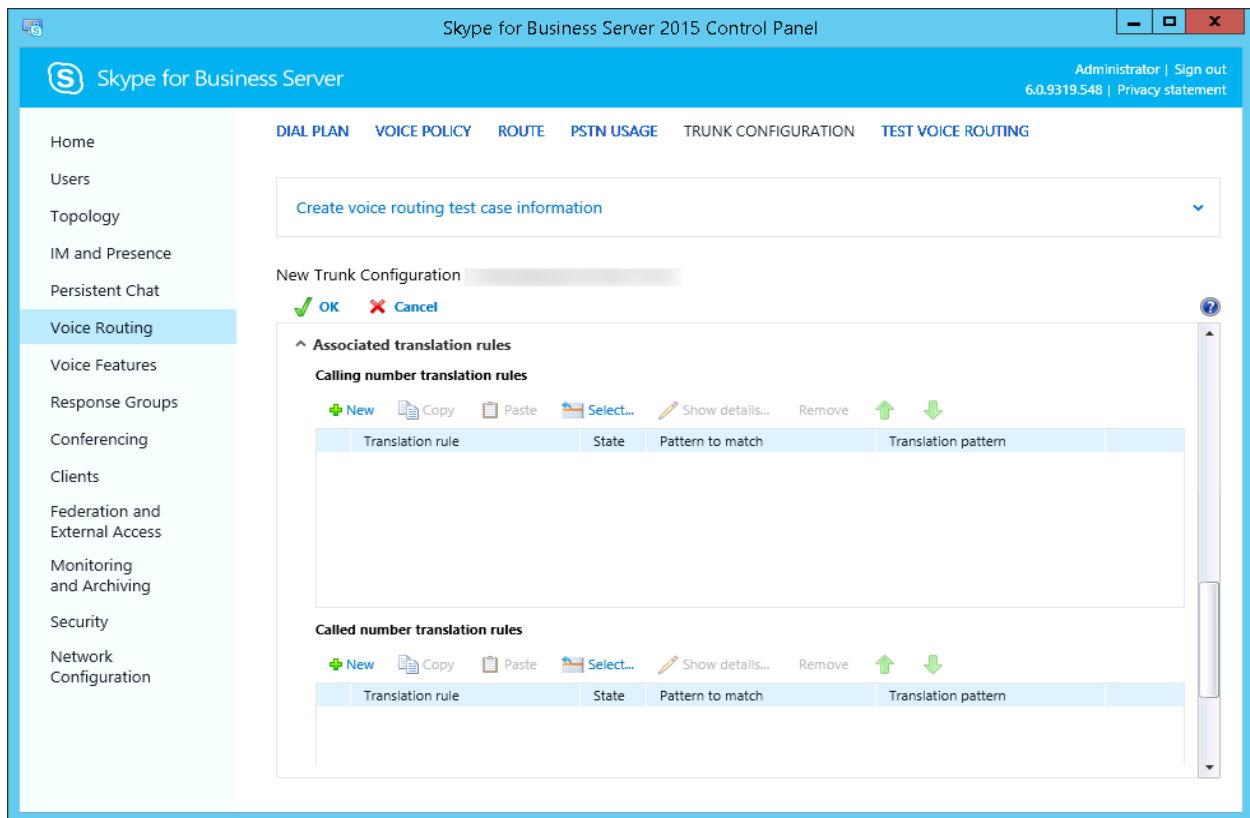


Figure 22: Trunk Configuration Continuation

Click on 'OK' and commit all unsaved changes.

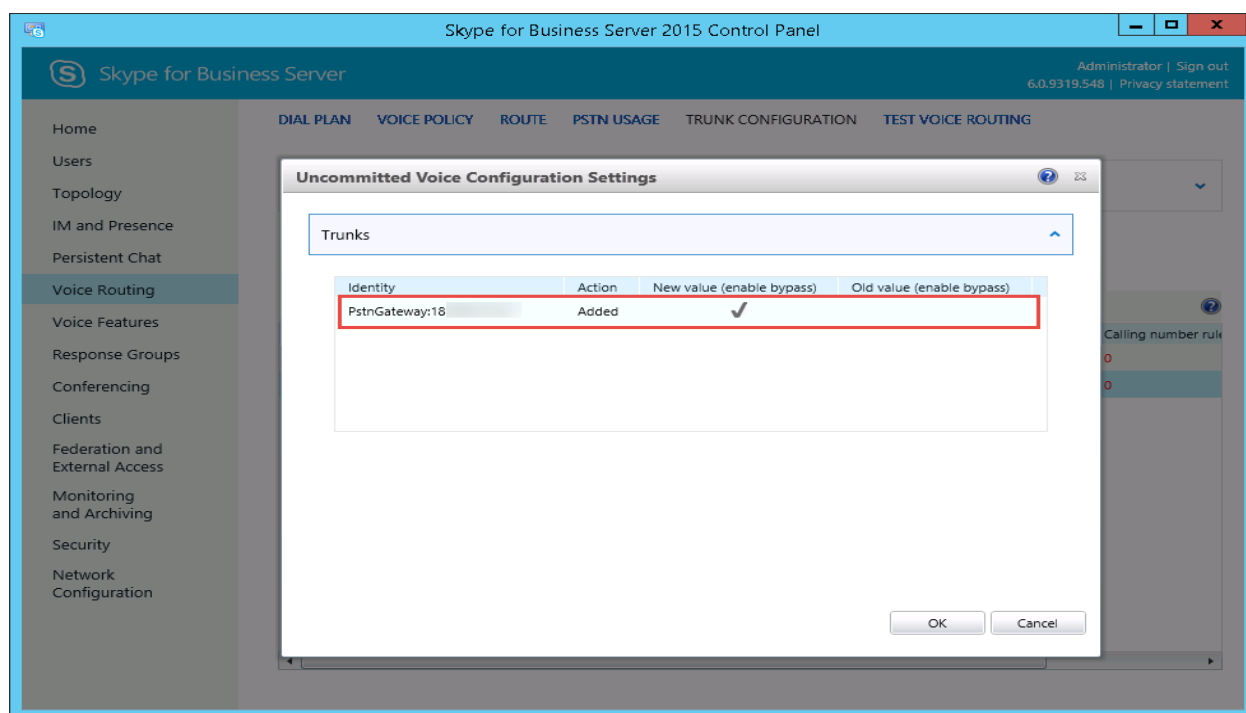


Figure 23: Uncommitted Trunk Configuration settings

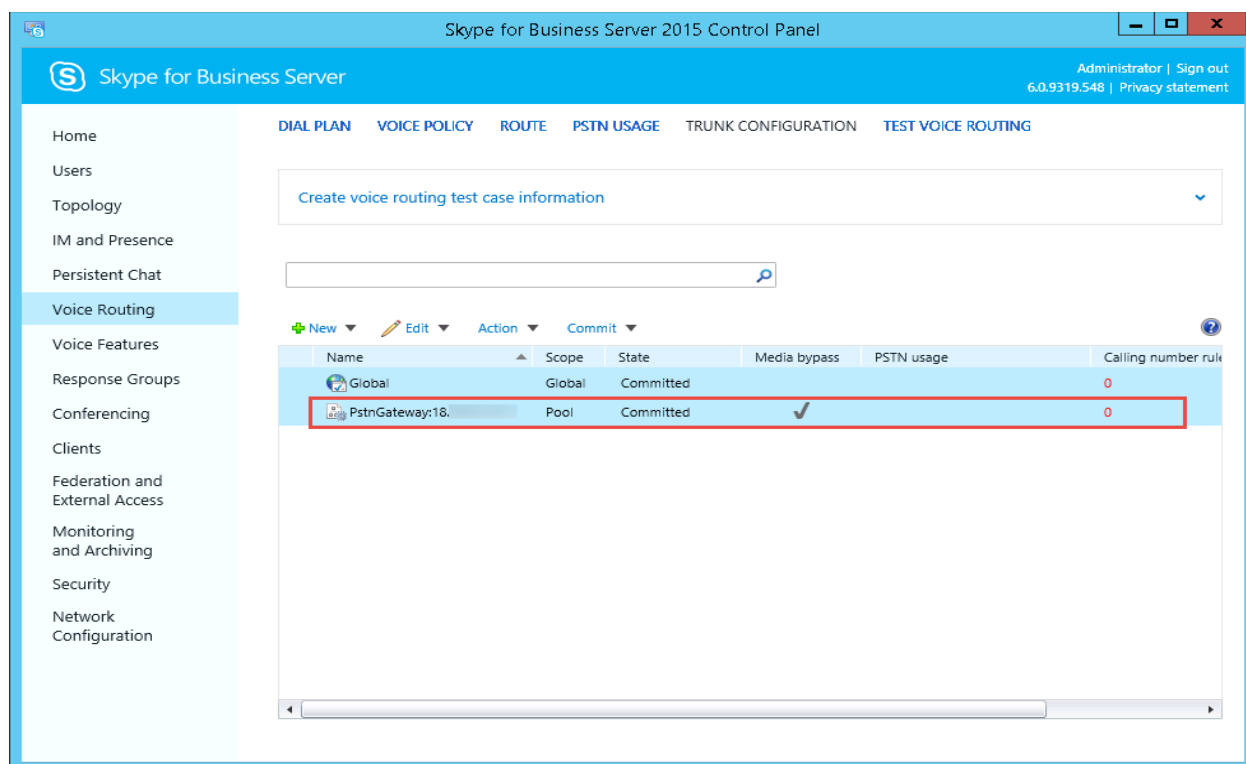
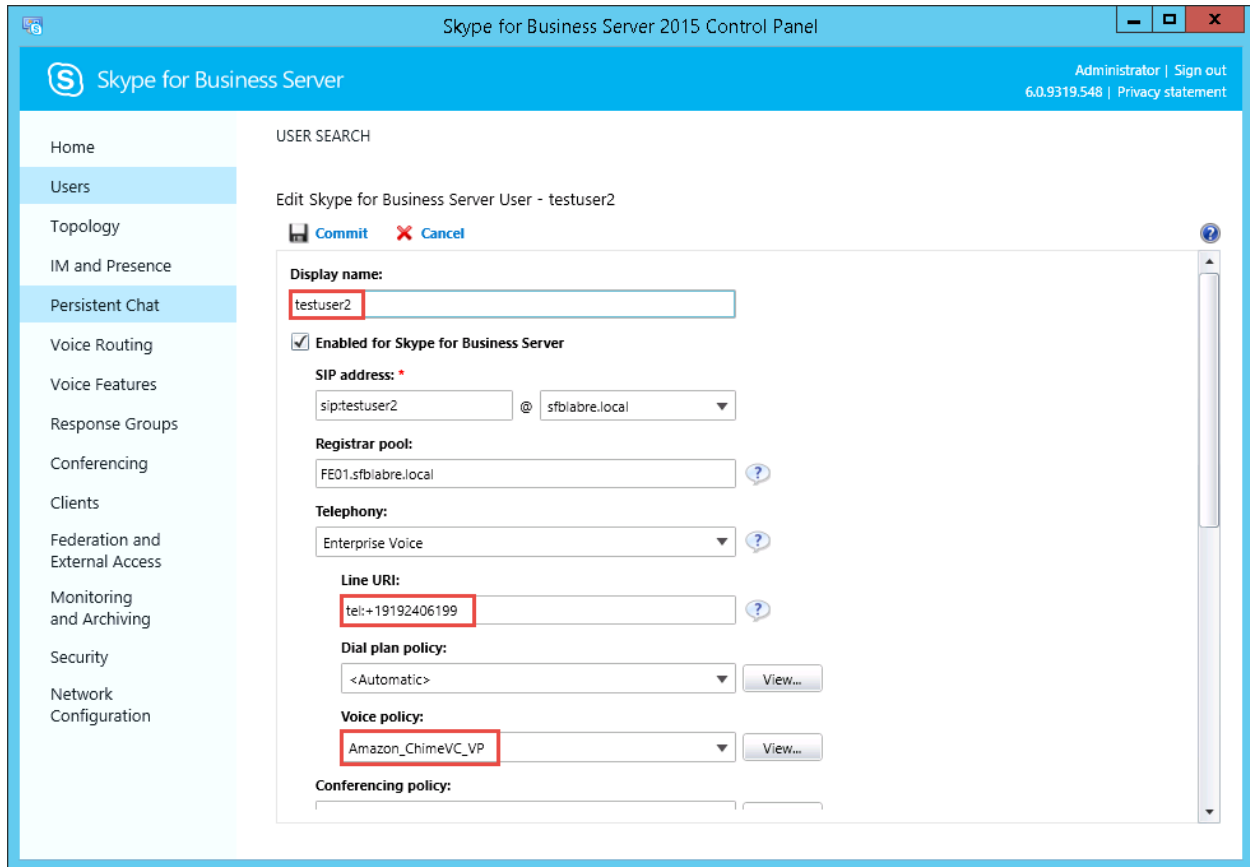


Figure 24: Trunk Configuration after committed changes

4.2.4 User Configuration

The user configuration step involves enabling users for Skype for Business with a sip address and Line URI. Amazon Chime Voice Connector DID is associated with the users and appropriate voice policy is selected for the user.



The screenshot displays the 'Skype for Business Server 2015 Control Panel' interface. On the left is a navigation menu with options: Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing, Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, Network, and Configuration. The 'Users' section is selected. The main area is titled 'USER SEARCH' and 'Edit Skype for Business Server User - testuser2'. It includes 'Commit' and 'Cancel' buttons. The configuration fields are as follows:

- Display name:** testuser2
- ☒ **Enabled for Skype for Business Server**
- SIP address:** sip:testuser2 @ sfblabre.local
- Registrar pool:** FE01.sfbblabre.local
- Telephony:** Enterprise Voice
- Line URI:** tel:+19192406199
- Dial plan policy:** <Automatic> (with a 'View...' button)
- Voice policy:** Amazon_ChimeVC_VP (with a 'View...' button)
- Conferencing policy:** (empty field)

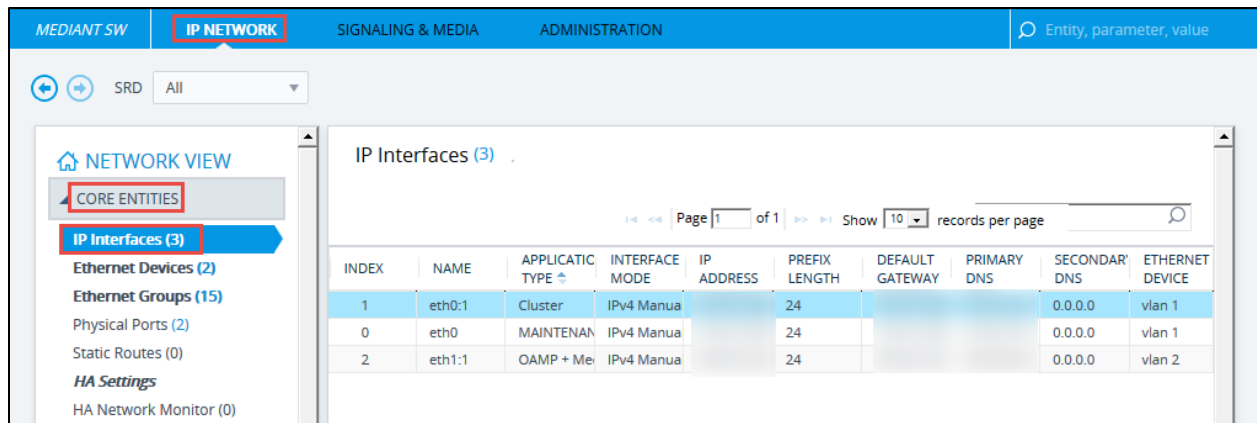
Figure 25: User Configuration

4.3 AudioCodes CE Configuration

The AudioCodes CE is configured with one trunk pointing to Skype for Business server and another trunk pointing to Amazon Chime Voice Connector. The steps involved in configuring the IP and Trunks are shown below

4.3.1 Network IP Interface configuration

Navigate to '*SETUP*', '*IP NETWORK*' and expand '*CORE ENTITIES*'. Click '*IP Interfaces*' and the below figure shows the interfaces that are been used.



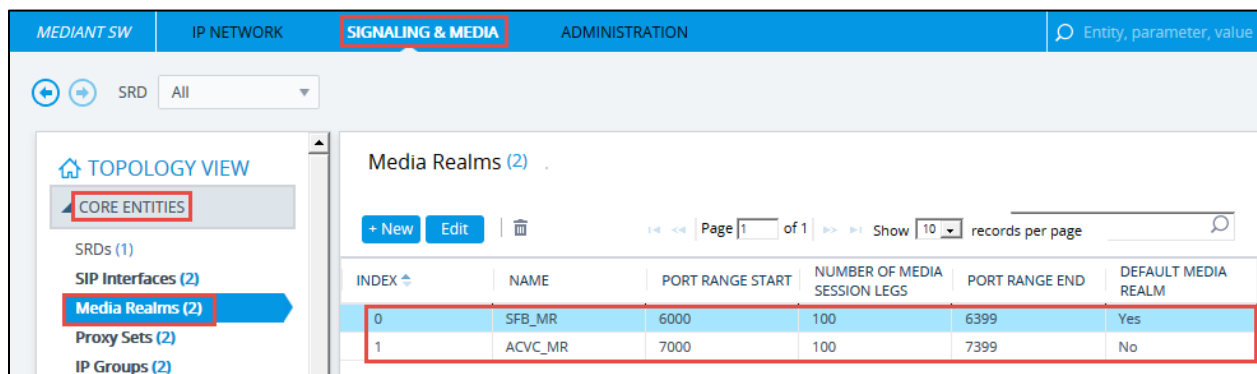
The screenshot shows the 'IP NETWORK' tab in the configuration interface. The left sidebar has 'CORE ENTITIES' expanded, and 'IP Interfaces (3)' is selected. The main area displays a table of IP interfaces.

INDEX	NAME	APPLICATIC TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDAR DNS	ETHERNET DEVICE
1	eth0:1	Cluster	IPv4 Manua		24			0.0.0.0	vlan 1
0	eth0	MAINTENAN	IPv4 Manua		24			0.0.0.0	vlan 1
2	eth1:1	OAMP + Mei	IPv4 Manua		24			0.0.0.0	vlan 2

Figure 26: IP Interfaces

4.3.2 Media Realm configuration

Two media realms are created, one is associated to Skype for Business and another is associated with Amazon Chime Voice Connector. To configure media realm, navigate to '*SETUP*' and select '*SIGNALING & MEDIA*'. Expand '*CORE ENTITIES*' and select '*Media Realms*'.



The screenshot shows the 'SIGNALING & MEDIA' tab in the configuration interface. The left sidebar has 'CORE ENTITIES' expanded, and 'Media Realms (2)' is selected. The main area displays a table of media realms.

INDEX	NAME	PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	PORT RANGE END	DEFAULT MEDIA REALM
0	SFB_MR	6000	100	6399	Yes
1	ACVC_MR	7000	100	7399	No

Figure 27: Media Realms Table

Enter the name of the Media Realm, *Port Range Start* value and *Number of Media Session Legs*. Select the appropriate IPv4 Interface Name for Skype for Business.

Media Realms [SFB_MR]	
GENERAL	
Index	0
Name	SFB_MR
Topology Location	Down
Remote IPv4 Interface Name	#0 [eth1] View
Remote IPv6 Interface Name	-- View
Port Range Start	6000
Number Of Media Session Legs	100
Port Range End	6399
Default Media Realm	Yes
QUALITY OF EXPERIENCE	
QoE Profile	-- View
Bandwidth Profile	-- View
Cancel APPLY	

Figure 28: Media Realm for Skype for Business

Enter the name of the Media Realm, *Port Range Start* value and *Number of Media Session Legs*. Select the appropriate IPv4 Interface Name for Amazon Chime Voice Connector.

Media Realms [ACVC_MR]	
GENERAL	
Index	1
Name	ACVC_MR
Topology Location	Up
Remote IPv4 Interface Name	#0 [eth1] View
Remote IPv6 Interface Name	-- View
Port Range Start	7000
Number Of Media Session Legs	100
Port Range End	7399
Default Media Realm	No
QUALITY OF EXPERIENCE	
QoE Profile	-- View
Bandwidth Profile	-- View
Cancel APPLY	

Figure 29: Media Realm for Amazon Chime Voice Connector

4.3.3 SRD configuration

To configure SRD, navigate to 'SETUP' and select 'SIGNALING & MEDIA'. Expand 'CORE ENTITIES' and select SRDs.

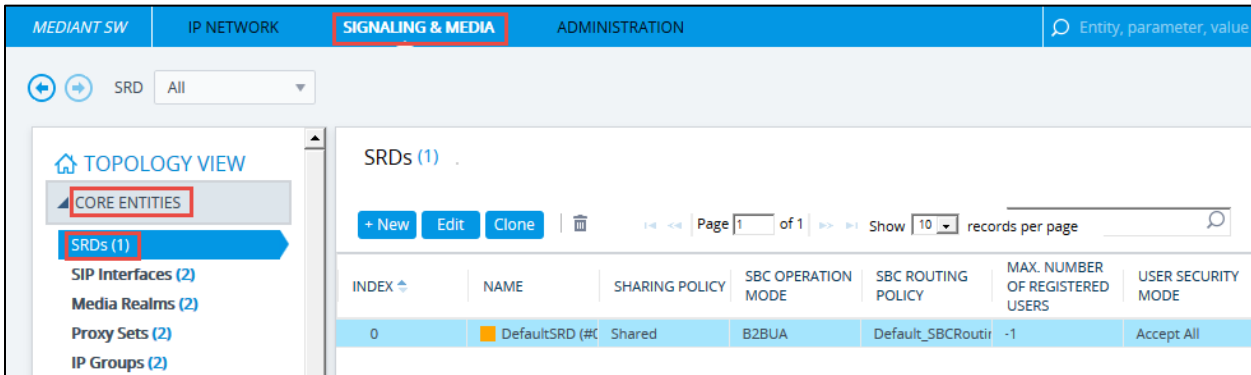


Figure 30: Default SRD

The default SRD configuration is used.

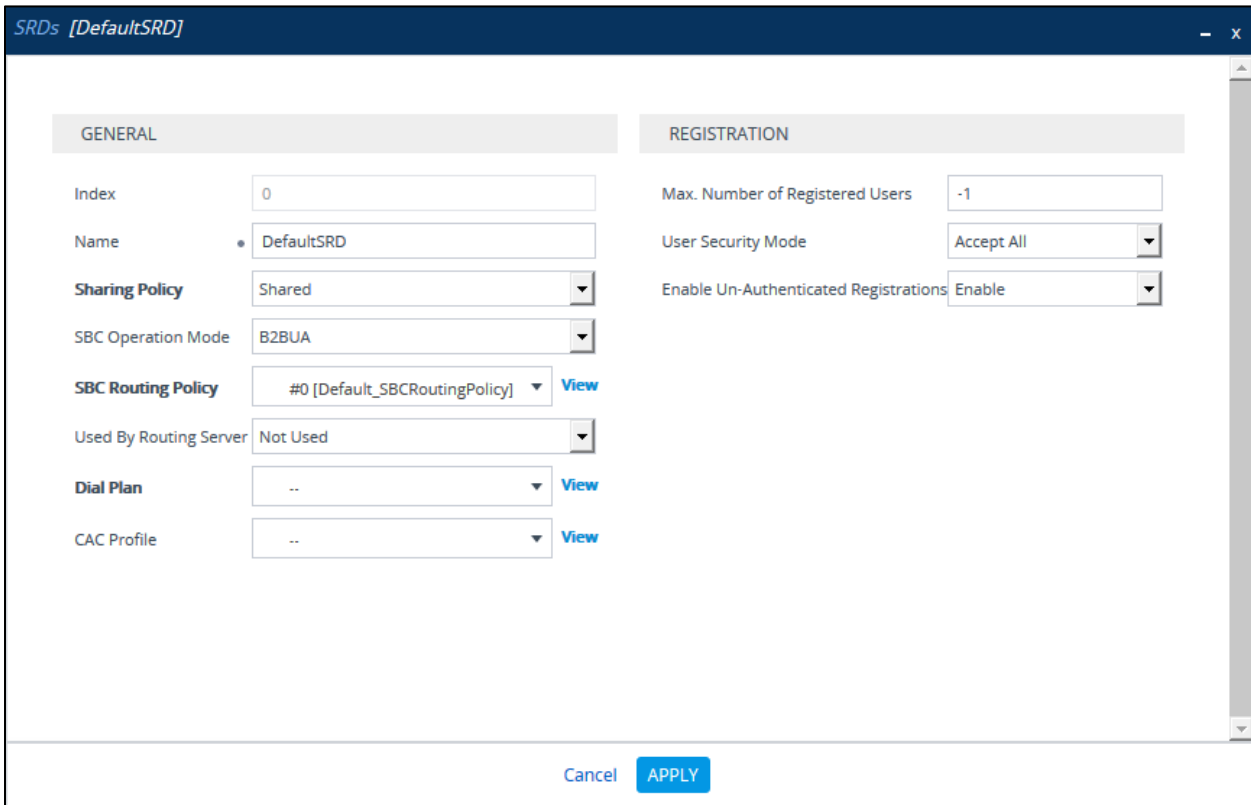


Figure 31: SRD Table Details

4.3.4 SIP Interface configuration

Navigate to 'SETUP' and select 'SIGNALING & MEDIA'. Expand 'CORE ENTITIES' and select 'Sip Interfaces'. Two SIP Interfaces are created, one is for Skype for Business and the other is for Amazon Chime Voice Connector.

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATION PROTOCOL	MEDIA REALM
0	SFB_SipInt	DefaultS	eth1:1	SBC	0	5060	0	No encapsu	SFB_MR
1	ACVC_SipInt	DefaultS	eth1:1	SBC	5060	0	5061	No encapsu	ACVC_MR

Figure 32: SIP Interfaces

Network Interface, Media Realm, SRD and Port numbers are associated to Skype for Business SIP Interface and the remaining parameters are set to default.

SRD: #0 [DefaultSRD]

GENERAL

Index: 0

Name: SFB_SipInt

Topology Location: Down

Network Interface: #2 [eth1:1]

Application Type: SBC

UDP Port: 0

TCP Port: 5060

TLS Port: 0

Additional UDP Ports:

Additional UDP Ports Mode: Always Open

MEDIA

Media Realm: #0 [SFB_MR]

Direct Media: Disable

SECURITY

TLS Context Name: ..

TLS Mutual Authentication:

Message Policy: ..

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

Buttons: Cancel, APPLY

Figure 33: SIP Interface for Skype for Business

SIP Interfaces [SFB_SipInt]

TCP Port	5060	Message Policy	.. View
TLS Port	0	User Security Mode	Not Configured
Additional UDP Ports		Enable Un-Authenticated Registrations	Not configured
Additional UDP Ports Mode	Always Open	Max. Number of Registered Users	-1
Encapsulating Protocol	No encapsulation		
Enable TCP Keepalive	Disable		
Used By Routing Server	Not Used		
Pre-Parsing Manipulation Set	.. View		
CAC Profile	.. View		

CLASSIFICATION

Classification Failure Response Type	500
Pre-classification Manipulation Set ID	-1
Call Setup Rules Set ID	-1

[Cancel](#)
[APPLY](#)

Figure 34: SIP Interface for Skype for Business Continuation

Network Interface, Media Realm, SRD and Port numbers are associated to Amazon Chime Voice Connector SIP Interface and the remaining parameters are set to default.

SIP Interfaces [ACVC_SipInt]

SRD **#0 [DefaultSRD]**

GENERAL	MEDIA
Index	1
Name	ACVC_SipInt
Topology Location	Up
Network Interface	#2 [eth1:1] View
Application Type	SBC
UDP Port	5060
TCP Port	0
TLS Port	5061
Additional UDP Ports	
Additional UDP Ports Mode	Always Open
	MEDIA
	Media Realm
	#1 [ACVC_MR] View
	Direct Media
	Disable
	SECURITY
	TLS Context Name
	.. View
	TLS Mutual Authentication
	Message Policy
	.. View
	User Security Mode
	Not Configured
	Enable Un-Authenticated Registrations
	Not configured
	Max. Number of Registered Users
	-1

[Cancel](#)
[APPLY](#)

Figure 35: SIP Interface for Amazon Chime Voice Connector

SIP Interfaces [ACVC_SipInt]

TCP Port: 0
 TLS Port: 5061
 Additional UDP Ports:
 Additional UDP Ports Mode: Always Open
 Encapsulating Protocol: No encapsulation
 Enable TCP Keepalive: Disable
 Used By Routing Server: Not Used
 Pre-Parsing Manipulation Set: -- View
 CAC Profile: -- View

Message Policy: -- View
 User Security Mode: Not Configured
 Enable Un-Authenticated Registrations: Not configured
 Max. Number of Registered Users: -1

CLASSIFICATION

Classification Failure Response Type: 500
 Pre-classification Manipulation Set ID: -1
 Call Setup Rules Set ID: -1

Cancel APPLY

Figure 36: SIP Interface for Amazon Chime Voice Connector Continuation

4.3.5 Proxy Sets configuration

Navigate to 'SETUP' and select 'SIGNALING & MEDIA'. Expand 'CORE ENTITIES' and select 'Proxy Sets'. Destination address or FQDN is configured in Proxy Sets. Two Proxy Sets are created, one for Skype for Business and other for Amazon Chime Voice Connector.

SIGNALING & MEDIA

Entity, parameter, value

SRD: All

TOPOLOGY VIEW

- CORE ENTITIES
 - SRDs (1)
 - SIP Interfaces (2)
 - Media Realms (2)
 - Proxy Sets (2)**
 - IP Groups (2)

Proxy Sets (2)

+ New Edit

Page 1 of 1 Show 10 records per page

INDEX	NAME	SRD	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	SFB_ProxySet	DefaultSRD (#)	SFB_SipInt	60		Disable
1	ACVC_ProxySet	DefaultSRD (#)	ACVC_SipInt	60		Disable

Figure 37: Proxy Sets table

Select *SBC IPv4 SIP Interface* and enable *Proxy Keep-Alive* for Skype for Business Proxy Set.

Proxy Sets [SFB_ProxySet]

SRD #0 [DefaultSRD]

GENERAL

Index: 0

Name: SFB_ProxySet

SBC IPv4 SIP Interface: #0 [SFB_SipInt] [View](#)

TLS Context Name: -- [View](#)

REDUNDANCY

Redundancy Mode: [Dropdown]

Proxy Hot Swap: Disable

Proxy Load Balancing Method: Disable

Min. Active Servers for Load Balancing: 1

KEEP ALIVE

Proxy Keep-Alive: Using OPTIONS

Proxy Keep-Alive Time [sec]: 60

Keep-Alive Failure Responses: [Dropdown]

Success Detection Retries: 1

Success Detection Interval: 10

ADVANCED

Classification Input: IP Address only

DNS Resolve Method: [Dropdown]

[Cancel](#) [APPLY](#)

Figure 38: Proxy Set table for Skype for Business

Click on 'Proxy Address 0 items' link in bottom to add *Proxy Address* and *Transport Type*.

Proxy Sets [#0] > Proxy Address (1)

[+ New](#) [Edit](#) [Delete](#)

Page 1 of 1 Show 10 records per page

INDEX	PROXY ADDRESS	TRANSPORT TYPE
0	14.	TCP

#0 [Edit](#)

GENERAL

Proxy Address: [Dropdown]

Transport Type: TCP

Proxy Priority: 0

Proxy Random ...: 0

Figure 39: Proxy Address for Skype for Business

Select *SBC IPv4 SIP Interface* and enable *Proxy Keep-Alive* for Amazon Chime Voice Connector Proxy Set.

The screenshot shows the 'Proxy Sets [ACVC_ProxySet]' configuration window. At the top, there is a dropdown for 'SRD' set to '#0 [DefaultSRD]'. Below this are four tabs: 'GENERAL', 'REDUNDANCY', 'KEEP ALIVE', and 'ADVANCED'. In the 'GENERAL' tab, 'Index' is 1, 'Name' is 'ACVC_ProxySet', 'SBC IPv4 SIP Interface' is '#1 [ACVC_SipInt]' (highlighted with a red box), and 'TLS Context Name' is '..'. In the 'REDUNDANCY' tab, 'Redundancy Mode' is a dropdown, 'Proxy Hot Swap' is 'Disable', 'Proxy Load Balancing Method' is 'Disable', and 'Min. Active Servers for Load Balancing' is 1. In the 'KEEP ALIVE' tab, 'Proxy Keep-Alive' is 'Using OPTIONS' (highlighted with a red box), 'Proxy Keep-Alive Time [sec]' is 60, 'Keep-Alive Failure Responses' is empty, and 'Success Detection Retries' is 1. In the 'ADVANCED' tab, 'Classification Input' is 'IP Address only' and 'DNS Resolve Method' is a dropdown. At the bottom are 'Cancel' and 'APPLY' buttons.

Figure 40: Proxy Set table for Amazon Chime Voice Connector

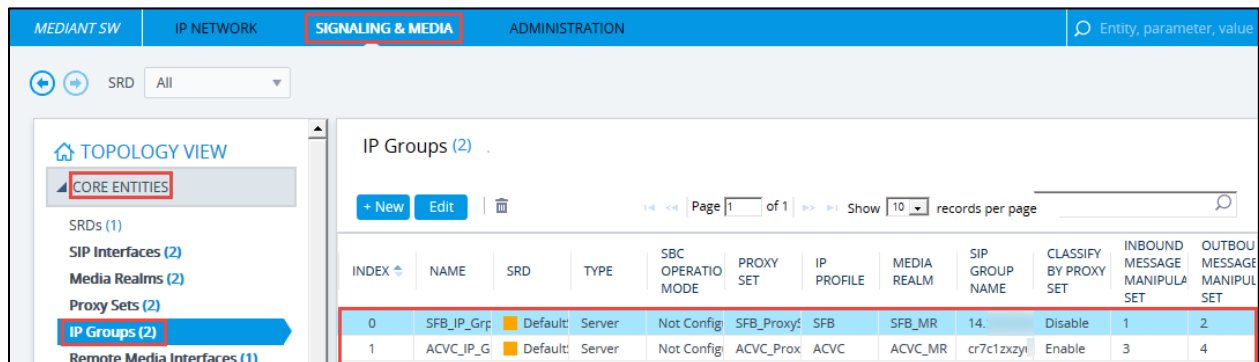
Click on '*Proxy Address 0 items*' link in bottom to add Proxy Address and Transport Type.

The screenshot shows the 'Proxy Sets [#1] > Proxy Address (1)' page. At the top, there are '+ New', 'Edit', and a trash icon. Below this is a table with columns 'INDEX', 'PROXY ADDRESS', and 'TRANSPORT TYPE'. The table has one row with index '0', proxy address 'cr7c1xzy' (highlighted with a red box), and transport type 'UDP'. Below the table is a section for '#0' with an 'Edit' button. Under the '#0' section is a 'GENERAL' tab with fields: 'Proxy Address' (cr7c1xzy), 'Transport Type' (UDP), 'Proxy Priority' (0), and 'Proxy Random ...' (0).

Figure 41: Proxy Address for Amazon Chime Voice Connector

4.3.6 IP Group Table configuration

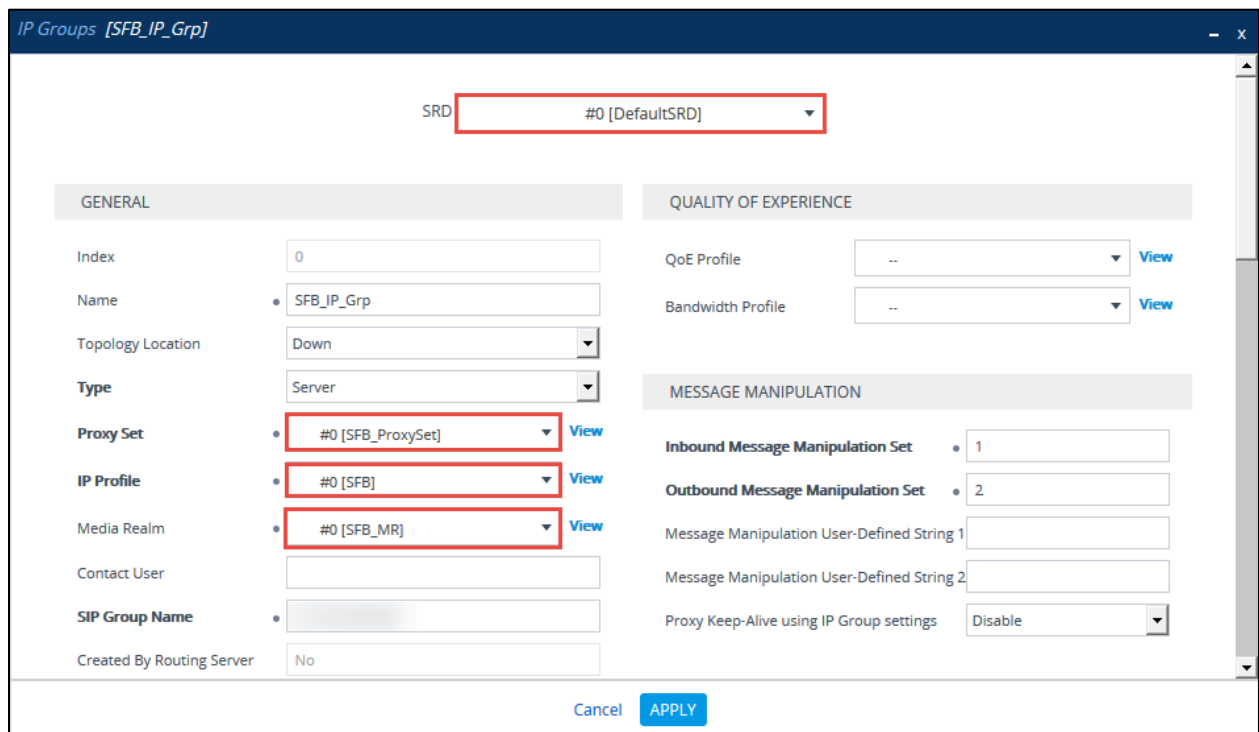
Navigate to '*SETUP*' and select '*SIGNALING & MEDIA*'. Expand '*CORE ENTITIES*' and select '*IP Groups*'. IP Groups are configured for denoting source and destination in IP-to-IP routing rules. IP Groups created for Skype for Business and AudioCodes CE.



INDEX	NAME	SRD	TYPE	SBC OPERATIO MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULA SET	OUTBOU MESSAGE MANIPUL SET
0	SFB_IP_Grp	Default	Server	Not Config	SFB_Proxy	SFB	SFB_MR	14.	Disable	1	2
1	ACVC_IP_G	Default	Server	Not Config	ACVC_Prox	ACVC	ACVC_MR	cr7c1zxzyi	Enable	3	4

Figure 42: IP Group Table

Enter the name of the IP Groups for Skype for Business and associate *Proxy Set*, *IP Profile*, *Media Realm* and the remaining parameters are set to default.



IP Groups [SFB_IP_Grp]

SRD: #0 [DefaultSRD]

GENERAL

Index: 0

Name: SFB_IP_Grp

Topology Location: Down

Type: Server

Proxy Set: #0 [SFB_ProxySet] [View](#)

IP Profile: #0 [SFB] [View](#)

Media Realm: #0 [SFB_MR] [View](#)

Contact User:

SIP Group Name:

Created By Routing Server: No

QUALITY OF EXPERIENCE

QoE Profile: -- [View](#)

Bandwidth Profile: -- [View](#)

MESSAGE MANIPULATION

Inbound Message Manipulation Set: 1

Outbound Message Manipulation Set: 2

Message Manipulation User-Defined String 1:

Message Manipulation User-Defined String 2:

Proxy Keep-Alive using IP Group settings: Disable

[Cancel](#) [APPLY](#)

Figure 43: IP Group Table for Skype for Business

IP Groups [SFB_IP_Grp]

Used By Routing Server: Not Used

Proxy Set Connectivity: Connected

SBC GENERAL

Classify By Proxy Set: Disable

SBC Operation Mode: Not Configured

SBC Client Forking Mode: Sequential

CAC Profile: -- [View](#)

ADVANCED

Local Host Name:

UUI Format: Disable

SBC REGISTRATION AND AUTHENTICATION

Max. Number of Registered Users: -1

Registration Mode: User Initiates Registration

User Stickiness: Disable

User UDP Port Assignment: Disable

Authentication Mode: User Authenticates

Authentication Method List:

SBC Server Authentication Type: According to Global Parameter

OAuth HTTP Service: -- [View](#)

Username:

Password:

GW GROUP STATUS

GW Group Registered IP Address:

[Cancel](#) [APPLY](#)

Figure 44: IP Group table for Skype for Business Continuation

IP Groups [SFB_IP_Grp]

UUI Format: Disable

Always Use Src Address: No

SBC ADVANCED

Source URI Input:

Destination URI Input:

SIP Connect: No

SBC PSAP Mode: Disable

Route Using Request URI Port: Disable

DTLS Context: -- [View](#)

Keep Original Call-ID: No

Dial Plan: -- [View](#)

Call Setup Rules Set ID: -1

Tags:

GW Group Registered IP Address:

GW Group Registered Status: Not Registered

[Cancel](#) [APPLY](#)

Figure 45: IP Group table for Skype for Business Continuation

Enter the name of the IP Groups for Amazon Chime Voice Connector and associate *Proxy Set*, *IP Profile*, *Media Realm* and the remaining parameters are set to default.

IP Groups [ACVC_IP_Grp]

SRD#0 [DefaultSRD]

GENERAL

Index

1

Name

ACVC_IP_Grp

Topology Location

Up

Type

Server

Proxy Set

#1 [ACVC_ProxySet]

View

IP Profile

#1 [ACVC]

View

Media Realm

#1 [ACVC_MR]

View

Contact User

SIP Group Name

cr7c1xzyuaaeqeuews5s1.voiceconnector.ch

Created By Routing Server

No

QUALITY OF EXPERIENCE

QoE Profile

--

View

Bandwidth Profile

--

View

MESSAGE MANIPULATION

Inbound Message Manipulation Set

3

Outbound Message Manipulation Set

4

Message Manipulation User-Defined String 1

Message Manipulation User-Defined String 2

Proxy Keep-Alive using IP Group settings

Disable

Cancel

APPLY

Figure 46: IP Group table for Amazon Chime Voice Connector

IP Groups [ACVC_IP_Grp]

Used By Routing Server

Not Used

Proxy Set Connectivity

Connected

SBC GENERAL

Classify By Proxy Set

Enable

SBC Operation Mode

Not Configured

SBC Client Forking Mode

Sequential

CAC Profile

--

View

ADVANCED

Local Host Name

UUI Format

Disable

SBC REGISTRATION AND AUTHENTICATION

Max. Number of Registered Users

-1

Registration Mode

User Initiates Registration

User Stickiness

Disable

User UDP Port Assignment

Disable

Authentication Mode

SBC as Client

Authentication Method List

SBC Server Authentication Type

According to Global Parameter

OAuth HTTP Service

--

View

Username

Password

GW GROUP STATUS

GW Group Registered IP Address

Cancel

APPLY

Figure 47: IP Group table for Amazon Chime Voice Connector Continuation

IP Groups [ACVC_IP_Grp]

UII Format: Disable
 Always Use Src Address: No
 GW Group Registered IP Address:
 GW Group Registered Status: Not Registered

SBC ADVANCED

Source URI Input:
 Destination URI Input:
 SIP Connect: No
 SBC PSAP Mode: Disable
 Route Using Request URI Port: Disable
 DTLS Context: -- [View](#)
 Keep Original Call-ID: No
 Dial Plan: -- [View](#)
 Call Setup Rules Set ID: -1
 Tags:

Cancel **APPLY**

Figure 48: IP Group table for Amazon Chime Voice Connector Continuation

4.3.7 Coder Groups configuration

Navigate to 'SETUP' and select 'SIGNALING & MEDIA'. Expand 'CODERS & PROFILES' and select 'Coder Groups'. G711 U-law is configured in Coder Groups.

MEDIANT SW | **IP NETWORK** | **SIGNALING & MEDIA** | **ADMINISTRATION** | Entity, parameter, value

SRD All

TOPOLOGY VIEW

- CORE ENTITIES
- CODERS & PROFILES**
 - IP Profiles (2)
 - Coder Settings
 - Coder Groups**
 - Allowed Audio Coders Groups (1)
 - Allowed Video Coders Groups (0)
- SBC
- SIP DEFINITIONS
- MESSAGE MANIPULATION
- MEDIA
- INTRUSION DETECTION
- SIP RECORDING

Coder Groups

Coder Group Name: 0 : AudioCodersGroups_0 [Delete Group](#)

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.711U-law	20	64	0	Disabled	

Cancel **APPLY**

Figure 49: Coder Groups

Navigate to 'SETUP', 'SIGNALING & MEDIA', 'CODERS & PROFILES' and select 'Allowed Audio Coders Groups'. Click on 'New' button to create Allowed Audio

Coders Group and then Click on 'Allowed Audio Coders 0 items' link and click 'New' to add the coders.

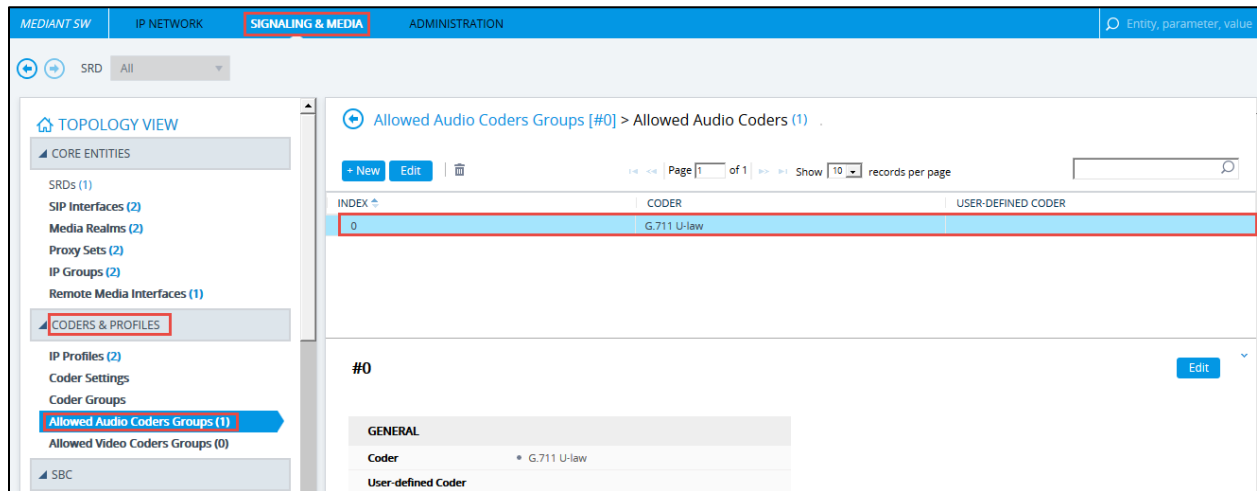


Figure 50: Allowed Audio Coders

4.3.8 IP Profile configuration

Navigate to 'SETUP' and select 'SIGNALING & MEDIA'. Expand 'CODERS & PROFILES' and select 'IP Profiles'. Two IP Profiles are created, one for Skype for Business and other for Amazon Chime Voice Connector.

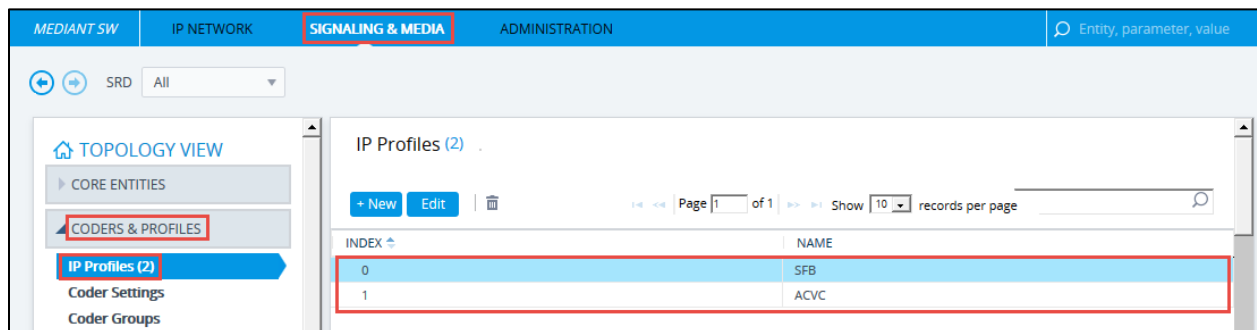


Figure 51: IP Profiles

In the IP Profile for Skype for Business, select the PRACK Mode as 'Optional', Session Expire Mode as 'Supported' and Remote Early Media RTP Detection Mode as 'By Media'. Extension Coder Group and Allowed Audio Coders are associated appropriately.

IP Profiles [SFB]

GENERAL	SBC SIGNALING
Index: 0	PRACK Mode : Optional
Name: SFB	P-Asserted-Identity Header Mode: As Is
Created by Routing Server: No	Diversion Header Mode: As Is
	History-Info Header Mode: As Is
	Session Expires Mode: Supported
MEDIA SECURITY	Remote Update Support: Supported
SBC Media Security Mode: As Is	Remote re-INVITE: Supported
Symmetric MKI: Disable	Remote Delayed Offer Support: Supported
MKI Size: 0	Remote Representation Mode: According to Operation Mode
SBC Enforce MKI Size: Don't enforce	Keep Incoming Via Headers: According to Operation Mode
SBC Media Security Method: SDES	Keep Incoming Routing Headers: According to Operation Mode
Reset SRTP Upon Re-key: Disable	Keep User-Agent Header: According to Operation Mode
Generate SRTP Keys Mode: Only If Required	Handle X-Detect: No
SBC Remove Crypto Lifetime in SDP: No	

Cancel APPLY

Figure 52: IP Profile for Skype for Business

IP Profiles [SFB]

SBC Remove Unknown Crypto: No	ISUP Body Handling: Transparent
	ISUP Variant: Itu92
	Max Call Duration [min]: 0
SBC EARLY MEDIA	SBC REGISTRATION
Remote Early Media: Supported	User Registration Time: 0
Remote Multiple 18x: Supported	NAT UDP Registration Time: -1
Remote Early Media Response Type: Transparent	NAT TCP Registration Time: -1
Remote Multiple Early Dialogs: According to Operation Mode	
Remote Multiple Answers Mode: Disable	SBC FORWARD AND TRANSFER
Remote Early Media RTP Detection Mode: By Media	Remote REFER Mode: Regular
Remote RFC 3960 Support: Not Supported	Remote Replaces Mode: Standard
Remote Can Play Ringback: Yes	Play RBT To Transferee: No
Generate RTP: None	Remote 3xx Mode: Transparent
SBC MEDIA	

Cancel APPLY

Figure 53: IP Profile for Skype for Business Continuation

IP Profiles [SFB]

Remote Early Media	Supported	SBC REGISTRATION	
Remote Multiple 18x	Supported	User Registration Time	0
Remote Early Media Response Type	Transparent	NAT UDP Registration Time	-1
Remote Multiple Early Dialogs	According to Operation Mode	NAT TCP Registration Time	-1
Remote Multiple Answers Mode	Disable	SBC FORWARD AND TRANSFER	
Remote Early Media RTP Detection Mode	By Media	Remote REFER Mode	Regular
Remote RFC 3960 Support	Not Supported	Remote Replaces Mode	Standard
Remote Can Play Ringback	Yes	Play RBT To Transferee	No
Generate RTP	None	Remote 3xx Mode	Transparent
SBC MEDIA		SBC HOLD	
Mediation Mode	RTP Mediation	Remote Hold Format	Transparent
Extension Coders Group	#0 [AudioCodersGroups_0]		
Allowed Audio Coders	#0 [G711] View		

Cancel **APPLY**

Figure 54: IP Profile for Skype for Business Continuation

IP Profiles [SFB]

Allowed Coders Mode	Restriction	Reliable Held Tone Source	Yes
Allowed Video Coders	-- View	Play Held Tone	No
Allowed Media Types		SBC FAX	
Direct Media Tag		Fax Coders Group	--
RFC 2833 Mode	As Is	Fax Mode	As Is
RFC 2833 DTMF Payload Type	0	Fax Offer Mode	All coders
Alternative DTMF Method	As Is	Fax Answer Mode	Single coder
Send Multiple DTMF Methods	Disable	Remote Renegotiate on Fax Detection	Transparent
Adapt RFC2833 BW to Voice coder BW	Disabled	Fax Rerouting Mode	Disable
SDP Ptime Answer	Remote Answer	MEDIA	
Preferred PTime	0	Broken Connection Mode	Disconnect
Use Silence Suppression	Transparent	Media IP Version Preference	Only IPv4
RTP Redundancy Mode	As Is		
RTCP Mode	Transparent		

Cancel **APPLY**

Figure 55: IP Profile for Skype for Business Continuation

IP Profiles [SFB]

Jitter Compensation	Disable	RTP Redundancy Depth	Disable
ICE Mode	Disable		
SDP Handle RTCP	Don't Care		
RTCP Mux	Not Supported		
RTCP Feedback	Feedback Off		
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-op	No		
Enhanced PLC	Disable		

LOCAL TONES

Local RingBack Tone Index	-1
Local Held Tone Index	-1

QUALITY OF SERVICE

RTP IP DiffServ	46
Signaling DiffServ	24

Cancel APPLY

Figure 56: IP Profile for Skype for Business Continuation

IP Profiles [SFB]

QUALITY OF SERVICE

RTP IP DiffServ	46
Signaling DiffServ	24

JITTER BUFFER

Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
Jitter Buffer Max Delay [msec]	300

VOICE

Echo Canceled	Line
Input Gain (-32 to 31 dB)	0
Voice Volume (-32 to 31 dB)	0

Cancel APPLY

Figure 57: IP Profile for Skype for Business Continuation

In the IP profile for Amazon Chime Voice Connector, select Session Expires Mode as 'Supported' and Remote Update Support as 'Not Supported'. Extension Coder Group and Allowed Audio Coders are associated appropriately.

The screenshot shows the 'IP Profiles [ACVC]' configuration window. The 'GENERAL' tab is active, displaying fields for Index (1), Name (ACVC), and Created by Routing Server (No). The 'MEDIA SECURITY' section includes SBC Media Security Mode (As Is), Symmetric MKI (Disable), MKI Size (0), SBC Enforce MKI Size (Don't enforce), SBC Media Security Method (SDS), Reset SRTP Upon Re-key (Disable), and Generate SRTP Keys Mode (Only If Required). The 'SBC SIGNALING' tab is also visible, showing PRACK Mode (Transparent), P-Asserted-Identity Header Mode (As Is), Diversion Header Mode (As Is), History-Info Header Mode (As Is), Session Expires Mode (Supported), Remote Update Support (Not Supported), Remote re-INVITE (Supported), Remote Delayed Offer Support (Supported), Remote Representation Mode (According to Operation Mode), Keep Incoming Via Headers (According to Operation Mode), Keep Incoming Routing Headers (According to Operation Mode), and Keep User-Agent Header (According to Operation Mode). The 'Supported' and 'Not Supported' options are highlighted with red boxes. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 58: IP Profile for Amazon Chime Voice Connector

The screenshot shows the 'IP Profiles [ACVC]' configuration window, continuing from the previous figure. The 'SBC EARLY MEDIA' section includes Remote Early Media (Supported), Remote Multiple 18x (Supported), Remote Early Media Response Type (Transparent), Remote Multiple Early Dialogs (According to Operation Mode), Remote Multiple Answers Mode (Disable), Remote Early Media RTP Detection Mode (By Signaling), Remote RFC 3960 Support (Not Supported), Remote Can Play Ringback (Yes), and Generate RTP (None). The 'SBC REGISTRATION' section includes User Registration Time (0), NAT UDP Registration Time (-1), and NAT TCP Registration Time (-1). The 'SBC FORWARD AND TRANSFER' section includes Remote REFER Mode (Regular), Remote Replaces Mode (Standard), and Play RBT To Transferee (No). At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 59: IP Profile for Amazon Chime Voice Connector Continuation

IP Profiles [ACVC]

SBC MEDIA		Remote 3xx Mode	Transparent
Mediation Mode	RTP Mediation	SBC HOLD	
Extension Coders Group	#0 [AudioCodersGroups_0]	Remote Hold Format	Transparent
Allowed Audio Coders	#0 [G711] View	Reliable Held Tone Source	Yes
Allowed Coders Mode	Restriction	Play Held Tone	No
Allowed Video Coders	-- View	SBC FAX	
Allowed Media Types		Fax Coders Group	--
Direct Media Tag		Fax Mode	As Is
RFC 2833 Mode	As Is	Fax Offer Mode	All coders
RFC 2833 DTMF Payload Type	0	Fax Answer Mode	Single coder
Alternative DTMF Method	As Is	Remote Renegotiate on Fax Detection	Transparent
Send Multiple DTMF Methods	Disable	Fax Rerouting Mode	Disable
Adapt RFC2833 BW to Voice coder BW	Disabled		
SDP Ptime Answer	Remote Answer		

Cancel **APPLY**

Figure 60: IP Profile for Amazon Chime Voice Connector Continuation

IP Profiles [ACVC]

SDP Ptime Answer	Remote Answer	Fax Rerouting Mode	Disable
Preferred PTime	0	MEDIA	
Use Silence Suppression	Transparent	Broken Connection Mode	Disconnect
RTP Redundancy Mode	As Is	Media IP Version Preference	Only IPv4
RTCP Mode	Transparent	RTP Redundancy Depth	Disable
Jitter Compensation	Disable	LOCAL TONES	
ICE Mode	Disable	Local RingBack Tone Index	-1
SDP Handle RTCP	Don't Care	Local Held Tone Index	-1
RTCP Mux	Not Supported		
RTCP Feedback	Feedback Off		
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-op	No		
Enhanced PLC	Disable		

Cancel **APPLY**

Figure 61: IP Profile for Amazon Chime Voice Connector Continuation

IP Profiles [ACVC]

QUALITY OF SERVICE

RTP IP DiffServ: 46

Signaling DiffServ: 24

JITTER BUFFER

Dynamic Jitter Buffer Minimum Delay [msec]: 10

Dynamic Jitter Buffer Optimization Factor: 10

Jitter Buffer Max Delay [msec]: 300

VOICE

Echo Canceled: Line

Input Gain (-32 to 31 dB): 0

Voice Volume (-32 to 31 dB): 0

Cancel APPLY

Figure 62: IP Profile for Amazon Chime Voice Connector Continuation

4.3.9 IP-to-IP Routing

Navigate to **'SETUP'** and select **'SIGNALING & MEDIA'**. Expand **'SBC'** and select **'IP-to-IP Routing'**. Routing rules are defined for forwarding SIP messages based on IP Groups from source to destination.

IP-to-IP Routing (3)

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	OPTIONS	Default_SBCR	Route Row	Any	OPTIONS	*	*	Dest Address	--	--	internal
1	SFB to ACVC	Default_SBCR	Route Row	SFB_IP_Grp	All	*	*	IP Group	ACVC_IP_Grp	--	
2	ACVC to SFB	Default_SBCR	Route Row	ACVC_IP_Grp	All	*	*	IP Group	SFB_IP_Grp	--	

Figure 63: IP-to-IP Routing

IP to IP routing for OPTIONS message

#0[OPTIONS] Edit

GENERAL	
Name	• OPTIONS
Alternative Rout...	Route Row
MATCH	
Source IP Group	• Any View
Request Type	• OPTIONS
Source Userna...	*
Source Host	*
Source Tag	
Destination Use...	*
Destination Host	*
Destination Tag	
Message Condi...	-- View
Call Trigger	Any
ReRoute IP Group	• Any View
ACTION	
Destination Type	• Dest Address
Destination IP ...	-- View
Destination SIP ...	-- View
Destination Ad...	• internal
Destination Port	0
Destination Tra...	
IP Group Set	-- View
Call Setup Rule...	-1
Group Policy	Sequential
Cost Group	-- View
Routing Tag Na...	default
Internal Action	

Figure 64: IP-to-IP Routing for OPTIONS

IP to IP routing from Skype for Business to Amazon Chime Voice Connector.

#1[SFB to ACVC] Edit

GENERAL	
Name	• SFB to ACVC
Alternative Rout...	Route Row
MATCH	
Source IP Group	• SFB_IP_Grp View
Request Type	All
Source Userna...	*
Source Host	*
Source Tag	
Destination Use...	*
Destination Host	*
Destination Tag	
Message Condi...	-- View
Call Trigger	Any
ReRoute IP Group	• Any View
ACTION	
Destination Type	IP Group
Destination IP ...	• ACVC_IP_Grp View
Destination SIP ...	-- View
Destination Ad...	
Destination Port	0
Destination Tra...	
IP Group Set	-- View
Call Setup Rule...	-1
Group Policy	Sequential
Cost Group	-- View
Routing Tag Na...	default
Internal Action	

Figure 65: IP-to-IP Routing from Skype for Business to Amazon Chime Voice Connector

IP to IP routing from Amazon Chime Voice Connector to Skype for Business.

#2[ACVC to SFB] Edit

GENERAL		ACTION	
Name	• ACVC to SFB	Destination Type	IP Group
Alternative Rout...	Route Row	Destination IP ...	• SFB_IP_Grp View
		Destination SIP ...	-- View
		Destination Ad...	
		Destination Port	0
		Destination Tra...	
		IP Group Set	-- View
		Call Setup Rule...	-1
		Group Policy	Sequential
		Cost Group	-- View
		Routing Tag Na...	default
		Internal Action	

MATCH	
Source IP Group	• ACVC_IP_Grp View
Request Type	All
Source Userna...	*
Source Host	*
Source Tag	
Destination Use...	*
Destination Host	*
Destination Tag	
Message Condi...	-- View
Call Trigger	Any
ReRoute IP Group	• Any View

Figure 66: IP-to-IP Routing from Amazon Chime Voice Connector to Skype for Business

4.3.10 TLS Configuration

TLS is configured between AudioCodes CE and Amazon Chime Voice Connector. Navigate to 'SETUP' and select 'IP NETWORK'. Expand 'SECURITY' and click on 'TLS Contexts'.

MEDIANT SW	IP NETWORK	SIGNALING & MEDIA	ADMINISTRATION	Entity, parameter, value										
SRD All														
NETWORK VIEW CORE ENTITIES SECURITY TLS Contexts (1) Firewall (0) Security Settings		TLS Contexts (1) + New Edit Page 1 of 1 Show 10 records per page <table> <tr> <th>INDEX</th><th>NAME</th><th>TLS VERSION</th><th>DTLS VERSION</th><th>CIPHER SERVER</th></tr> <tr> <td>0</td><td>default</td><td>TLSv1.2</td><td>Any</td><td>DEFAULT</td></tr> </table>			INDEX	NAME	TLS VERSION	DTLS VERSION	CIPHER SERVER	0	default	TLSv1.2	Any	DEFAULT
INDEX	NAME	TLS VERSION	DTLS VERSION	CIPHER SERVER										
0	default	TLSv1.2	Any	DEFAULT										

Figure 67: TLS Context list

Figure 68: TLS Context for Amazon Chime Voice Connector

Amazon Trust Root Certificate is to be installed in the Trusted Root Certificates list under TLS Context. In the TLS Context page, select the *TLS Context* for Amazon Chime Voice Connector and click '*Trusted Root Certificates*' link located in the bottom. Click on *Import* button and select the certificate file.

Figure 69: Trusted Root Certificate Import option

Amazon Chime Voice Connector Root Certificate can be downloaded from Amazon Chime Voice Connector account.

To configure media security, navigate to '*SETUP*' and select '*SIGNALING & MEDIA*'. Expand '*MEDIA*' and click on '*Media Security*'. Under General section, set *Media Security* as Enable.

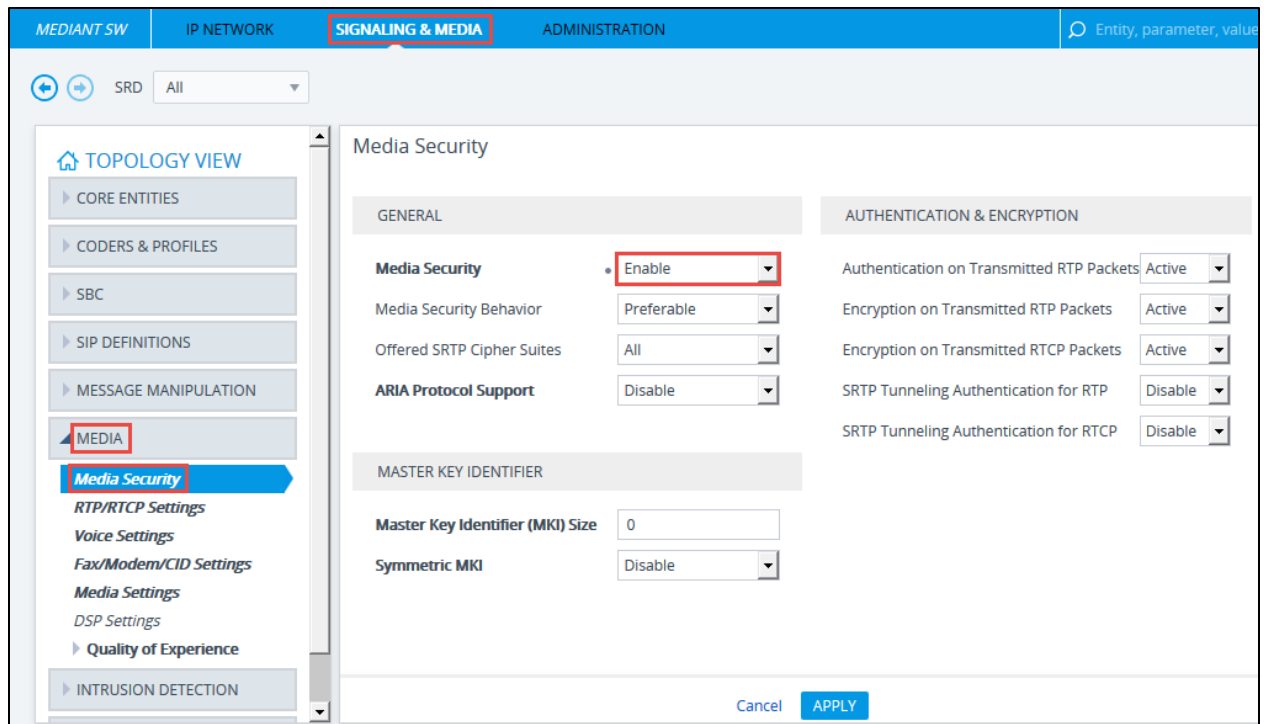


Figure 70: Media Security

In the IP Profile for Amazon Chime Voice Connector SRTP has to be enabled.

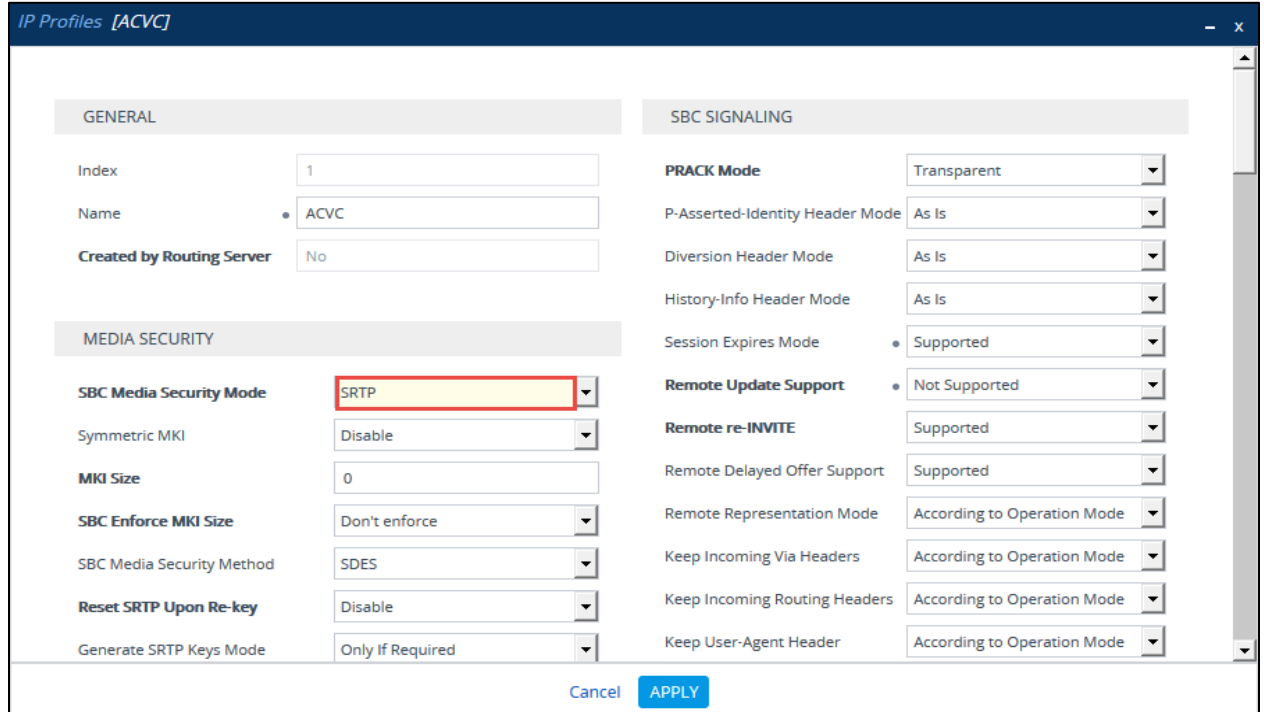


Figure 71: SRTP option in IP Profile

4.3.11 Message Manipulation configuration

SIP message manipulation rules are created to modify SIP headers for each IP entity based on manipulation sets enabled in IP Groups. The following are the message manipulation created for interoperability between Skype for Business and Amazon Chime Voice Connector.

#0[Change From header towards SFB] Edit

GENERAL		ACTION	
Name	• Change From header towards SFB	Action Subject	• Header.From.URL.Host
Manipulation Set ID	• 2	Action Type	• Modify
Row Role	Use Current Condition	Action Value	• ' '

MATCH

Message Type	• Any.Request
Condition	

Figure 72: From header modification Skype for Business

#1[Change From header towards AVSC] Edit

GENERAL		ACTION	
Name	• Change From header towards AVSC	Action Subject	• Header.From.URL.Host
Manipulation Set ID	• 4	Action Type	• Modify
Row Role	Use Current Condition	Action Value	• ' '

MATCH

Message Type	• Any.Request
Condition	

Figure 73: From header Modification Amazon Chime Voice Connector

#2[Change OPTIONS RURI towards ACVC] Edit

GENERAL	
Name	• Change OPTIONS RURI towards ACVC
Manipulation Set ID	• 4
Row Role	Use Current Condition

ACTION	
Action Subject	• Header.Request-URI.URL.Host
Action Type	• Modify
Action Value	• 'cr7c1zxzy'

MATCH	
Message Type	• Options
Condition	• Param.Message.address.dst.SIPInterface=='1'

Figure 74: OPTIONS RURI modification

#3[Change OPTIONS to URI towards ACVC-To] Edit

GENERAL	
Name	• Change OPTIONS to URI towards ACVC-To
Manipulation Set ID	• 4
Row Role	Use Current Condition

ACTION	
Action Subject	• Header.To.URL.Host
Action Type	• Modify
Action Value	• 'cr7c1zxzy'

MATCH	
Message Type	• Options
Condition	• Param.Message.Address.Dst.SIPInterface=='1'

Figure 75: OPTIONS To header modification

#4[Change PAI towards ACVC] Edit

GENERAL	
Name	• Change PAI towards ACVC
Manipulation Set ID	• 4
Row Role	Use Current Condition

ACTION	
Action Subject	• Header.P-Asserted-Identity.URL.Host
Action Type	• Modify
Action Value	• ' ' ,

MATCH	
Message Type	• Invite.Request
Condition	• Header.P-Asserted-Identity exists

Figure 76: PAI header modification

#5[Add UPDATE support] Edit

GENERAL	
Name	• Add UPDATE support
Manipulation Set ID	• 2
Row Role	Use Current Condition
MATCH	
Message Type	• invite
Condition	• header.Allow lexists

ACTION	
Action Subject	• Header.Allow
Action Type	Add
Action Value	• 'UPDATE'

Figure 77: Add Allow header-UPDATE

#6[Modify Session Expires] Edit

GENERAL	
Name	• Modify Session Expires
Manipulation Set ID	• 2
Row Role	Use Current Condition
MATCH	
Message Type	• Invite.Response.2xx
Condition	• Header.Session-Expires exists

ACTION	
Action Subject	• Header.Session-Expires.Time
Action Type	• Modify
Action Value	• '900'

Figure 78: Session Expires Timer modification