



# **Crestron UC-PHONE and UC-PHONE-PLUS**

**Connecting Microsoft Teams  
Direct Routing using AudioCodes  
Mediant Virtual Edition (VE),  
Avaya Aura v7.1 and  
Skype for Business 2015(Hybrid)**

**September 2019**

## Document History

<b>Rev. No.</b>	<b>Date</b>	<b>Document Owner</b>	<b>Description</b>
1.0	Sep-18-2019	tekVizion	Draft - Configuration Guide
1.1	Sep-23-2019	tekVizion	Updated after Internal Review

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# 1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring **Crestron UC-PHONE and UC-PHONE-PLUS with Microsoft Teams Direct Routing using AudioCodes Mediant VE SBC, Avaya Aura v7.1 as Customer PBX and Skype for Business 2015 (Hybrid)**.

## 1.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services help service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

For more information on tekVizion and its practice areas, please visit tekVizion Labs website at [www.tekVizion.com](http://www.tekVizion.com)

## 2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representation of Crestron UC-PHONE and UC-PHONE-PLUS connected O365 Cloud with Microsoft Teams Direct Routing to Avaya Aura v7.1 environment and Skype for Business 2015(Hybrid) using AudioCodes Mediant VE SBC and PSTN Gateway for PSTN connectivity. Media bypass enables Configured teams side used in this topology.

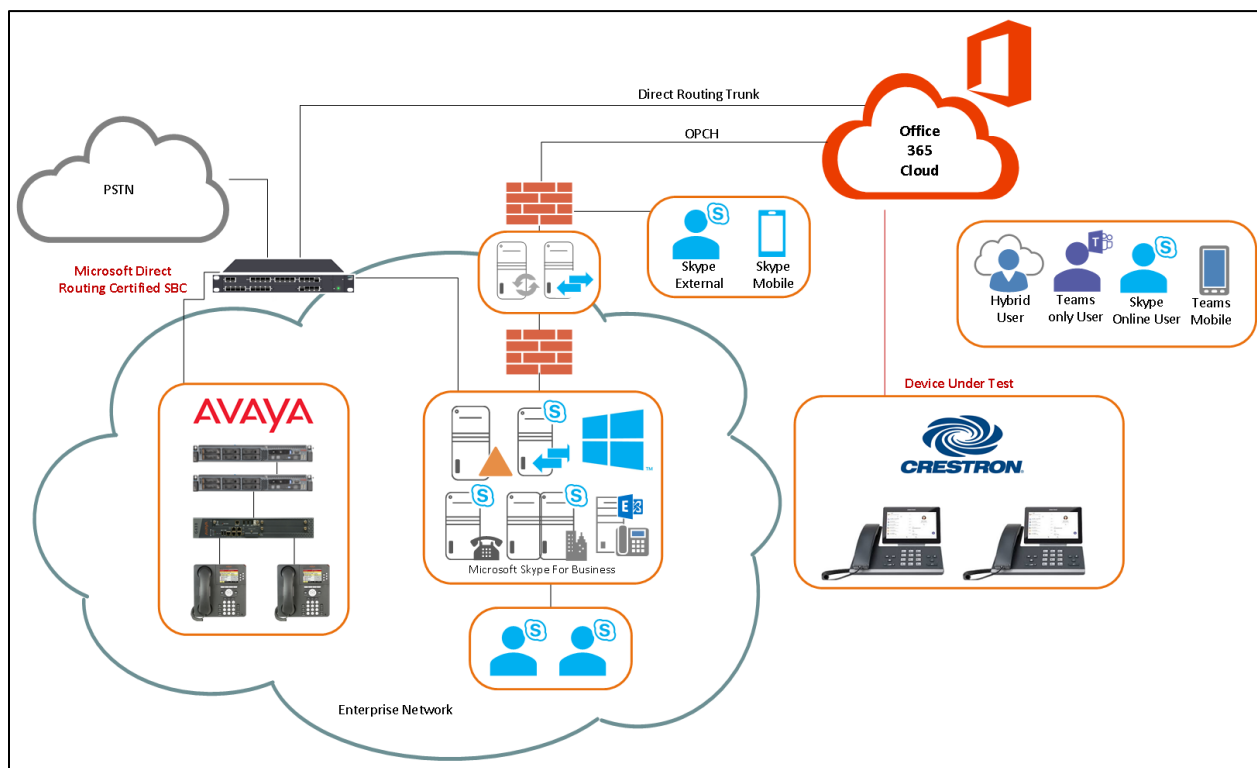


Figure 1 Network Topology

### Numbering Plan

- Avaya users are configured with 4 digit extension 75XX
- Teams users are configured with E164 numbers +197259809XX
- Skype for Business Server 2015 with E164 numbers +197259809XX

### Dialing Plan

- Teams users and Avaya users call PSTN either doing 10 digits 11 digits dialing or E164 dialing
- Teams users call Avaya users by dialing 75XX
- Avaya users call Teams users by dialing 8XXX and AudioCodes will include the prefix +1972XXX and will send to Teams.
- Teams user calls Pure on-prem user via SIP URI dialing
- On-Prem user calls Teams user via SIP URI dialing

## 2.1 Hardware Components

- Microsoft Office 365 tenant with E5 without Audio Conferencing assigned to Teams users
- AudioCodes Mediant VE SBC for Teams Direct Routing serves as the demarcation point between customer's network and O365 WAN network
- Crestron UC-PHONE-PLUS and Crestron UC-PHONE phones
- Skype For Business 2015 on System running Windows 2012 R2
- Avaya Aura Communication Manager Configuration
- Avaya Aura Session Manager Configuration
- Avaya SBCE Configuration
- PSTN Gateway

## 2.2 Software Requirements

- AudioCodes Mediant VE SBC v7.20A.250.003
- Skype For Business 2015 Version (6.0.9319)
- Avaya Aura Communication Manager Configuration v7.1.3.1
- Avaya Aura Session Manager Configuration v7.1.3.1
- Avaya SBCE Configuration v7.1.3.1
- Crestron UC-PHONE-PLUS v58.15.91.15

# 3 Features

## 3.1 Features Supported

- Basic Inbound and Basic Outbound
- Call hold and resume
- Call transfer (semi-attended and consultative)
- Conference
- Call forward (all, no answer)
- Busy On Busy



- Simultaneous ring
- Calling line identification restriction
- DTMF relay both directions (RFC2833)

### 3.2 Caveats and Limitations

- Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license.
- The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.

## 4 Configuration

### 4.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure **Microsoft Teams, Avaya SBCE, Avaya Aura Session Manager, Avaya Aura Communication Manager, Skype for Business 2015 and AudioCodes** for SIP Trunking with **Microsoft Teams Direct Routing**.

*Table 1 – PBX Configuration Steps*

Steps	Description	Reference
Step 1	Microsoft Skype for Business Hybrid Configuration	<a href="#">Section 4.3</a>
Step 1	Microsoft Teams Configuration	<a href="#">Section 4.3</a>
Step 2	AudioCodes VE SBC Configuration	<a href="#">Section 4.4</a>
Step 3	Avaya Aura Communication Manager	<a href="#">Section 4.5</a>
Step 4	Avaya Aura Session Manager	<a href="#">Section 4.6</a>
Step 5	Avaya SBCE	<a href="#">Section 4.7</a>

### 4.2 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Component	Lab Value
<b>AudioCodes</b>	
LAN IP Address	10.64.3.10
LAN Subnet Mask	255.255.255.0
WAN IP Address	192.XX.XX.XX
WAN Subnet Mask	255.255.255.128
<b>Avaya Aura Communication Manager</b>	
IP Address	10.89.26.4 (Signaling)/10.89.26.14 (Media)
Subnet Mask	255.255.255.0
<b>Avaya Aura Session Manager</b>	
LAN IP Address	10.89.26.7
LAN Subnet Mask	255.255.255.0
<b>Avaya SBCE</b>	
LAN IP Address	10.89.17.3
LAN Subnet Mask	255.255.255.0
WAN IP Address	10.64.5.57
WAN Subnet Mask	255.255.255.0
<b>Skype for Business 2015</b>	
Skype for Business - FQDN	Fe0101.tekvizionlabs.com
Edge Server - FQDN	accessedge02.tekvizionlabs.com
Exchange - FQDN	exum.tekvizionlabs.com

*Table 2 – IP Addresses*

### 4.3 Microsoft Skype for Business Hybrid Configuration

Configure hybrid connectivity between Skype for Business Server and Teams or Skype for Business Online. Hybrid connectivity enables you to move your on-premises users to Teams or Skype for Business Online, and enables your users signed in Teams using Crestron UC-PHONE to take advantage of cloud services. The scenario assumes with this guide that an Edge Server is already in production and operational.

A SIP address (Session Initiation Protocol) is an identifier that must be unique for each user. In Hybrid mode, it is necessary to configure the Office365 tenant in a shared mode for the SIP domain used with Skype for Business 2015 on-premises.

#### 4.3.1 Create tenant account for Office 365

Follow these steps to set up an Office 365 Enterprise tenant if customer does not have one set up already.

1. Navigate to <https://www.microsoft.com/en-us/microsoft-365/>
2. Select the O365 Plan - O365 tend to fall into 5 main categories: Small Business, Midsize Business, Enterprise, Education and Government. Most of these categories have trial accounts, and all of these can be converted to regular licensed accounts if required.
3. Enter the Correct information - **Once set up, the Tenant account name cannot be changed.** When administrator first creates the Tenant account, it will be in the form of .onmicrosoft.com, but administrator can add in and use your own registered domain name once the Tenant account is created.
4. Complete the sign-in process by validating the text message or phone call.

#### 4.3.2 Add on-prem domain to O365

To add, modify or remove domains the engineer must be a Global Administrator of a business or enterprise plan. These changes affect the whole tenant, customized administrators or regular users won't be able to make these changes.

1. Go to the admin center at <https://admin.microsoft.com>.
2. Go to the **Setup** → **Domains** page.
3. Select Add domain and enter the domain name to be added. **tekvizionlabs.com** is of the domain used for this setup
4. Select **Next** and **Finish**
5. Wait at least five minutes for replication to complete.

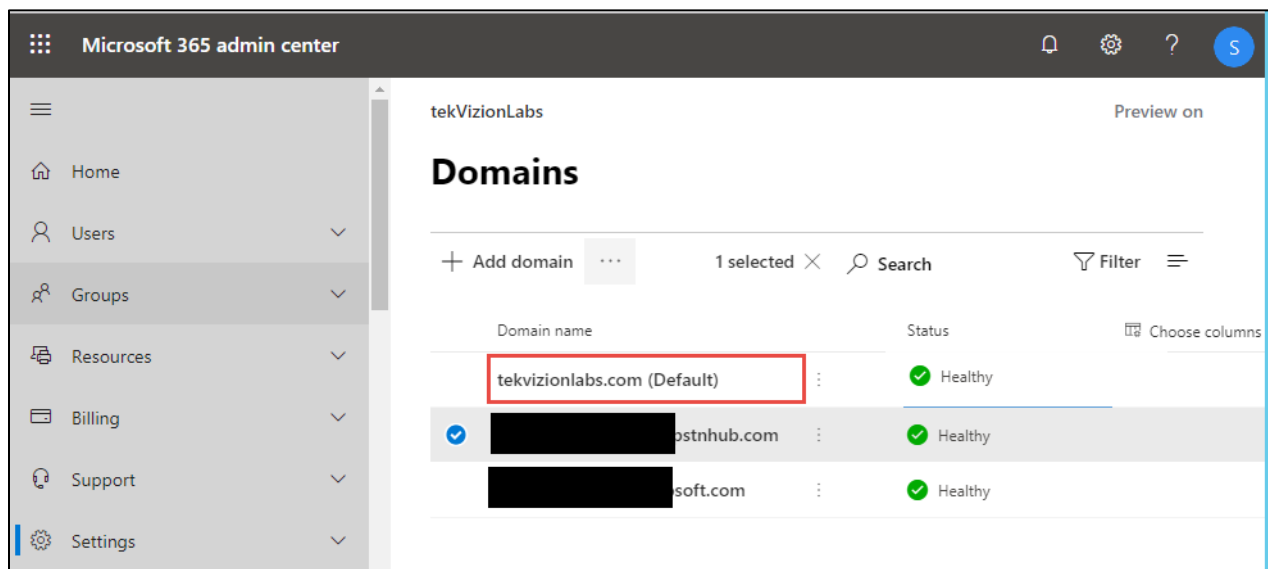


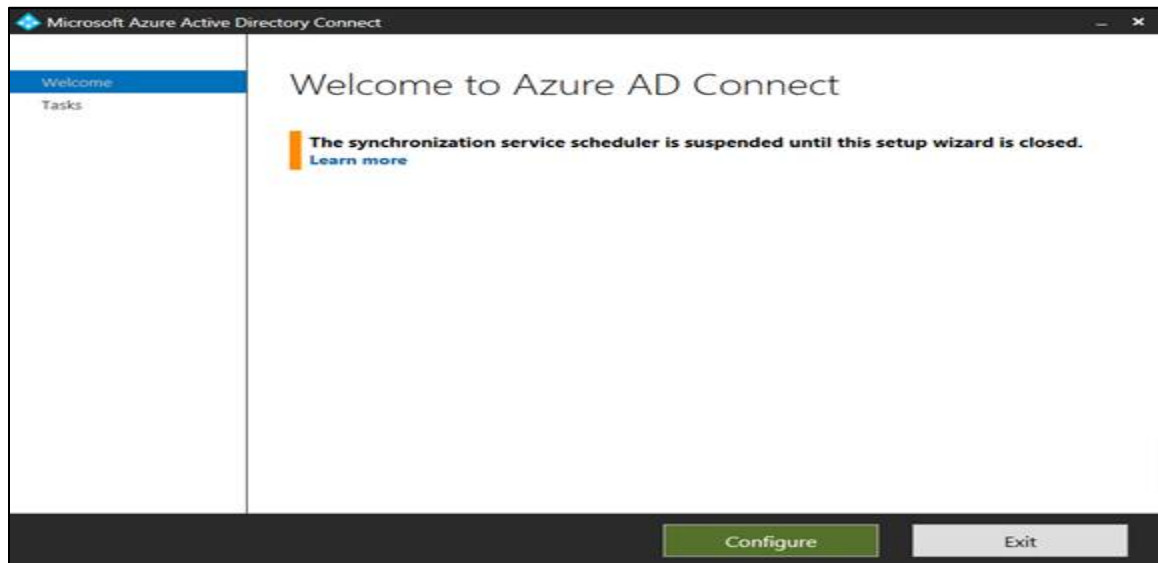
Figure 2 – Add Domain

### 4.3.3 Setup AD synchronization

Active Directory synchronization keeps on-premises Active Directory continuously synchronized with Office 365. This lets you to create a synchronized version of each user account and group.

The configuration steps below guides the administrator to setup the Azure AD Connect tool downloaded from Microsoft Site and provision in the on-prem server. On the Connect to Azure AD screen, use a global admin account and password.

Recommendation is to use an account in the default **onmicrosoft.com** domain, which comes with Azure AD tenant.



*Figure 3 - Azure AD - Sync Process*

1. On the Connect to Azure AD screen, use a global admin account and password. Recommendation is to use an account in the default **onmicrosoft.com** domain, which comes with Azure AD tenant.

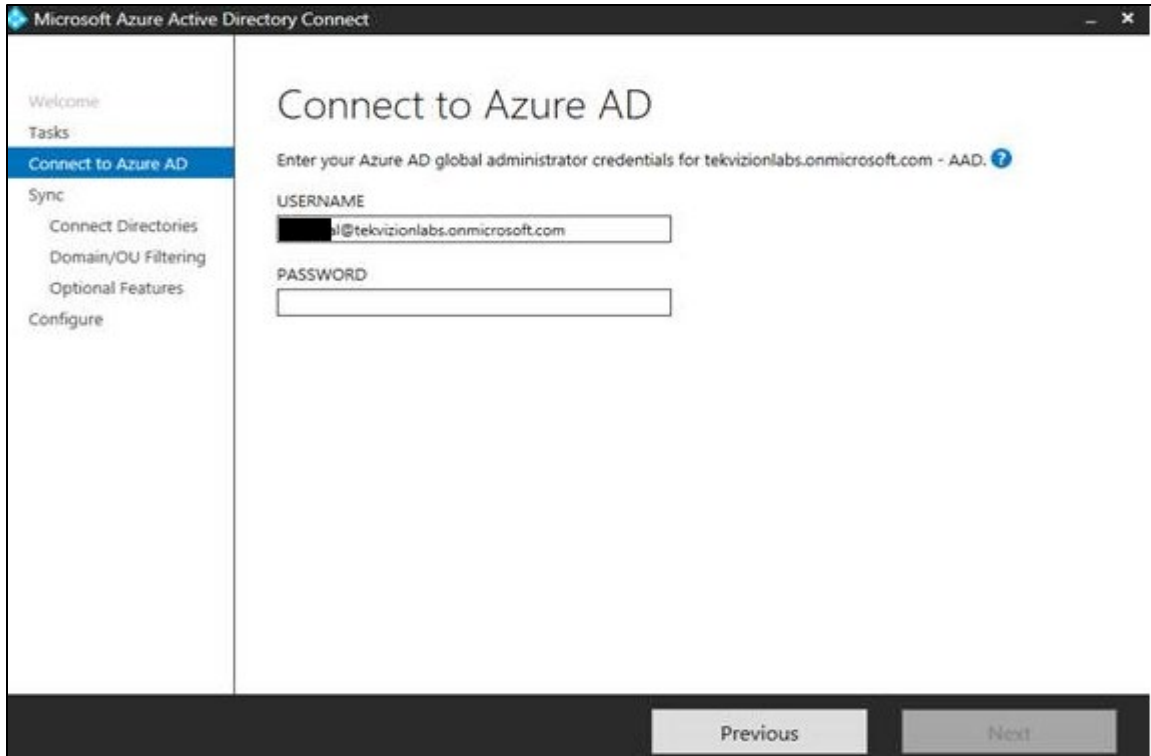


Figure 4 – Azure AD – Sync Process

2. Select **Active Directory** Type
3. Select **tekvizionlabs.com** from **Forest** Drop Down

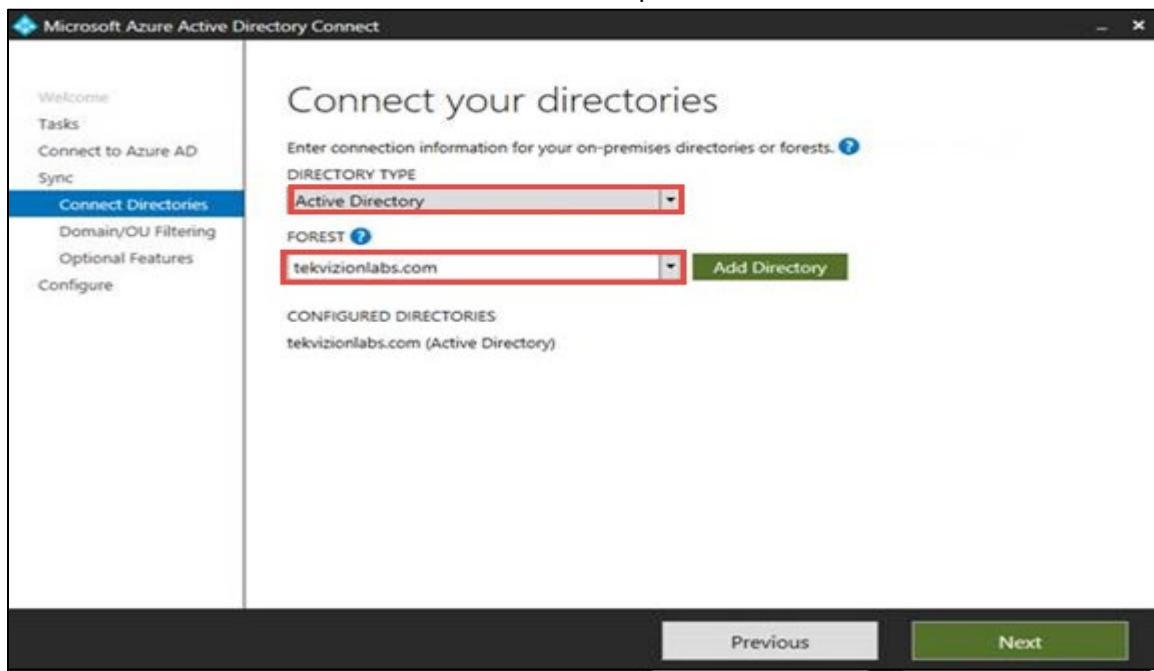


Figure 5 – Azure AD – Sync Process

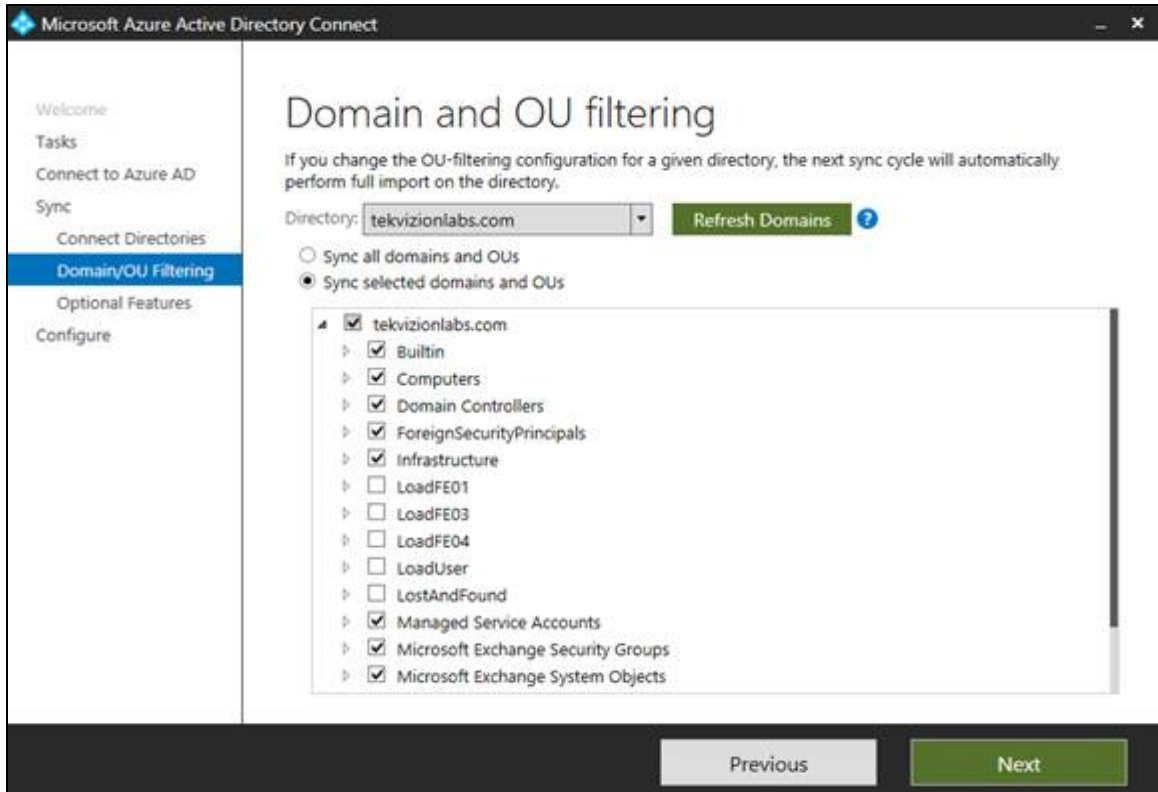


Figure 6 – Azure AD – Sync Process

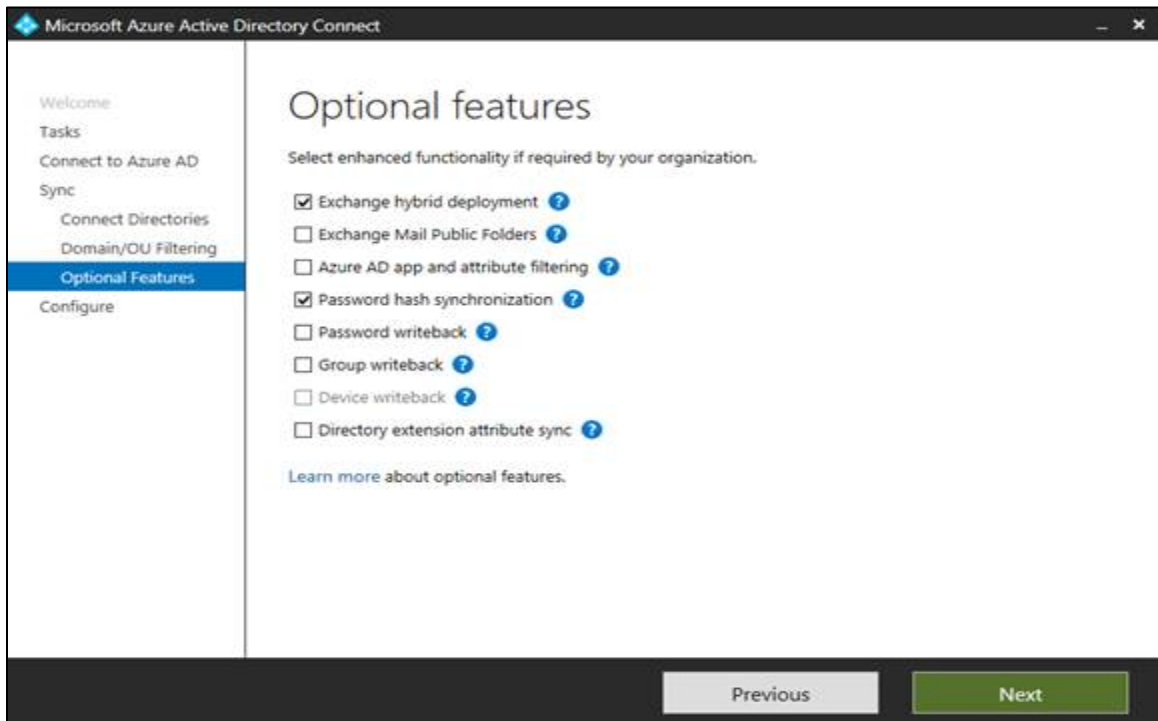


Figure 7 – Azure AD – Sync Process

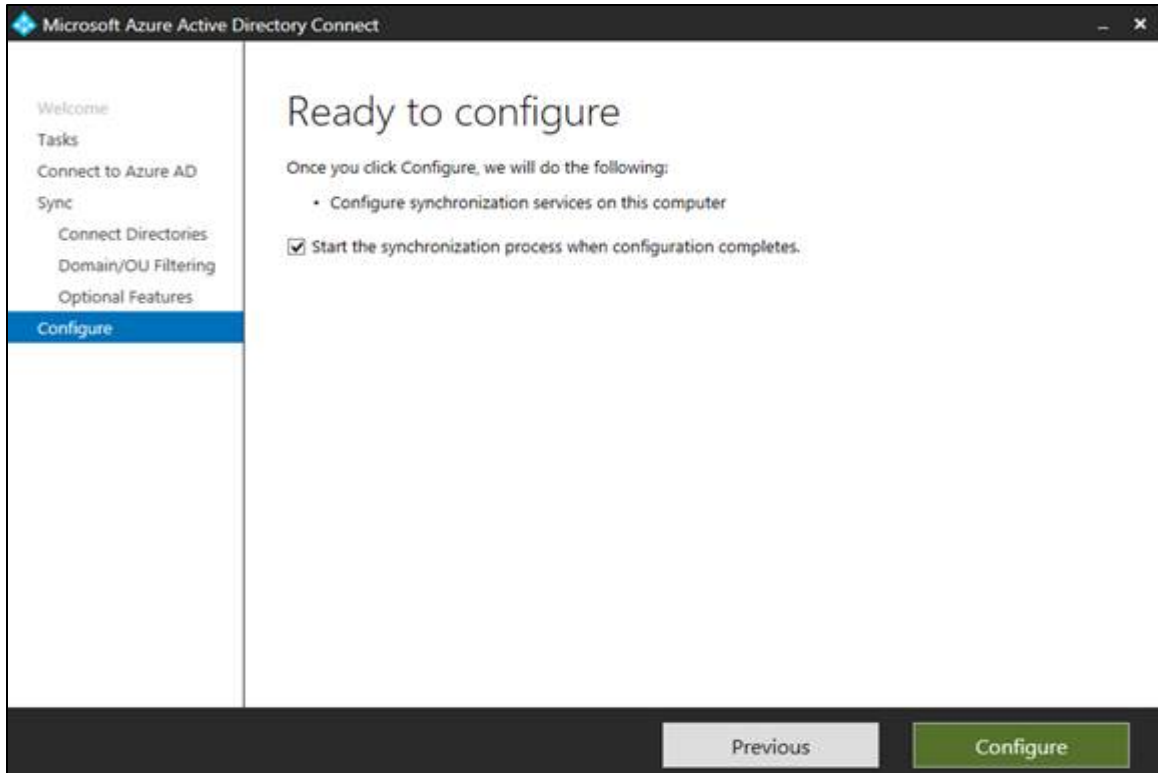


Figure 8 – Azure AD – Sync Process

4. Once the Synchronization to O365 completed successfully, login to <https://www.office.com/>
5. Select O365 **Admin**

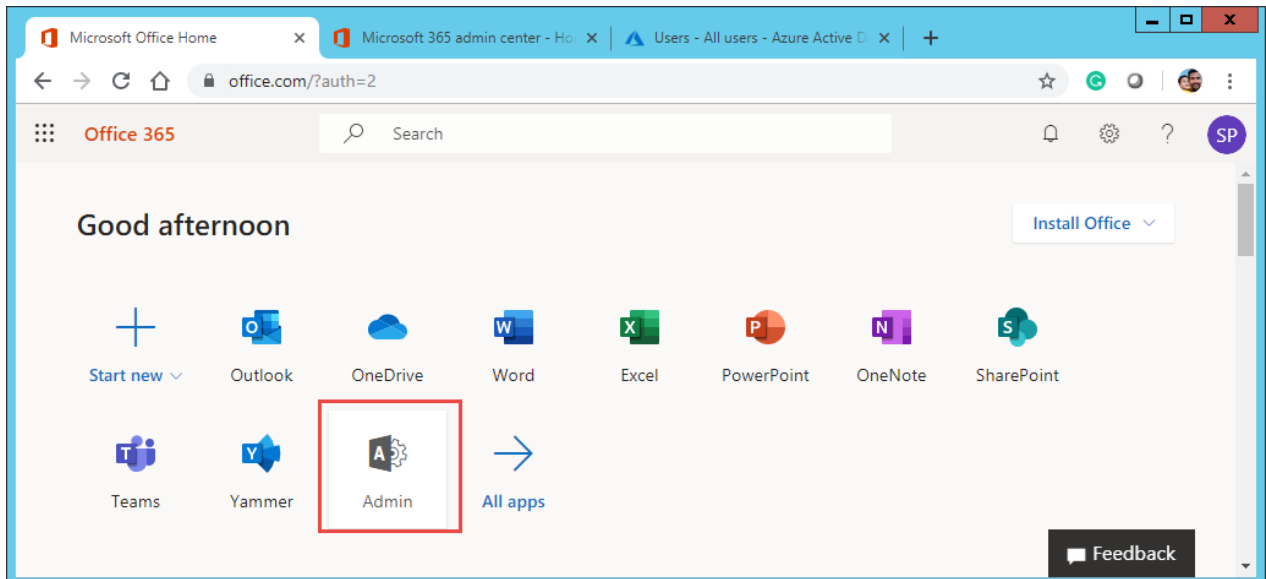


Figure 9 – Azure AD – Sync Process

6. Select **Azure Active Directory** from Microsoft 365 admin center

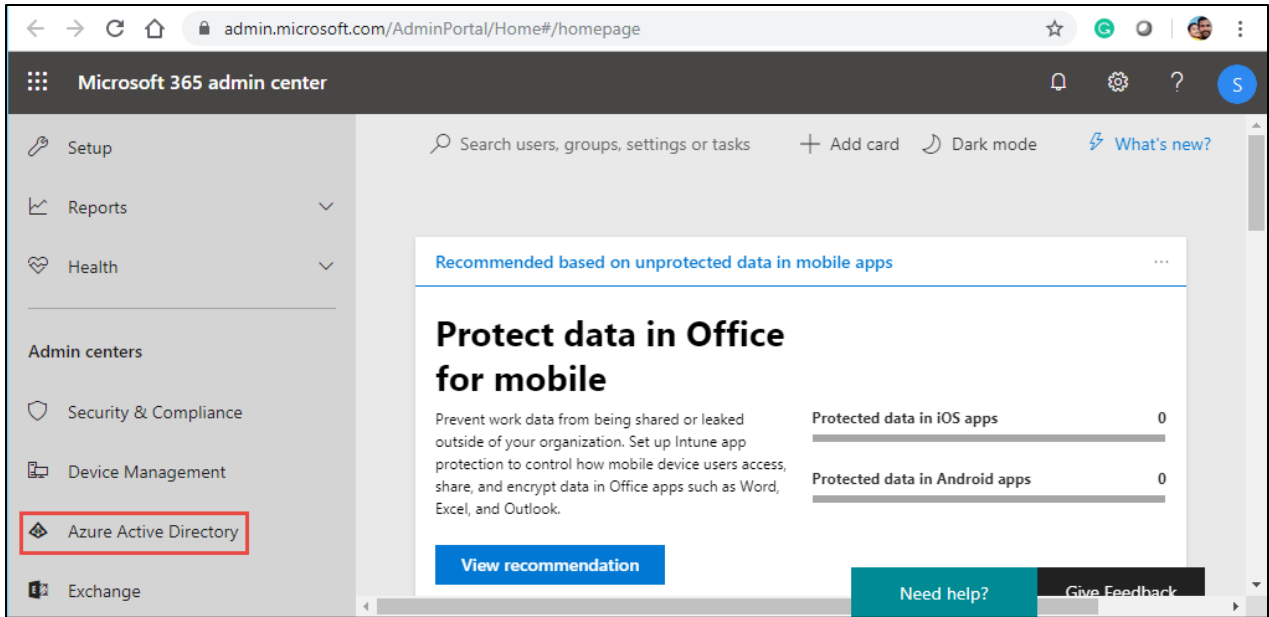


Figure 10 – Azure AD – Sync Process

7. **Azure Active Directory admin center** window will appear as below, search the users to make sure the on-prem users are listed

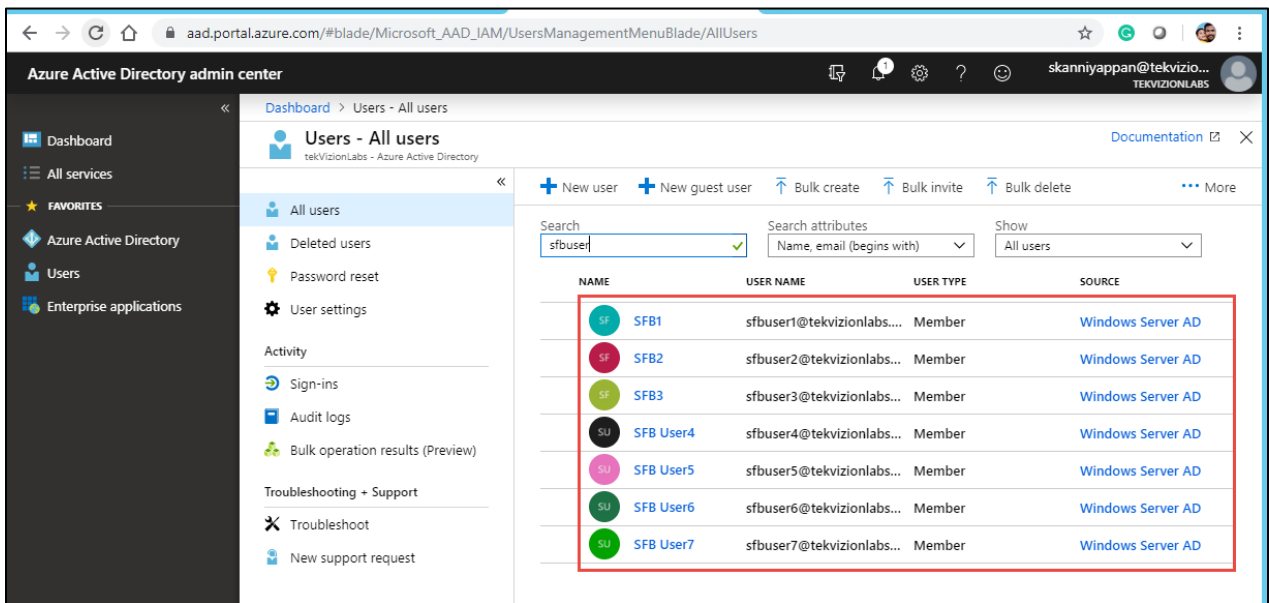


Figure 11 – Azure AD – Sync Process

#### 4.3.4 Configure skype for Business hybrid

The below steps shows how to configure hybrid connectivity between Skype for Business Server and Teams or Skype for Business Online. Hybrid connectivity enables the ability to



move on-premises users to Teams or Skype for Business Online, and enable users to take advantage of cloud services.

#### 4.3.4.1 Configure your on-premises Edge service to federate with Office 365

Federation allows users in on-premises deployment to communicate with Office 365 users in organization. To configure federation, run the following cmdlet in the Skype for Business Server Management Shell:

#### **Set-CsAccessEdgeConfiguration -AllowOutsideUsers 1 -AllowFederatedUsers 1 - EnablePartnerDiscovery 1 -UseDnsSrvRouting**

```
PS C:\Users\administrator.TEKVIZIONLABS> Get-CsAccessEdgeConfiguration

Identity                : Global
AllowAnonymousUsers     : True
AllowFederatedUsers     : True
AllowOutsideUsers       : True
BeClearingHouse         : False
EnablePartnerDiscovery  : True
DiscoveredPartnerVerificationLevel : UseSourceVerification
EnableArchivingDisclaimer : True
EnableUserReplicator    : False
KeepCr1sUpToDateForPeers : True
MarkSourceVerifiableOnOutgoingMessages : True
OutgoingTlsCountForFederatedPartners : 4
DnsSrvCacheRecordCount  : 131072
DiscoveredPartnerStandardRate : 20
EnabledDiscoveredPartnerContactsLimit : True
MaxContactsPerDiscoveredPartner : 1000
DiscoveredPartnerReportPeriodMinutes : 60
MaxAcceptedCertificatesStored : 1000
MaxRejectedCertificatesStored : 500
CertificatesDeletedPercentage : 20
SkypeSearchUrl          : https://skypegraph.skype.com/search/v1.0
RoutingMethod           : UseDnsSrvRouting

PS C:\Users\administrator.TEKVIZIONLABS>
```

Figure 12 – Edge Federation

#### 4.3.4.2 Configure your on-premises environment to enable shared SIP address space with Office 365

Configure your on-premises environment to trust Office 365 and enable shared SIP address space with Office 365. Office 365 can potentially host user accounts for the same set of SIP domains as your on-premises environment, and messages can be routed between users hosted on premises and online. This is achieved by configuring a hosting provider with **ProxyFqdn=sipfed.online.lync.com** as described below.

Create a new hosting provider using the **New-CsHostingProvider** cmdlet as follows:

#### **New-CsHostingProvider -Identity Office365 -ProxyFqdn "sipfed.online.lync.com" - Enabled \$true -EnabledSharedAddressSpace \$true -HostsOCSUsers \$true -**

**VerificationLevel UseSourceVerification -IsLocal \$false -AutodiscoverUrl  
https://webdir.online.lync.com/Autodiscover/AutodiscoverService.svc/root**

The below is the output taken from on-prem Skype for Business Server

```
PS C:\Users\administrator.TEKVIZIONLABS> Get-CsHostingProvider
Identity           : LyncOnline
Name               : LyncOnline
ProxyFqdn          : sipfed.online.lync.com
VerificationLevel  : AlwaysVerifiable
Enabled            : True
EnabledSharedAddressSpace : True
HostsOCSUsers      : True
IsLocal            : False
AutodiscoverUrl    : https://webdir1a.online.lync.com/Autodiscover/AutodiscoverService.svc/root

PS C:\Users\administrator.TEKVIZIONLABS>
```

Figure 13 – Hosting Provider

#### 4.3.4.3 Enable shared SIP address space in your Office 365 tenant

In addition to the change made in on-premises deployment, make the corresponding change in Office 365 tenant to enable shared SIP address space with on-premises deployment. A SIP address (Session Initiation Protocol) is an identifier that must be unique for each user. In Hybrid mode, it is necessary to configure the Office365 tenant in a shared mode for the SIP domain used with Skype for Business 2015 on-premises.

#### **Set-CsTenantFederationConfiguration -SharedSipAddressSpace \$true**

The SharedSipAddressSpace attribute needs to remain "True" until migration to online is completed, and no users remain on-premises.

#### 4.3.4.4 Skype for Business 2015 Hybrid Mode Configuration

Skype for Business control panel provides an option to configure the Hybrid Mode .

- 1) In Control Panel home page, select the link « Set up hybrid with Teams and Skype for Business Online

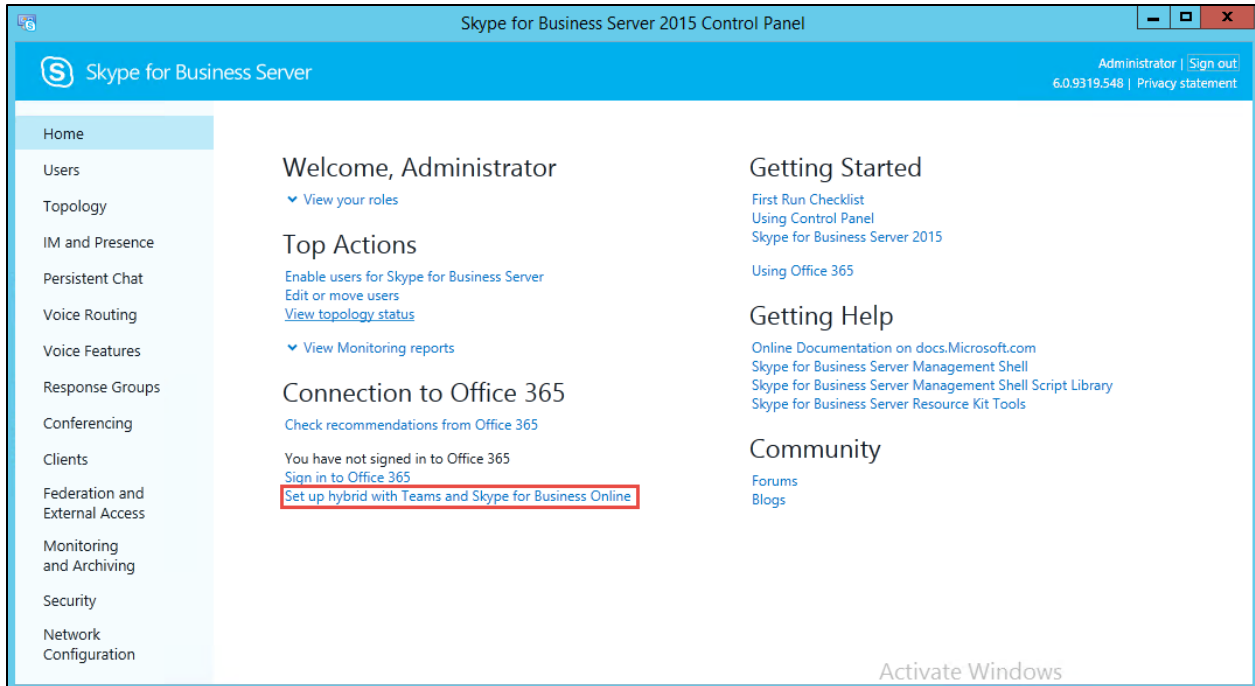


Figure 14 – Hybrid with Teams and SFB

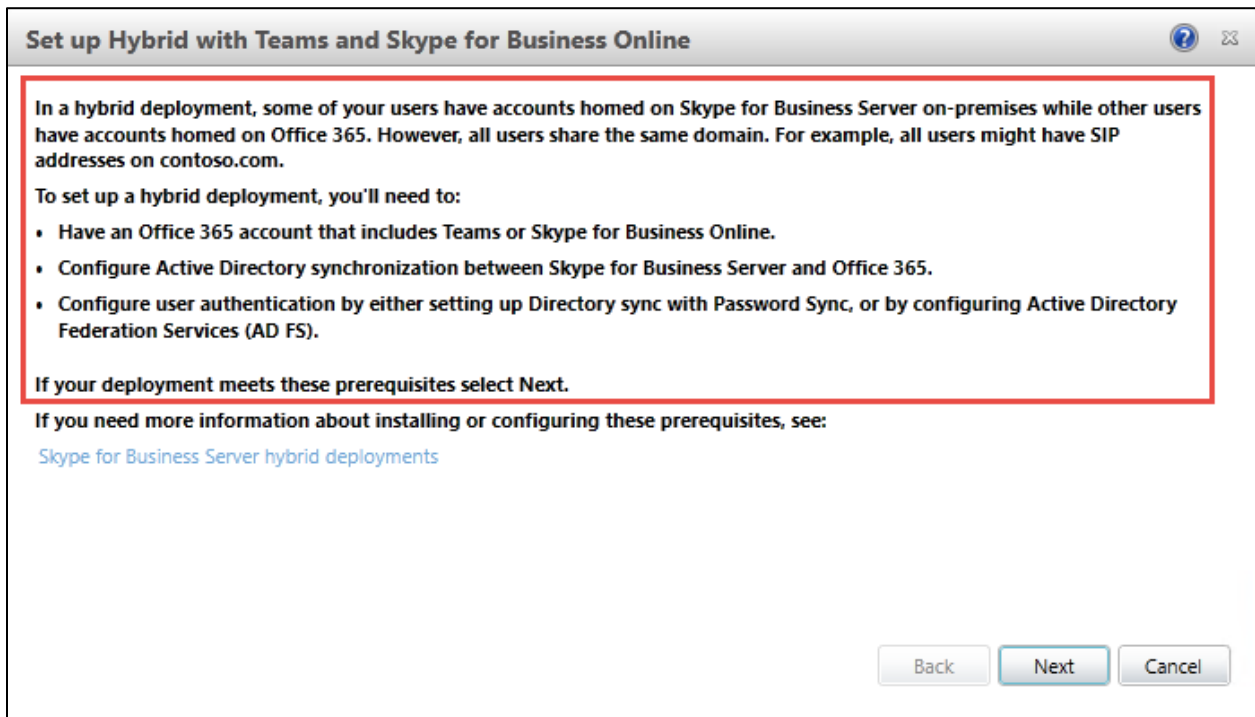


Figure 15 – Hybrid with Teams and SFB Contd.

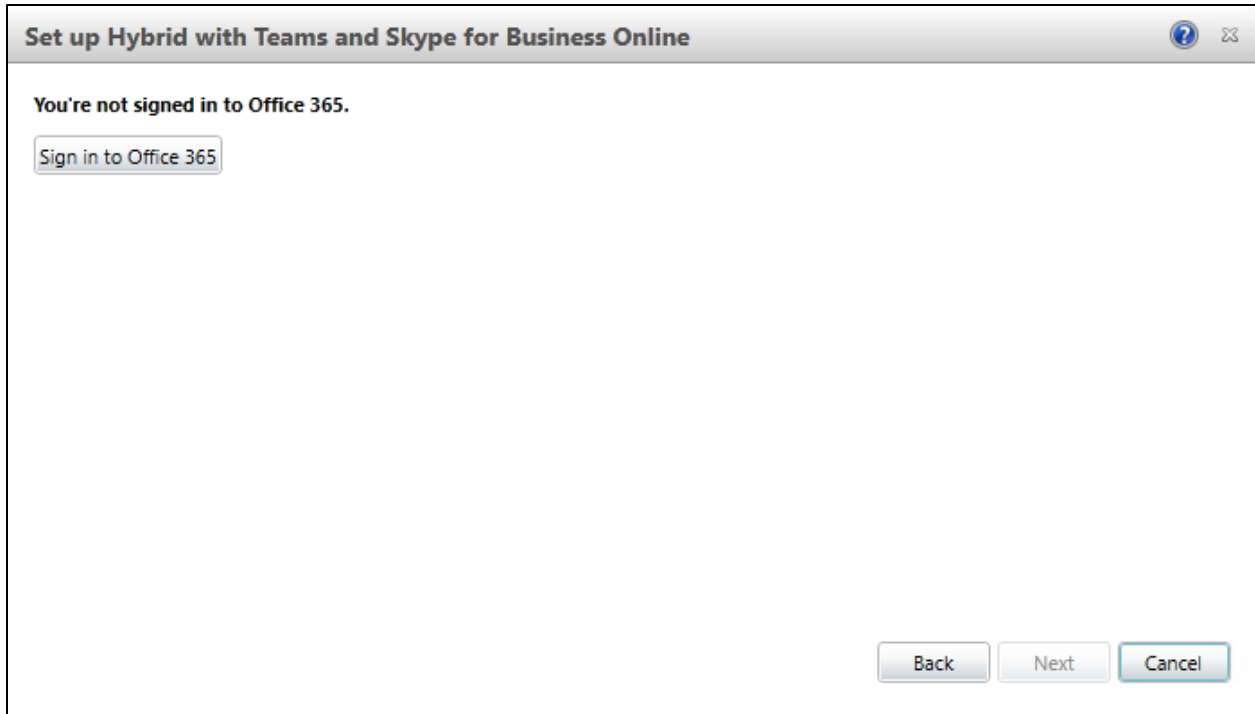


Figure 16 – Hybrid with Teams and SFB Contd.

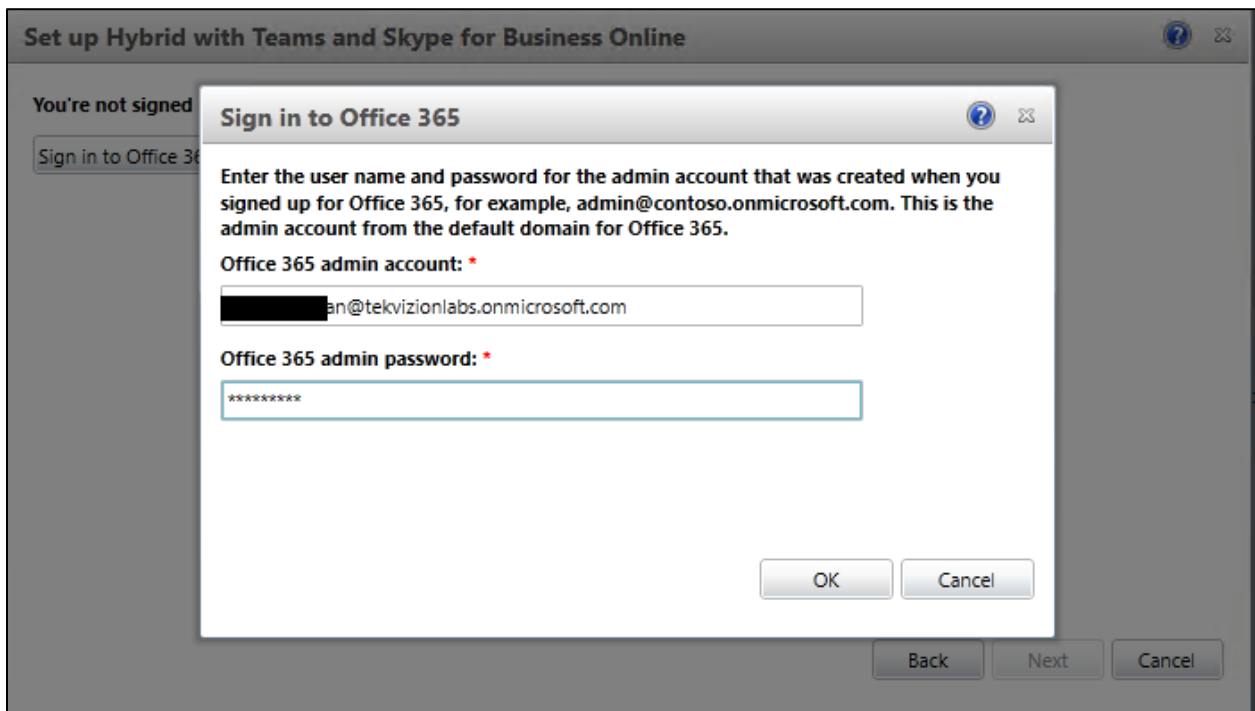


Figure 17 – Hybrid with Teams and SFB Contd.

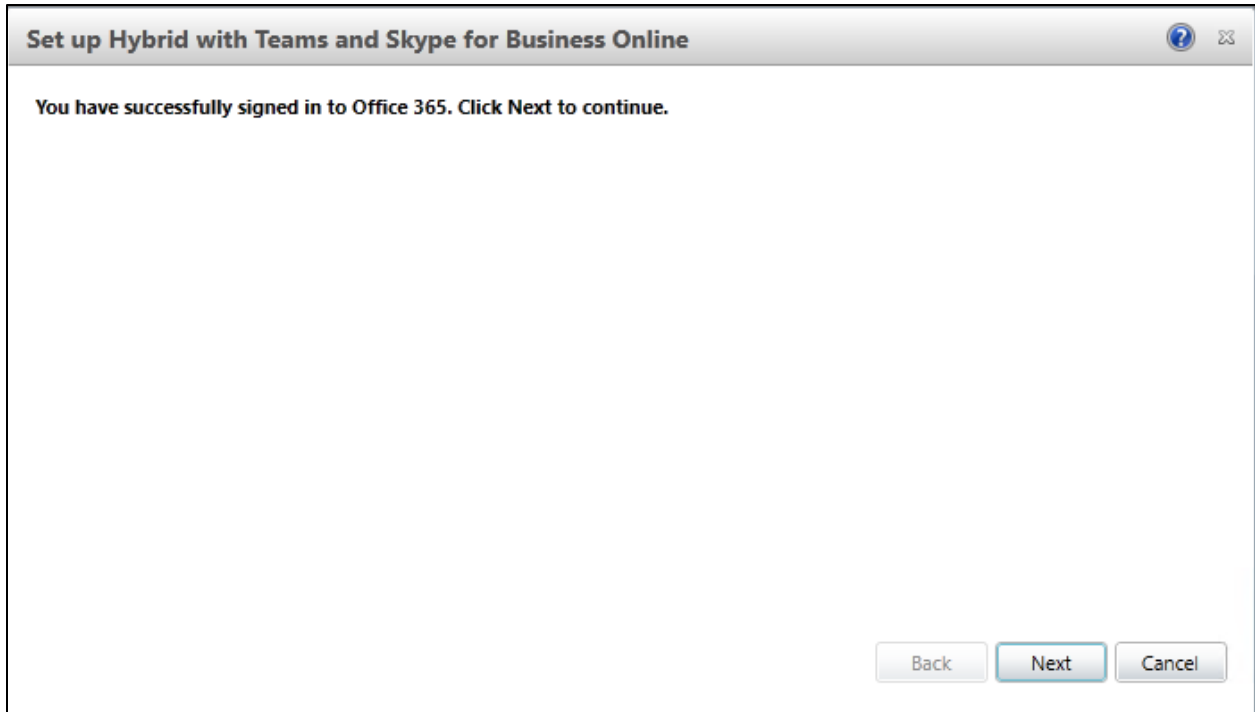


Figure 18 – Hybrid with Teams and SFB Contd.

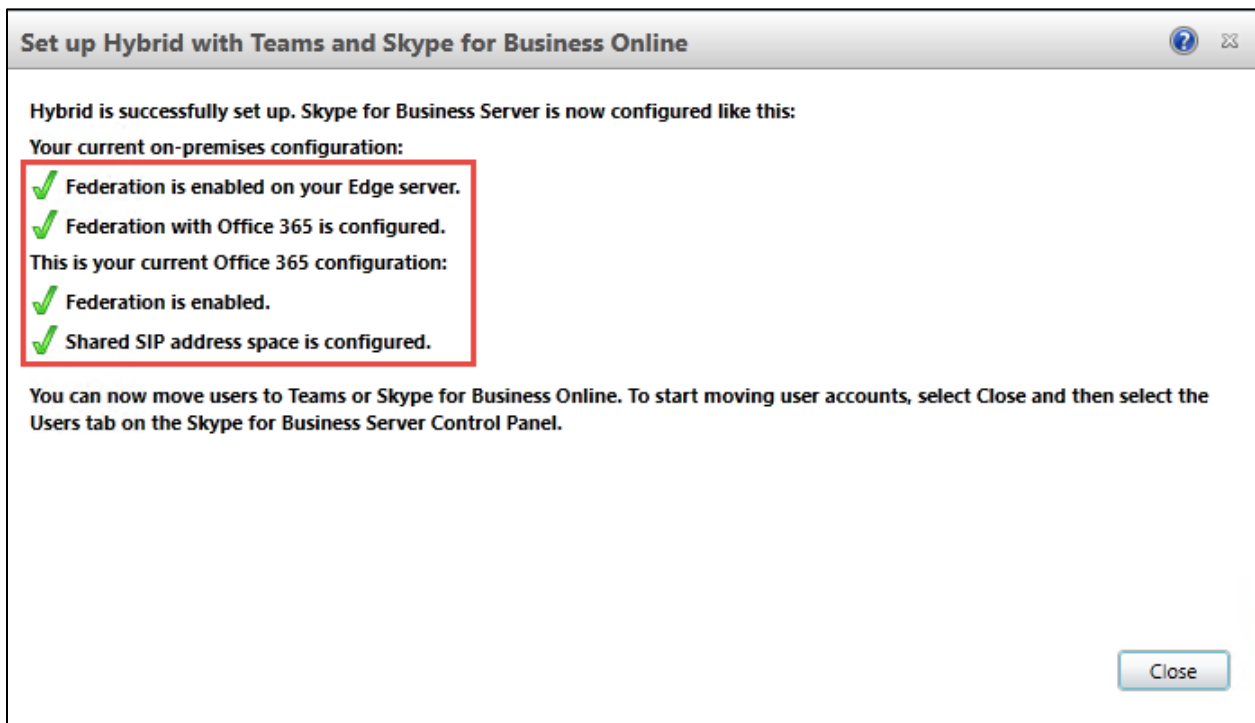


Figure 19 – Hybrid with Teams and SFB Contd.

#### 4.3.4.5 Move to Teams using Skype for Business Server Control Panel

In an on-premises deployment of Skype for Business Server that is enabled for hybrid, users can be moved between on-premises environment and cloud (either to Microsoft Teams or to Skype for Business Online).

When a user is moved online, the user is allowed to use Skype for Business Online or Teams Only or both (Islands mode). Microsoft strongly recommends that users moved online to be configured in Teams only mode, which will ensure that routing of all incoming chats and calls lands in their Teams client. This is configured in this setup.

1. Open the Skype for Business Server Control Panel app.
2. In the left navigation, choose **Users**.
3. Use **Find** to locate the user(s) you would like to move to Teams.
4. Select the user(s), and then, from the **Action** dropdown above the list, choose **Move selected users to Teams**.
5. In the wizard, click **Next**.
6. If prompted, sign in to Office 365, with an account that ends in .onmicrosoft.com and has sufficient permissions (tenant user with Global Admin role).
7. Click **Next**, and then **Next** one more time to move the user.
8. Note that status messages regarding success or failure are provided at the top of the main Control Panel app, not in the wizard.

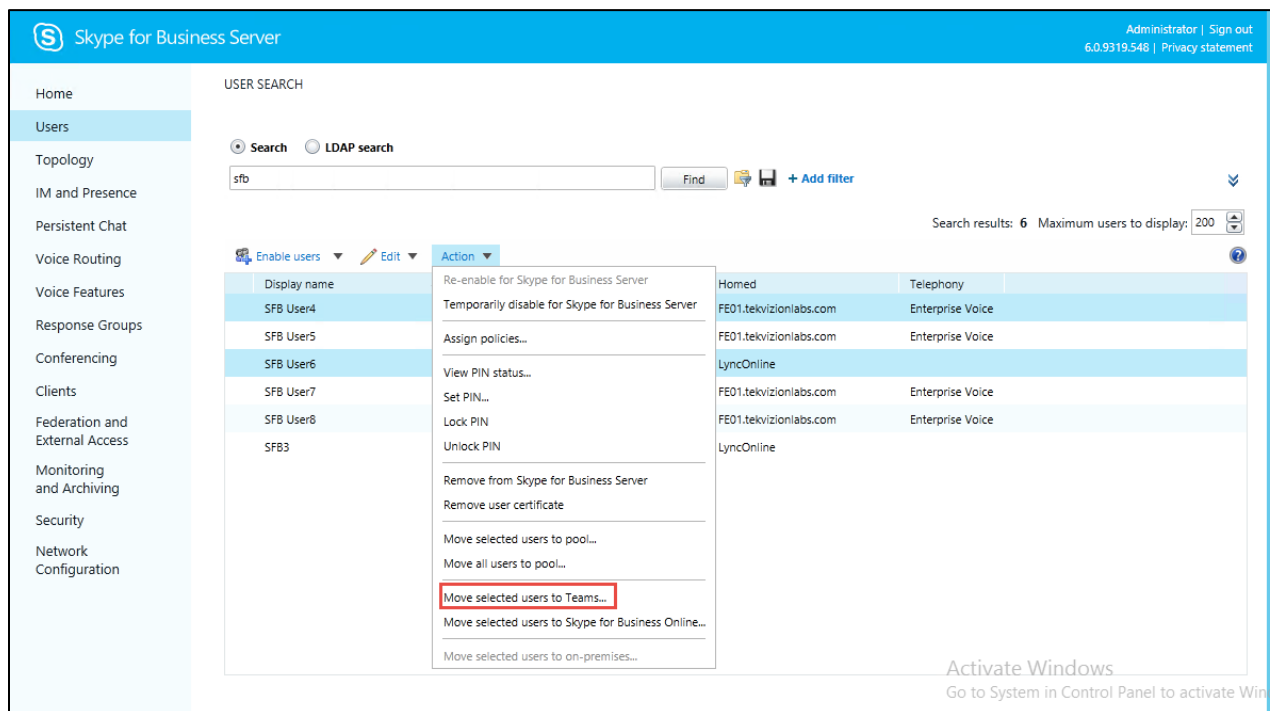
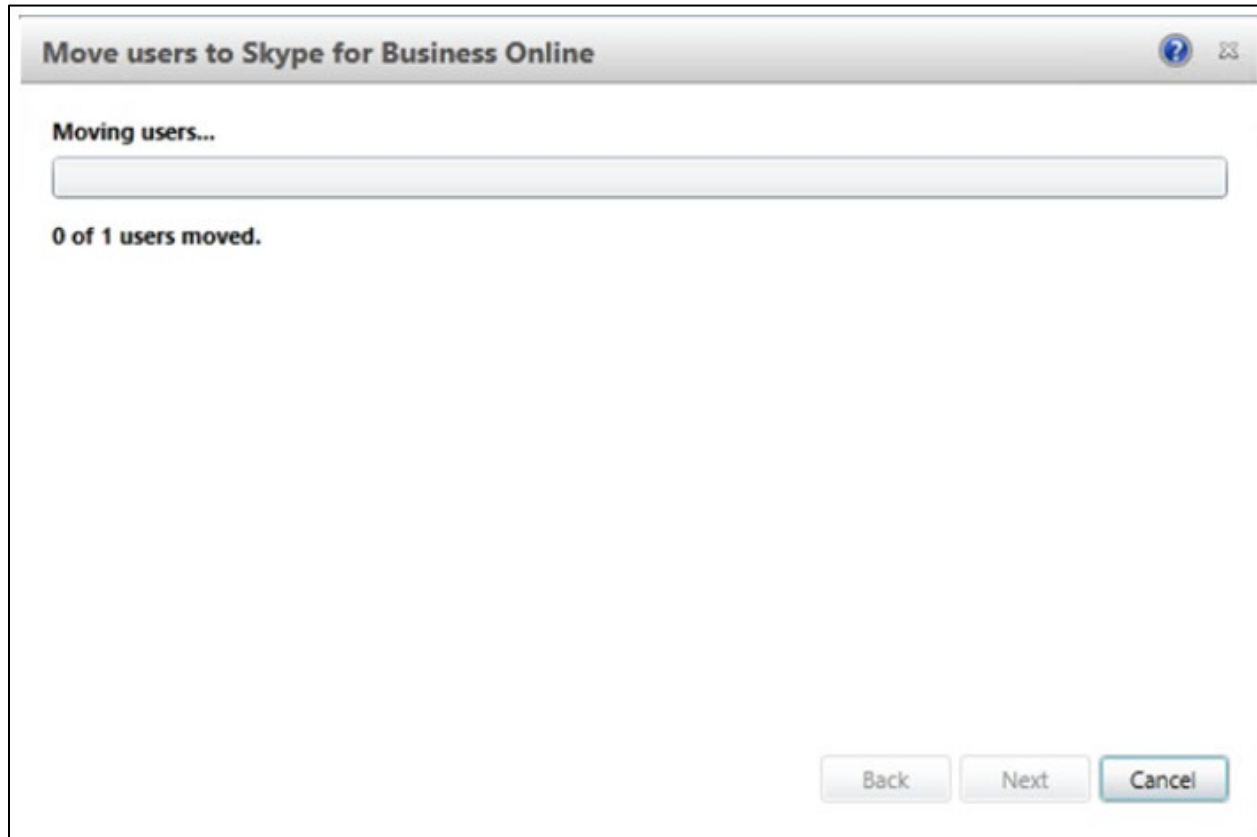


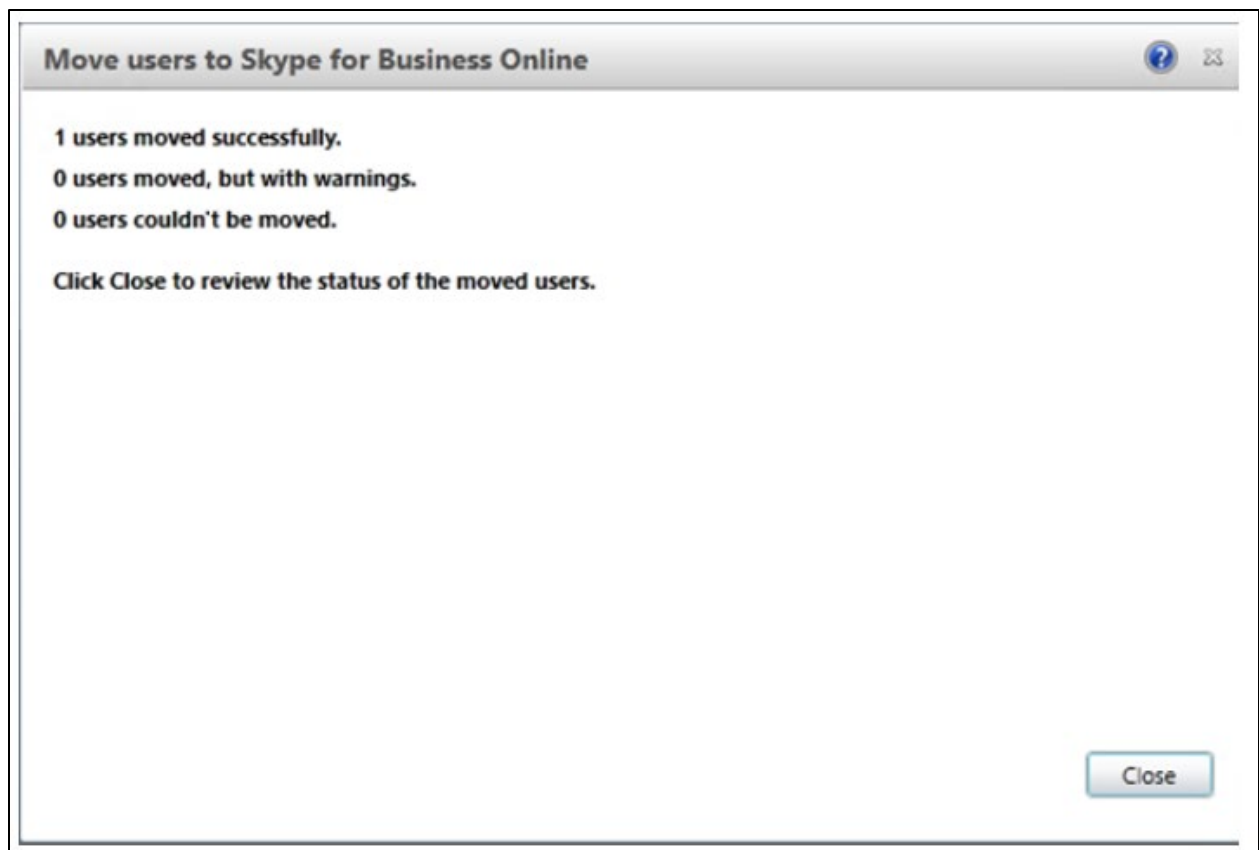
Figure 20 – Move Users to Teams

A window allows you to track the progress of your application to change the host to the selected user.



*Figure 21 - Move Users to Teams Contd.*

The migration of the selected user is complete.



*Figure 22 – Move Users to Teams Contd.*



## Administration of the user 'sfbuser6' from the on-site control panel

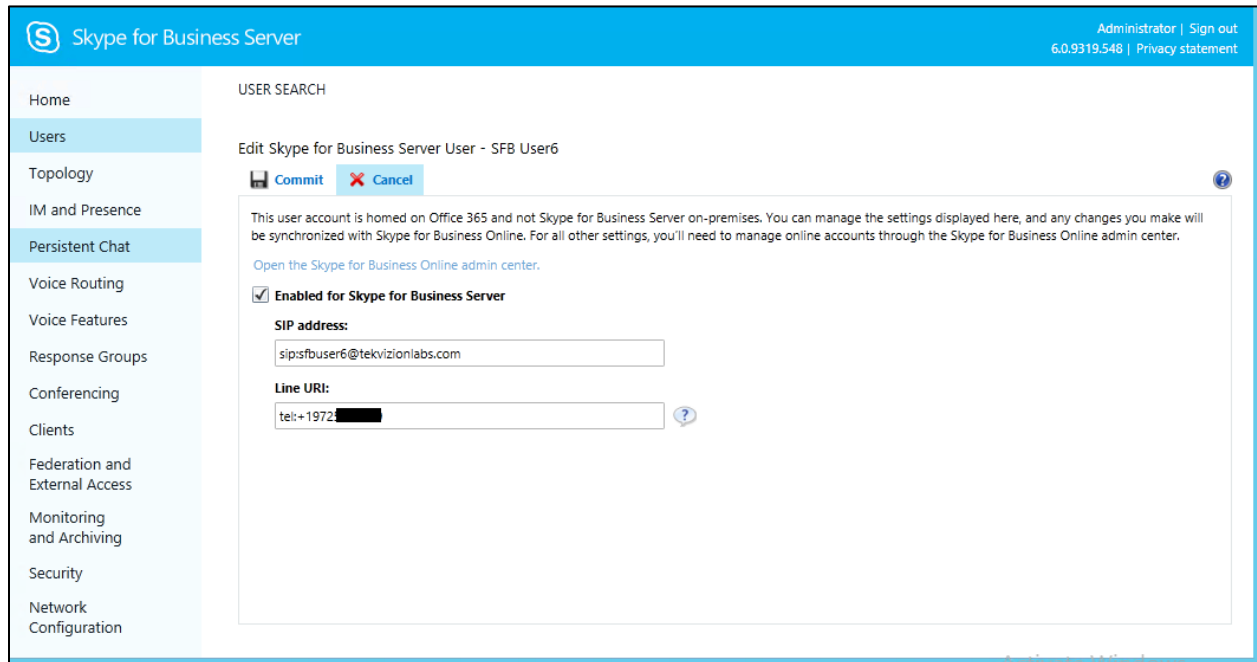


Figure 23 – Move Users to Teams

## 4.4 Microsoft Teams Configuration

This section with screen shots taken from Office 365 Portal and PowerShell Commands used for the interoperability testing gives a general overview of the Microsoft Teams Configuration.

### 4.4.1 Teams User Configuration

Below are the steps to create a pure online Teams user in office 365 portal.

1. Login into **http://portal.office.com/** using your office 365 tenant administrator credentials.

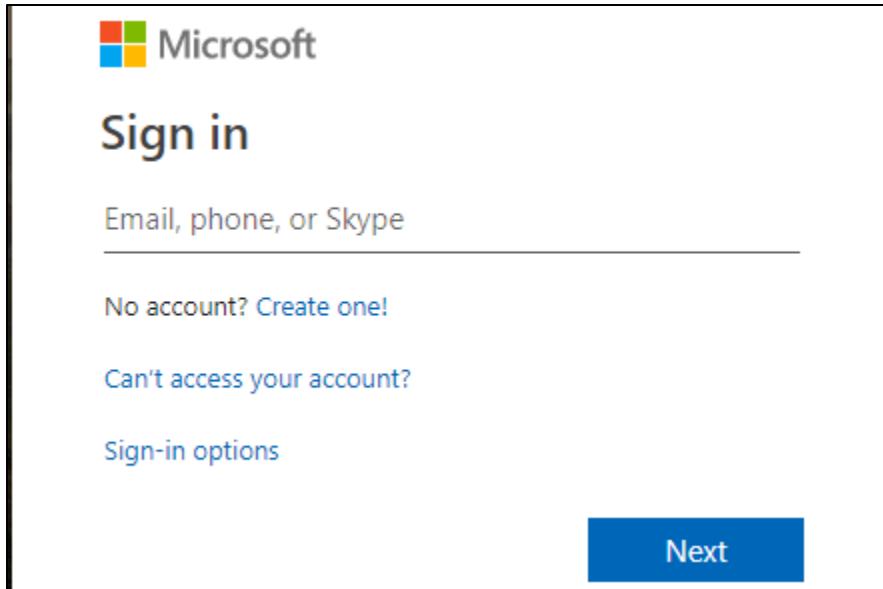


Figure 24: Office 365 Portal Login

2. Select the Office 365 Admin Icon to login Office 365 Admin Center as shown below.

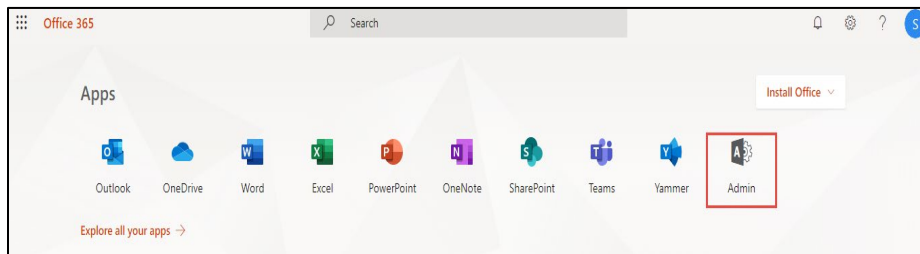


Figure 25: Office 365 Portal Login

3. Select "Add a user" from the Microsoft 365 Admin Center as shown below.

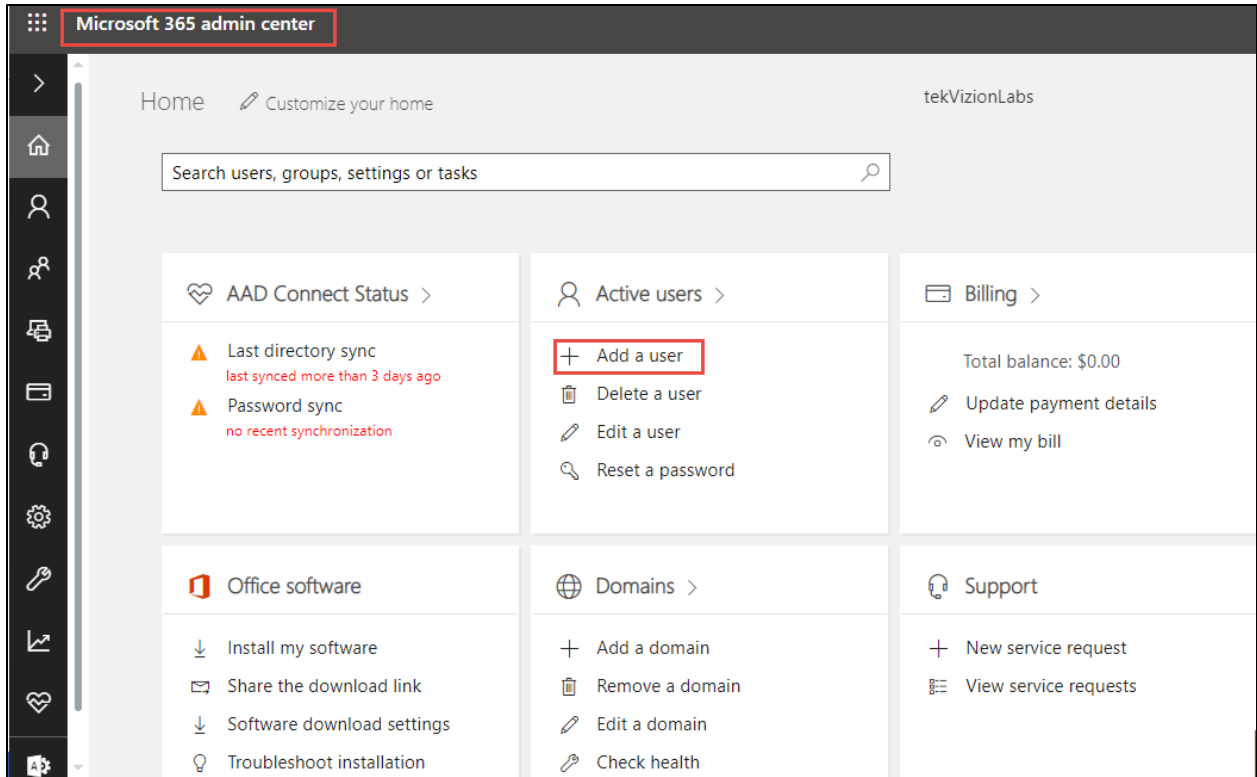


Figure 26: Teams User Creation

4. Enter the user details, password and assign required license to the users and Click Add

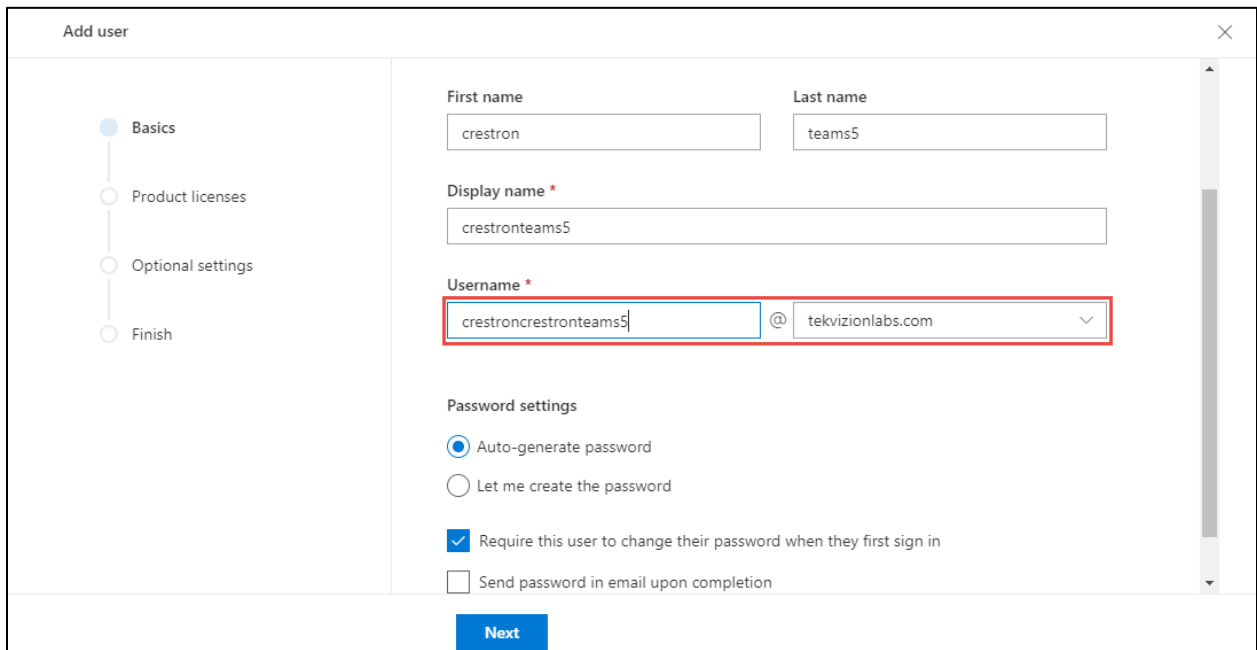


Figure 27: Teams User Creation – Contd.

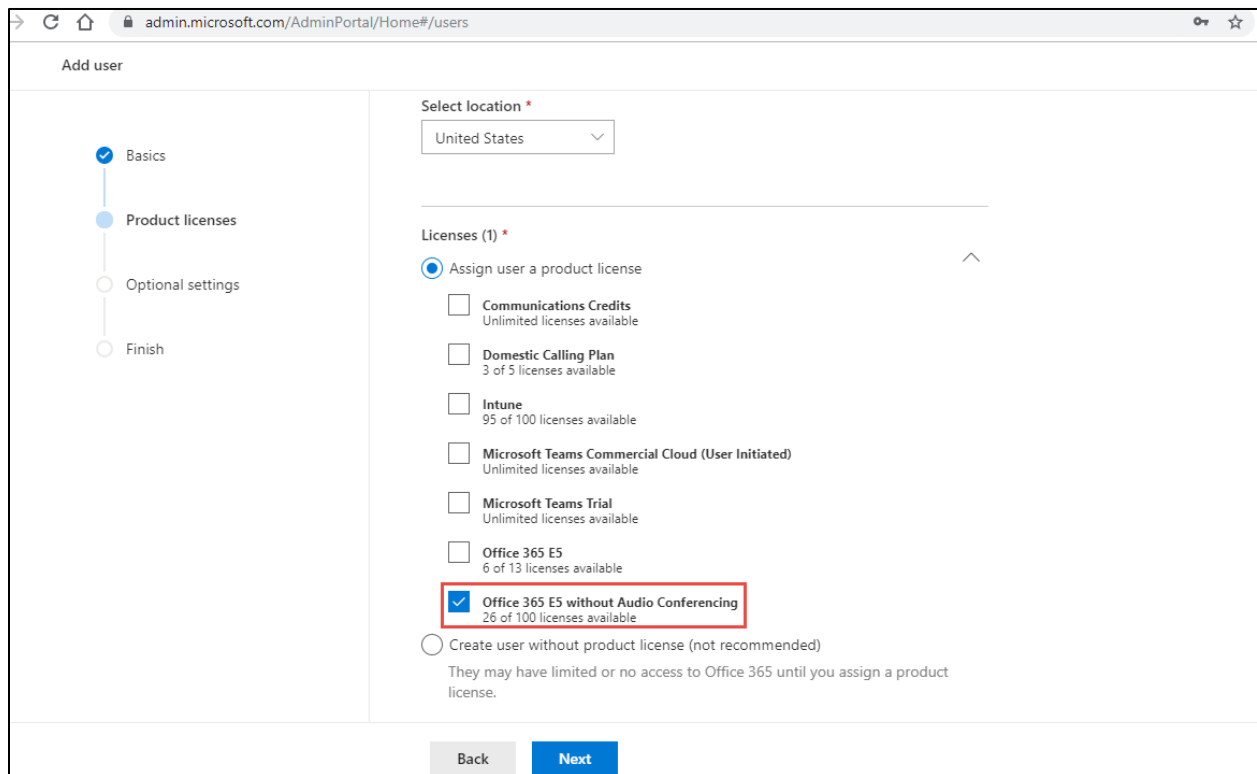


Figure 28: Teams User Creation – Contd.

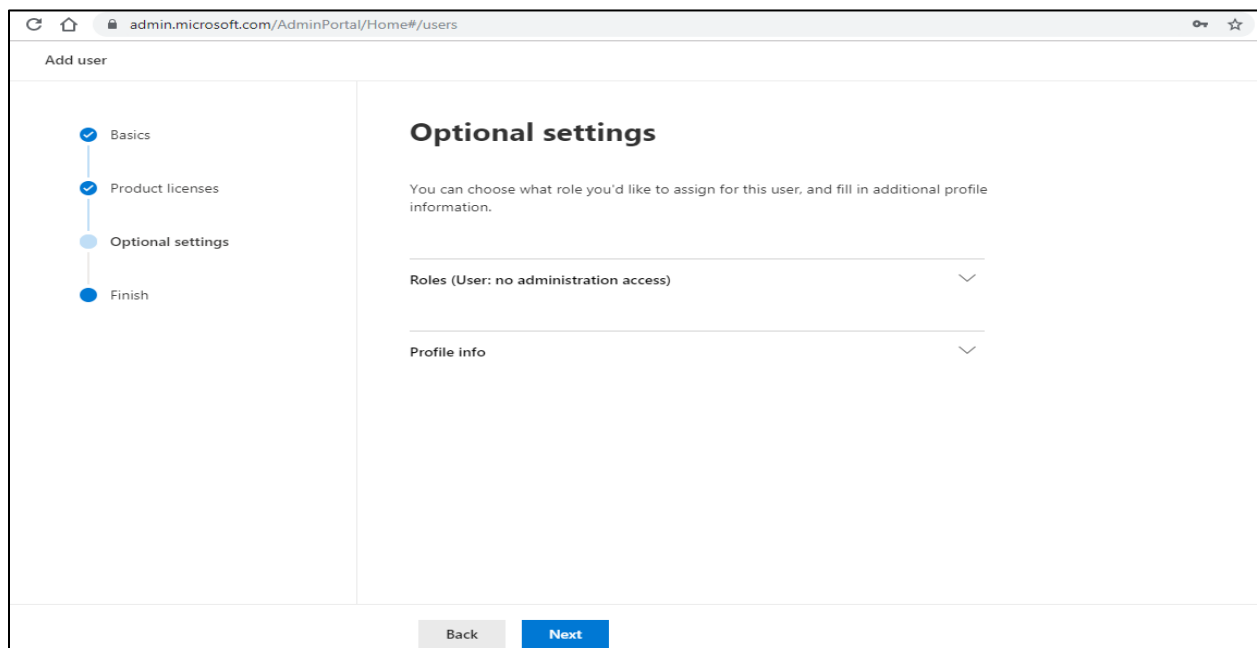


Figure 29: Teams User Creation – Contd.

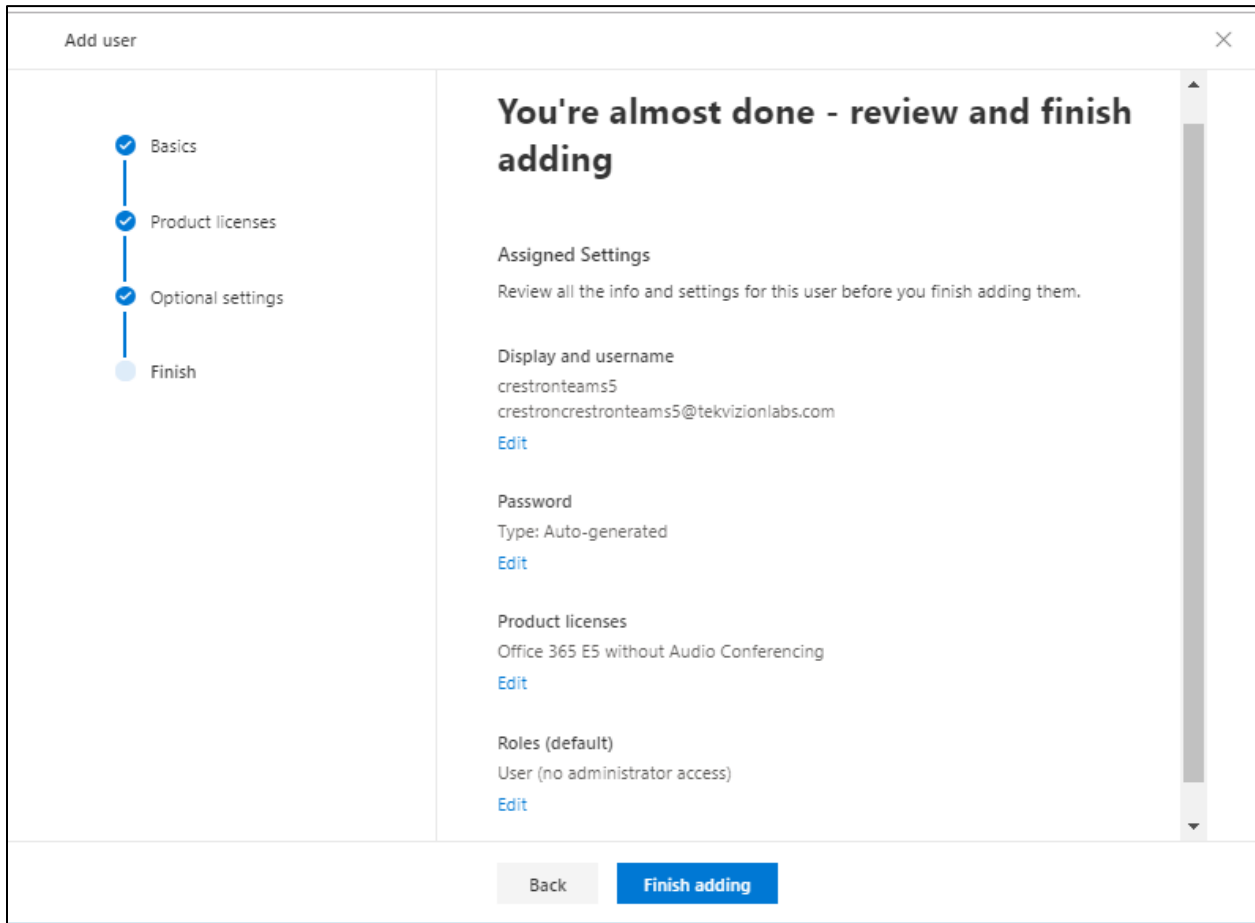


Figure 30: Teams User Creation – Contd.

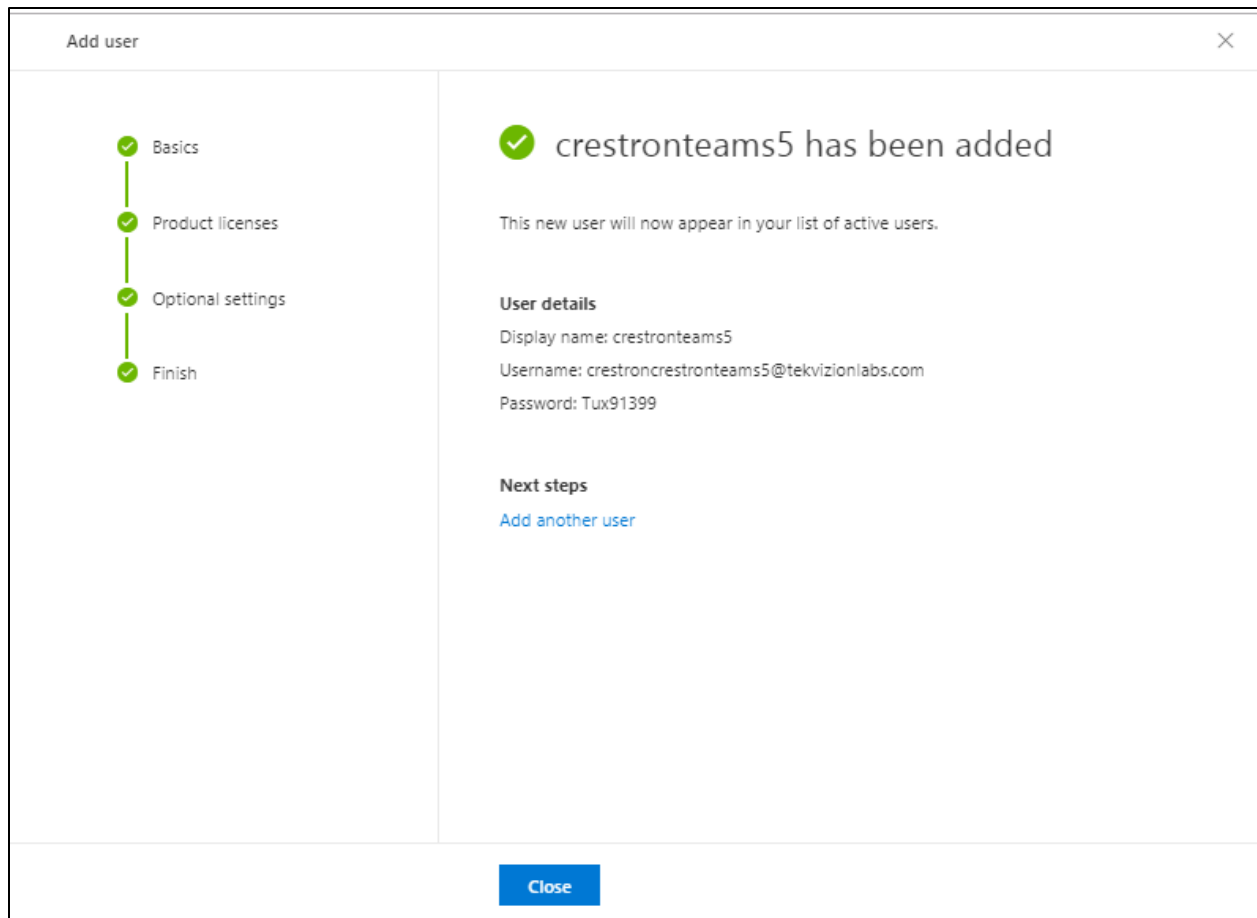


Figure 31: Teams User Creation – Contd.

5. Select the Admin icon from the Microsoft 365 Administrator Home page and navigate to Microsoft Teams admin center as shown below.

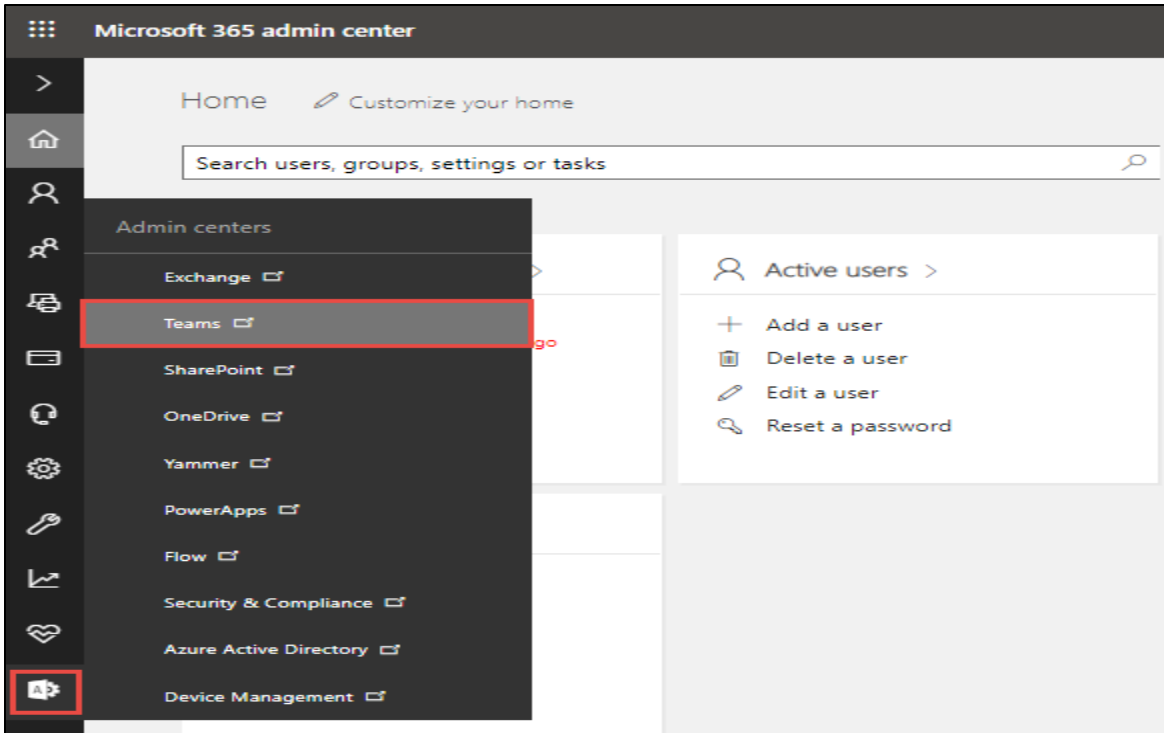


Figure 32: Microsoft O365 admin

6. Select Users from the Microsoft Teams Admin Center to view the list of available users.

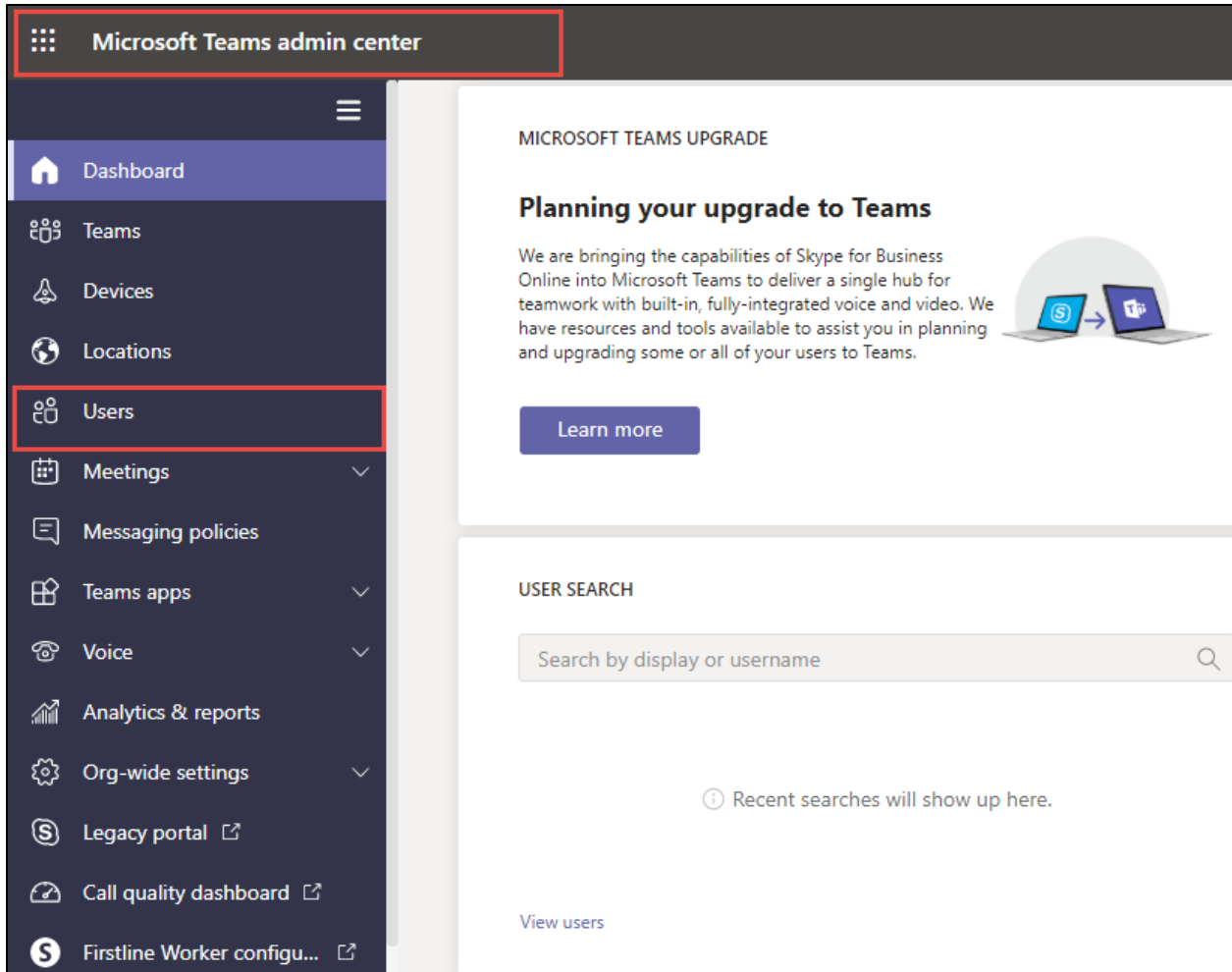


Figure 33: Microsoft O365 admin

7. Search for the user created above and click on the user display name to view user properties.



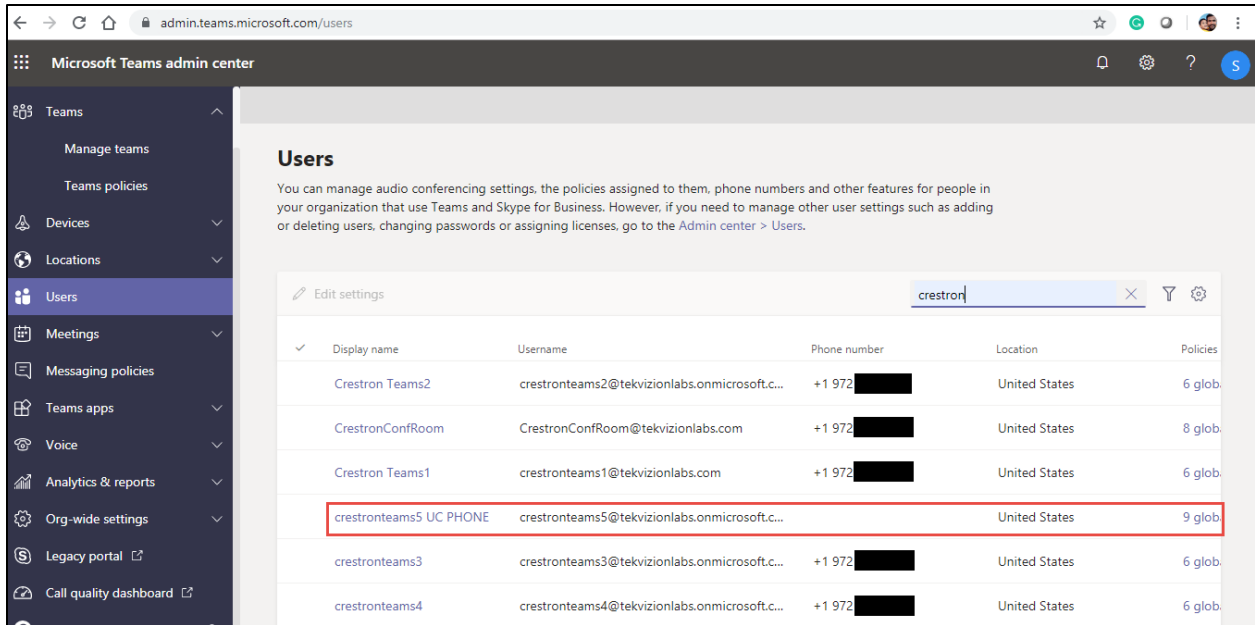


Figure 34: Microsoft O365 admin

- Under user properties, navigate to Account and set the teams upgrade mode to Teams only as shown below.

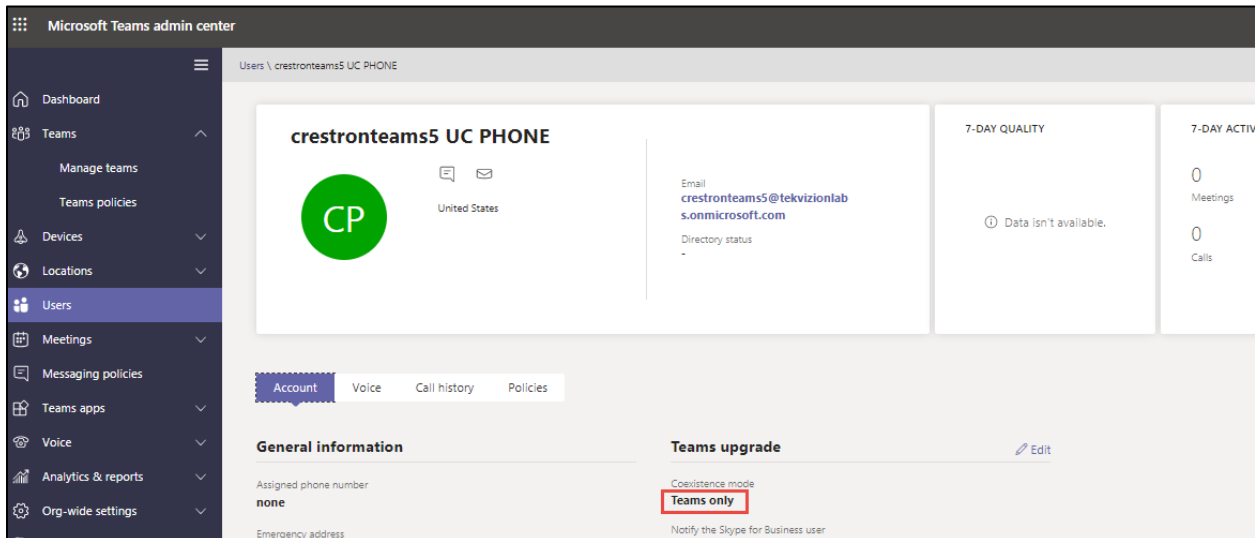


Figure 35: Teams User

## 4.4.2 Configure Calling policy to Users

- 1) Under user properties, navigate to Policies and set the Calling Policy as shown below. Here in the below example custom policy “Busy on Busy enabled” is assigned to user. Procedure to create custom policy is shown in the next section.

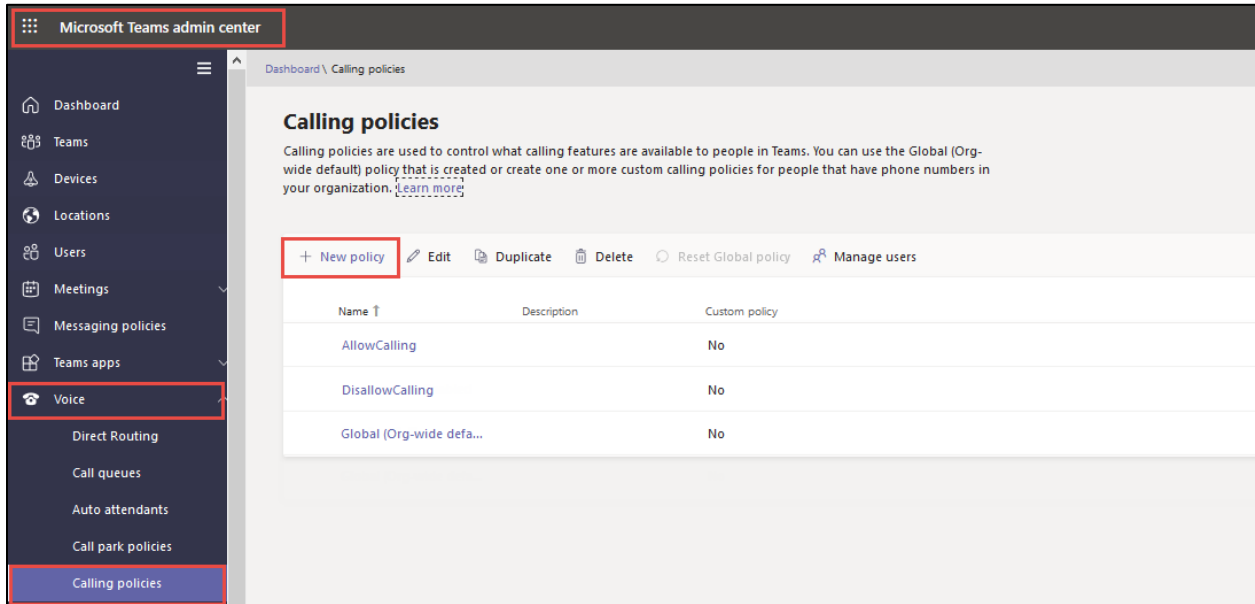


Figure 36 – Calling Policy

2. Below calling policy is created to turn on Busy on Busy. Click save to complete the configuration.

Dashboard \ Calling policies \ New policy

## Busy on Busy Enabled

Description

Make private calls  On

Call forwarding and simultaneous ringing to people in your organization  On

Call forwarding and simultaneous ringing to external phone numbers  On

Voicemail is available for routing inbound calls

Inbound calls can be routed to call groups  On

Allow delegation for inbound and outbound calls  On

Prevent toll bypass and send calls through the PSTN  Off

**Busy on busy is available when in a call  On**

Figure 37 – Calling Policy

#### 4.4.3 Configure user parameters.

Using the Remote PowerShell connect to Microsoft office 365 Tenant. Use the below commands to set DID and enable Enterprise Voice, Hosted Voicemail for Teams users.

```
Set-CsUser -identity "crestronteam5@tekvisionlabs.com" -EnterpriseVoiceEnabled $true -HostedVoicemail $true
```

```
Set-CsUser -identity "crestronteam5@tekvisionlabs.com" -OnPremLineURI tel:+197259800xx
```

#### 4.4.4 Create Online PSTN Gateway

Use the below command to pair the SBC to the tenant.

```
New-CsOnlinePSTNGateway -Fqdn <SBC FQDN> -SipSignallingPort <SBC SIP Port>
```

```
-ForwardCallHistory $true -ForwardPai $true -MaxConcurrentSessions <Max Concurrent Sessions the SBC can handle> -Enabled $true -MediaBypass $true
```

```

PS C:\Users\spandian> Get-CsOnlinePSTNGateway -Identity sbc4.tekvizionlabs.com

Identity           : sbc4.tekvizionlabs.com
Fqdn               : sbc4.tekvizionlabs.com
SipSignallingPort  : 5061
FailoverTimeSeconds : 10
ForwardCallHistory : True
ForwardPai        : True
SendsipOptions    : True
MaxConcurrentSessions : 100
Enabled           : True
MediaBypass       : True
GatewaySiteId     : 
GatewaySiteLbrEnabled : False
FailoverResponseCodes : 408, 503, 504
GenerateRingingWhileLocatingUser : True
PidfloSupported   : True
MediaRelayRoutingLocationOverride : 
ProxySbc          : 
BypassMode        : None

```

Figure 38 - Online PSTN Gateway

#### 4.4.5 Configure Online PSTN Usage

Use the below command to add a new PSTN usage.

**Set-CsOnlinePstnUsage -identity Global -Usage @{Add="<usage name>"}**

After creating Online PSTN usage use the command "**(Get-CsOnlinePstnUsage).usage**" to view the online pstn usage created. Example is shown below.

```

PS C:\WINDOWS\system32> (Get-CsOnlinePstnUsage).usage
US and Canada
Test
CCE
Non E.164
ThinkTel
sbc3
sbc4

```

Figure 39 - Microsoft Teams - Online PSTN usage reference

#### 4.4.6 Configure Online Voice Route

Use the below command to add a new online Voice Route.

```

New-CsOnlineVoiceRoute -Identity "<Route name>" -NumberPattern ".*"
-OnlinePstnGatewayList "<SBCFQDN>" -Priority 1 -OnlinePstnUsages "<PSTN usage
name>"}

```

```
PS C:\WINDOWS\system32> Get-CsOnlineVoiceRoute -Identity sbc4

Identity           : sbc4
Priority            : 5
Description        :
NumberPattern      : .*
OnlinePstnUsages   : {sbc4}
OnlinePstnGatewayList : {sbc4.tekvizionlabs.com}
Name               : sbc4
```

Figure 40 - Microsoft Teams - Online PSTN Voice Route reference

#### 4.4.7 Configure Online Voice Route Policy

Create a new online Voice Routing Policy using the below command.

```
New-CsOnlineVoiceRoutingPolicy "<policy name>" -OnlinePstnUsages "<pstn usage name>"
```

```
PS C:\WINDOWS\system32> Get-CsOnlineVoiceRoutingPolicy

Identity           : Tag:sbc4
OnlinePstnUsages   : {sbc4}
Description        :
RouteType          : BYOT
```

Figure 41 - Microsoft Teams - Online Voice Route Policy

#### 4.4.8 Configure Online Voice Route Policy to user

Assign a online Voice Routing Policy to user using the below command.

```
Grant-CsOnlineVoiceRoutingPolicy -Identity "<Teams User>" -PolicyName "<PSTN Usage>"
```

```
> Grant-CsOnlineVoiceRoutingPolicy -Identity "crestronteam5" -PolicyName "sbc4"
```

Figure 42 - Microsoft Teams - Online Voice Route Policy to User

#### 4.4.9 Configure Tenant Dial Plan

Tenant dial plan added to provision custom dial plan to user. Example is shown below

```
New-CsTenantDialPlan -Identity <dial plan name> -Description "For Extension Calling"
```

```
> Get-CsTenantDialPlan -Identity crestron

Identity           : Tag:crestron
Description        : For Extention Dialing
```

```
NormalizationRules :  
{Description=crestron;Pattern=^(.*)$;Translation=$1;Name=crestron;IsInternalExtension=False}  
ExternalAccessPrefix :  
SimpleName : crestron  
OptimizeDeviceDialing : False
```

Figure 43 - Microsoft Teams – Configure Tenant Dial Plan

#### 4.4.10 Create Normalization Rule

Create a new Voice Normalization Rule using the below command.

```
$rule1 = New-CsVoiceNormalizationRule -Parent Global -Description "description" -  
Pattern '^(.*)$' -Translation '$1' -Name <dial plan name> -IsInternalExtension $false  
-InMemory
```

```
> $rule1 = New-CsVoiceNormalizationRule -Parent Global -Description "crestron" -Pattern '^(.*)$' -Translation '$1' -Name crestron -IsInternalExtension $false -InMemory
```

Figure 44 - Microsoft Teams – Normalization Rule

#### 4.4.11 Associate Normalization rule to tenant dial plan

Associate the Voice Normalization Rule to tenant dial plan created earlier using the below command.

```
Set-CsTenantDialPlan -Identity <dial plan name> -NormalizationRules  
@{add=$rule1}
```

```
> Set-CsTenantDialPlan -Identity crestron -NormalizationRules @{add=$rule1}
```

Figure 45 - Microsoft Teams – Normalization Rule to tenant dial plan

#### 4.4.12 Associate tenant Dial plan to user

Assign the Tenant dial plan to the user using below command.

```
Grant-CsTenantDialPlan -identity <username> -PolicyName <dial plan name>
```

```
> Grant-CsTenantDialPlan -identity crestronteam5 -PolicyName crestron
```

Figure 46 - Microsoft Teams – tenant dial plan to user

#### 4.4.13 Calling Line Identity Policy

Calling Line Identity Policy is used to present/restrict users Caller ID.

```
New-CsCallingLineIdentity -Identity anonymous_policy -Description "clid restricted" -CallingIDSubstitute Anonymous -EnableUserOverride $true
```

Use the command **Get-CsCallingLineIdentity** to view the Calling Line Identity policy created.

```
PS C:\WINDOWS\system32> Get-CsCallingLineIdentity -Identity anonymous_policy

Identity           : Tag:Anonymous_policy
Description        : clid restricted
EnableUserOverride : True
ServiceNumber     :
CallingIDSubstitute : Anonymous
BlockIncomingPstnCallerID : False
```

*Figure 47 – Privacy Policy*

Associate the policy created above to the users using the below command.

```
Grant-CsCallingLineIdentity -Identity "crestrontteams5@tekvizionlabs.com" -PolicyName anonymous_policy
```

User associated with the above policy gets an additional Option as “Caller ID” in their Teams Client.

Navigate to Settings -> Calls -> Caller ID in users Teams client, Check **“Hide my phone number and profile information”** to restrict caller ID.

## 4.5 AudioCodes VE SBC Configuration

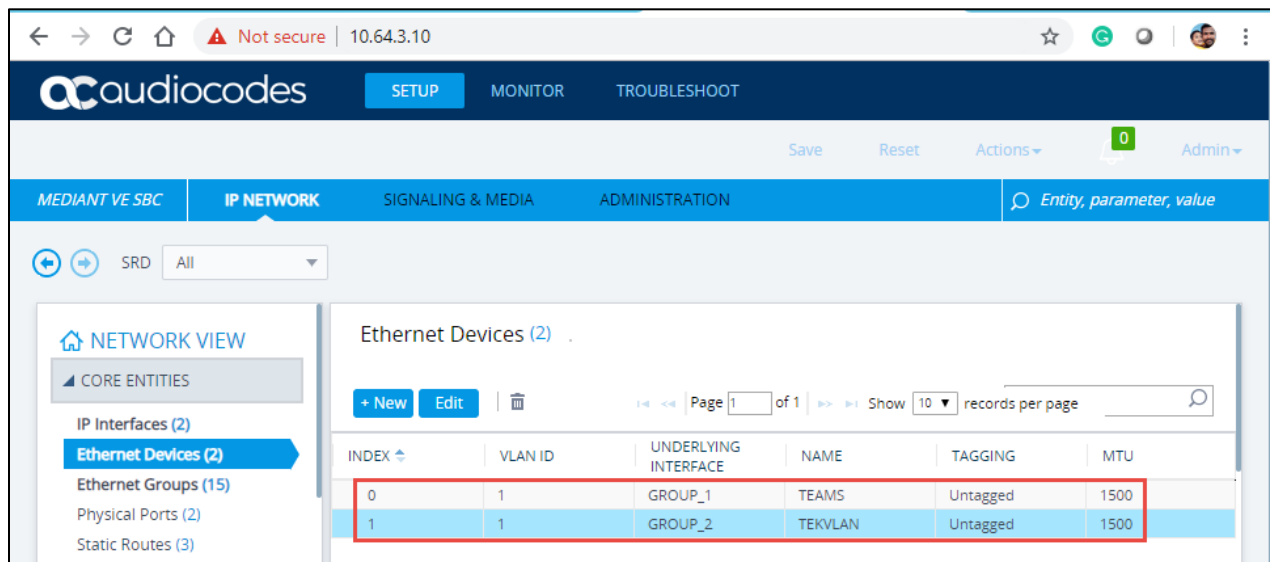
### 4.5.1 General

AudioCodes Mediant 1000 SBC was used as it can meet the requirements and support the enhancements for Microsoft Teams Direct Routing. PSTN Gateway SIP Trunk is a non-registering trunk that connects to E-SBC using UDP. Avaya SBCE SIP Trunk that connects to E-SBC using UDP. The SBC must be configured to perform back to back User Agent (B2BUA) functionality. For the B2BUA configuration, it is recommended that Physical interfaces are connected with two different customer WAN networks.

### 4.5.2 Configure VLANs

To configure VLANs, navigate to **IP Network tab** → **Core Entities menu** → **Ethernet Devices**

Add an entry with VLAN ID for underlying Teams and CenturyLink Voice Complete® interface Groups configured.



The screenshot shows the AudioCodes Mediant VE SBC configuration interface. The browser address bar displays "10.64.3.10". The interface includes a navigation menu with "IP NETWORK" selected. The "Ethernet Devices (2)" section is active, showing a table with two entries. The table has columns for INDEX, VLAN ID, UNDERLYING INTERFACE, NAME, TAGGING, and MTU. The first entry (INDEX 0) is for "TEAMS" with VLAN ID 1 and MTU 1500. The second entry (INDEX 1) is for "TEKVLAN" with VLAN ID 1 and MTU 1500. A red box highlights the second entry.

INDEX	VLAN ID	UNDERLYING INTERFACE	NAME	TAGGING	MTU
0	1	GROUP_1	TEAMS	Untagged	1500
1	1	GROUP_2	TEKVLAN	Untagged	1500

Figure 48 – Ethernet Devices



### 4.5.3 Configure IP Network Interfaces

To configure IP Network interfaces, navigate to the **IP Network tab->Core Entities menu ->Interfaces Table**.

Configure the WAN and LAN interface (interface towards Teams and LAN) as shown below:

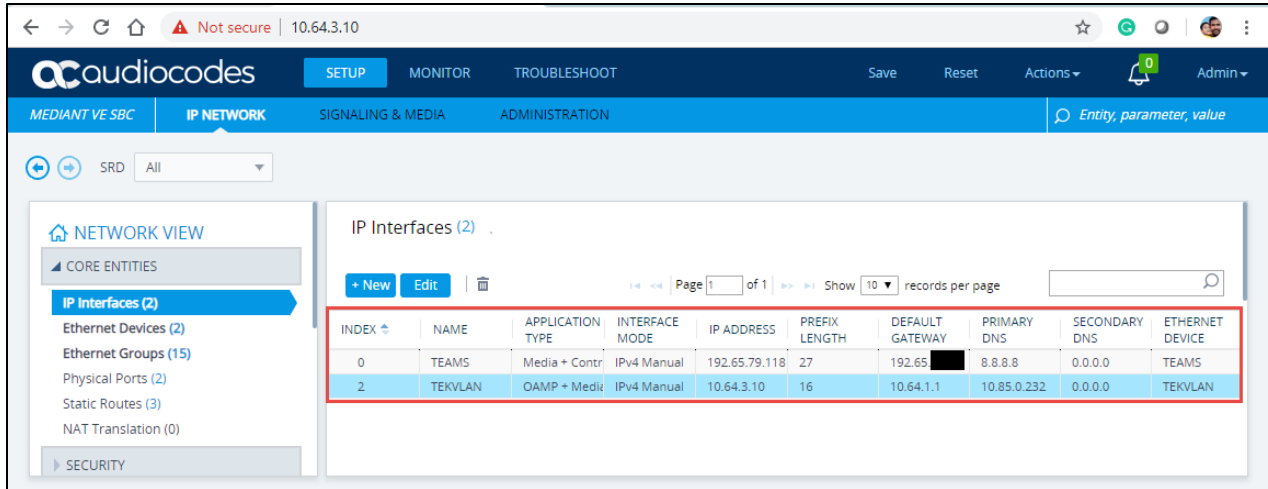


Figure 49 – IP interface Devices

### IP interface TEAMS

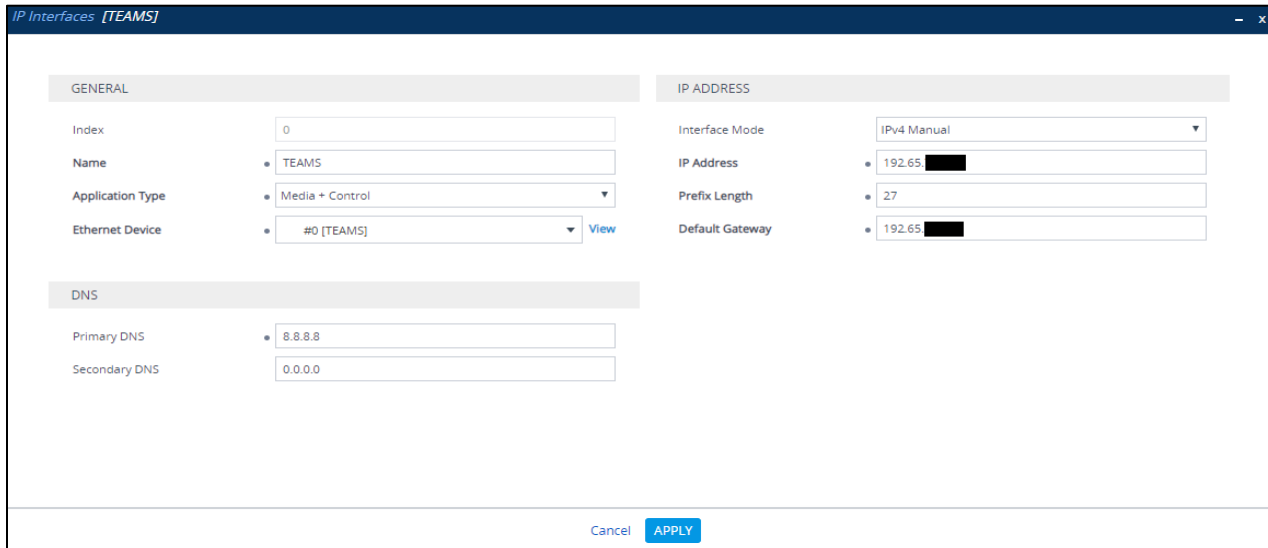


Figure 50 – IP interface Devices

## IP Interfaces – TEKVLAN

The screenshot shows the configuration interface for an IP interface named 'TEKVLAN'. The 'GENERAL' tab is active, showing the following settings:

- Index: 2
- Name: TEKVLAN
- Application Type: OAMP + Media + Control
- Ethernet Device: #1 [TEKVLAN]

The 'IP ADDRESS' tab is also visible, showing the following settings:

- Interface Mode: IPv4 Manual
- IP Address: 10.64.3.10
- Prefix Length: 16
- Default Gateway: 10.64.1.1

The 'DNS' section at the bottom left shows:

- Primary DNS: 10.85.0.232
- Secondary DNS: 0.0.0.0

At the bottom right, there are 'Cancel' and 'APPLY' buttons.

Figure 51 – IP interface Devices

### 4.5.4 Configure DNS SRV Records

Microsoft Teams Direct Routing uses primary, secondary and tertiary datacenters for call routing.

AudioCodes Mediant 1000 SBC uses internal SRV records to resolve the FQDN of these datacenters.

To configure DNS SRV records, navigate to the **IP Network tab** → **DNS menu** → **Internal SRV Table**.

Configure a DNS SRV records as shown below and associate it under proxy set towards Teams

GENERAL		2ND ENTRY	
Domain Name	teams.local	DNS Name 2	sip2.pstnhub.microsoft.com
Transport Type	TLS	Priority 2	2
1ST ENTRY		Weight 2	1
DNS Name 1	sip.pstnhub.microsoft.com	Port 2	5061
Priority 1	1	3RD ENTRY	
Weight 1	1	DNS Name 3	sip3.pstnhub.microsoft.com
Port 1	5061	Priority 3	3
		Weight 3	1
		Port 3	5061

Figure 52 – DNS SRV Records

#### 4.5.5 Configure SRTP

By default, SRTP is disabled.

To enable SRTP, navigate to **Setup → Signaling and Media → Media → Media → Media Security**. Set the parameter 'Media Security' to Enable; configure the other parameters as shown below

The screenshot shows the Audiocodes web interface for configuring Media Security. The left sidebar shows a navigation menu with 'Media Security' selected. The main content area is titled 'Media Security' and is divided into 'GENERAL' and 'AUTHENTICATION & ENCRYPTION' sections. In the 'GENERAL' section, the 'Media Security' dropdown is set to 'Enable' (highlighted with a red box), and 'Media Security Behavior' is set to 'Preferable - Single me' (also highlighted with a red box). Other settings include 'Offered SRTP Cipher Suites' set to 'All', 'Aria Protocol Support' set to 'Disable', 'Master Key Identifier (MKI) Size' set to '0', and 'Symmetric MKI' set to 'Disable'. In the 'AUTHENTICATION & ENCRYPTION' section, 'Authentication On Transmitted RTP Packets' is set to 'Active', 'Encryption On Transmitted RTP Packets' is set to 'Active', 'Encryption On Transmitted RTCP Packets' is set to 'Active', 'SRTP Tunneling Authentication for RTP' is set to 'Disable', and 'SRTP Tunneling Authentication for RTCP' is set to 'Disable'. At the bottom right, there are 'Cancel' and 'APPLY' buttons.

Figure 53 – Media Security

#### 4.5.6 Configure TLS contexts

Microsoft Teams Direct Routing allows only TLS connections from SBCs for SIP traffic with a certificate signed by one of the trusted Certification Authorities. Currently, supported Certification Authorities are:

- AffirmTrust
- AddTrust External CA Root
- Baltimore CyberTrust Root
- Buypass
- Cybertrust
- Class 3 Public Primary Certification Authority
- Comodo Secure Root CA
- Deutsche Telekom
- DigiCert Global Root CA
- DigiCert High Assurance EV Root CA
- Entrust
- GlobalSign
- Go Daddy
- GeoTrust
- Verisign, Inc.
- Starfield
- Symantec Enterprise Mobile Root for Microsoft
- SwissSign
- Thawte Timestamping CA
- Trustwave
- TeliaSonera
- T-Systems International GmbH (Deutsche Telekom)
- QuoVadis

Please refer to the below URL for latest Certification Authorities trusted by Microsoft Teams Direct Routing. <https://docs.microsoft.com/en-us/microsoftteams/direct-routing-plan>

To configure TLS contexts, navigate to **IP Network** tab → **Security** menu → **TLS Contexts**. Create a new TLS context for Teams as shown below.

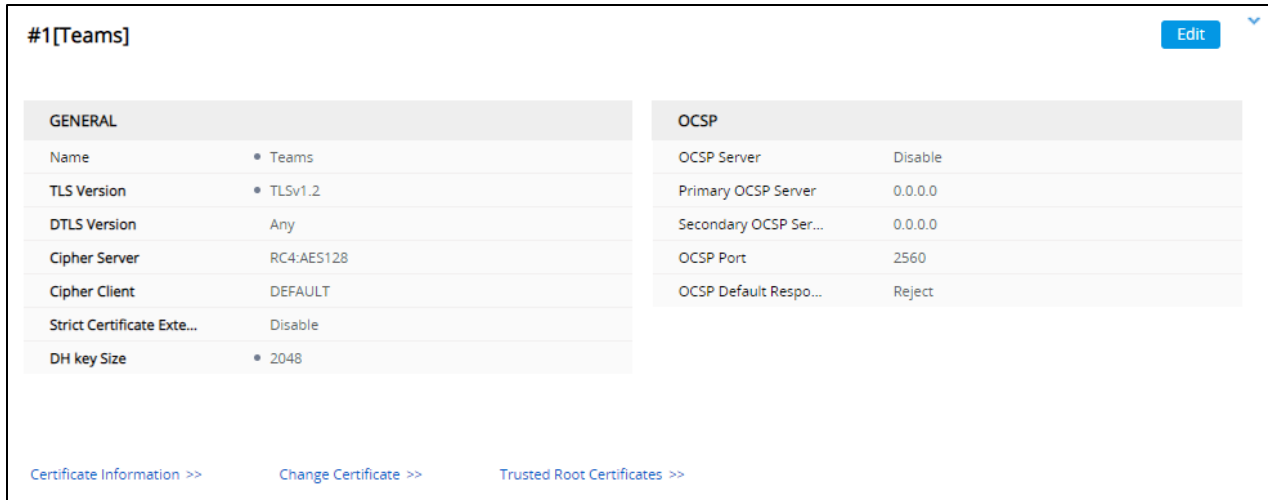


Figure 54 – Teams TLS

Once TLS context is configured, click on the change certificate and generate a CSR. Get the CSR signed from a CA trusted by direct routing and upload it to the same TLS context under change certificates. Import the root and intermediate Certificates to the trusted root certificates shown above.

Note: Root certificate used by Microsoft Direct Routing has to be uploaded to the SBC trusted root certificates.

#### 4.5.7 Configure Media Realms

To configure Media Realm, navigate to **Signaling & Media** tab -> **Core Entities** menu -> **Media Realms**.

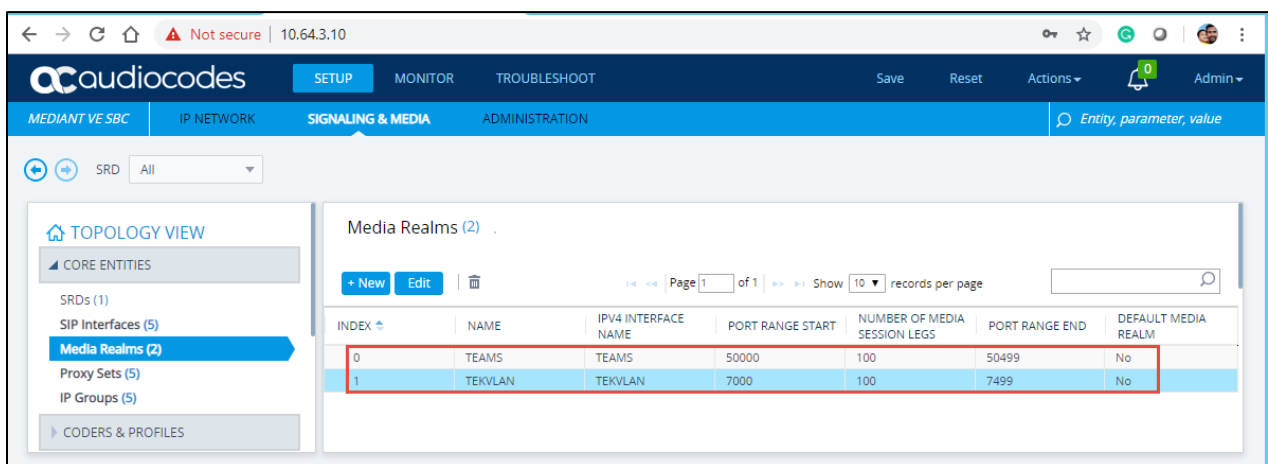


Figure 55 – Media Realms

Configure a Media Realm for WAN traffic – “Teams” as shown below:

The screenshot shows the 'Media Realms [TEAMS]' configuration window. It is divided into two main sections: 'GENERAL' and 'QUALITY OF EXPERIENCE'. In the 'GENERAL' section, the 'Name' field is highlighted with a red box and contains the text 'TEAMS'. Other fields include 'Index' (0), 'Topology Location' (Down), 'IPv4 Interface Name' (#0 [TEAMS]), 'Port Range Start' (50000), 'Number Of Media Session Legs' (100), 'Port Range End' (50499), and 'Default Media Realm' (No). The 'QUALITY OF EXPERIENCE' section has 'QoE Profile' and 'Bandwidth Profile' both set to '..'. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 56 – Teams

Configure a Media Realm for LAN traffic – “TEKVLAN” as shown below:

The screenshot shows the 'Media Realms [TEKVLAN]' configuration window. It is divided into two main sections: 'GENERAL' and 'QUALITY OF EXPERIENCE'. In the 'GENERAL' section, the 'Name' field is highlighted with a red box and contains the text 'TEKVLAN'. Other fields include 'Index' (1), 'Topology Location' (Up), 'IPv4 Interface Name' (#2 [TEKVLAN]), 'Port Range Start' (7000), 'Number Of Media Session Legs' (100), 'Port Range End' (7499), and 'Default Media Realm' (No). The 'QUALITY OF EXPERIENCE' section has 'QoE Profile' and 'Bandwidth Profile' both set to '..'. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 57 – LAN LAB

#### 4.5.8 Configure the SRD

To configure Signaling Routing Domains (SRD), navigate to **Signaling & Media tab** → **Core Entities menu** → **SRD Table**

Here the default SRD is used as shown below.

The screenshot shows the configuration page for the default SRD. It is divided into two main sections: GENERAL and REGISTRATION. The GENERAL section includes fields for Name (DefaultSRD), Sharing Policy (Shared), SBC Operation (B2BUA), SBC Routing Policy (Default\_SBCRoutingPolicy), Used By Routine (Not Used), Dial Plan ([-]), and CAC Profile ([-]). The REGISTRATION section includes Max. Number of Registered Users (-1), User Security Mode (Accept All), and Enable Un-Authenticated Registrations (Enable). Each field has a 'View' link next to it.

GENERAL	
Name	• DefaultSRD
Sharing Policy	Shared
SBC Operation ...	B2BUA
SBC Routing Pol...	• # [Default_SBCRoutingPolicy] <a href="#">View</a>
Used By Routin...	Not Used
Dial Plan	# [-] <a href="#">View</a>
CAC Profile	# [-] <a href="#">View</a>

REGISTRATION	
Max. Number o...	-1
User Security M...	Accept All
Enable Un-Auth...	Enable

Figure 58 – Default SRD

#### 4.5.9 Configure SIP Signaling Interface

For this test, three external SIP interfaces were configured on the SBC. To configure SIP interfaces, navigate to **Signaling & Media** tab → **Core Entities** menu → **SIP Interface Table**.

Configure a SIP interface for the WAN (towards Teams) as shown below.

The screenshot shows the configuration page for a SIP interface named 'TEAMS'. It is divided into three main sections: GENERAL, MEDIA, and SECURITY. The GENERAL section includes fields for Index (0), Name (TEAMS), Topology Location (Down), Network Interface (#0 [TEAMS]), Application Type (SBC), UDP Port (5060), TCP Port (0), TLS Port (5061), Additional UDP Ports, Additional UDP Ports Mode (Always Open), and Encapsulating Protocol (No encapsulation). The MEDIA section includes Media Realm (#0 [TEAMS]), Direct Media (Disable), and TLS Context Name (#1 [Teams]). The SECURITY section includes TLS Mutual Authentication (Enable), Message Policy (--), User Security Mode (Not Configured), Enable Un-Authenticated Registrations (Not configured), and Max. Number of Registered Users (-1). Several fields are highlighted with red boxes.

GENERAL	
Index	0
Name	• TEAMS
Topology Location	Down
Network Interface	• #0 [TEAMS] <a href="#">View</a>
Application Type	SBC
UDP Port	5060
TCP Port	• 0
TLS Port	5061
Additional UDP Ports	
Additional UDP Ports Mode	Always Open
Encapsulating Protocol	No encapsulation

MEDIA	
Media Realm	• #0 [TEAMS] <a href="#">View</a>
Direct Media	Disable

SECURITY	
TLS Context Name	• #1 [Teams] <a href="#">View</a>
TLS Mutual Authentication	• Enable
Message Policy	-- <a href="#">View</a>
User Security Mode	Not Configured
Enable Un-Authenticated Registrations	Not configured
Max. Number of Registered Users	-1

Figure 59 – Teams

Enable TCP Keepalive: Enable

Used By Routing Server: Not Used

Pre-Parsing Manipulation Set: .. View

CAC Profile: .. View

**CLASSIFICATION**

Classification Failure Response Type: 0

Pre-classification Manipulation Set ID: -1

Call Setup Rules Set ID: -1

Cancel APPLY

Figure 60 – Teams

Configure a SIP interface for the LAN (towards PSTN Gateway) as shown below.

SIP Interfaces [PSTNGW]

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 1

Name: PSTNGW

Topology Location: Up

Network Interface: #2 [TEKVLAN] View

Application Type: SBC

UDP Port: 5060

TCP Port: 0

TLS Port: 0

Additional UDP Ports:

Additional UDP Ports Mode: Always Open

Encapsulating Protocol: No encapsulation

**MEDIA**

Media Realm: #1 [TEKVLAN] 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

Figure 61 – PSTN

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 62 – PSTN

Configure a SIP interface for the LAN (towards Avaya SBCE) as shown below.



SIP Interfaces [AVAYA]

SRD #0 [DefaultSRD]

**GENERAL**

Index: 3

Name: AVAYA

Topology Location: Down

Network Interface: #2 [TEKVLAN] View

Application Type: SBC

UDP Port: 5064

TCP Port: 0

TLS Port: 0

Additional UDP Ports:

Additional UDP Ports Mode: Always Open

Encapsulating Protocol: No encapsulation

**MEDIA**

Media Realm: #1 [TEKVLAN] View

Direct Media: Disable

**SECURITY**

TLS Context Name: #0 [default] View

TLS Mutual Authentication:

Message Policy: .. View

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

Figure 63 – Avaya

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 64 – Avaya

#### 4.5.10 Configure Proxy Sets

The Proxy Set defines the destination address (IP address or FQDN) of the SIP entity server.

For the test, three Proxy Sets were configured: one for the Microsoft Teams, PSTN Gateway and another one towards Avaya SBCE. These proxy sets were later associated with IP Groups.

To configure Proxy Sets, navigate to **Signaling & Media** tab → **Core Entities** menu → **Proxy Sets Table**

Configure a Proxy Set for the Teams as shown below.

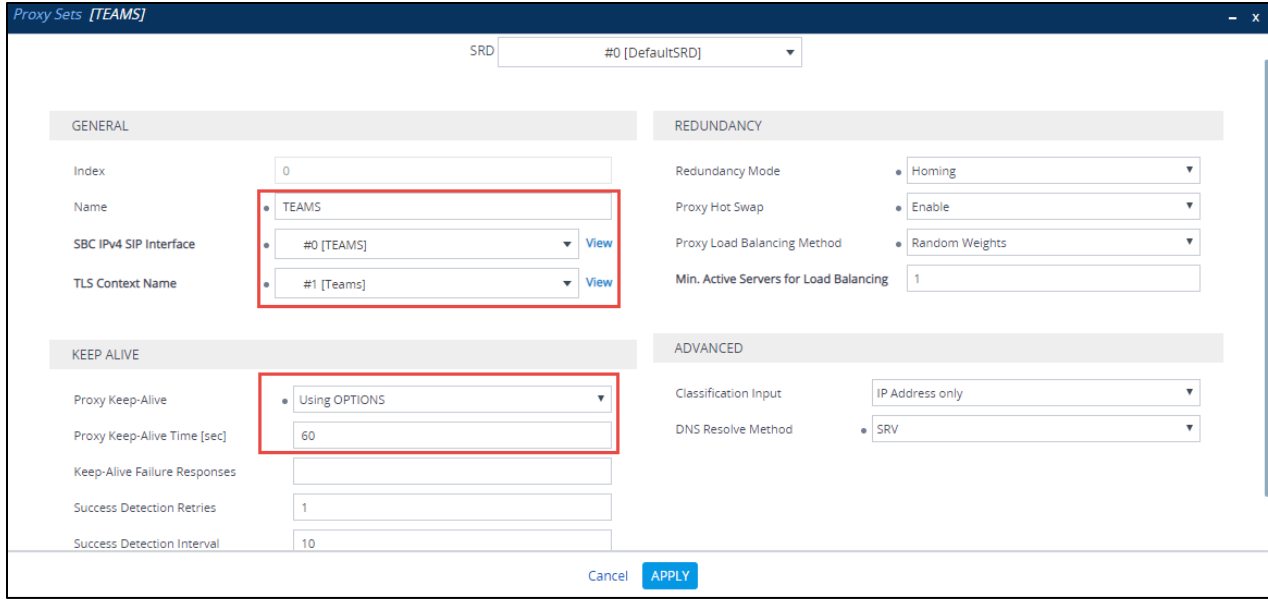


Figure 65 – Teams

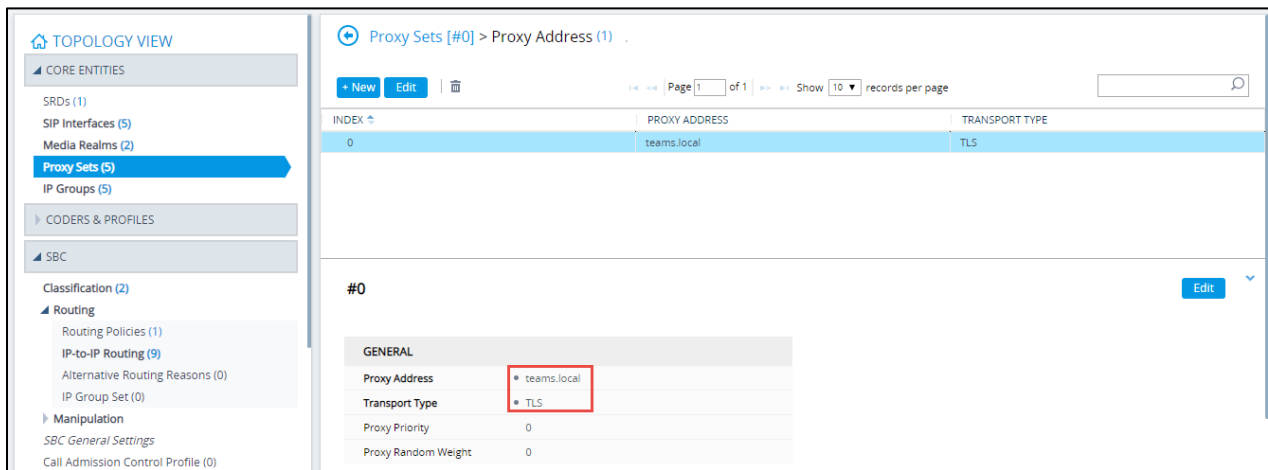


Figure 66 – Teams

Configure a Proxy Set for the PSTN Gateway as shown below.

Proxy Sets [PSTNGW]

SRD #0 [DefaultSRD]

**GENERAL**

Index: 1

Name: PSTNGW

SBC IPv4 SIP Interface: #1 [PSTNGW] [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: [Text Box]

Success Detection Retries: 1

Success Detection Interval: 10

**ADVANCED**

Classification Input: IP Address only

DNS Resolve Method: [Dropdown]

Figure 67 – PSTN Gateway

Keep-Alive Failure Responses: [Text Box]

Success Detection Retries: 1

Success Detection Interval: 10

Cancel [APPLY](#)

Figure 68 – PSTN Gateway

Configure a Proxy Set for the Avaya SBCE as shown below.

Proxy Sets [AVAYA]

SRD #0 [DefaultSRD]

**GENERAL**

Index: 3

Name: AVAYA

SBC IPv4 SIP Interface: #3 [AVAYA] [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: [Text Box]

Success Detection Retries: 1

Success Detection Interval: 10

**ADVANCED**

Classification Input: IP Address only

DNS Resolve Method: [Dropdown]

Figure 69 – Avaya

Success Detection Interval	<input type="text" value="10"/>
Failure Detection Retransmissions	<input type="text" value="-1"/>
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>	

Figure 70 – Avaya

### 4.5.11 Configure IP Groups

The IP Group represents an IP entity on the network with which the SBC communicates. For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting the source and destination of the call.

For the test, IP Groups were configured for the following IP entities:

- Microsoft Teams
- PSTN Gateway – SIP Trunk
- Avaya SBCE – SIP Trunk

To configure IP groups, navigate to **Signaling & Media** tab → **Core Entities** menu → **IP Group Table**

Configure an IP Group for Microsoft Teams as shown below

The screenshot shows the configuration page for an IP Group named 'TEAMS'. The 'GENERAL' section includes fields for Index (0), Name (TEAMS), Topology Location (Down), Type (Server), Proxy Set (#0 [TEAMS]), IP Profile (#1 [TEAMS\_Profile]), Media Realm (#0 [TEAMS]), Contact User, SIP Group Name (sbc4.tekvizionlabs.com), Created By Routing Server (No), and Used By Routing Server (Not Used). The 'MESSAGE MANIPULATION' section includes Inbound Message Manipulation Set (1) and Outbound Message Manipulation Set (2). The 'QUALITY OF EXPERIENCE' section includes QoE Profile and Bandwidth Profile. The 'SBC REGISTRATION AND AUTHENTICATION' section is partially visible at the bottom.

Figure 71 – IP Group – Teams – Contd.

IP Groups [TEAMS]

Proxy Set Connectivity: Connected

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: Admin

Password: .....

**SBC GENERAL**

Classify By Proxy Set: Disable

SBC Operation Mode: Not Configured

SBC Client Forking Mode: Sequential

CAC Profile: .. View

**ADVANCED**

Local Host Name: sbc4.tekvizionlabs.com

UII Format: Disable

Always Use Src Address: No

**GW GROUP STATUS**

GW Group Registered IP Address:

GW Group Registered Status: Not Registered

Figure 72 – IP Group – Teams – Contd.

**SBC ADVANCED**

Source URI Input:

Destination URI Input:

SIP Connect: No

SBC PSAP Mode: Disable

Route Using Request URI Port: Disable

DTLS Context: #1 [Teams] View

Keep Original Call-ID: No

Dial Plan: .. View

Call Setup Rules Set ID: -1

Tags:

Cancel APPLY

Figure 73 – IP Group – Teams

Configure an IP Group for PSTN Gateway as shown below

IP Groups [PSTNGW] SRD #0 [DefaultSRD]

GENERAL		QUALITY OF EXPERIENCE	
Index	1	QoE Profile	-- <a href="#">View</a>
Name	PSTNGW	Bandwidth Profile	-- <a href="#">View</a>
Topology Location	Up	MESSAGE MANIPULATION	
Type	Server	Inbound Message Manipulation Set	0
Proxy Set	#1 [PSTNGW] <a href="#">View</a>	Outbound Message Manipulation Set	3
IP Profile	#2 [PSTNGW_Profile] <a href="#">View</a>	Message Manipulation User-Defined String 1	
Media Realm	#1 [TEKVLAN] <a href="#">View</a>	Message Manipulation User-Defined String 2	
Contact User		Proxy Keep-Alive using IP Group settings	Enable
SIP Group Name	10.64.1.72	SBC REGISTRATION AND AUTHENTICATION	
Created By Routing Server	No		
Used By Routing Server	Not Used		

Figure 74 – IP Group – PSTN – Contd.

IP Groups [PSTNGW] Proxy Set Connectivity Connected

SBC GENERAL		ADVANCED		SBC REGISTRATION AND AUTHENTICATION	
Classify By Proxy Set	Enable	Local Host Name		Max. Number of Registered Users	-1
SBC Operation Mode	Not Configured	UUI Format	Disable	Registration Mode	User Initiates Registration
SBC Client Forking Mode	Sequential	Always Use Src Address	No	User Stickiness	Disable
CAC Profile	-- <a href="#">View</a>			User UDP Port Assignment	Disable
				Authentication Mode	User Authenticates
				Authentication Method List	
				SBC Server Authentication Type	According to Global Parameter
				OAuth HTTP Service	-- <a href="#">View</a>
				Username	Admin
				Password	.....
				GW GROUP STATUS	
				GW Group Registered IP Address	
				GW Group Registered Status	Not Registered

Figure 75 – IP Group – PSTN – Contd.

SBC ADVANCED	
Source URI Input	<input type="text"/>
Destination URI Input	<input type="text"/>
SIP Connect	No
SBC PSAP Mode	Disable
Route Using Request URI Port	Disable
DTLS Context	#0 [default] <a href="#">View</a>
Keep Original Call-ID	No
Dial Plan	-- <a href="#">View</a>
Call Setup Rules Set ID	-1
Tags	<input type="text"/>

Cancel [APPLY](#)

Figure 76 – IP Group

Configure an IP Group for Avaya SBCE as shown below

*IP Groups [AVAYA]*

SRD

GENERAL	QUALITY OF EXPERIENCE
Index: <input type="text" value="3"/>	QoE Profile: <input type="text" value="--"/> <a href="#">View</a>
Name: <input type="text" value="AVAYA"/>	Bandwidth Profile: <input type="text" value="--"/> <a href="#">View</a>
Topology Location: <input type="text" value="Down"/>	
Type: <input type="text" value="Server"/>	
Proxy Set: <input type="text" value="#3 [AVAYA]"/> <a href="#">View</a>	
IP Profile: <input type="text" value="#4 [AVAYA_Profile]"/> <a href="#">View</a>	
Media Realm: <input type="text" value="#1 [TEKVLAN]"/> <a href="#">View</a>	
Contact User: <input type="text"/>	
SIP Group Name: <input type="text" value="10.64.5.57"/>	
Created By Routing Server: No	

MESSAGE MANIPULATION	
Inbound Message Manipulation Set: <input type="text" value="6"/>	
Outbound Message Manipulation Set: <input type="text" value="7"/>	
Message Manipulation User-Defined String 1: <input type="text"/>	
Message Manipulation User-Defined String 2: <input type="text"/>	
Proxy Keep-Alive using IP Group settings: <input type="text" value="Disable"/>	

Figure 77 – IP Group – Avaya

The screenshot shows the configuration interface for an IP Group in Avaya. The window title is "IP Groups [AVAYA]".

- Used By Routing Server:** Not Used
- Proxy Set Connectivity:** NA
- SBC GENERAL:**
  - Classify By Proxy Set: Enable
  - SBC Operation Mode: Not Configured
  - SBC Client Forking Mode: Sequential
  - CAC Profile: -- (with a View link)
- ADVANCED:**
  - Local Host Name: (empty text box)
  - UUI Format: Disable
  - Always Use Src Address: No
- 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: (empty text box)
  - SBC Server Authentication Type: According to Global Parameter
  - OAuth HTTP Service: -- (with a View link)
  - Username: Admin
  - Password: \*
- GW GROUP STATUS:**
  - GW Group Registered IP Address: (empty text box)

Figure 78 – IP Group – Avaya – Contd.

The screenshot shows the "SBC ADVANCED" configuration window. The title bar includes "GW Group Registered Status" with a dropdown set to "Not Registered".

- SBC ADVANCED:**
  - Source URI Input: (empty text box)
  - Destination URI Input: (empty text box)
  - SIP Connect: No
  - SBC PSAP Mode: Disable
  - Route Using Request URI Port: Disable
  - DTLS Context: #0 [default] (with a View link)
  - Keep Original Call-ID: No
  - Dial Plan: -- (with a View link)
  - Call Setup Rules Set ID: -1
  - Tags: (empty text box)

At the bottom, there are "Cancel" and "APPLY" buttons.

Figure 79 – IP Group – Avaya – Contd

#### 4.5.12 Configure IP Profile

The IP Profile defines a set of call capabilities relating to signaling.

For this test, IP Profiles were configured for the following IP entities:

- Microsoft Teams
- PSTN Gateway – SIP Trunk
- Avaya SBCE – SIP Trunk



To configure IP profiles, navigate to **Signaling & Media** tab → **Coders and Profiles** → **IP Profile Settings**.

Click **Add**.

Configure the IP Profile for the Microsoft Teams as shown below.

The screenshot shows the configuration for the IP Profile 'TEAMS\_Profile'. The 'GENERAL' tab is active, showing the profile name and routing server status. The 'MEDIA SECURITY' tab shows SBC Media Security Mode set to SRTP, Symmetric MKI set to Disable, MKI Size set to 1, SBC Enforce MKI Size set to Don't enforce, SBC Media Security Method set to SDES, Reset SRTP Upon Re-key set to Disable, and Generate SRTP Keys Mode set to Always. The 'SBC SIGNALING' tab shows PRACK Mode set to Optional, P-Asserted-Identity Header Mode, Diversion Header Mode, and History-Info Header Mode all set to As Is, Session Expires Mode set to Transparent, Remote Update Support, Remote re-INVITE, and Remote Delayed Offer Support all set to Not Supported, and Remote Representation Mode, Keep Incoming Via Headers, Keep Incoming Routing Headers, and Keep User-Agent Header all set to According to Operation Mode.

Figure 80 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the IP Profile 'TEAMS\_Profile' in the 'SBC EARLY MEDIA' tab. Remote Early Media and Remote Multiple 18x are set to Supported, Remote Early Media Response Type is set to Transparent, Remote Multiple Early Dialogs is set to According to Operation Mode, Remote Multiple Answers Mode is set to Disable, Remote Early Media RTP Detection Mode is set to By Media, Remote RFC 3960 Support is set to Not Supported, Remote Can Play Ringback is set to No, and Generate RTP is set to None. The 'SBC REGISTRATION' tab shows User Registration Time set to 0, NAT UDP Registration Time set to -1, and NAT TCP Registration Time set to -1. The 'SBC FORWARD AND TRANSFER' tab shows Remote REFER Mode set to Regular, Remote Replaces Mode set to Standard, and Play RBT To Transferee set to Yes.

Figure 81 – IP Profile – Teams – Contd.

IP Profiles [TEAMS\_Profile]

SBC MEDIA

Mediation Mode: RTP Mediation

Extension Coders Group: #0 [AudioCodersGroups\_0]

Allowed Audio Coders: #0 [AllowedAudioCodersGroup\_TEAMS]

Allowed Coders Mode: Preference

Allowed Video Coders: ..

Allowed Media Types:

Direct Media Tag:

RFC 2833 Mode: As Is

RFC 2833 DTMF Payload Type: 101

Alternative DTMF Method: As Is

Send Multiple DTMF Methods: Disable

Adapt RFC2833 BW to Voice coder BW: Disabled

SDP Ptime Answer: Preferred Value

Remote 3xx Mode: Handle Locally

SBC HOLD

Remote Hold Format: Inactive

Reliable Held Tone Source: Yes

Play Held Tone: No

SBC FAX

Fax Coders Group: ..

Fax Mode: As Is

Fax Offer Mode: All coders

Fax Answer Mode: Single coder

Remote Renegotiate on Fax Detection: Transparent

Fax Rerouting Mode: Disable

Figure 82 – IP Profile – Teams – Contd.

IP Profiles [TEAMS\_Profile]

Preferred PTime: 20

Use Silence Suppression: Add

RTP Redundancy Mode: As Is

RTCP Mode: Generate Always

Jitter Compensation: Disable

ICE Mode: Lite

SDP Handle RTCP: Don't Care

RTCP Mux: Supported

RTCP Feedback: Feedback Off

Voice Quality Enhancement: Disable

Max Opus Bandwidth: 0

Generate No-op: No

Enhanced PLC: Disable

MEDIA

Broken Connection Mode: Disconnect

Media IP Version Preference: Only IPv4

RTP Redundancy Depth: Disable

GATEWAY

Coders Group: #0 [AudioCodersGroups\_0]

LOCAL TONES

Local RingBack Tone Index: -1

Local Held Tone Index: -1

Figure 83 – IP Profile – Teams – Contd.

IP Profiles [TEAMS\_Profile]

**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 Canceler: Line

Input Gain (-32 to 31 dB): 0

Voice Volume (-32 to 31 dB): 0

Cancel APPLY

Figure 84 – IP Profile – Teams – Contd.

Configure the IP Profile for the PSTN Gateway as shown below.

IP Profiles [PSTNGW\_Profile]

**GENERAL**

Index: 2

Name: PSTNGW\_Profile

Created by Routing Server: No

**MEDIA SECURITY**

SBC Media Security Mode: RTP

Symmetric MKI: Disable

MKI Size: 0

SBC Enforce MKI Size: Don't enforce

SBC Media Security Method: SDES

Reset SRTP Upon Re-key: Disable

Generate SRTP Keys Mode: Only If Required

**SBC SIGNALING**

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: Supported Only After Connect

Remote re-INVITE: Supported only with SDP

Remote Delayed Offer Support: Not Supported

Remote Representation Mode: According to Operation Mode

Keep Incoming Via Headers: According to Operation Mode

Keep Incoming Routing Headers: According to Operation Mode

Keep User-Agent Header: According to Operation Mode

Figure 85 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW\_Profile]

SBC Remove Crypto Lifetime in SDP	No	Handle X-Detect	No
SBC Remove Unknown Crypto	No	ISUP Body Handling	Transparent
<b>SBC EARLY MEDIA</b>		ISUP Variant	Itu92
Remote Early Media	Supported	Max Call Duration [min]	0
Remote Multiple 18x	Supported	<b>SBC REGISTRATION</b>	
Remote Early Media Response Type	Transparent	User Registration Time	0
Remote Multiple Early Dialogs	According to Operation Mode	NAT UDP Registration Time	-1
Remote Multiple Answers Mode	Disable	NAT TCP Registration Time	-1
Remote Early Media RTP Detection Mode	By Signaling	<b>SBC FORWARD AND TRANSFER</b>	
Remote RFC 3960 Support	Not Supported	Remote REFER Mode	Handle Locally
Remote Can Play Ringback	Yes	Remote Replaces Mode	Handle Locally
Generate RTP	None	Play RBT To Transferee	Yes

Figure 86 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW\_Profile]

<b>SBC MEDIA</b>		Remote 3xx Mode	Transparent
Mediation Mode	RTP Mediation	<b>SBC HOLD</b>	
Extension Coders Group	..	Remote Hold Format	Transparent
Allowed Audio Coders	#1 [AllowedAudioCodersGroup_PSTNGW]	Reliable Held Tone Source	Yes
Allowed Coders Mode	Restriction	Play Held Tone	No
Allowed Video Coders	..	<b>SBC FAX</b>	
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	101	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	Preferred Value		

Figure 87 – IP Profile – PSTN Gateway – Contd.

**IP Profiles [PSTNGW\_Profile]**

Preferred PTime	20	<b>MEDIA</b>	
Use Silence Suppression	Add	Broken Connection Mode	Disconnect
RTP Redundancy Mode	As Is	Media IP Version Preference	Only IPv4
RTCP Mode	Generate Always	RTP Redundancy Depth	Disable
Jitter Compensation	Disable	<b>GATEWAY</b>	
ICE Mode	Disable	Coders Group	#0 [AudioCodersGroups_0]
SDP Handle RTCP	Don't Care	<b>LOCAL TONES</b>	
RTCP Mux	Not Supported	Local RingBack Tone Index	-1
RTCP Feedback	Feedback Off	Local Held Tone Index	-1
Voice Quality Enhancement	Disable		
Max Opus Bandwidth	0		
Generate No-op	No		
Enhanced PLC	Disable		

Figure 88 – IP Profile – PSTN Gateway – Contd.

**IP Profiles [PSTNGW\_Profile]**

<b>QUALITY OF SERVICE</b>	
RTP IP DiffServ	46
Signaling DiffServ	24
<b>JITTER BUFFER</b>	
Dynamic Jitter Buffer Minimum Delay [msec]	10
Dynamic Jitter Buffer Optimization Factor	10
Jitter Buffer Max Delay [msec]	300
<b>VOICE</b>	
Echo Canceler	Line
Input Gain (-32 to 31 dB)	0
Voice Volume (-32 to 31 dB)	0

Cancel **APPLY**

Figure 89 – IP Profile – PSTN Gateway

Configure the IP Profile for the Avaya as shown below.

IP Profiles [AVAYA\_Profile]

GENERAL		SBC SIGNALING	
Index	4	PRACK Mode	Transparent
Name	AVAYA_Profile	P-Asserted-Identity Header Mode	Add
Created by Routing Server	No	Diversion Header Mode	Add
MEDIA SECURITY		History-Info Header Mode	As Is
SBC Media Security Mode	RTP	Session Expires Mode	Supported
Symmetric MKI	Disable	Remote Update Support	Supported
MKI Size	0	Remote re-INVITE	Supported
SBC Enforce MKI Size	Don't enforce	Remote Delayed Offer Support	Supported
SBC Media Security Method	SDES	Remote Representation Mode	According to Operation Mode
Reset SRTP Upon Re-key	Disable	Keep Incoming Via Headers	According to Operation Mode
Generate SRTP Keys Mode	Only If Required	Keep Incoming Routing Headers	According to Operation Mode
		Keep User-Agent Header	According to Operation Mode

Figure 90 – IP Profile – Avaya – Contd.

IP Profiles [AVAYA\_Profile]

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

Figure 91 – IP Profile – Avaya – Contd.

**IP Profiles [AVAYA\_Profile]**

<b>SBC MEDIA</b>		Remote 3xx Mode	Transparent
Mediation Mode	RTP Mediation	<b>SBC HOLD</b>	
Extension Coders Group	..	Remote Hold Format	Transparent
Allowed Audio Coders	#1 [AllowedAudioCodersGroup_PSTNGW]	Reliable Held Tone Source	Yes
Allowed Coders Mode	Restriction	Play Held Tone	No
Allowed Video Coders	..	<b>SBC FAX</b>	
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		

Figure 92 – IP Profile – Avaya – Contd.

**IP Profiles [AVAYA\_Profile]**

Use Silence Suppression	Transparent	<b>MEDIA</b>	
RTP Redundancy Mode	As Is	Broken Connection Mode	Disconnect
RTCP Mode	Transparent	Media IP Version Preference	Only IPv4
Jitter Compensation	Disable	RTP Redundancy Depth	Disable
ICE Mode	Disable	<b>GATEWAY</b>	
SDP Handle RTCP	Don't Care	Coders Group	#0 [AudioCodersGroups_0]
RTCP Mux	Not Supported	<b>LOCAL TONES</b>	
RTCP Feedback	Feedback Off	Local RingBack Tone Index	-1
Voice Quality Enhancement	Disable	Local Held Tone Index	-1
Max Opus Bandwidth	0		
Generate No-op	No		
Enhanced PLC	Disable		

Figure 93 – IP Profile – Avaya – Contd.

IP Profiles [AVAYA\_Profile]

**QUALITY OF SERVICE**

RTP IP DiffServ

Signaling DiffServ

**JITTER BUFFER**

Dynamic Jitter Buffer Minimum Delay [msec]

Dynamic Jitter Buffer Optimization Factor

Jitter Buffer Max Delay [msec]

**VOICE**

Echo Canceler

Input Gain (-32 to 31 dB)

Voice Volume (-32 to 31 dB)

Cancel **APPLY**

Figure 94 – IP Profile – Avaya

#### 4.5.13 Configure SIP Definition and General Setting

The screenshot below captures the configuration of the **SIP Definitions General Settings** that were used during the test for the successful test execution



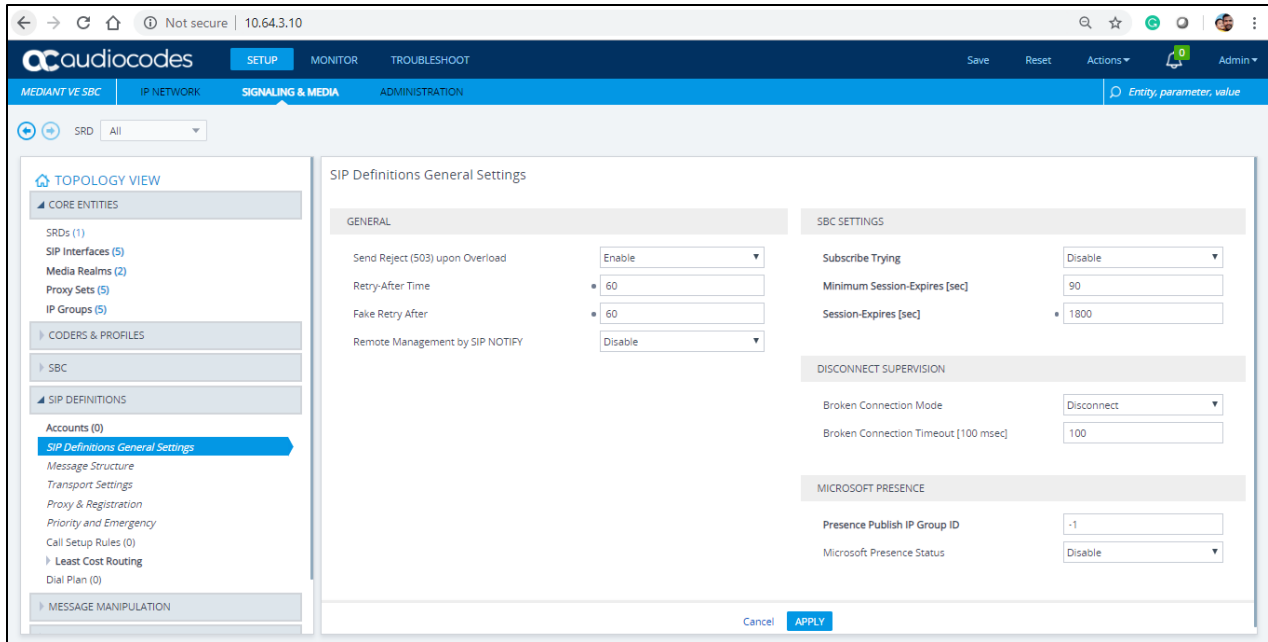


Figure 95 – SIP Definition

#### 4.5.14 Configure SBC General Settings

The screenshot below captures the configuration of the **SBC General Parameters** that was used during the test for the successful test execution.

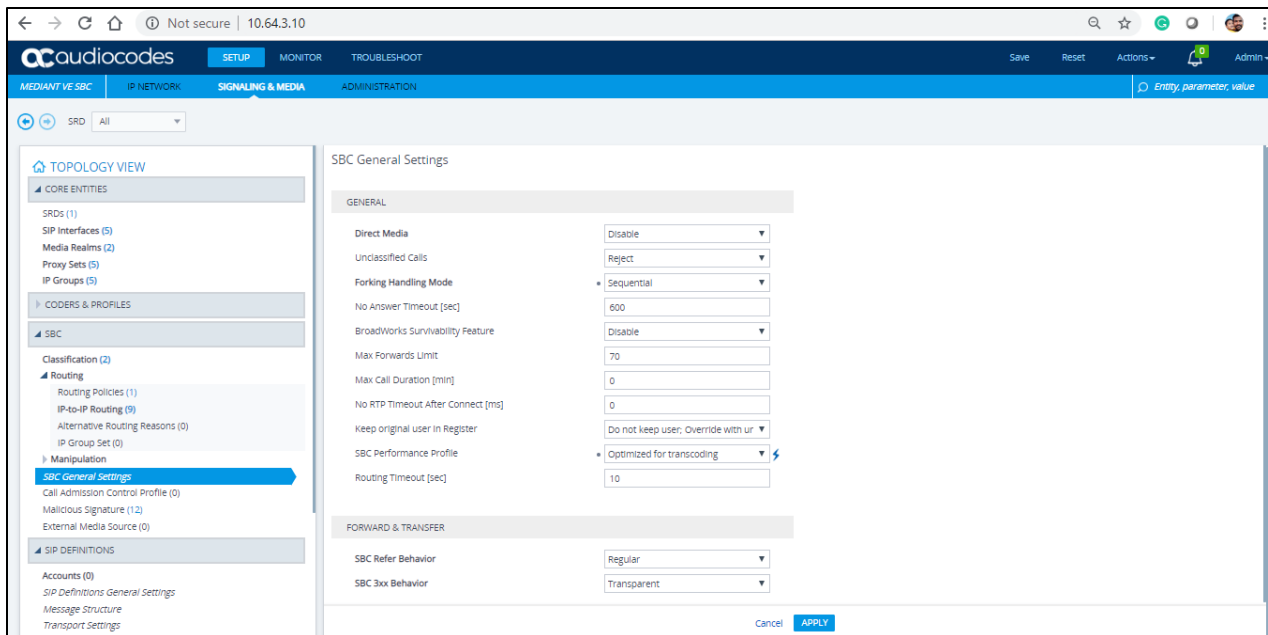


Figure 96 – SBC General Setting – Contd.

## 4.5.15 Configure IP-to-IP Routing Rules

This section describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call.

For the test, the following IP-To-IP Routing rules were configured to route calls between the Teams and CenturyLink

- Calls from Teams to PSTN Gateway
- Calls from PSTN Gateway to Teams
- Calls from Teams to Avaya
- Calls from Avaya to Teams

To configure IP-to-IP routing rules, navigate to **Signaling & Media** tab → **SBC** menu → **Routing** → **IP-to-IP Routing Table**. Click **Add**.

Calls from Teams to PSTN Gateway

The screenshot shows the configuration window for an IP-to-IP Routing rule titled "TEAMS -> PSTN". The window is divided into two main sections: GENERAL and ACTION.

**GENERAL Section:**

- Index:** 4
- Name:** TEAMS -> PSTN
- Alternative Route Options:** Route Row
- MATCH Section:**
  - Source IP Group:** #0 [TEAMS]
  - Request Type:** All
  - Source Username Pattern:** \*
  - Source Host:** \*
  - Source Tag:** (empty)

**ACTION Section:**

- Destination Type:** IP Group
- Destination IP Group:** #1 [PSTNGW]
- Destination SIP Interface:** #1 [PSTNGW]
- Destination Address:** (empty)
- Destination Port:** 0
- Destination Transport Type:** (empty)
- IP Group Set:** ..
- Call Setup Rules Set ID:** -1
- Group Policy:** Sequential
- Cost Group:** ..

At the bottom of the window, there are "Cancel" and "APPLY" buttons.

Figure 97 – Teams to PSTN – Contd.

Destination Username Pattern	*	Routing Tag Name	default
Destination Host	*	Internal Action	<input type="text"/> Editor
Destination Tag			
Message Condition	..		View
Call Trigger	Any		
ReRoute IP Group	Any		View

Cancel **APPLY**

Figure 98 – Teams to PSTN

### Calls from PSTN Gateway to Teams

IP-to-IP Routing [PSTNGW\_to\_TEAMS]

Routing Policy #0 [Default\_SBCRoutingPolicy]

GENERAL	ACTION
Index: 6	Destination Type: IP Group
Name: PSTNGW_to_TEAMS	Destination IP Group: #0 [TEAMS] View
Alternative Route Options: Route Row	Destination SIP Interface: #0 [TEAMS] View
<b>MATCH</b>	
Source IP Group: #1 [PSTNGW] View	Destination Address:
Request Type: All	Destination Port: 0
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: .. View
Source Tag:	Call Setup Rules Set ID: .1
Destination Username Pattern: *	Group Policy: Sequential
	Cost Group: .. View
	Routing Tag Name: default

Figure 99 – PSTN to Teams – Contd.

Destination Username Pattern	*	Routing Tag Name	default
Destination Host	*	Internal Action	<input type="text"/> Editor
Destination Tag			
Message Condition	..		View
Call Trigger	Any		
ReRoute IP Group	Any		View

Cancel **APPLY**

Figure 100 – PSTN to Teams

### Calls from Teams to Avaya

IP-to-IP Routing [Teams -> Avaya]

Routing Policy: #0 [Default\_SBCRoutingPolicy]

GENERAL	ACTION
Index: 3	Destination Type: IP Group
Name: Teams -> Avaya	Destination IP Group: #3 [AVAYA] <a href="#">View</a>
Alternative Route Options: Route Row	Destination SIP Interface: #3 [AVAYA] <a href="#">View</a>
<b>MATCH</b>	
Source IP Group: #0 [TEAMS] <a href="#">View</a>	Destination Address:
Request Type: All	Destination Port: 0
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: .. <a href="#">View</a>
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: .. <a href="#">View</a>

Figure 101 –Teams to Avaya.

Destination Username Pattern: 7	Routing Tag Name: default
Destination Host: *	Internal Action: <a href="#">Editor</a>
Destination Tag:	
Message Condition: .. <a href="#">View</a>	
Call Trigger: Any	
ReRoute IP Group: Any <a href="#">View</a>	
<a href="#">Cancel</a> <a href="#">APPLY</a>	

Figure 102 –Teams to Avaya Contd.

IP-to-IP Routing [Avaya -> Teams]

Routing Policy #0 [Default\_SBCRoutingPolicy]

GENERAL	ACTION
Index: 8	Destination Type: IP Group
Name: Avaya -> Teams	Destination IP Group: #0 [TEAMS] <a href="#">View</a>
Alternative Route Options: Route Row	Destination SIP Interface: #0 [TEAMS] <a href="#">View</a>
<b>MATCH</b>	
Source IP Group: #3 [AVAYA] <a href="#">View</a>	Destination Address:
Request Type: All	Destination Port: 0
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: -- <a href="#">View</a>
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: -- <a href="#">View</a>

Figure 103 –Avaya to Teams.

Destination Username Pattern: *	Routing Tag Name: default
Destination Host: *	Internal Action: <a href="#">Editor</a>
Destination Tag:	
Message Condition: -- <a href="#">View</a>	
Call Trigger: Any	
ReRoute IP Group: Any <a href="#">View</a>	
<a href="#">Cancel</a> <a href="#">APPLY</a>	

Figure 104 –Avaya to Teams – Contd.

## 4.5.16 IP Group

### IP Group – Teams

The screenshot shows the configuration page for an IP Group named 'TEAMS'. The SRD is set to '#0 [DefaultSRD]'. The 'GENERAL' section includes fields for Index (0), Name (TEAMS), Topology Location (Down), Type (Server), Proxy Set (#0 [TEAMS]), IP Profile (#1 [TEAMS\_Profile]), Media Realm (#0 [TEAMS]), Contact User, SIP Group Name (sbc4.tekvizionlabs.com), and Created By Routing Server (No). The 'QUALITY OF EXPERIENCE' section includes QoE Profile and Bandwidth Profile, both set to '..'. The 'MESSAGE MANIPULATION' section includes Inbound and Outbound Message Manipulation Sets (1 and 2), Message Manipulation User-Defined Strings, and Proxy Keep-Alive settings (Enable).

Figure 105 – IP Groups Teams – Contd.

The screenshot shows the configuration page for an IP Group named 'TEAMS', continuing from the previous page. The 'SBC REGISTRATION AND AUTHENTICATION' section includes 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, Username (Admin), and Password (.....). The 'ADVANCED' section includes Local Host Name (sbc4.tekvizionlabs.com), UUI Format (Disable), and Always Use Src Address (No). The 'GW GROUP STATUS' section includes GW Group Registered IP Address.

Figure 106 – IP Groups Teams – Contd.

SBC ADVANCED	
Source URI Input	<input type="text"/>
Destination URI Input	<input type="text"/>
SIP Connect	No
SBC PSAP Mode	Disable
Route Using Request URI Port	Disable
DTLS Context	#1 [Teams] <a href="#">View</a>
Keep Original Call-ID	No
Dial Plan	.. <a href="#">View</a>
Call Setup Rules Set ID	-1
Tags	<input type="text"/>

Cancel [APPLY](#)

Figure 107 – IP Groups Teams

## IP Group – PSTN Gateway

IP Groups [PSTNGW] - x

SRD

GENERAL	QUALITY OF EXPERIENCE
index: 1	QoE Profile: .. <a href="#">View</a>
Name: PSTNGW	Bandwidth Profile: .. <a href="#">View</a>
Topology Location: Up	
Type: Server	
Proxy Set: #1 [PSTNGW] <a href="#">View</a>	
IP Profile: #2 [PSTNGW_Profile] <a href="#">View</a>	
Media Realm: #1 [TEKVLAN] <a href="#">View</a>	
Contact User: <input type="text"/>	
SIP Group Name: 10.64.1.72	
Created By Routing Server: No	

MESSAGE MANIPULATION	
Inbound Message Manipulation Set: 0	
Outbound Message Manipulation Set: 3	
Message Manipulation User-Defined String 1: <input type="text"/>	
Message Manipulation User-Defined String 2: <input type="text"/>	
Proxy Keep-Alive using IP Group settings: Enable	

Figure 108 – IP Groups PSTN – Contd.

IP Groups [PSTNGW] - x

Used By Routing Server: Not Used		<b>SBC REGISTRATION AND AUTHENTICATION</b>
Proxy Set Connectivity: Connected		Max. Number of Registered Users: -1
<b>SBC GENERAL</b>		Registration Mode: User Initiates Registration
Classify By Proxy Set: Enable		User Stickiness: Disable
SBC Operation Mode: Not Configured		User UDP Port Assignment: Disable
SBC Client Forking Mode: Sequential		Authentication Mode: User Authenticates
CAC Profile: .. <a href="#">View</a>		Authentication Method List: <input type="text"/>
<b>ADVANCED</b>		SBC Server Authentication Type: According to Global Parameter
Local Host Name: <input type="text"/>		OAuth HTTP Service: .. <a href="#">View</a>
UI Format: Disable		Username: Admin
Always Use Src Address: No		Password: .....
		<b>GW GROUP STATUS</b>
		GW Group Registered IP Address: <input type="text"/>

Figure 109 – IP Groups PSTN – Contd.

SBC ADVANCED	
Source URI Input	<input type="text"/>
Destination URI Input	<input type="text"/>
SIP Connect	No
SBC PSAP Mode	Disable
Route Using Request URI Port	Disable
DTLS Context	#0 [default] <a href="#">View</a>
Keep Original Call-ID	No
Dial Plan	.. <a href="#">View</a>
Call Setup Rules Set ID	-1
Tags	<input type="text"/>

Cancel [APPLY](#)

Figure 110 – IP Groups PSTN

## IP Group – Avaya

*IP Groups [AVAYA]* - x

SRD

GENERAL	QUALITY OF EXPERIENCE
Index: <input type="text" value="3"/>	QoE Profile: <input type="text" value=".."/> <a href="#">View</a>
Name: <input style="border: 2px solid red;" type="text" value="AVAYA"/>	Bandwidth Profile: <input type="text" value=".."/> <a href="#">View</a>
Topology Location: <input type="text" value="Down"/>	
Type: <input type="text" value="Server"/>	
Proxy Set: <input style="border: 2px solid red;" type="text" value="#3 [AVAYA]"/> <a href="#">View</a>	
IP Profile: <input type="text" value="#4 [AVAYA_Profile]"/> <a href="#">View</a>	
Media Realm: <input type="text" value="#1 [TEKVLAN]"/> <a href="#">View</a>	
Contact User: <input type="text"/>	
SIP Group Name: <input style="border: 2px solid red;" type="text" value="10.64.5.57"/>	
Created By Routing Server: <input type="text" value="No"/>	

MESSAGE MANIPULATION

Inbound Message Manipulation Set:

Outbound Message Manipulation Set:

Message Manipulation User-Defined String 1:

Message Manipulation User-Defined String 2:

Proxy Keep-Alive using IP Group settings:

Figure 111 – IP Groups Avaya



Figure 112 – IP Groups Avaya – Contd.

Figure 113 – IP Groups Avaya – Contd.

#### 4.5.17 Message Manipulation

A Message Manipulation rule defines a manipulation sequence for SIP messages. SIP message manipulation enables the normalization of SIP messaging fields between communicating network segments. SIP message manipulations can also be implemented to resolve incompatibilities between SIP devices inside the enterprise network.

Each Message Manipulation rule is configured with a Manipulation Set ID. Groups (sets) of Message Manipulation rules can be created by assigning each of the relevant Message Manipulation rules to the same Manipulation Set ID.

The SIP message manipulation feature supports the following:

- Manipulation on SIP message type (Method, Request/Response, and Response type)
- Addition of new SIP headers
- Removal of SIP headers
- Modification of SIP header components such as values, header values (e.g., URI value of the P-Asserted-Identity header can be copied to the From header), call's parameter values
- Deletion of SIP body (e.g., if a message body is not supported at the destination network this body is removed)
- Translating one SIP response code to another
- Topology hiding (generally present in SIP headers such as Via, Record Route, Route and Service-Route).
- Configurable identity hiding (information related to identity of subscribers, for example P-Asserted-Identity, Referred-By, Identity and Identity-Info)

To configure Message Manipulation rules, navigate to **Signaling & Media** tab → **Message Manipulations** menu → **Message Manipulations**.

Click **Add** and populate the required fields in the screen that appears as below:

GENERAL		ACTION	
Index	1	Action Subject	<input type="text"/> Editor
Name	<input type="text"/>	Action Type	Add
Manipulation Set ID	0	Action Value	<input type="text"/> Editor
Row Role	Use Current Condition		
MATCH			
Message Type	<input type="text"/> Editor		
Condition	<input type="text"/> Editor		

Figure 114 – SIP Message Manipulation

Then click **Add** again, once the parameters have been configured.

For this test, the following message manipulations were configured and assigned to one manipulation set ID.

Manipulation set ID = 1: Manipulation from Teams

Manipulation set ID = 2: Manipulation to Teams

Manipulation set ID = 3: Manipulation to PSTN

Manipulation set ID = 6: Manipulation from Avaya

Manipulation set ID = 7: Manipulation to Avaya

Manipulation from Teams

- To Remove "Privacy" header: To Remove Privacy Header from Teams

The screenshot shows a configuration window titled "Message Manipulations [Filter Privacy ID except for Anonymous]". It is divided into two main sections: GENERAL and ACTION. In the GENERAL section, the Index is 28, the Name is "Filter Privacy ID except for Anonymous", and the Manipulation Set ID is 1. In the MATCH section, the Message Type is "Invite.Request" and the Condition is "Header.From.URL.Host contains '.com'". In the ACTION section, the Action Subject is "header.privacy", the Action Type is "Remove", and the Action Value is empty. Red boxes highlight the Name, Manipulation Set ID, Message Type, Condition, Action Subject, and Action Type fields.

Figure 115 – SIP Message Manipulation - Privacy

Manipulation to Teams

- To Modify "PAI" header: To display an FQDN instead of IP address for outbound calls towards Teams

The screenshot shows a configuration window titled "Message Manipulations [modify pai host towards teams]". It is divided into two main sections: GENERAL and ACTION. In the GENERAL section, the Index is 21, the Name is "modify pai host towards teams", and the Manipulation Set ID is 2. In the MATCH section, the Message Type is "Invite" and the Condition is empty. In the ACTION section, the Action Subject is "header.P-Asserted-Identity.URL.Host", the Action Type is "Modify", and the Action Value is "'sbcd.tekvizionlabs.com'". Red boxes highlight the Name, Manipulation Set ID, Message Type, Condition, Action Subject, Action Type, and Action Value fields.

Figure 116 – SIP Message Manipulation - PAI

- To Modify "TO" header: To display an FQDN instead of IP address for outbound calls towards Teams

The screenshot shows the configuration for a SIP message manipulation named "modify to towards teams". The interface is divided into several sections:

- GENERAL:**
  - Index: 19
  - Name: modify to towards teams
  - Manipulation Set ID: 2
  - Row Role: Use Current Condition
- MATCH:**
  - Message Type: Invite.request
  - Condition: (empty)
- ACTION:**
  - Action Subject: header.to.uri.host
  - Action Type: Modify
  - Action Value: 'sip.pstnhub.microsoft.com'

Buttons for "Cancel" and "APPLY" are visible at the bottom.

Figure 117 – SIP Message Manipulation - To

- To Modify "FROM" header: To display an FQDN instead of IP address for outbound calls towards Teams

The screenshot shows the configuration for a SIP message manipulation named "Towards Teams FROM". The interface is divided into several sections:

- GENERAL:**
  - Index: 0
  - Name: Towards Teams FROM
  - Manipulation Set ID: 2
  - Row Role: Use Current Condition
- MATCH:**
  - Message Type: Options
  - Condition: param.message.address.dst.sipinterface=='0'
- ACTION:**
  - Action Subject: Header.From.URL
  - Action Type: Modify
  - Action Value: 'sip.admin@sbcd.tekvizionlabs.com'

Buttons for "Cancel" and "APPLY" are visible at the bottom.

Figure 118 – SIP Message Manipulation - From

- To Modify "CONTACT" header: To display an FQDN instead of IP address for outbound calls towards Teams

Message Manipulations [Towards Teams Contact]

**GENERAL**

Index: 1

Name: towards Teams Contact

Manipulation Set ID: 2

Row Role: Use Current Condition

**MATCH**

Message Type: Options

Condition: param.Message.Address.Dst.SIPinterface==0

**ACTION**

Action Subject: Header.Contact.URL.Host

Action Type: Modify

Action Value: 'sbc4.tekvizionlabs.com'

Buttons: Cancel, APPLY

Figure 119 – SIP Message Manipulation - Contact

- To Modify “FROM” header: To display an FQDN instead of IP address for outbound calls towards Teams

Message Manipulations [Towards Teams]

**GENERAL**

Index: 2

Name: Towards Teams

Manipulation Set ID: 2

Row Role: Use Current Condition

**MATCH**

Message Type: invite:Request

Condition:

**ACTION**

Action Subject: Header.From.URL.host

Action Type: Modify

Action Value: 'sbc4.tekvizionlabs.com'

Buttons: Cancel, APPLY

Figure 120 – SIP Message Manipulation - From

### Manipulation to PSTN

- To Modify “TO” header: To display an IP for an PSTN Gateway

Message Manipulations [towards PSTNGW TO]

GENERAL		ACTION	
Index	3	Action Subject	header.to.url.host Editor
Name	towards PSTNGW TO	Action Type	Modify
Manipulation Set ID	3	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Options Editor		
Condition	Param.Message.Address.dst.SIPInterface=='1' Editor		

Cancel APPLY

Figure 121 – SIP Message Manipulation – To

- To Modify "FROM" header: To display an IP for an AudioCodes

Message Manipulations [Towards PSTNGW FROM]

GENERAL		ACTION	
Index	4	Action Subject	Header.From.URL.host Editor
Name	Towards PSTNGW FROM	Action Type	Modify
Manipulation Set ID	3	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Options Editor		
Condition	Param.Message.Address.dst.SIPInterface=='1' Editor		

Cancel APPLY

Figure 122 – SIP Message Manipulation – From

- To Modify "Referred-By" header: To display an IP for an AudioCodes in Referred by

The screenshot shows a configuration window titled "Message Manipulations [Referred-By to PSTNGW]". It is divided into three main sections: GENERAL, MATCH, and ACTION.

- GENERAL:**
  - Index: 5
  - Name: Referred-By to PSTNGW
  - Manipulation Set ID: 3
  - Row Role: Use Current Condition
- MATCH:**
  - Message Type: Invite
  - Condition: Header.Referred-By exists
- ACTION:**
  - Action Subject: Header.Referred-By.url.host
  - Action Type: Modify
  - Action Value: '10.64.3.10'

Buttons for "Cancel" and "APPLY" are located at the bottom center.

Figure 123 – SIP Message Manipulation – Referred - By

- To Modify "FROM" header: To display an IP for an AudioCodes in From

The screenshot shows a configuration window titled "Message Manipulations [Towards PSTNGW Invite]". It is divided into three main sections: GENERAL, MATCH, and ACTION.

- GENERAL:**
  - Index: 6
  - Name: Towards PSTNGW Invite
  - Manipulation Set ID: 3
  - Row Role: Use Current Condition
- MATCH:**
  - Message Type: Invite Request
  - Condition: (empty)
- ACTION:**
  - Action Subject: Header.From.URL.Host
  - Action Type: Modify
  - Action Value: '10.64.3.10'

Buttons for "Cancel" and "APPLY" are located at the bottom center.

Figure 124 – SIP Message Manipulation – From

### Manipulation to Avaya

- To Modify "Diversion" header: To display AudioCodes IP

Message Manipulations [Teams -> Avaya Modify Diversion header]

GENERAL		ACTION	
Index	22	Action Subject	header.Diversion.url.host Editor
Name	Teams -> Avaya Modify Diversion header	Action Type	Add
Manipulation Set ID	7	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	invite.request Editor		
Condition	Header.Diversion exists Editor		

Cancel APPLY

Figure 125 – SIP Message Manipulation – Diversion

- To Modify "FROM" header: To display an IP for an AudioCodes in From

Message Manipulations [Modify SBC IP Teams -> Avaya]

GENERAL		ACTION	
Index	18	Action Subject	Header.From.URL.Host Editor
Name	Modify SBC IP Teams -> Avaya	Action Type	Modify
Manipulation Set ID	7	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Invite.Request Editor		
Condition	Editor		

Cancel APPLY

Figure 126 – SIP Message Manipulation – From

- To Modify "Referred-By" header: To display an IP for an AudioCodes in Referred by



Message Manipulations [Referred-By Teams -> Avaya]

GENERAL		ACTION	
Index	17	Action Subject	Header.Referred-By.url.host Editor
Name	Referred-By Teams -> Avaya	Action Type	Modify
Manipulation Set ID	7	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Invite Editor		
Condition	Header.Referred-By exists Editor		

Cancel APPLY

Figure 127 – SIP Message Manipulation – Referred By

- To Modify “FROM” header: To display an IP for an AudioCodes in From

Message Manipulations [From header Teams -> Avaya]

GENERAL		ACTION	
Index	15	Action Subject	Header.From.URL.host Editor
Name	From header Teams -> Avaya	Action Type	Modify
Manipulation Set ID	7	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Options Editor		
Condition	Param.Message.Address.dst.SIPInterface== Editor		

Cancel APPLY

Figure 128 – SIP Message Manipulation – From

- To Modify “TO” header: To display an IP for an AudioCodes in to

Message Manipulations [To header Teams -> Avaya]

GENERAL		ACTION	
Index	16	Action Subject	Header.To.URL.host <a href="#">Editor</a>
Name	To header Teams -> Avaya	Action Type	Modify
Manipulation Set ID	7	Action Value	'10.64.3.10' <a href="#">Editor</a>
Row Role	Use Current Condition		
MATCH			
Message Type	Options <a href="#">Editor</a>		
Condition	Param.Message.Address.dst.SIPInterface== <a href="#">Editor</a>		

Cancel [APPLY](#)

Figure 129 – SIP Message Manipulation – to

## 4.6 Avaya Aura Communication Manager Configuration

### 4.6.1 Version

Execute **swversion** to find the version for Avaya Aura Communication Manager

```
admin@lab126-cm7>
admin@lab126-cm7>
admin@lab126-cm7> swversion
  Operating system:  Linux 3.10.0-514.6.2.el7.AV1.x86_64 x86_64 x86_64
                   Built:  Feb 27 10:13 2017

  Contains:  01.0.532.0
  CM Reports as:  R017x.01.0.532.0
  CM Release String:  vcm-017-01.0.532.0
  RTS Version:  CM 7.1.3.1.0.532.24811
  Publication Date:  25 June 2015
  VMwaretools version:  9.10.2.48224 (build-2822639)

UPDATES:
Update ID                Status      Type  Update description
-----
01.0.532.0-24184         unpacked   cold  7.1.2.0.0-FP2
01.0.532.0-24515         unpacked   cold  7.1.3.0.0-FP3
01.0.532.0-24811         activated  cold  7.1.3.1.0-FP3SP1
01.0.532.0-25082         unpacked   cold  7.1.3.3.0-FP3SP3

Platform/Security ID      Status      Type  Update description
-----

CM Translation Saved:    2019-09-25 22:00:52
CM License Installed:    2019-09-25 13:41:42
```

Figure 130 - Version

### 4.6.2 IP Node Name

Use the **change node-names ip** command to verify that node names have been properly defined for Communication Manager (procr) and Session Manager (ASM7 in this test). These node names will be needed for configuring a Signaling Group later.

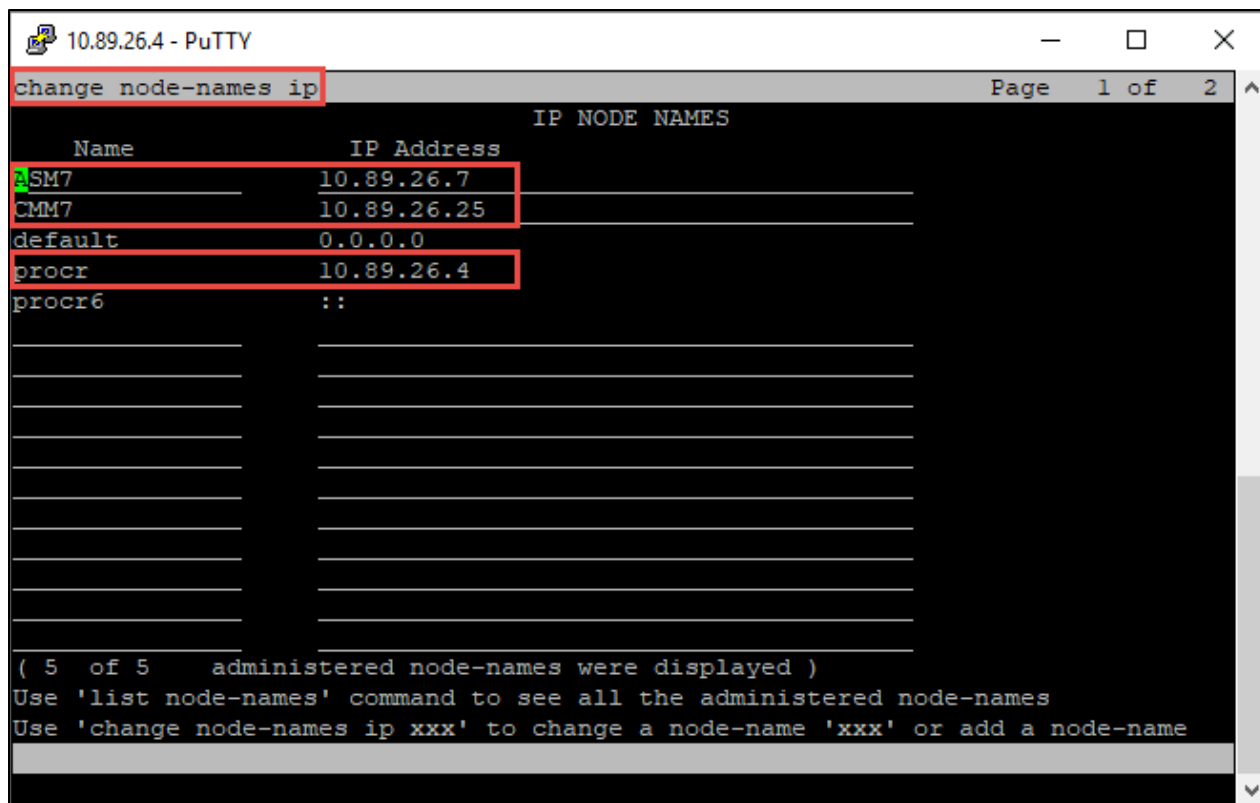


Figure 131 - IP Node Name

### 4.6.3 IP Codec Set

Use **change ip-codec-set <n>** command to define a list of codecs for calls from Avaya Aura

1. Set **Audio Codec**: G.711MU is entered
2. Leave other fields at default values

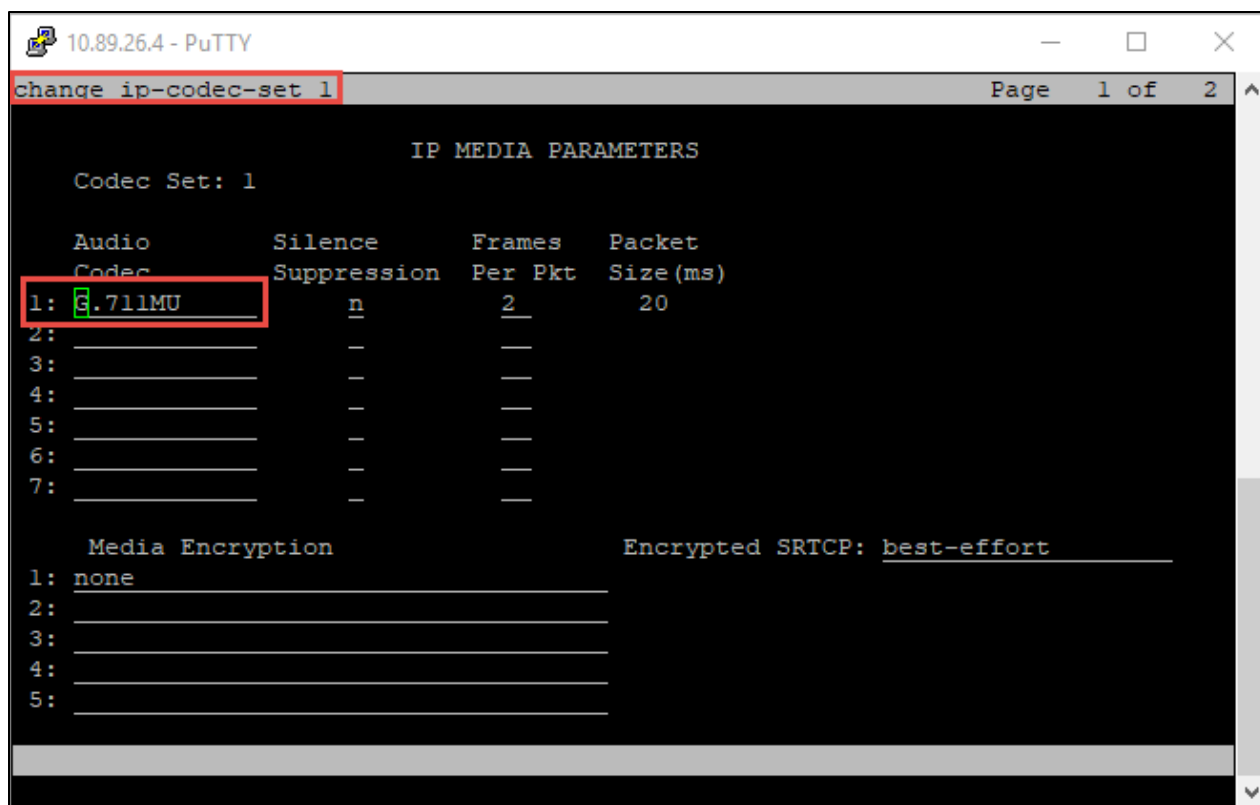


Figure 132 - IP Codec Set

#### 4.6.4 IP Network Region

IP Network Region 1 is utilized. Command **change ip-network-region 1** is issued

1. Set **Codec Set**: 1, which is programmed in the previous step
2. Set **Intra-region IP-IP Direct Audio**: yes
3. Set **Inter-region IP-IP Direct Audio**: yes
4. Leave other fields at default values

```
10.89.26.4 - PuTTY
change ip-network-region 1 Page 1 of 20
IP NETWORK REGION
Region: 1 NR Group: 1
Location: Authoritative Domain: lab.tekvizion.com
Name: Stub Network Region: n
MEDIA PARAMETERS
Codec Set: 1 Intra-region IP-IP Direct Audio: yes
Inter-region IP-IP Direct Audio: yes
UDP Port Min: 2048 IP Audio Hairpinning? n
UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
H.323 IP ENDPOINTS AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y RSVP Enabled? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

Figure 133 - IP Network Region

#### 4.6.5 Signaling Groups

Signaling group is configured for SIP trunk.

Command **add signaling-group x** was used to create Signaling Group, command **change signaling-group <x>** is used to modify an existing Signaling Group. Signaling Group 1 is used for the SIP trunk.

1. Set **Group Type**: sip
2. Set **Transport Method**: tcp
3. Set **Peer Detection Enable**: y
4. Set **Near-end Node Name**: procr
5. Set **Near-end Listen Port**: 5060
6. Set **Far-end Node Name**: ASM7
7. Set **Far-end Listen Port**: 5060
8. Set **Far-end Network Region**: 1
9. Set **DTMF over IP**: rtp-payload
10. Set **Direct IP-IP Audio Connections?**: y
11. Leave other fields as default value

```
10.89.26.4 - PuTTY
change signaling-group 1 Page 1 of 2
SIGNALING GROUP
Group Number: 1 Group Type: sip
IMS Enabled?  Transport Method: tcp
Q-SIP? 
IP Video?  Enforce SIPS URI for SRTP? 
Peer Detection Enabled?  Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? 
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? 
Alert Incoming SIP Crisis Calls? 
Near-end Node Name: procr Far-end Node Name: ASM7
Near-end Listen Port: 5060 Far-end Listen Port: 5062
Far-end Network Region: 1
Far-end Domain: lab.tekvizion.com
Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? 
RFC 3389 Comfort Noise? 
DTMF over IP: rtp-payload Direct IP-IP Audio Connections? 
Session Establishment Timer (min): 3 IP Audio Hairpinning? 
Enable Layer 3 Test? 
Alternate Route Timer (sec): 6
```

Figure 134 - Signaling Group

#### 4.6.6 Trunk Groups

Similar to Signaling Group, Trunk Group is related for this setup, Trunk Group 1 is for the SIP Trunk. Command **change trunk-group 1**.

1. Set **Group Type**: sip
2. Set **Group Name**: SIP Trunk, for example
3. Set **TAC**: #001, this value is given based on the system dial plan
4. Set **Direction**: two-way
5. Set **Service Type**: public-ntwrk
6. Set **Member Assignment Method**: auto
7. Set **Signaling Group**: 1
8. Set **Number of Members**: Enter a number between 1 and the max number of licensed SIP trunks

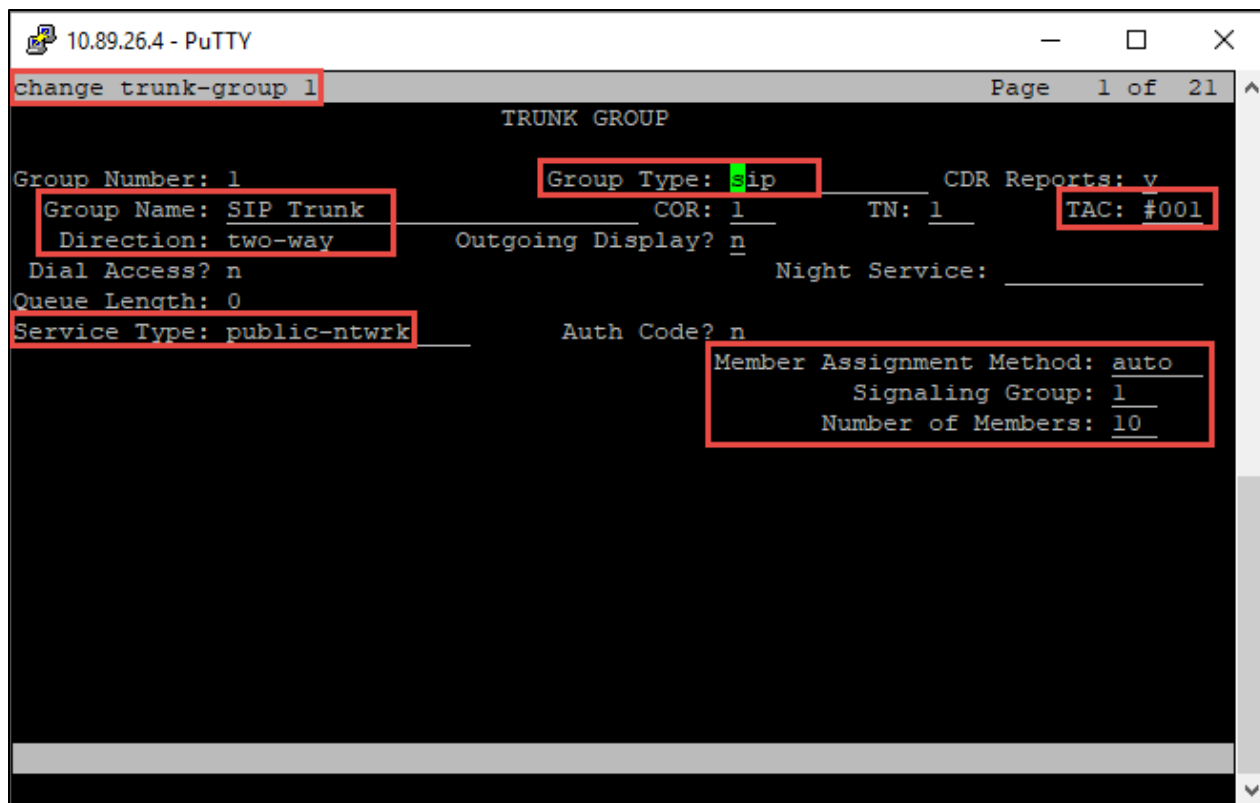


Figure 135 - Trunk Group

#### 4.6.7 Route Pattern

Use **change route-pattern <x>** command to specify the routing preference, Route pattern 1 is for SIP Trunk.

1. Set **Pattern Name:** to ASM7
2. Set **Grp No:** Trunk group 1 is given here
3. Set **FRL:** 0 is given as it has the least restriction
4. Set **Numbering Format:** unk-unk
5. Leave all other fields at default values



```

10.89.26.4 - PuTTY
change route-pattern 1 Page 1 of 3
Pattern Number: 1 Pattern Name: ASM7
SCCAN? n Secure SIP? n Used for SIP stations? n

Grp FRL NPA Pfx Hop Toll No. Inserted DCS/ IXC
No Mrk Lmt List Del Digits QSIG Intw
1: 1 0
2:
3:
4:
5:
6:

BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM Sub Numbering LAR
0 1 2 M 4 W Request Dgts Format
1: y y y y y n n rest unk-unk none
2: y y y y y n n rest none
3: y y y y y n n rest none
4: y y y y y n n rest none
5: y y y y y n n rest none
6: y y y y y n n rest none

```

Figure 136 - Route Pattern

#### 4.6.8 Outbound Call Routing

For outbound call to PSTN through AudioCodes, AAR is used. Use command **change aar analysis <x>** to configure the routing table. Here is an example to configure the AAR to call to Teams user

1. Set **Dialed String**: 8 is given for calling Teams user.
2. Set **Min**: 5 is given here
3. Set **Max**: 5 is given here
4. Set **Route Pattern**: The previously configured Route Pattern 1 is given here
5. Set **Call Type**: aar is given here

10.89.26.4 - PuTTY

change aar analysis 8 Page 1 of 2

AAR DIGIT ANALYSIS TABLE  
Location: all Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
8	7	7	254	aar		n
800	5	5	1	aar		n
8000	4	4	1	aar		n
9	7	7	254	aar		n
						n
						n
						n
						n
						n
						n
						n
						n
						n
						n

Figure 137 - Outbound Call Routing

#### 4.6.9 Inbound Call Termination

For inbound call to Avaya Communication Manager, the following configuration is made. Use command **change private-numbering <x>** to map the incoming number to extension. Here is an example to configure the incoming call termination.

1. Set **Ext code**: 7500 or 7501 is given for calling Teams user.
2. Set **Trk Grp(s)**: 3 is given here
3. Set **Private Prefix**: 7500 and 7501
4. Set **Total Len**: 4

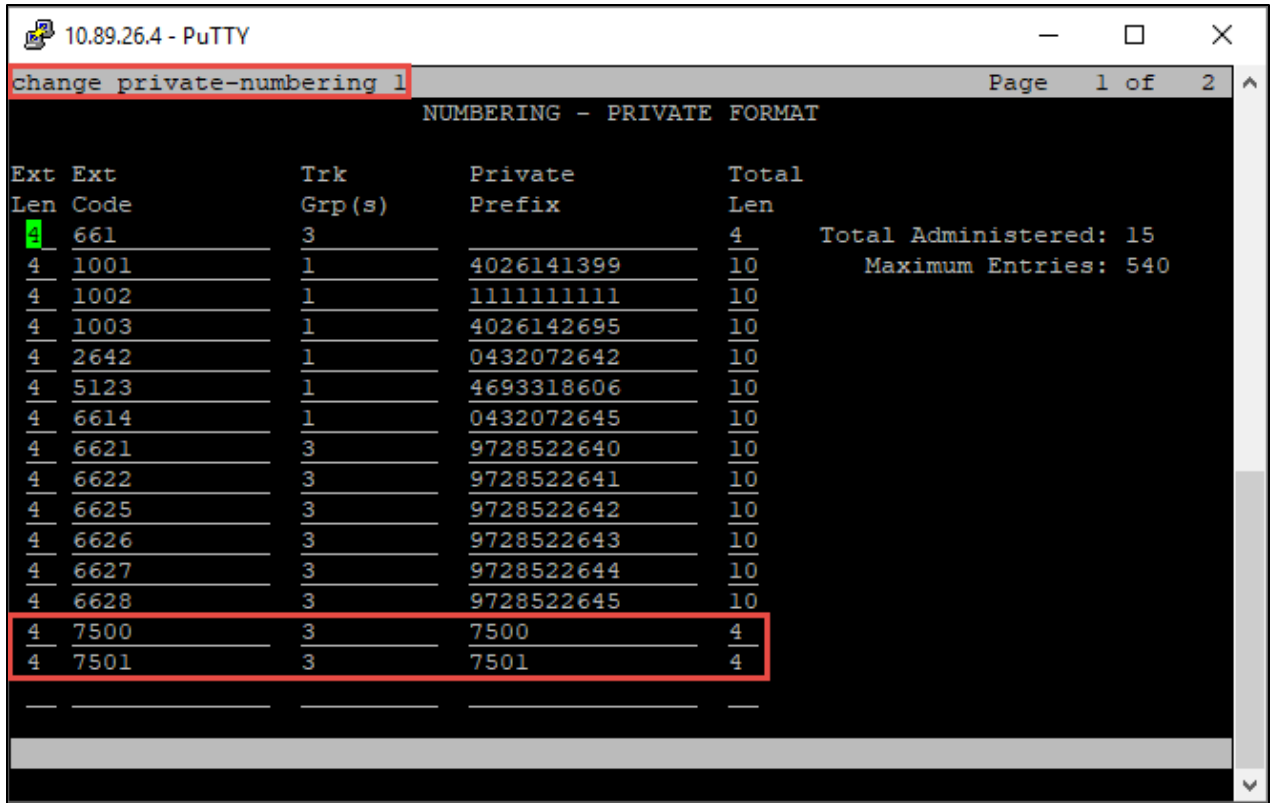


Figure 138 - Inbound Call Routing

## 4.7 Avaya Aura Session Manager Configuration

Avaya Aura Session Manager Configuration is accomplished through the Avaya Aura System Manager.

1. Access Avaya Aura System Manager Web login screen via <https://<IP Address/FQDN>>, the IP address is 10.89.26.3 in our lab
2. Use admin as User ID and associated password
3. Click Log On

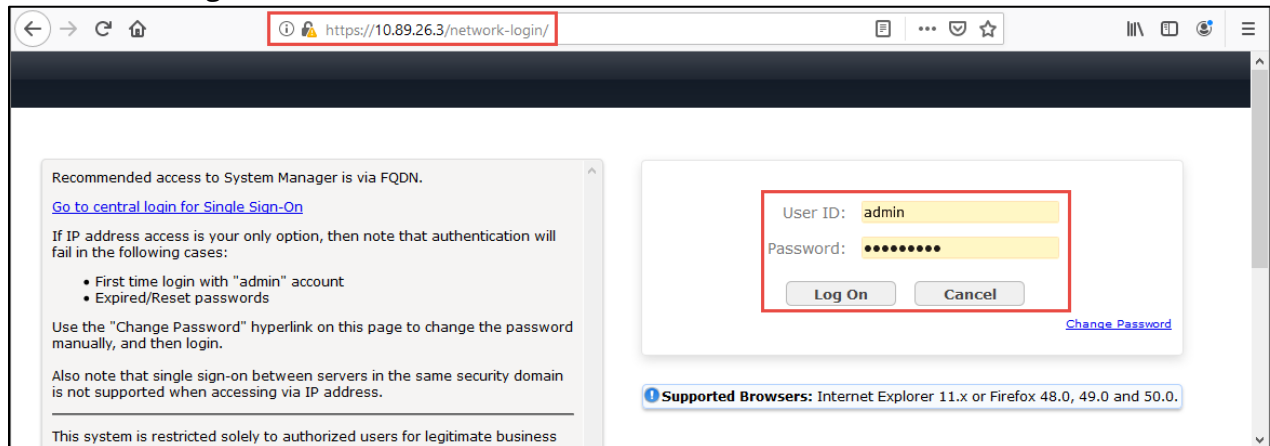


Figure 139 - Log into Avaya Aura System Manager

## Navigate to Elements → Routing

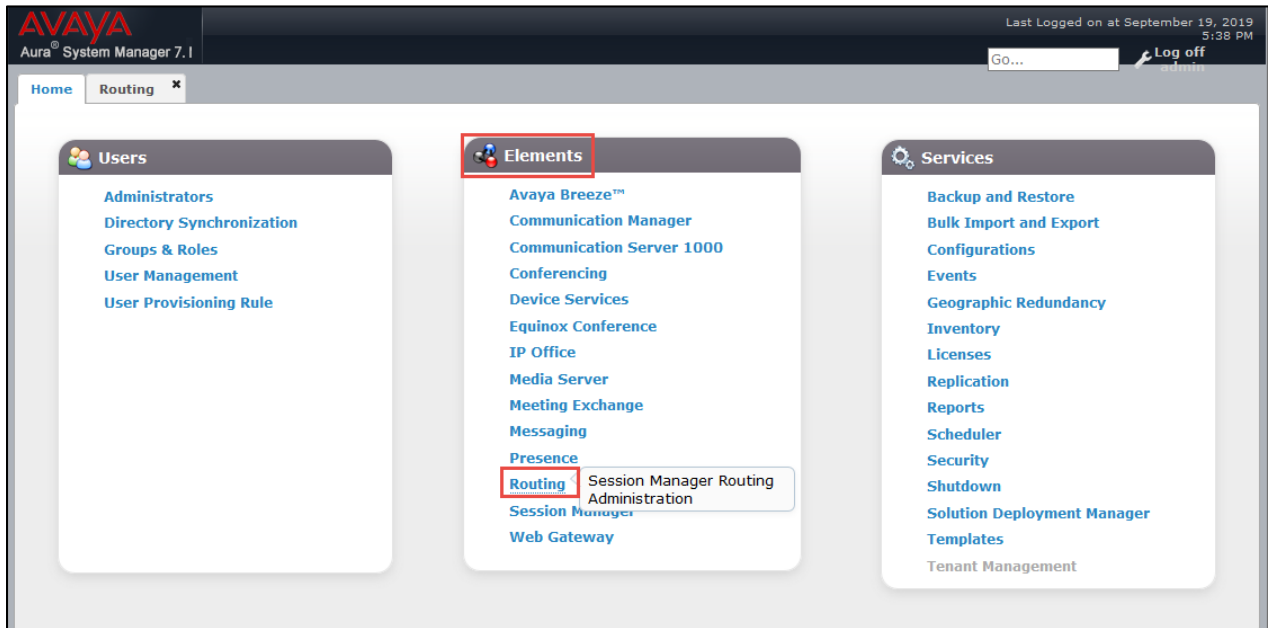


Figure 140 - Routing

### 4.7.1 Version

The version of Avaya System Manager used for the testing is given below

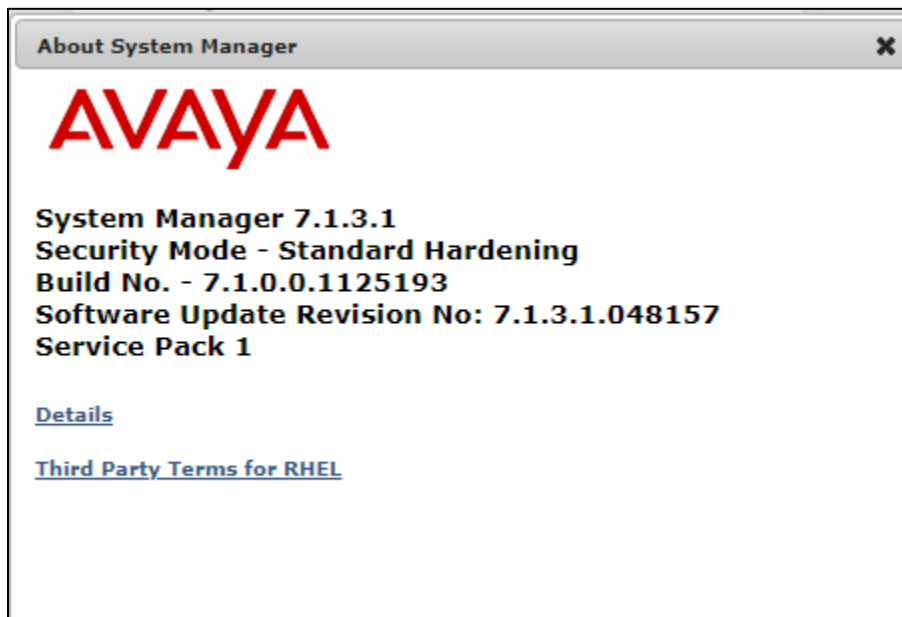


Figure 141 – Version

## 4.7.2 Domains

1. Navigate to **Routing -> Domains**
2. Click **New**

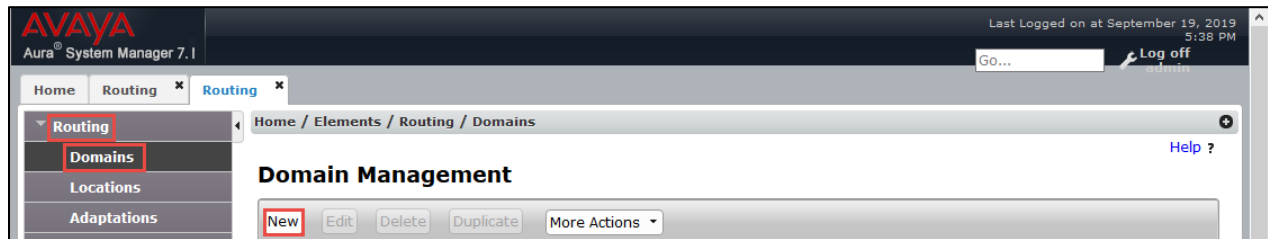


Figure 142 – Add Domain

3. Set **Name**: Enter the domain name of Avaya Aura PBX, lab.tekvizion.com is given for the test
4. Set **Type**: sip
5. Click **Commit**

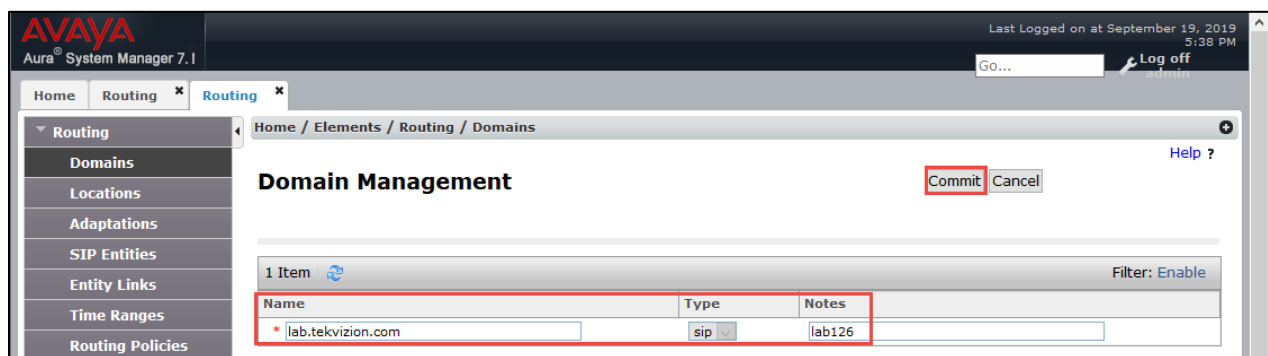


Figure 143 - Domain

## 4.7.3 Locations

1. Navigate to **Routing → Locations**
2. Select **New**
3. Set **Name**: Enter the name of your location, Lab126-Plano is set here
4. Under Location Pattern, select **Add** to add IP Address Patterns for different networks that communication within the location
5. Set **IP Address Pattern**: 10.89.26.\*
6. Leave all other fields at default values
7. Click **Commit**

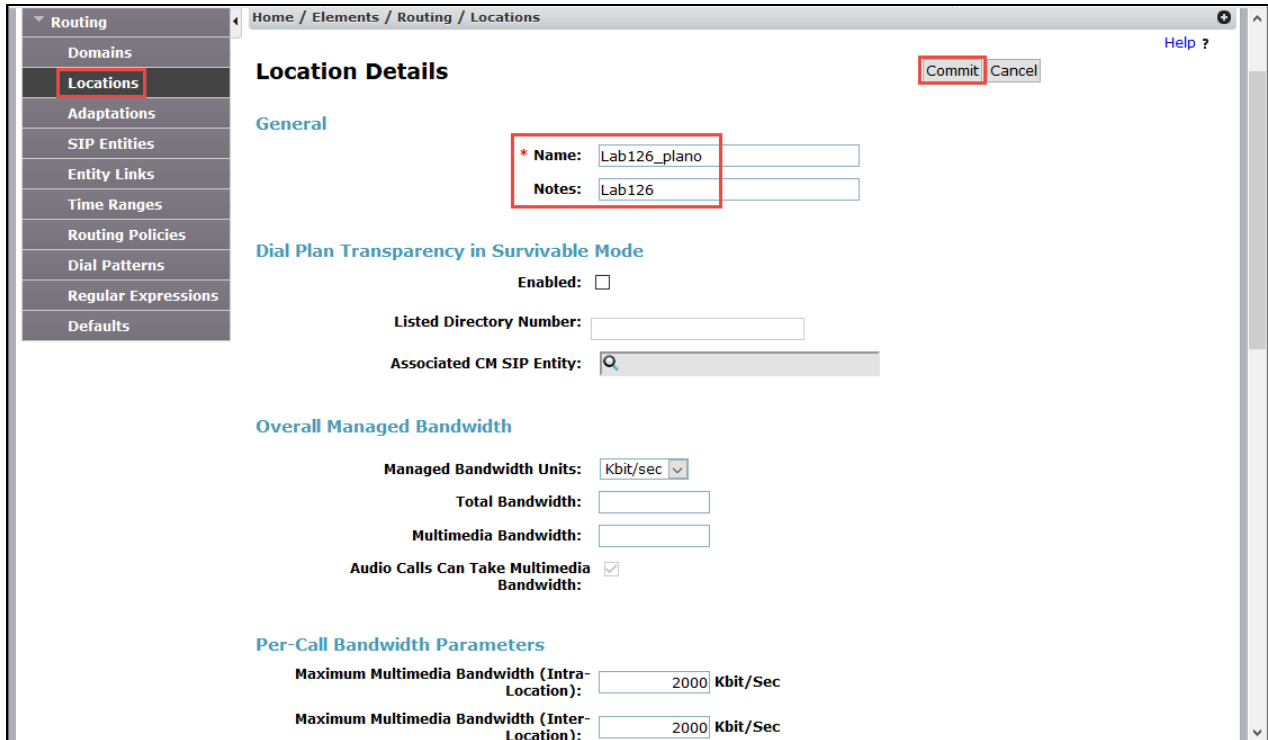


Figure 144- Add Location

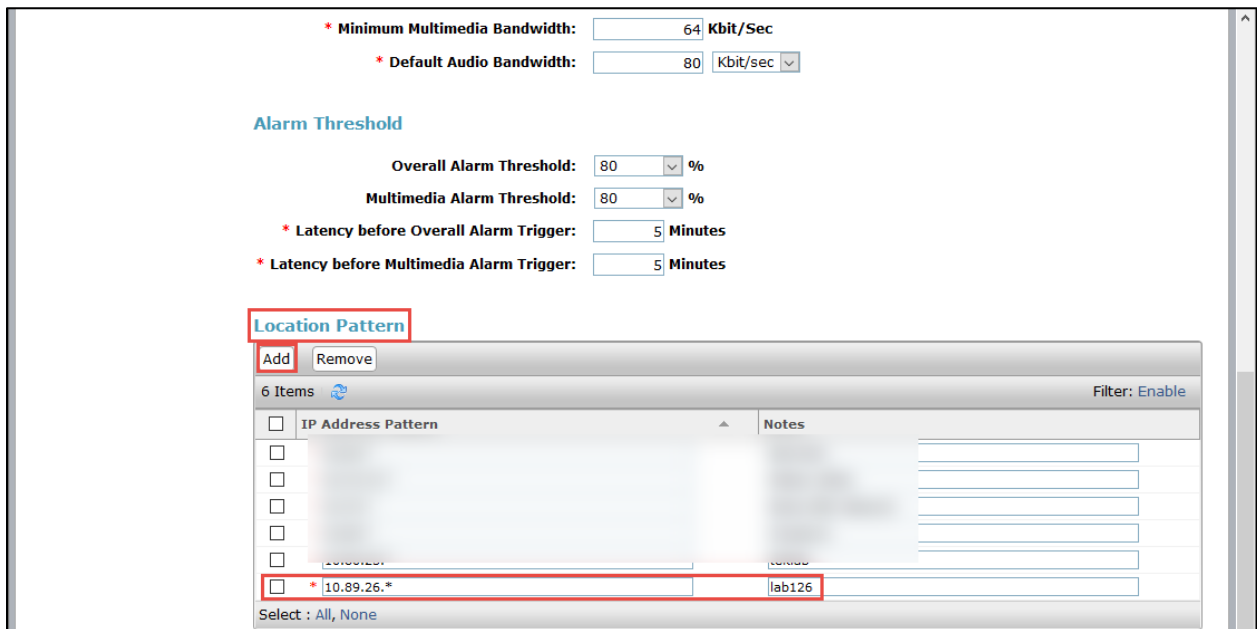


Figure 145 - Add Location

#### 4.7.4 Adaptation

Adaptation was created at the Session Manager for Avaya CM

1. Navigate to **Routing** → **Adaptations**. Click New
2. Set **Adaptation Name**: adaptation CM, for example

3. Set **Module Name**: DigitConversionAdapter
4. Set **Module Parameter Type**: Name-Value Parameter is selected from the drop down, Click **Add**
5. Set **Name/Value**: fromto/true
6. Leave all other fields at default values
7. Click **Commit**

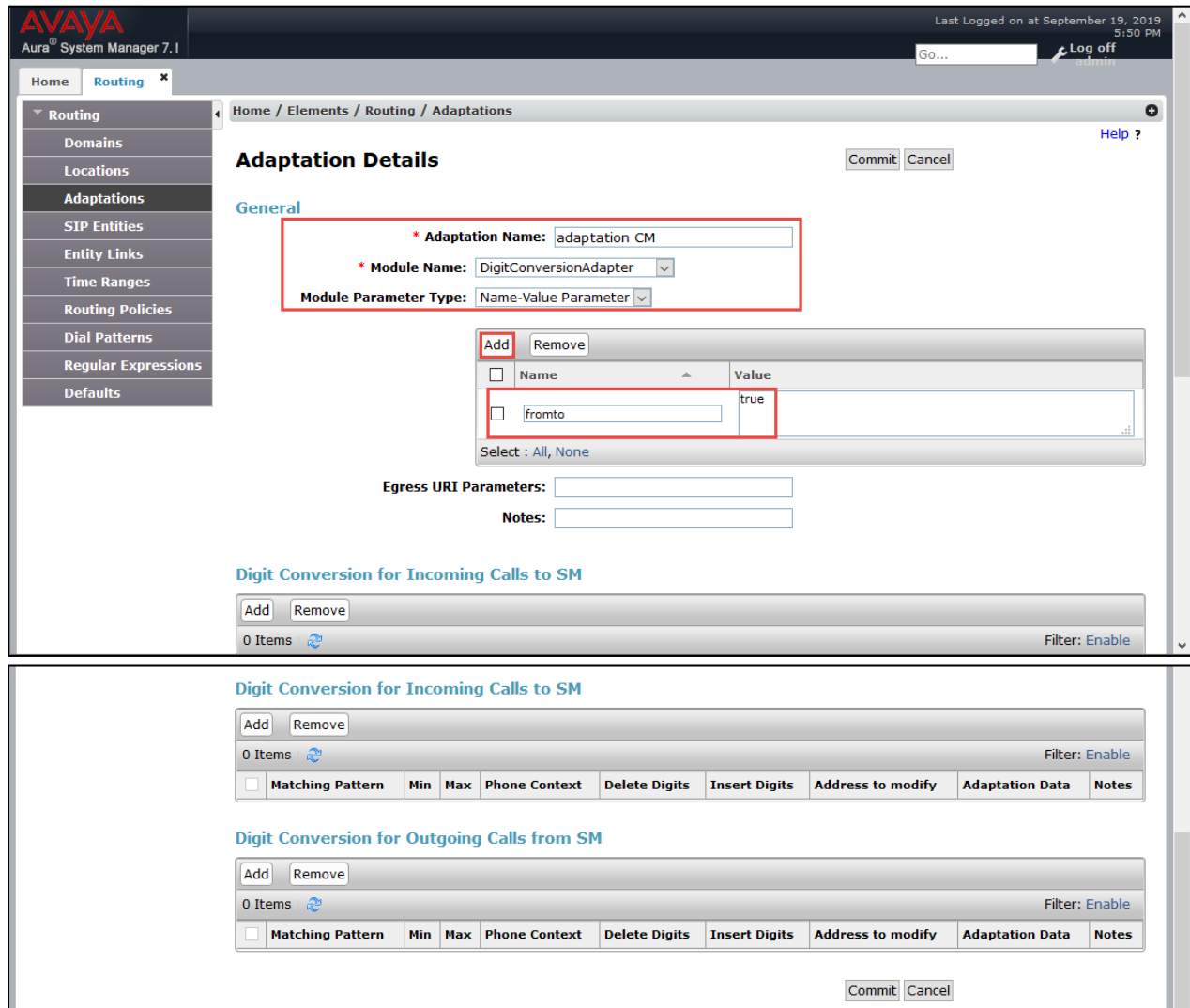


Figure 146 - Add Adaptation

#### 4.7.5 SIP Entities and Entity Links

Navigate to: **Routing** → **SIP Entities**. Click **New**

##### 4.7.5.1 SIP Entity for Avaya Aura Session Manager

1. Navigate to: **Routing** → **SIP Entities**. Click **New**
2. SIP Entity for Avaya Aura Session Manager

3. Set **Name**: Enter name of the host, Lab126-SM7 is used here for example
4. Set **FQDN or IP Address**: Enter the SIP address of the Session Manager
5. Set **Type**: Session Manager is selected from the drop down
6. Set **Location**: Select the location configured in the previous step  
Under Listen Port:
7. Set **TCP/TLS Failover Port**: 5060/5061
8. Click **Add** to assign Domain lab.tekvizion.com for the following Ports and Protocols
9. Port **5060** and Protocol **TCP/UDP**
10. Port **5061** and Protocol **TLS**
11. Leave all other fields at default values
12. Click **Commit**

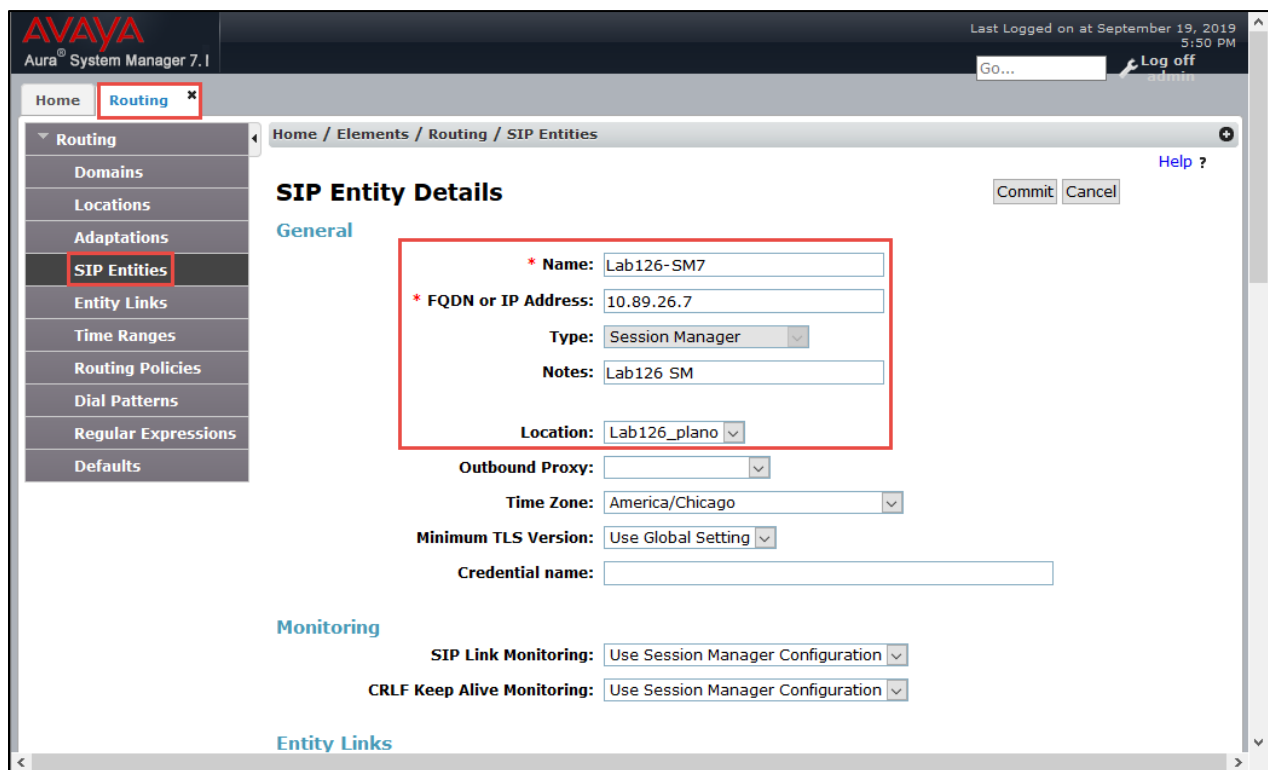


Figure 147 - SIP Entity: Avaya Aura Session Manager



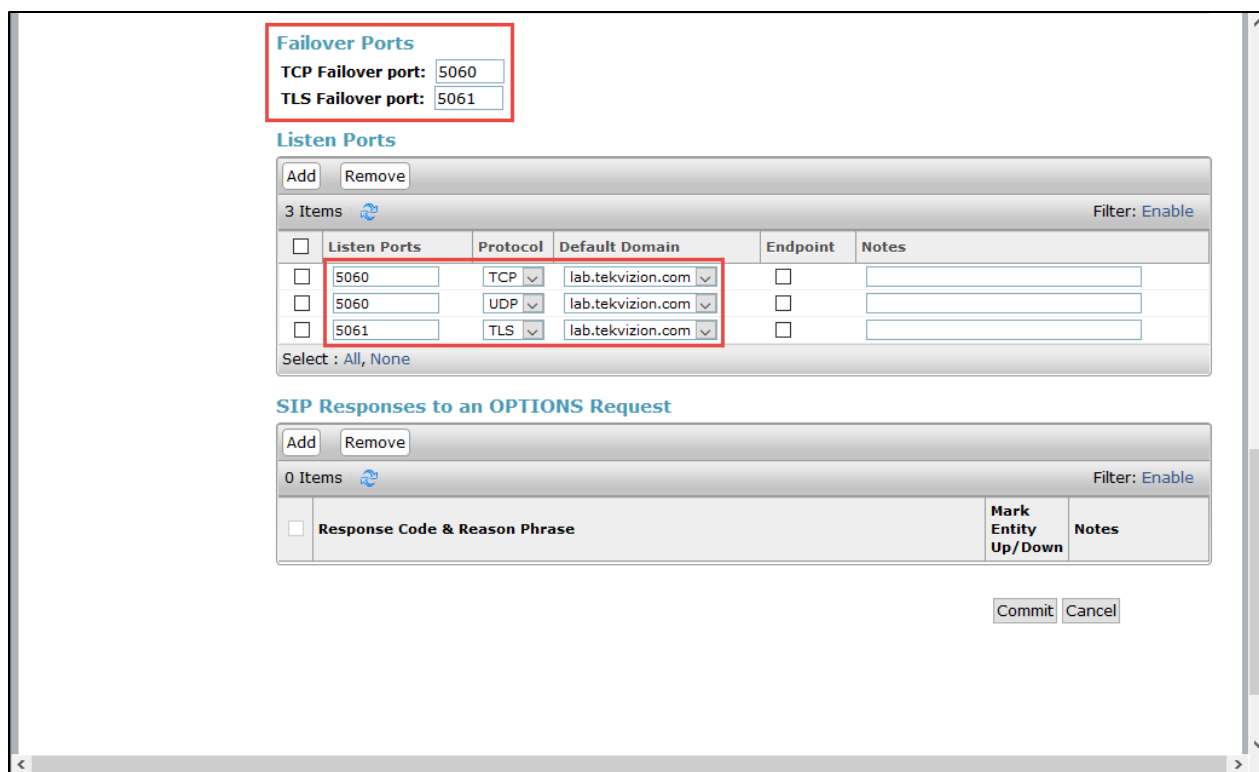


Figure 148 - SIP Entity: Avaya Aura Session Manager

#### 4.7.5.2 SIP Entity for Communication Manager SIP Trunk

1. Set **Name**: CM\_SIP\_Trunk
2. Set **FQDN or IP Address**: Enter the IP address of Avaya Aura Communication Manager
3. Set **Type**: CM
4. Set **Adaptation**: adaptation CM
5. Set **Location**: Select the location configured in previous step
6. Under **Entity Links**, Click **Add**
7. Set **SIP Entity 1**: Select the SIP entity Lab126-SM7 configured in previous step
8. Set **SIP Entity 2**: Select the SIP entity CM\_SIP\_Trunk
9. Set **Protocol**: TCP was used for this test
10. Set **Ports**: Set SIP Entity 1 Port to 5062 and SIP Entity 2 Port to 5060
11. Set **Connection Policy**: trusted
12. Leave all other fields at default values
13. Click **Commit**

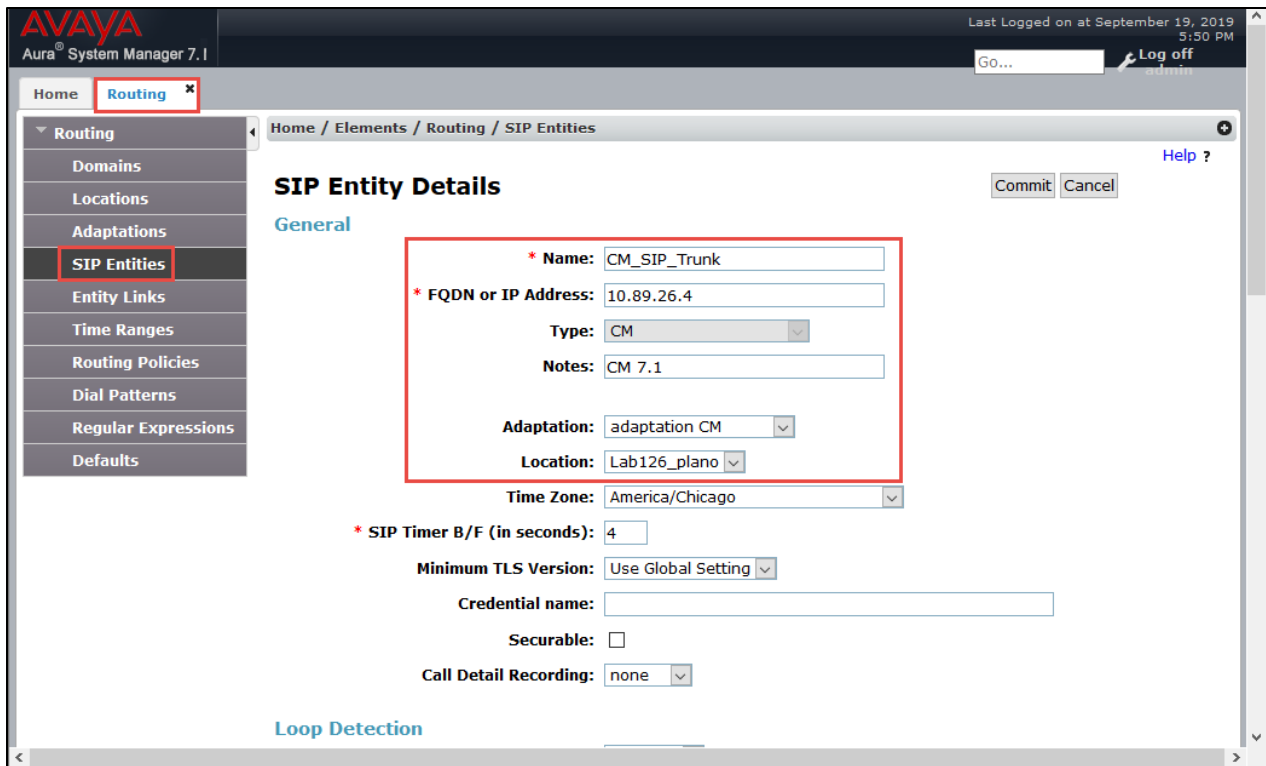


Figure 149- SIP Entity: Avaya Aura Communication Manager for SIP Trunk

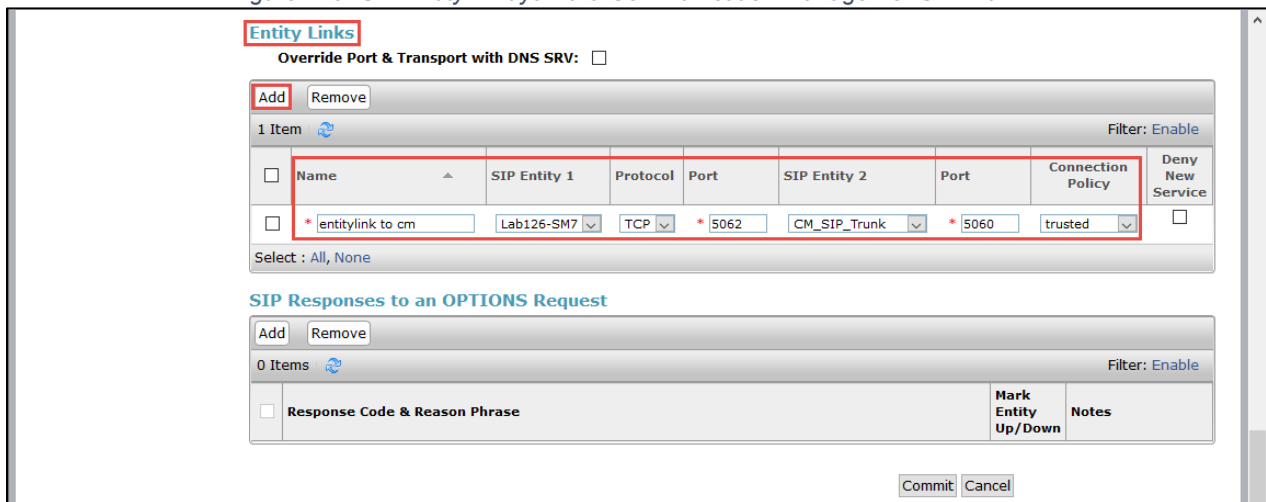


Figure 150 - SIP Entity: Avaya Aura Communication Manager for SIP Trunk

#### 4.7.5.3 SIP Entity for Avaya SBCE

1. Set **Name:** ASBC\_Crestron
2. Set **FQDN or IP Address:** Enter the IP address of Avaya SBCE interface facing Avaya Aura Session Manager
3. Set **Type:** SIP Trunk
4. Set **Location:** Select the location configured in the previous step
5. Under **Entity Links**, Click **Add**

6. Set **SIP Entity 1**: Select the SIP Entity Lab126-SM7 configured in previous step
7. Set **SIP Entity 2**: Select the SIP Entity ASBC\_Crestron
8. Set **Protocol**: TCP was used for this test
9. Set **Ports**: Set both Ports to 5060
10. Set **Connection Policy**: trusted
11. Leave all other fields at default values
12. Click **Commit**

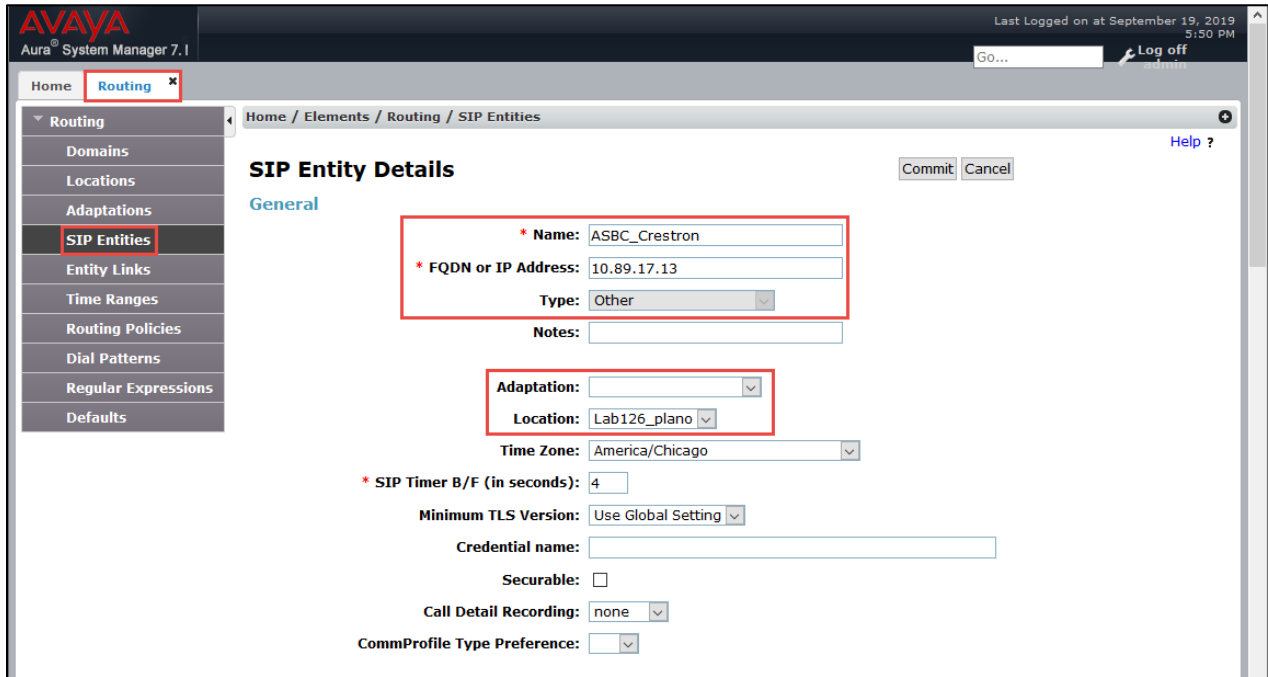


Figure 151 - SIP Entity: Avaya SBCE

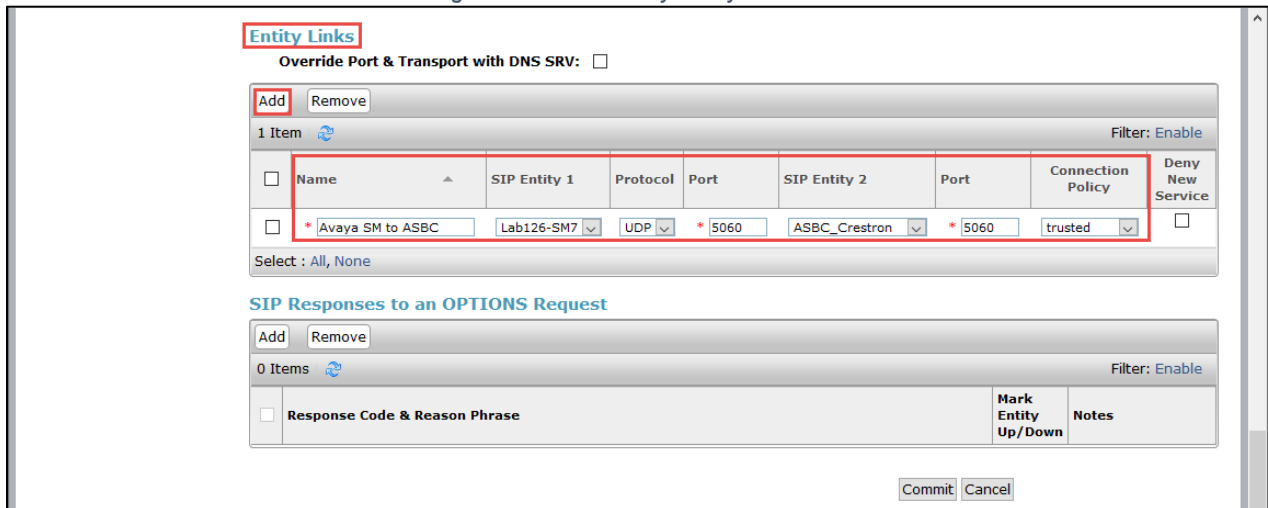


Figure 152 - SIP Entity: Avaya SBCE

#### 4.7.6 Routing Policies

Navigate to: **Routing** → **Routing Policies**. Click **New**

#### 4.7.6.1 Routing Policy to Avaya Aura Communication Manager

1. Set **Name**: SM\_to\_CM is given here
2. Click **Select** under SIP Entity as Destination and the SIP Entities window shows
3. Select **CM\_SIP\_Trunk** as destination SIP Entity (This is the SIP Entity configured for Avaya CM)
4. Click **Select** and return back to Routing Policy Details page
5. Leave all other fields at default values
6. Click **Commit**

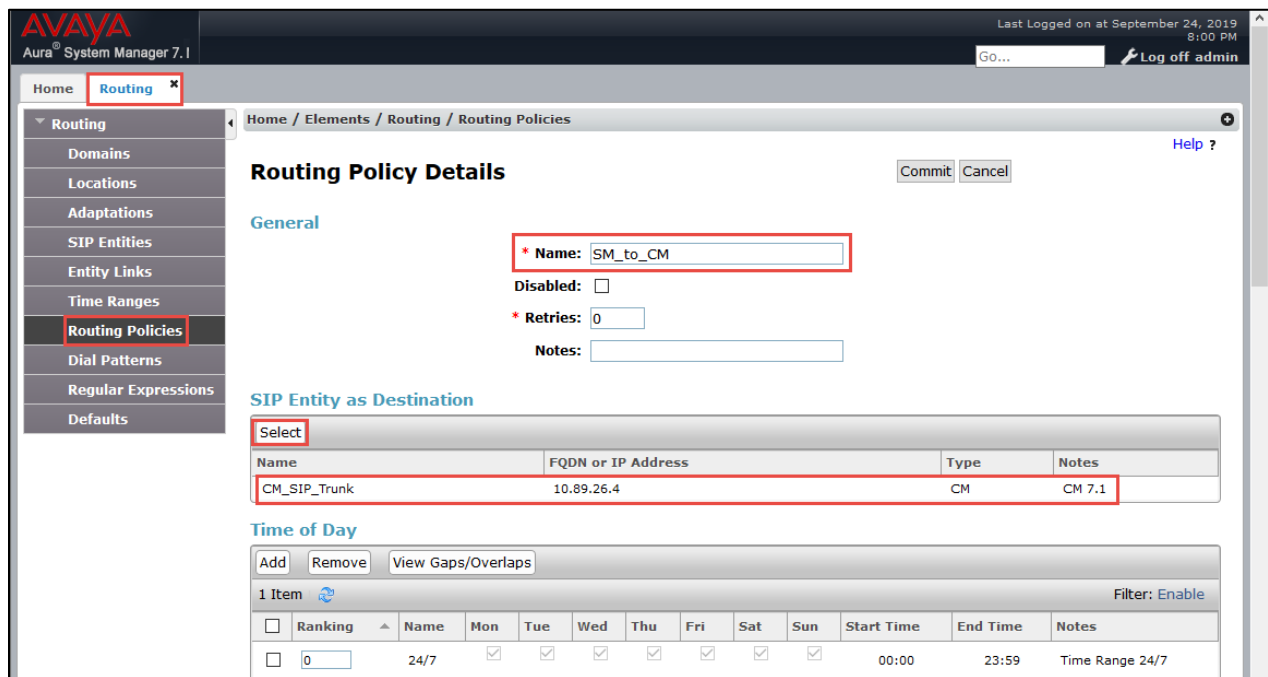


Figure 153 - Routing Policy to Avaya Aura Communication Manager

#### 4.7.6.2 Routing Policy to Avaya SBCE

1. Set **Name**: To\_ASBC is given here as an example
2. Click **Select** under SIP Entity as Destination and SIP Entities window shows
3. Select **ASBC\_Crestron** as destination SIP Entity (This is the SIP Entity configured for Avaya SBCE)
4. Click **Select** and return back to Routing Policy Details page
5. Leave all other fields at default values
6. Click **Commit**

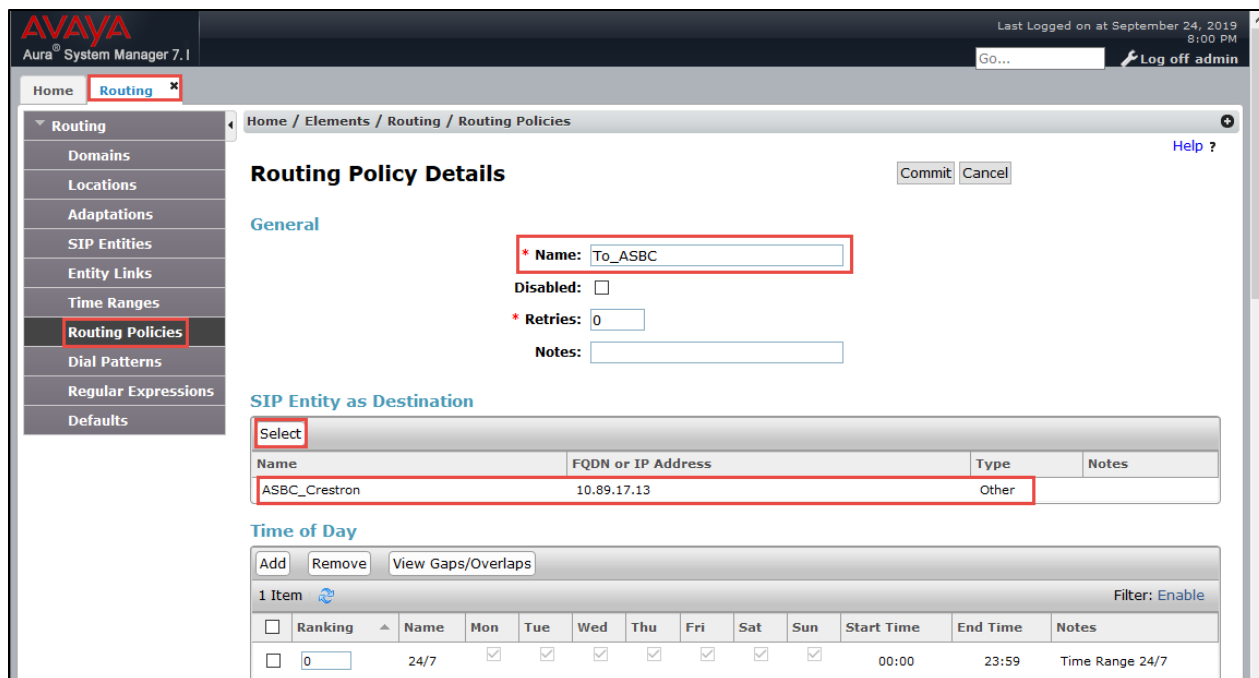


Figure 154 - Routing Policy to Avaya SBCE

## 4.7.7 Dial Patterns

Navigate to: **Routing** → **Dial Patterns**. Click **New**

### 4.7.7.1 Dial Pattern to Avaya Aura Communication Manager

1. Set **Pattern**: 75 - the leading Digits of the DID to be sent to Avaya CM for termination to extensions
2. Set **Min**: 4
3. Set **Max**: 4
4. Under **Originating Locations** and **Routing Policies**, Click **Add**, at the new window
5. **Originating Location**: Select your location, Lab126-Plano is used in this test
6. Check **SM\_to\_CM** as Routing Policy
7. Click **Select** to return to Dial Pattern Details page
8. Leave all other fields at default values.
9. Click **Commit**

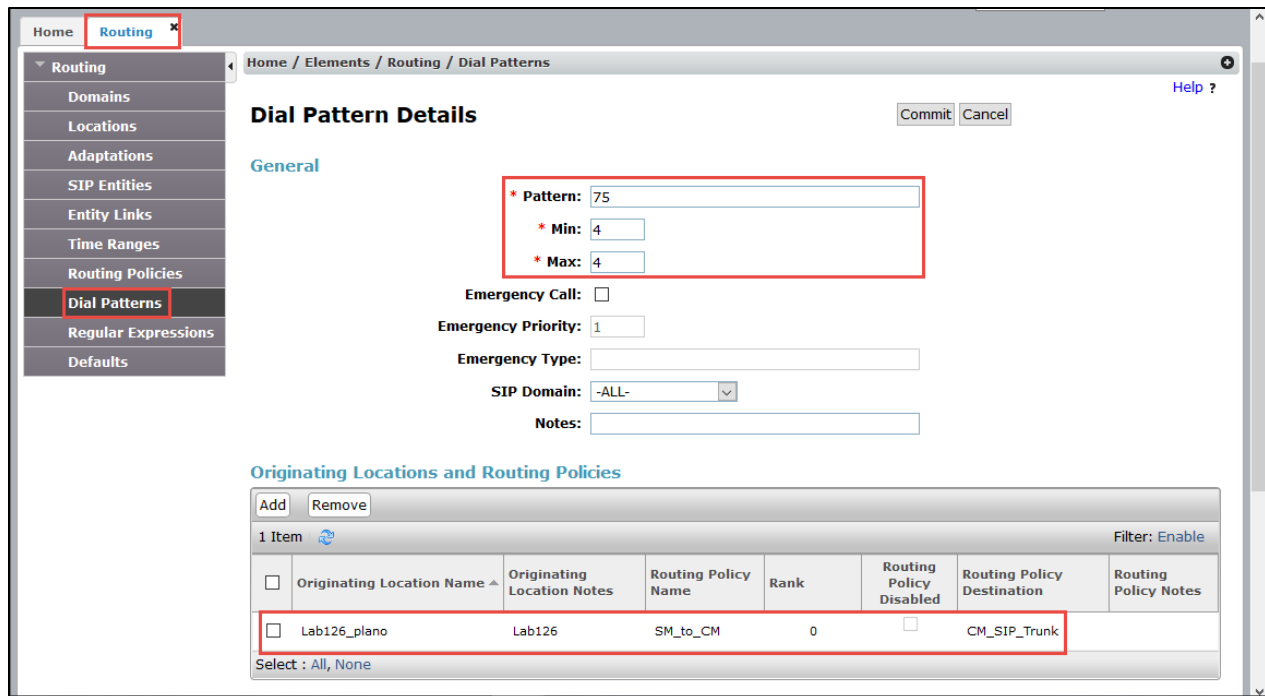


Figure 155 - Dial Pattern to Avaya Aura Communication Manager

#### 4.7.7.2 Dial Patterns to AudioCodes via Avaya SBCE

1. Set **Pattern:** 800 - the leading Digits of the Teams extensions to be dialed over the trunk
2. Set **Min:** 5
3. Set **Max:** 5
4. Under **Originating Locations** and **Routing Policies**, Click **Add**, at the new window
5. **Originating Location:** Select your location, Lab126-Plano is used in this test
6. Check **To\_ASBC** as Routing Policy
7. Click **Select** to return to Dial Pattern Details page
8. Leave all other fields at default values.
9. Click **Commit**

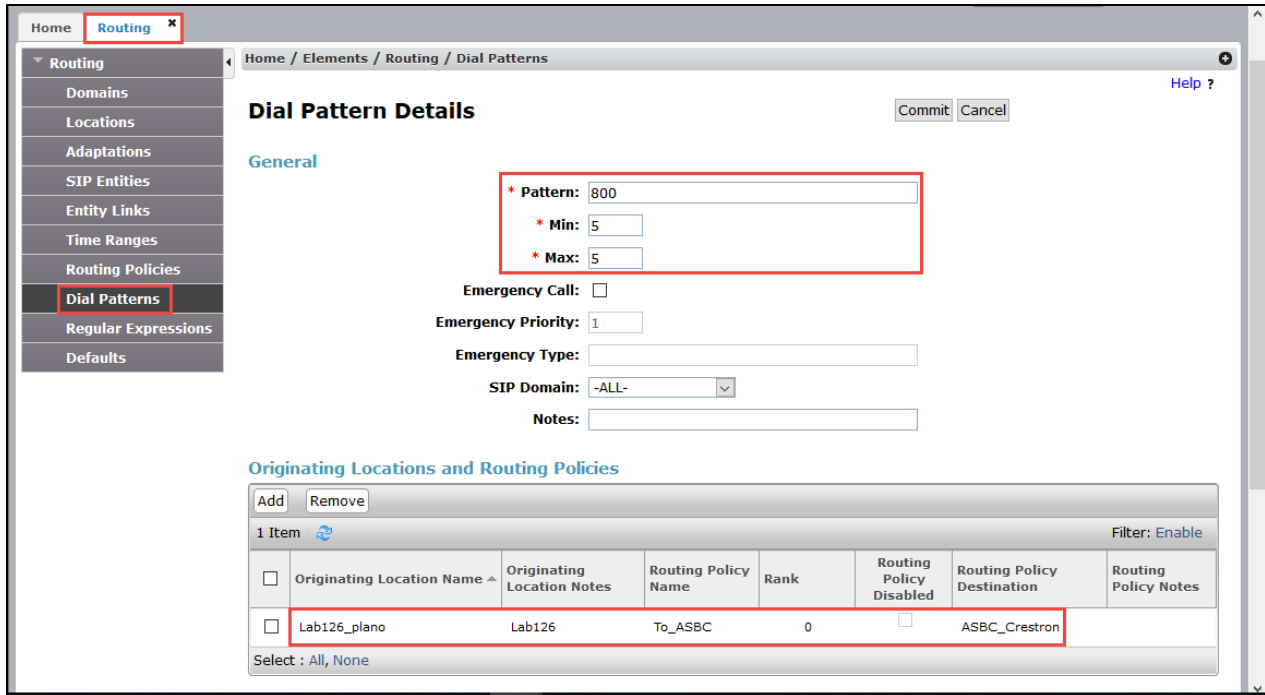


Figure 156 - Dial Pattern to Avaya SBCE

## 4.8 Avaya SBCE Configuration

### 4.8.1 Version

The following version of Avaya SBCE is used for this testing

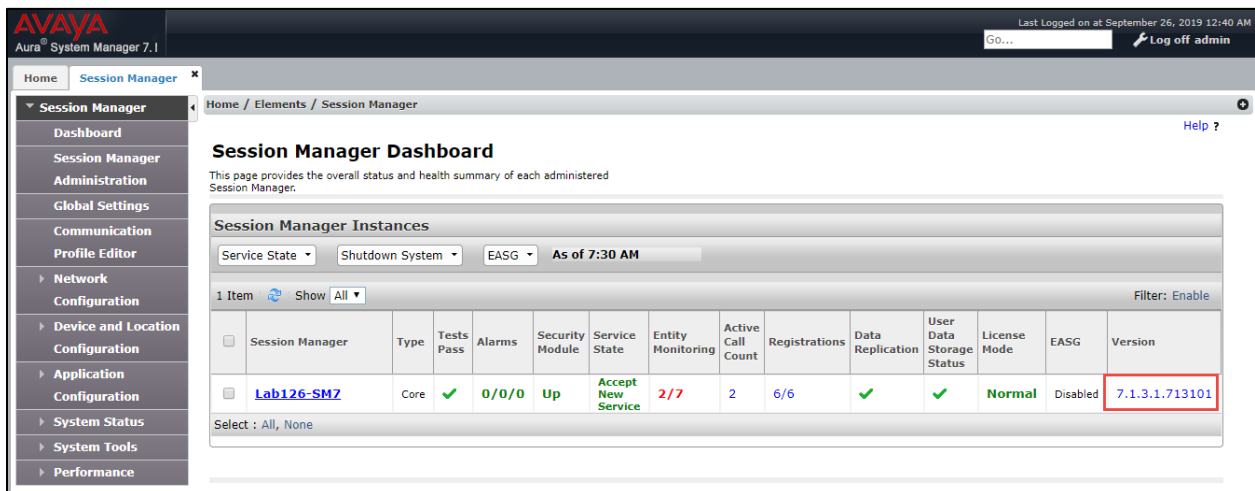


Figure 157 – Version

## 4.8.2 Configure Profiles and Services

### 4.8.2.1 Sever Interworking

1. Navigate to: **Configure Profiles → Server Interworking**
2. Select the predefined Interworking **Profile avaya-ru**, click **Clone**
3. Set Clone Name: **AASM7.1**, for example
4. Click **Finish**
5. Click newly cloned Profile **AASM7.1**, under tab General, click **Edit**
6. Keep all other parameters at default values and save

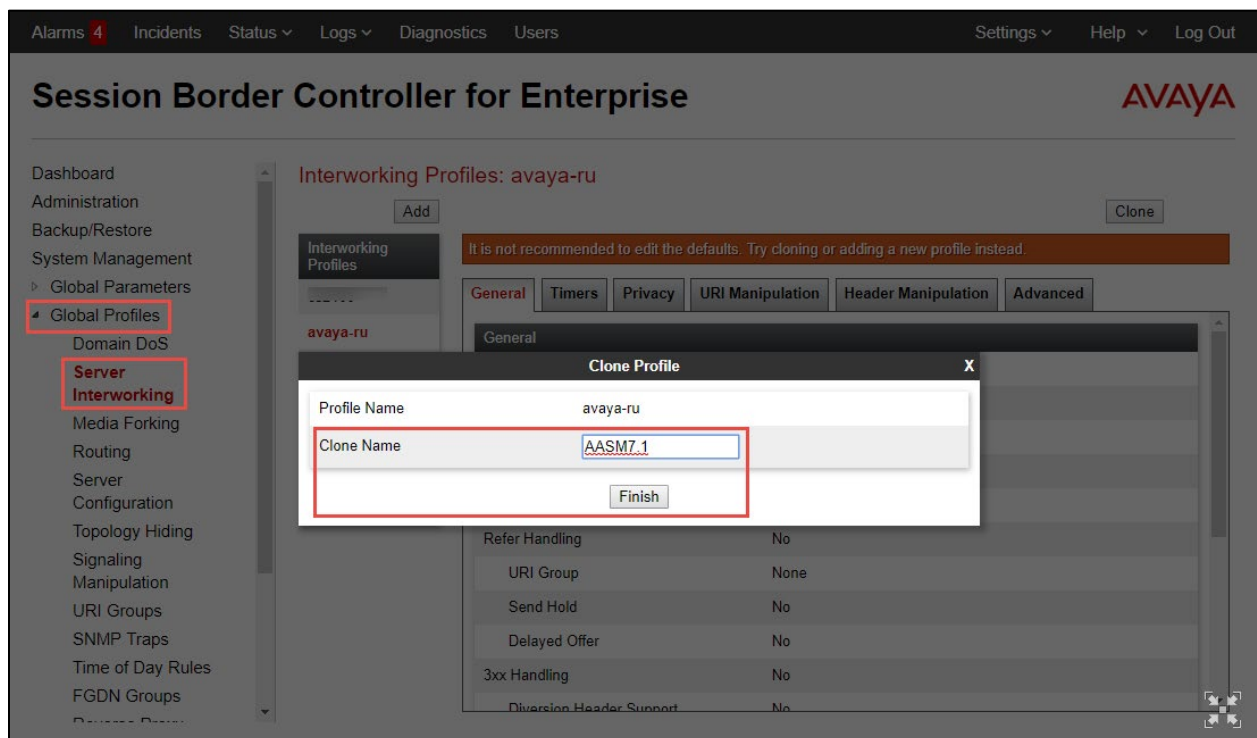


Figure 158 - Server Interworking for Avaya

### 4.8.2.2 SIP Servers – Avaya Aura Session Manager

1. Navigate to **Services → SIP Servers**
2. Click **Add**
3. Set Profile Name: **Avaya SM 7.1**
4. Click **Next**
5. Set **Server Type**: Select Trunk Server from the drop down
6. Set **IP Address/FQDN**: Enter the Avaya Aura Session Manager SIP IP Address
7. Set **Port**: 5060 is used in this setup
8. Set **Transport**: UDP is selected



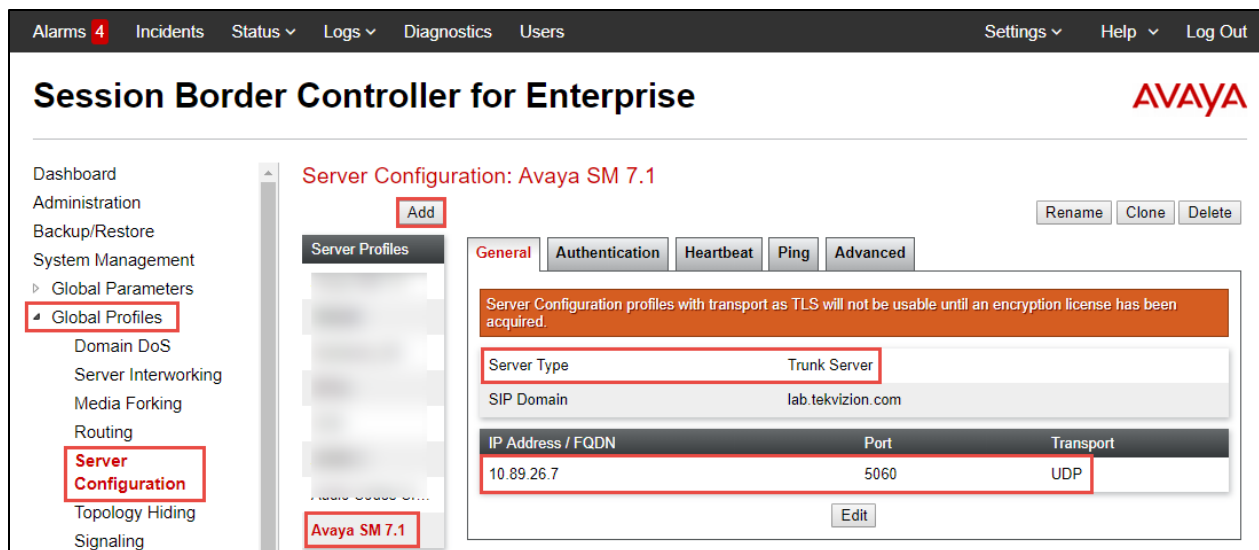


Figure 159- Add SIP Server – Avaya SM

9. Select **Authentication**

10. Keep the parameters at default values

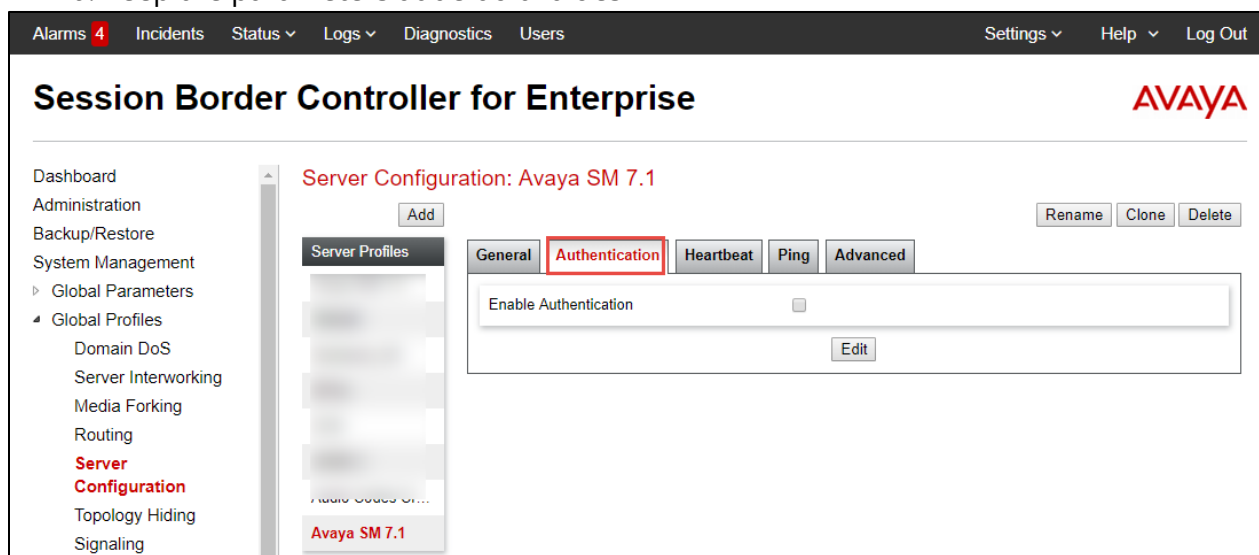


Figure 160 - Add SIP Server – Avaya SM

11. Select **Heartbeat**

12. Check **Enable Heartbeat**

13. Select **Method** as OPTIONS

14. Set **Frequency** as 30 seconds; **From URI** as ping@10.89.17.3, **To URI** as ping@10.89.26.7

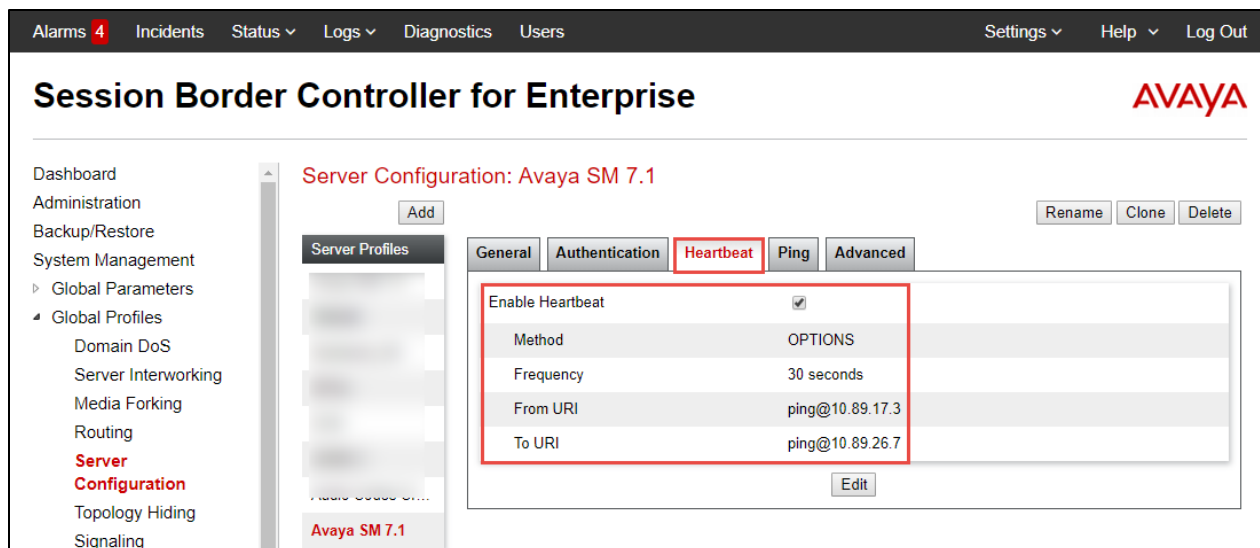


Figure 161 - Add SIP Server – Avaya SM

15. Select **Ping**

16. Keep the parameters at default values

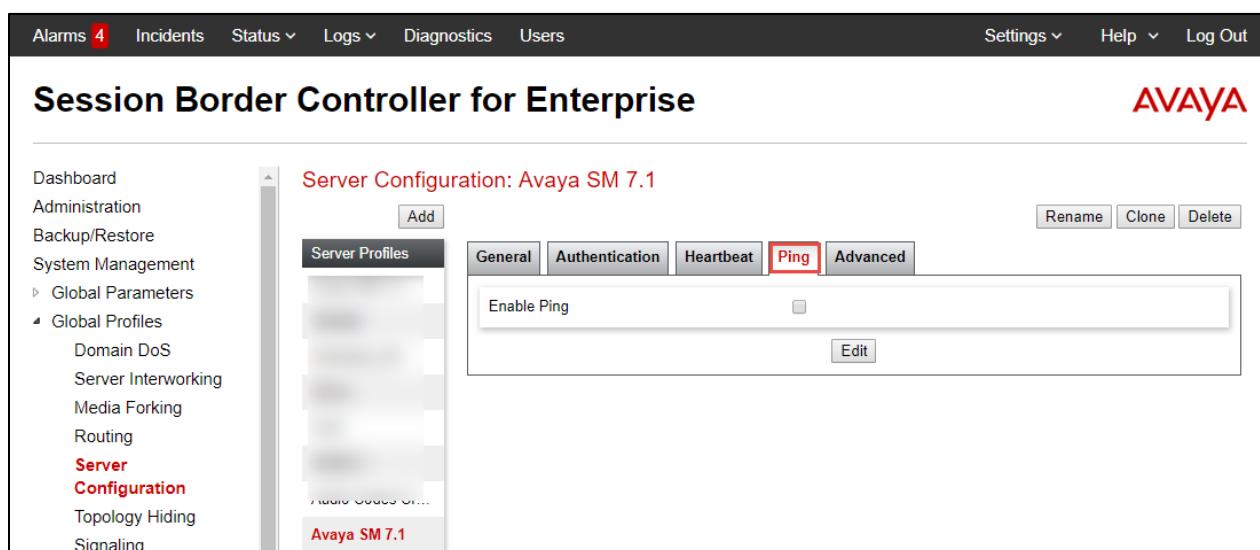


Figure 162 - Add SIP Server – Avaya SM

17. Select **Advanced**

18. Keep the parameters at default values

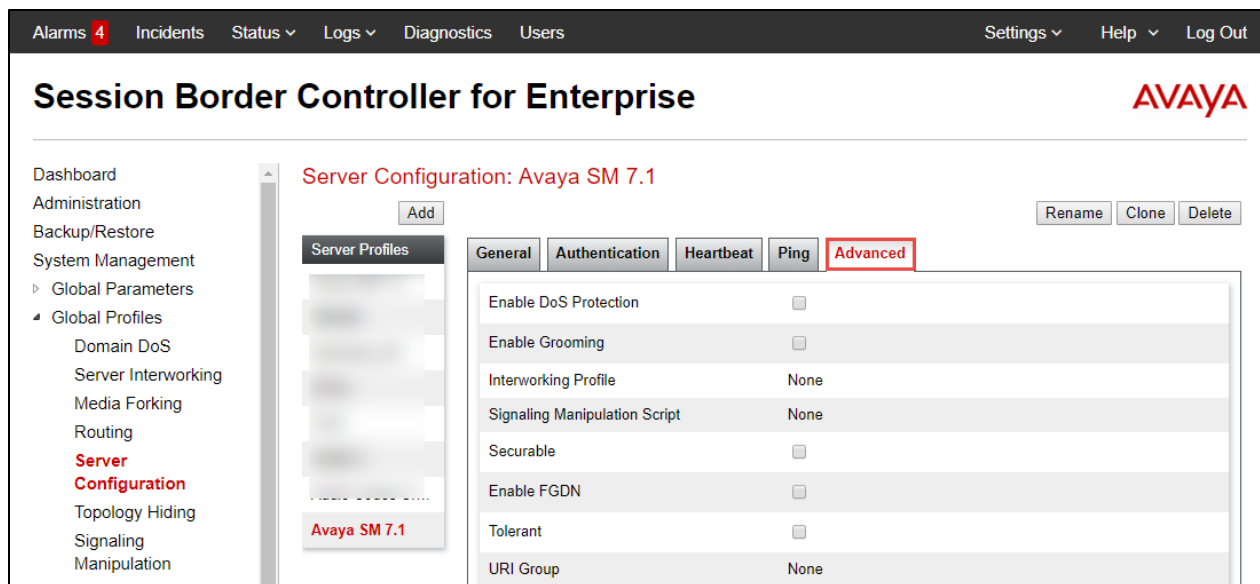


Figure 163 - Add SIP Server – Avaya SM

#### 4.8.2.3 SIP Servers – AudioCodes Crestron

1. Navigate to **Services** → **SIP Servers**
2. Click **Add**
3. Set Profile Name: **AudioCodes Crestron**
4. Click **Next**
5. Set **Server Type**: Select Trunk Server from the drop down
6. Set **IP Address/FQDN**: Enter the AudioCodes IP
7. Set **Port**: 5064 is used in this setup
8. Set **Transport**: UDP is selected

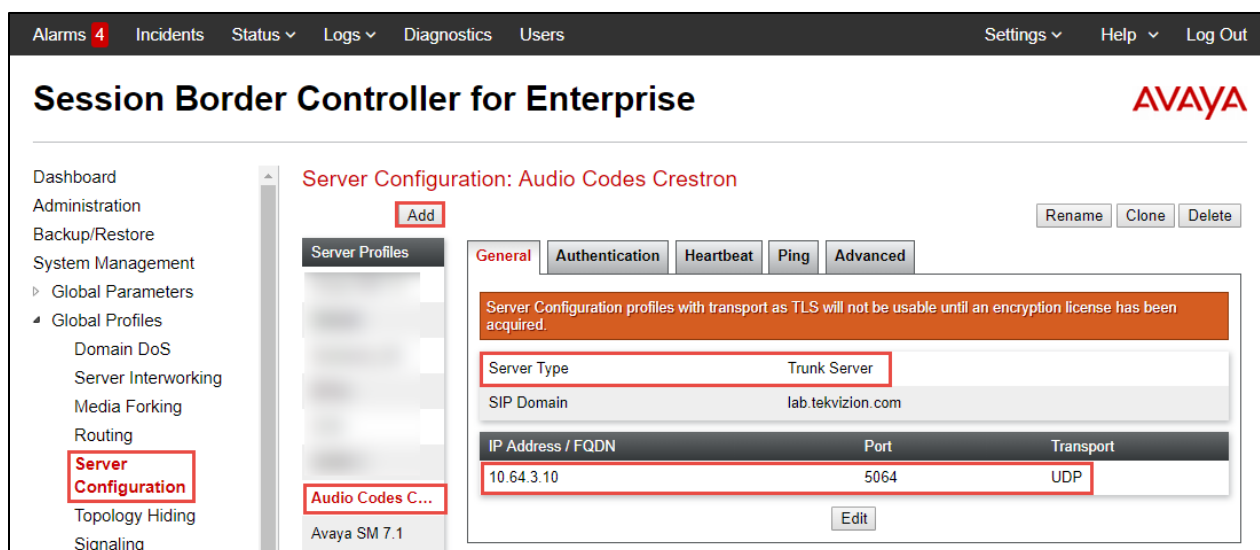


Figure 164 - Add SIP Server – AudioCodes

9. Select **Authentication**
10. Keep the parameters at default values

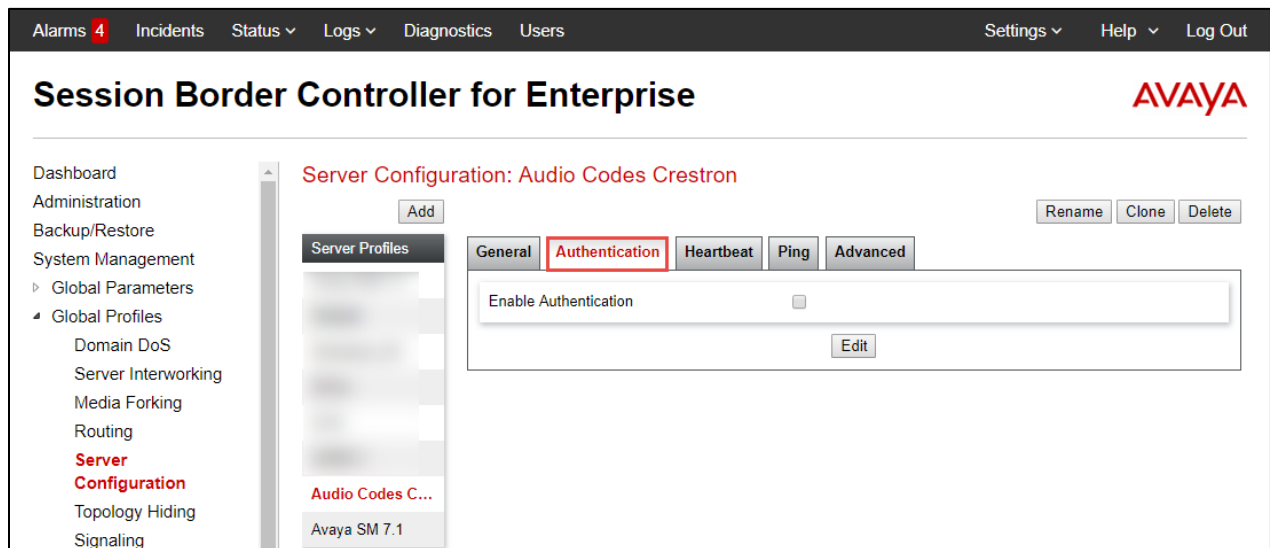


Figure 165 - Add SIP Server – AudioCodes

11. Select **Heartbeat**
12. Check **Enable Heartbeat**
13. Select **Method** as OPTIONS
14. Set **Frequency** as 30 seconds; **From URI** as ping@10.64.5.57, **To URI** as [ping@10.64.3.10](mailto:ping@10.64.3.10)

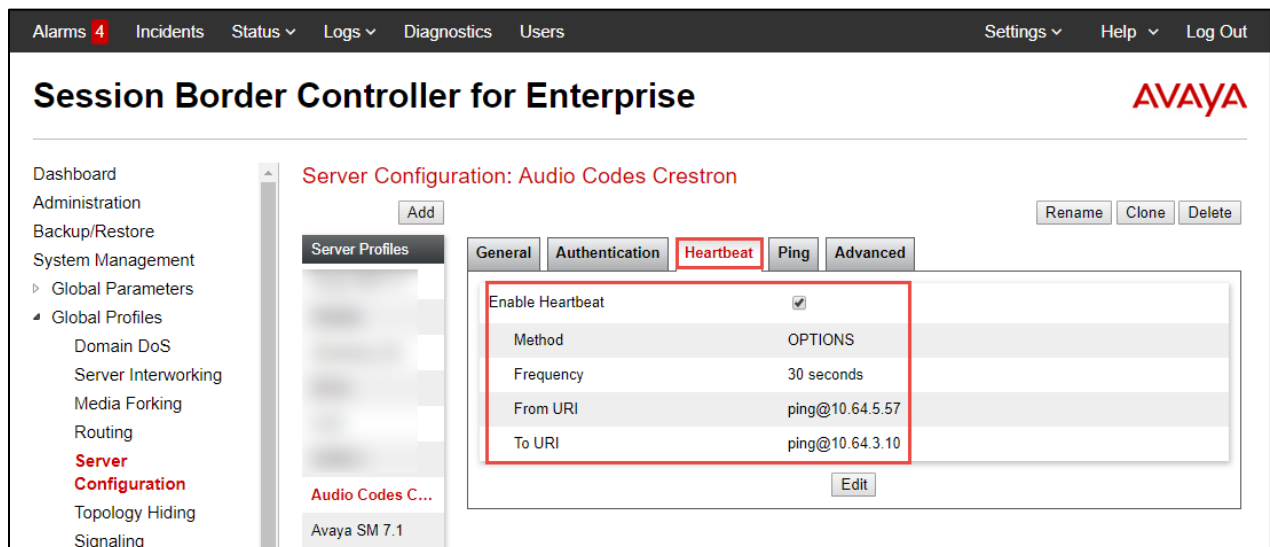


Figure 166 - Add SIP Server – AudioCodes

15. Select **Ping**
16. Keep the parameters at default values

Alarms 4 Incidents Status Logs Diagnostics Users Settings Help Log Out

# Session Border Controller for Enterprise

AVAYA

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    - Server Interworking
    - Media Forking
    - Routing
    - Server Configuration**
    - Topology Hiding
    - Signaling

## Server Configuration: Audio Codes Crestron

Add Rename Clone Delete

Server Profiles

- Audio Codes C...
- Avaya SM 7.1

General Authentication Heartbeat **Ping** Advanced

Enable Ping

Edit

Figure 167 - Add SIP Server – AudioCodes

### 17. Select **Advanced**

18. Keep the parameters at default values

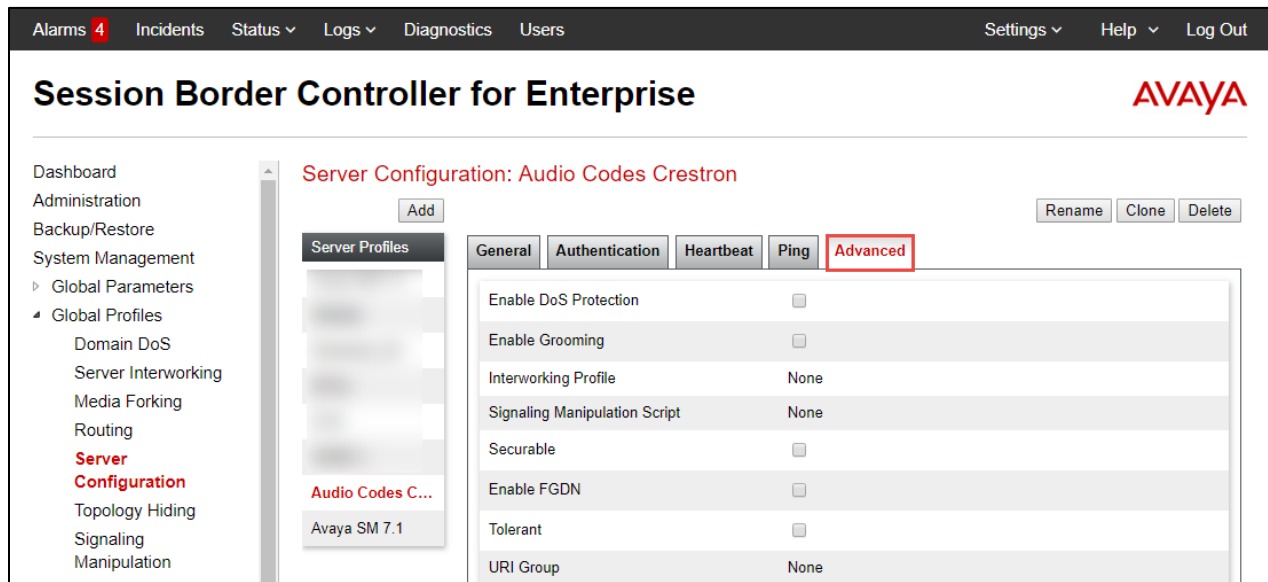


Figure 168 - Add SIP Server – AudioCodes

#### 4.8.2.4 Topology Hiding

Topology Hiding profiles were added for Avaya Session Manager and AudioCodes SBC to overwrite and hiding certain headers

1. Navigate to: **Configure Profiles → Topology Hiding**
2. Two profiles are used for the testing. One is default and another one is created as below.

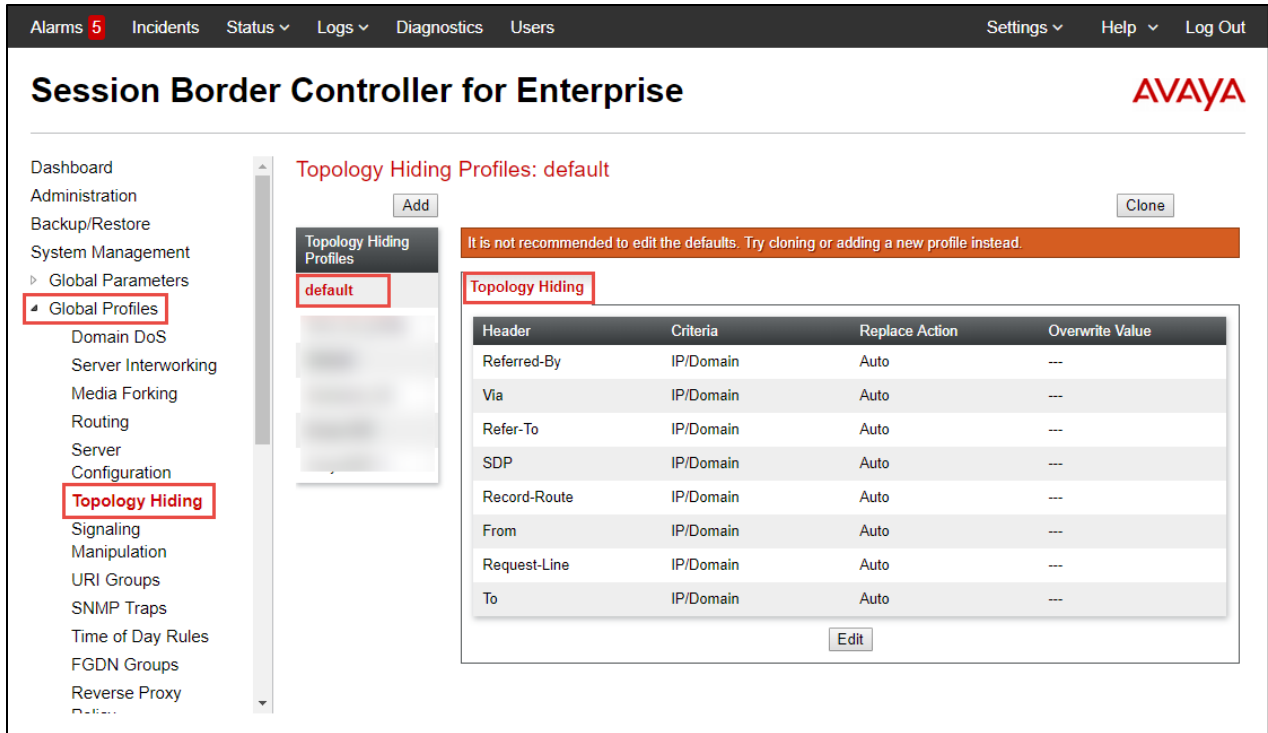


Figure 169 - Topology Hiding

3. Click **Add** and enter profile name
4. Add the following headers and keep Criteria and Replace Action with default values as below
5. Click **Finish**

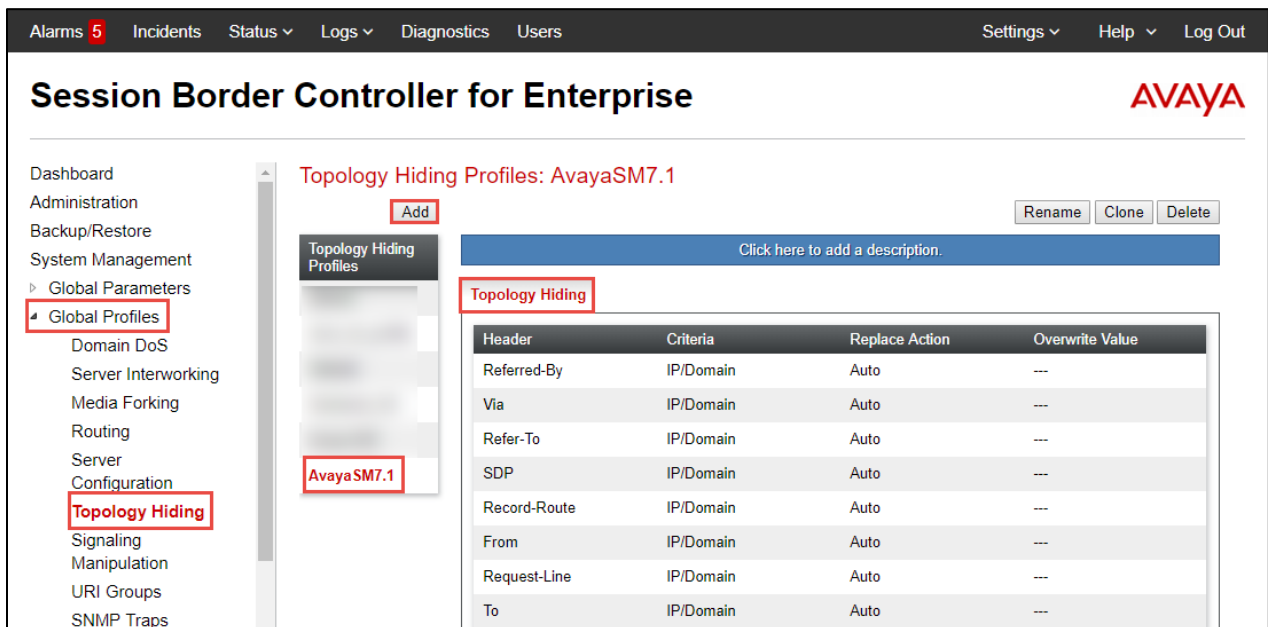


Figure 170 - Topology Hiding

#### 4.8.2.5 Routing

1. Navigate to: **Configuration Profiles → Routing**
2. Click **Add**
3. Set **Profile Name**: AASM7.1 is given here
4. Click **Next**

At Routing Profile Window, click **Add**

5. Set **Server Configuration**: Avaya SM 7.1 (which was configured under SIP Servers)
6. The Server IP, Port and Transport Protocol will populate automatically. Select UDP as Transport.
7. Leave all other fields as default
8. Click **Finish**

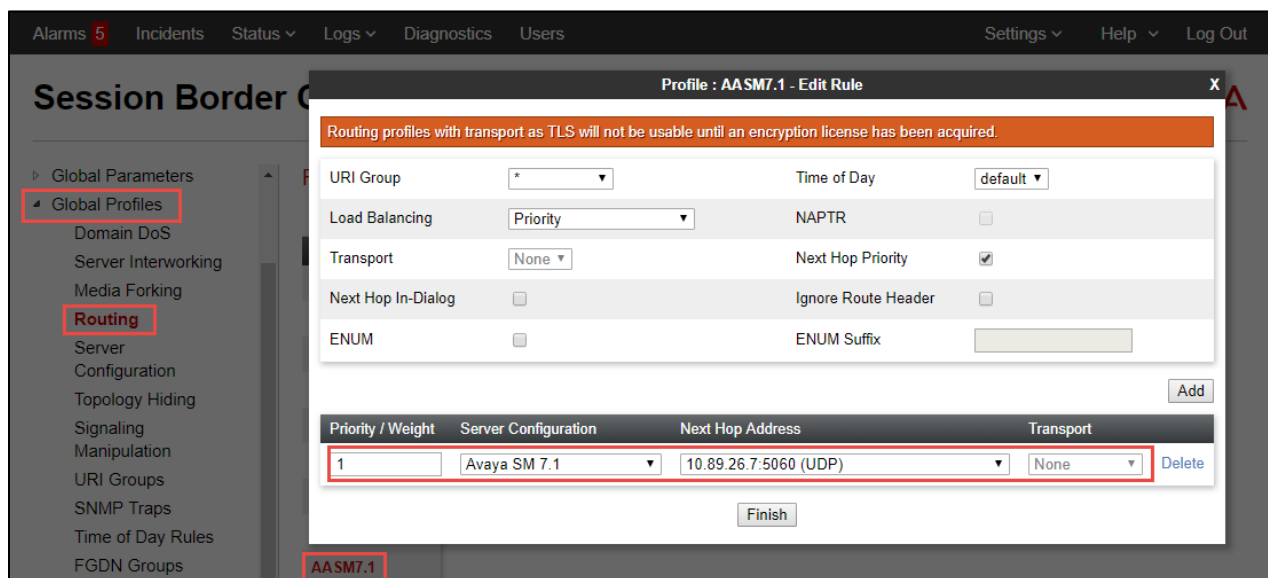


Figure 171 - Routing Profile – Avaya SM

9. Repeat same steps to create the Routing Profile AudioCodes for AudioCodes



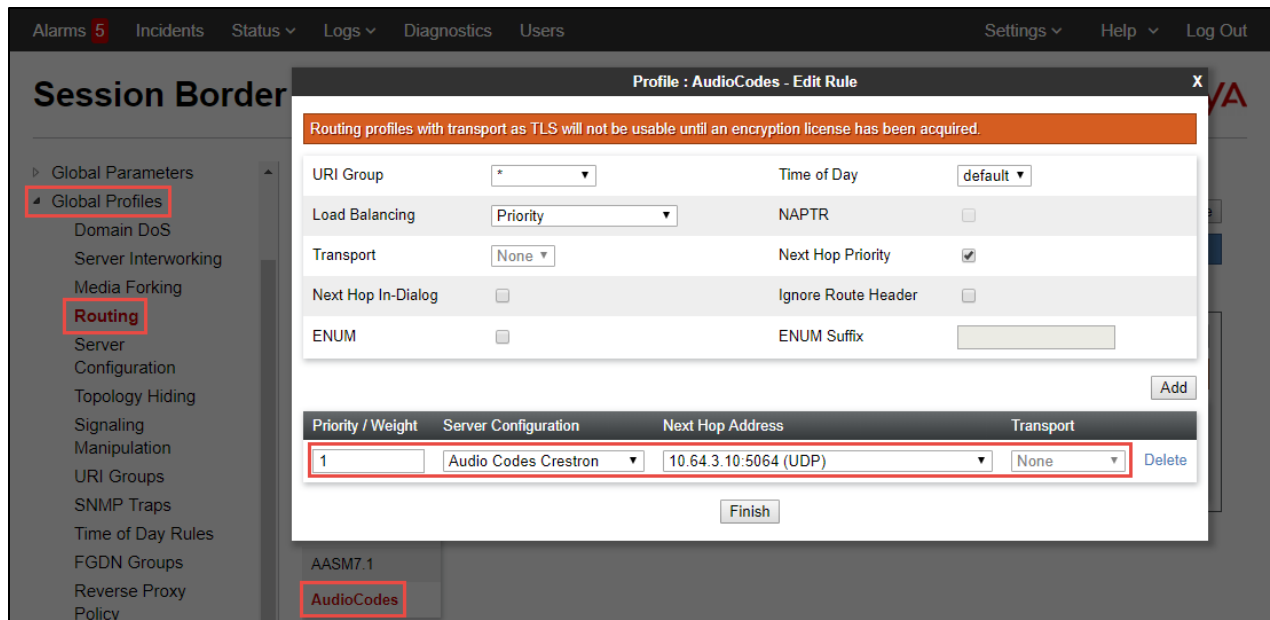


Figure 172 - Routing Profile – AudioCodes

## 4.8.3 Domain Policies

### 4.8.3.1 Signaling Rules

1. Navigate to: **Domain Policies -> Signaling Rules**
2. Select **default** under Signaling Rules, click **Clone**
3. Set **Name**: SM\_Rule is given in this test
4. Click **Finish**
5. Select the newly cloned Signaling Rule **SM\_Rule**, under tab Request Headers, click **Add In Header Control** and configure the setting as below
6. Click **Finish**

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    - Border Rules
    - Media Rules
    - Security Rules
    - Signaling Rules**
      - End Point Policy Groups
      - Session Policies
    - TLS Management
    - Device Specific Settings

### Signaling Rules: SM\_Rule

Add Filter By Device... Rename Clone Delete

Click here to add a description.

General Requests Responses **Request Headers** Response Headers Signaling QoS UCID

Add In Header Control Add Out Header Control

Row	Header Name	Method Name	Header Criteria	Action	Proprietary	Direction		
1	Alert-Info	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
2	Endpoint-View	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
3	P-AV-Message-ID	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-Charging-Vector	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-Location	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

Figure 173 - Signaling Rule – Avaya SM

7. Repeat the same for Response Headers also

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Border Rules  
Media Rules  
Security Rules  
**Signaling Rules**  
End Point Policy Groups  
Session Policies  
TLS Management  
Device Specific Settings

Signaling Rules: SM\_Rule

Add Filter By Device... Rename Clone Delete

Signaling Rules  
default  
No-Content-Typ...  
**SM\_Rule**  
Remove PAI

Click here to add a description.

General Requests Responses Request Headers **Response Headers** Signaling QoS UCID

Add In Header Control Add Out Header Control

Row	Header Name	Response Code	Method Name	Header Criteria	Action	Proprietary	Direction	Edit	Delete
1	Alert-Info	200	ALL	Forbidden	Remove Header	No	IN	Edit	Delete
2	Endpoint-View	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
3	Endpoint-View	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
4	P-AV-Message-ID	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
5	P-AV-Message-ID	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
6	P-Charging-Vector	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
7	P-Conference	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
8	P-Location	1XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
9	P-Location	200	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
10	P-Location	4XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete
11	P-Location	5XX	ALL	Forbidden	Remove Header	Yes	IN	Edit	Delete

Media Rules  
Security Rules  
**Signaling Rules**  
End Point Policy Groups  
Session Policies  
TLS Management  
Device Specific Settings

Figure 174- Signaling Rule – Avaya SM

#### 4.8.3.2 End Point Policy Groups

A new End Point Policy Group was created for Avaya Aura Session Manager. The default policy group was used for the AudioCodes side.

1. Navigate to: **Domain Policies -> End Point Policy Groups**
2. Two End Point Policy Groups are used for this testing. One is default-low and another one is created as below.

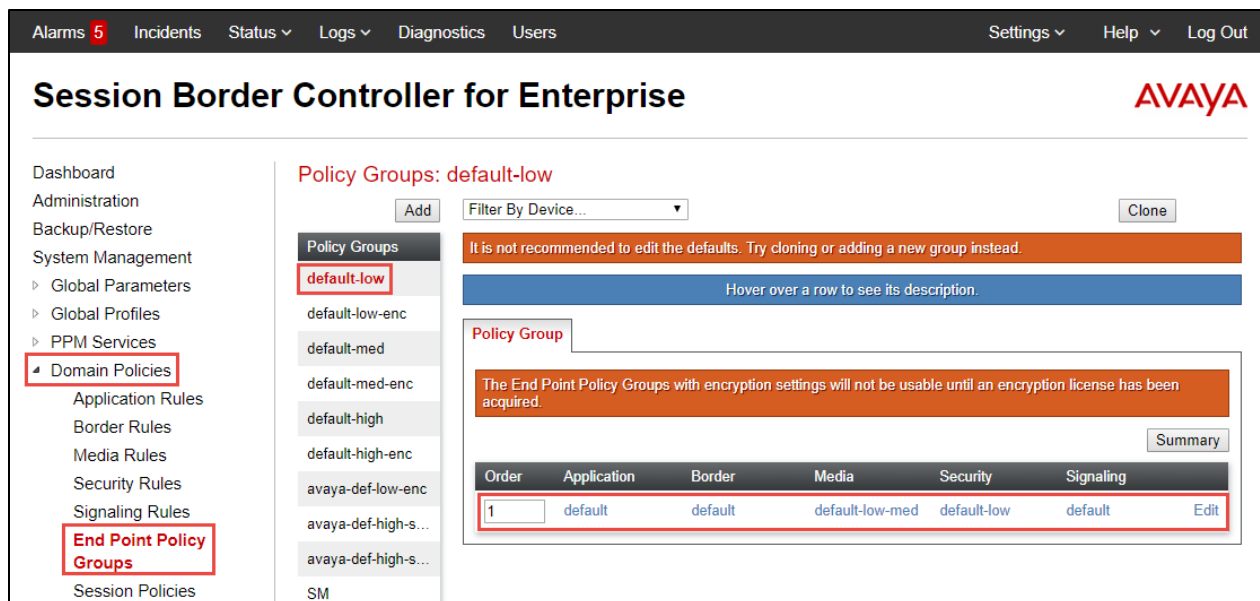


Figure 175- End Point Policy Group – Avaya SM

3. Select **default-low** under Policy Groups
4. Click **Clone**
5. Set Clone Name: **PG\_ASM7.1** is given
6. Click **Finish**

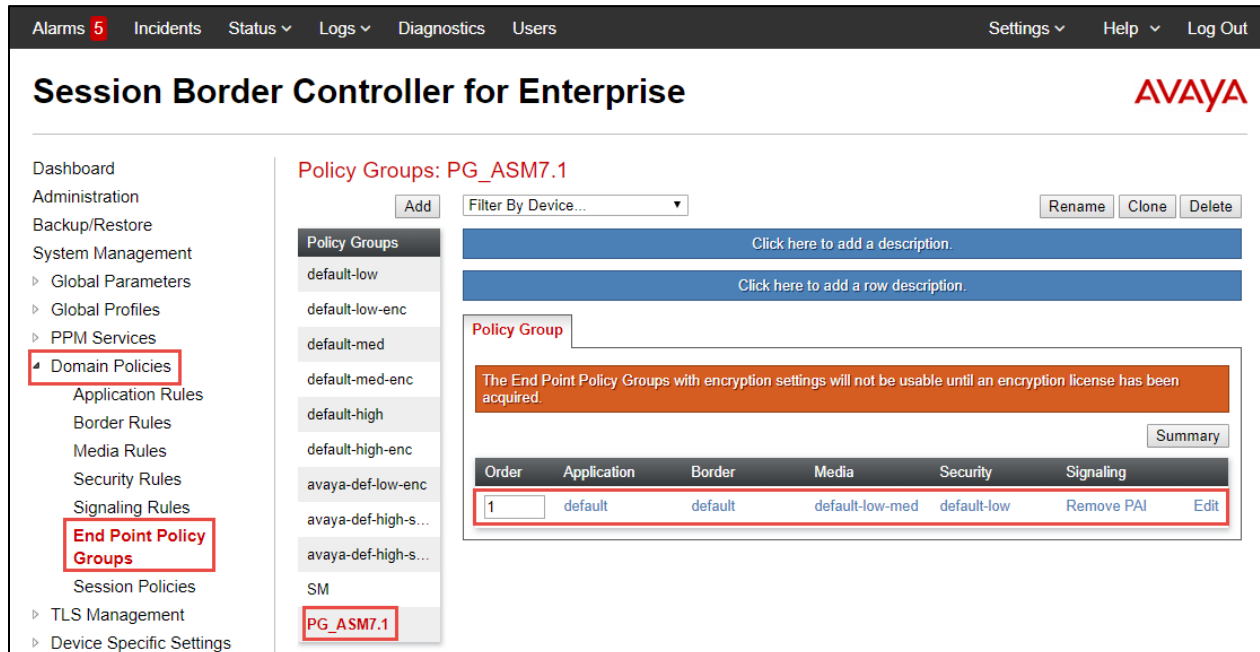


Figure 176 - End Point Policy Group – Avaya SM

## 4.8.4 Network & Flows

### 4.8.4.1 Media Interface

1. Navigate to: **Device Specific Settings** → **Media Interface**. Click **Add**
2. Set **Name**: SBC LAN is given here
3. Set **IP Address**: Select SBC LAN from the drop down and the IP address will populate automatically. The IP address for Interface facing Avaya Aura Session Manager is 10.89.17.13
4. Set **Port Range**: 35000-40000 is used for this setup
5. Click **Finish**
6. Repeat the same steps to create a Media Interface facing AudioCodes with the name SBC WAN

Name	Media IP Network	Port Range	TLS Profile	
SBC LAN	10.89.17.13 LAN-A1 (A1, VLAN 0)	35000 - 40000	None	Edit Delete
SBC WAN	10.64.5.57 WAN-B1 (B1, VLAN 0)	35000 - 40000	None	Edit Delete

Figure 177- Media Interface

### 4.8.4.2 Signaling Interface

1. Navigate to: **Network & Flows** → **Signaling Interface**. Click **Add**, new Add Signaling Interface window will appear
2. Set **Name**: SBC LAN is given for the interface facing Avaya Aura Session Manager
3. Set **IP Address**: Select the signaling IP which is the Avaya Aura Session Manager facing interface
4. Set **UDP Port**: 5060 is set
5. Set **UDP/TLS Port**: Leave the boxes empty as only UDP is used between Avaya Aura Session Manager and Avaya SBCE
6. Leave all other fields at default values
7. Click **Finish**

8. Repeat same steps to create the Signaling Interface facing AudioCodes. UDP is the protocol between Avaya SBCE and AudioCodes.

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‣ TLS Management  
‣ **Device Specific Settings**  
  Network Management  
  Media Interface  
  **Signaling Interface**  
  End Point Flows

Devices  
**Lab117-ASBCE**

**Signaling Interface**

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Add

Name	Signaling IP Network	TCP Port	UDP Port	TLS Port	TLS Profile	
SBC WAN	10.64.5.57 WAN-S1 (S1, VLAN 0)	---	5060	---	None	Edit Delete
SBC LAN	10.89.17.13 LAN-A1 (A1, VLAN 0)	---	5060	---	None	Edit Delete

Figure 178 - Signaling Interface

#### 4.8.4.3 Server Flows

1. Navigate to: **Network & Flows** → **End Point Flows** → **Server Flows**. Click **Add**
2. Set **Flow Name**: Avaya SM is given for enterprise
3. Set **SIP Server Profile**: Avaya SM 7.1 (created earlier)
4. Set **Transport**: UDP is selected here
5. Set **Receive Interface**: SIG\_WAN (created earlier)
6. Set **Signaling Interface**: SIG\_LAN (created earlier)
7. Set **Media Interface**: SIG\_LAN (created earlier)
8. Set **End Point Policy Group**: default-low (created earlier)
9. Set **Routing Profile**: AudioCodes (created earlier)
10. Set **Topology Hiding Profile**: Avaya\_SM (created earlier)
11. Leave all other fields at default values
12. Click **Finish**

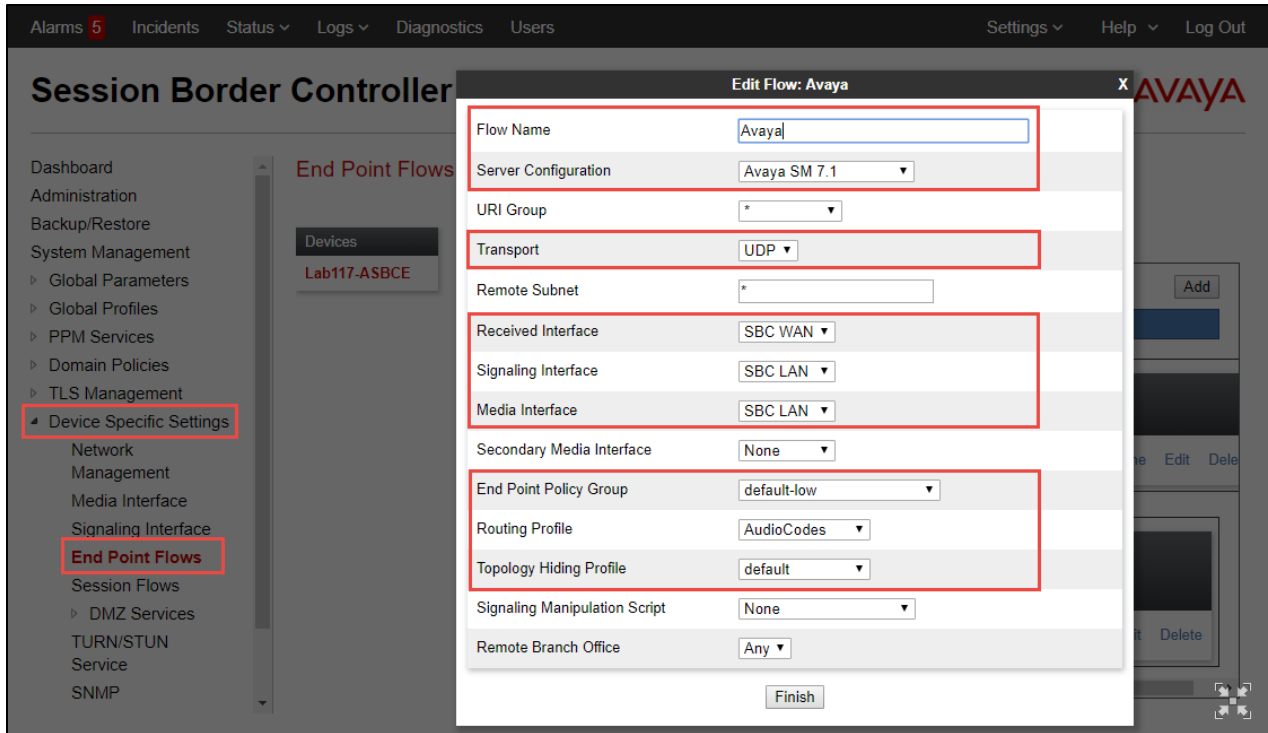


Figure 179 - Server Flow

13. Repeat the same steps for creating server flow for AudioCodes as below

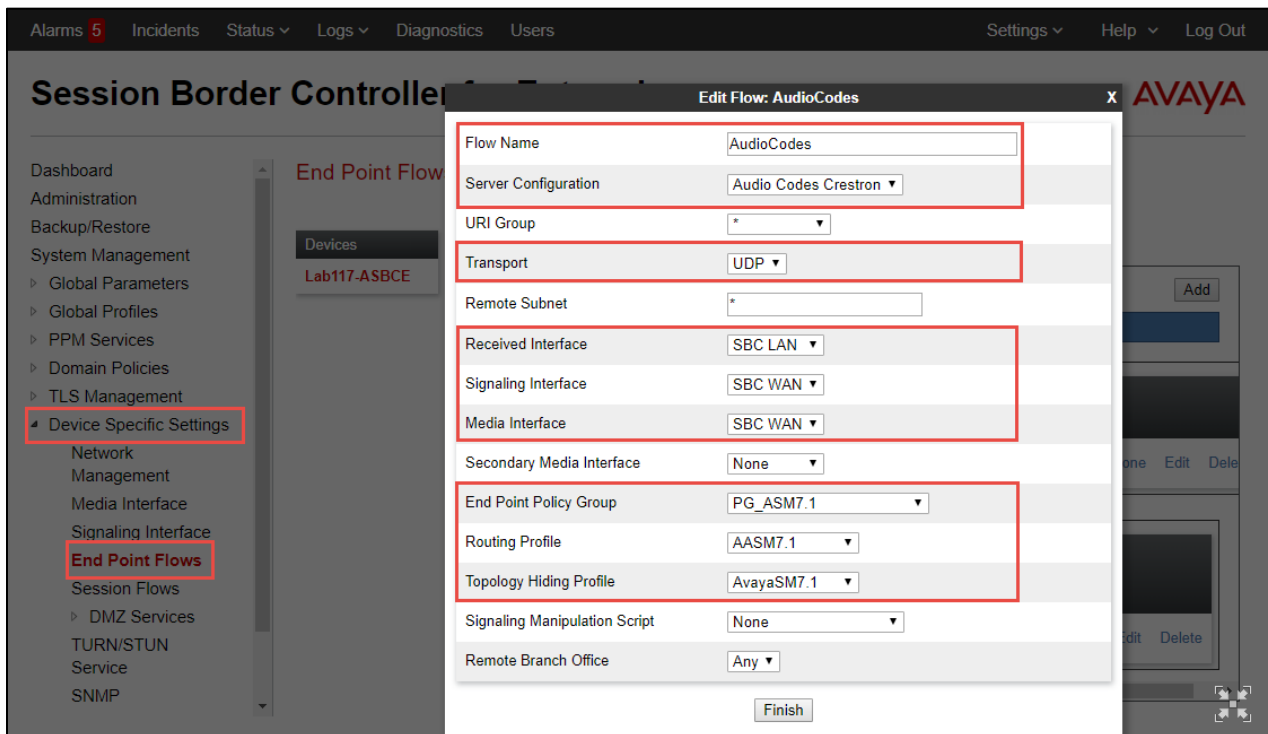


Figure 180 - Server Flow



## 5 Acronyms

Acronym	Definition
Avaya CM	Avaya Aura Communications Manager
Avaya SM	Avaya Aura Sessions Manager
Avaya SBCE	Avaya Session Border Controller for Enterprise
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
DNS	Domain Name Server
EXT	Extension
FQDN	Fully Qualified Domain Name
MRGL	Media Resource Group List
MTP	Media Termination Point
MWI	Message Waiting Indicator
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SRTP	Secure Real Time Protocol
SIP	Session Initiated Protocol
UDP	Uniform Dial Plan
VM	Voice Mail
B2BUA	Back to Back User Agent
SBC	Session Border Controller

## 6 Summary of Tests and Results

External ID	Title	Procedure	Expected Results	Status	Comments
1	Teams user Calls PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. PBX A user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	PASSED	
2	Teams user Calls PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. PBX B user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	PASSED	
3	Teams user Calls PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PSTN user hangs up the call 9. Verify call is cleared successfully	2. Call is disconnected		
4	Teams user Calls PBX A user and hangs up before answer	1. Make a voice call from Teams user to PBX A user 2. PBX A user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX A user is ringing 5. PBX A user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
5	Teams user Calls PBX B user and hangs up before answer	1. Make a voice call from Teams user to PBX B user 2. PBX B user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX B user is ringing 5. PBX B user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
6	Teams user Calls PSTN user and hangs up before answer	1. Make a voice call from Teams user to PSTN user 2. PSTN user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PSTN user is ringing	1. Call is disconnected before answer	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		5. PSTN user stops ringing 6. Verify call is cleared successfully			
7	PBX A user Calls Teams user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. PBX A user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. Teams user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	PASSED	
8	PBX B user Calls Teams user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. PBX B user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. Teams user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
9	PSTN user Calls Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. PSTN user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. Teams user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	PASSED	
10	PBX A user Calls Teams user and hangs up before answer	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PBX A user hears Ring back Tone</li> <li>4. PBX A user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	PASSED	
11	PBX B user Calls Teams user and hangs up before answer	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PBX B user hears Ring back Tone</li> <li>4. PBX B user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
12	PSTN user Calls Teams user and hangs up before answer	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PSTN user hears Ring back Tone</li> <li>4. PSTN user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	PASSED	
13	Teams user Calls PBX A user and performs hold/resume	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. Teams user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ol>	PASSED	
14	Teams user Calls PBX B user and performs hold/resume	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<p>on hold</p> <p>7. Teams user resumes the call</p> <p>8. Verify two way audio is re-established between the two end points</p> <p>9. Teams user hangs up the call</p> <p>10. Verify call is cleared successfully</p>	<p>present after call is resumed</p>		
15	Teams user Calls PSTN user and performs hold/resume	<p>1. Make a voice call from Teams user to PSTN user</p> <p>2. Teams user hears Ring back Tone</p> <p>3. PSTN user answers the call</p> <p>4. Verify two way audio</p> <p>5. Teams user initiates call hold</p> <p>6. Verify no audio is present while call is on hold</p> <p>7. Teams user resumes the call</p> <p>8. Verify two way audio is re-established between the two end points</p> <p>9. Teams user hangs up the call</p> <p>10. Verify call is cleared successfully</p>	<p>1. Call is placed on hold successfully</p> <p>2. No audio present during hold</p> <p>3. Call is resumed successfully</p> <p>4. Two way audio present after call is resumed</p>	PASSED	
16	PBX A user Calls Teams user and Teams user performs hold/resume	<p>1. Make a voice call from PBX A user to Teams user</p> <p>2. PBX A user hears Ring back Tone</p> <p>3. Teams user answers the call</p> <p>4. Verify two way audio</p> <p>5. Teams user initiates call hold</p> <p>6. Verify no audio is present while call is on hold</p> <p>7. Teams user resumes the call</p>	<p>1. Call is placed on hold successfully</p> <p>2. No audio present during hold</p> <p>3. Call is resumed successfully</p> <p>4. Two way audio present after call is resumed</p>	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using

External ID	Title	Procedure	Expected Results	Status	Comments
		<ul style="list-style-type: none"> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. PBX A user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ul>			receiver or speaker button.
17	PBX B user Calls Teams user and Teams user performs hold/resume	<ul style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. PBX B user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ul>	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.
18	PSTN user Calls Teams user and Teams performs hold/resume	<ul style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ul>	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.



External ID	Title	Procedure	Expected Results	Status	Comments
		9. PSTN user hangs up the call 10. Verify call is cleared successfully			
19	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from Teams user to PBX A user 1 2. Teams user hears Ring back Tone 3. PBX A user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 2 6. Verify PBX A user 1 is placed on hold 7. PBX A user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user 1 and PBX A user 2 11. PBX A user 1 hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
20	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PBX A user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		10. Verify two way audio between PBX A user and PBX B user 11. PBX A user hangs up the call 12. Verify call is cleared successfully			
21	Teams user Calls PBX A user, Teams user performs Attended Transfer to PSTN user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 6. Verify PBX A user is placed on hold 7. PSTN user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user and PSTN user 11. PBX A user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
22	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PBX B user 1 2. Teams user hears Ring back Tone 3. PBX B user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 2 6. Verify PBX B user 1 is placed on hold 7. PBX B user 2 answers the call	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<ul style="list-style-type: none"> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user 1 and PBX B user 2</li> <li>11. PBX B user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>			
23	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX A user	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PBX A user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PBX A user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	
24	Teams user Calls PBX B user, Teams user performs Attended	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	Transfer to PSTN user	<ul style="list-style-type: none"> <li>6. Verify PBX B user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PSTN user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>			
25	Teams user Calls PSTN user, Teams user performs Attended Transfer to PBX B user	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user</li> <li>6. Verify PSTN user is placed on hold</li> <li>7. PBX B user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PSTN user and PBX B user</li> <li>11. PSTN user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	
26	Teams user Calls PSTN user, Teams user performs	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	Attended Transfer to PBX A user	5. Teams user places a consultation call to PBX A user 6. Verify PSTN user is placed on hold 7. PBX A user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user and PBX A user 11. PSTN user hangs up the call 12. Verify call is cleared successfully			
27	Teams user Calls PSTN 1 user, Teams user performs Attended Transfer to PSTN 2 user	1. Make a voice call from Teams user to PSTN user 1 2. Teams user hears Ring back Tone 3. PSTN user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 2 6. Verify PSTN user 1 is placed on hold 7. PSTN user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user 1 and PSTN user 2 11. PSTN user 1 hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
28	PBX A user Calls Teams user, Teams	1. Make a voice call from PBX A user 1 to Teams user 2. PBX A user 1 hears Ring back Tone	1. Call is transferred successfully 2. Two way audio	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	user performs Attended Transfer to PBX A user	<ol style="list-style-type: none"> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user 2</li> <li>6. Verify PBX A user 1 is placed on hold</li> <li>7. PBX A user 2 answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user 1 and PBX A user 2</li> <li>11. PBX A user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	present after call is transferred		
29	PBX A user Calls Teams user, Teams user performs Attended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user</li> <li>6. Verify PBX A user is placed on hold</li> <li>7. PBX B user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user and PBX B user</li> <li>11. PBX A user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
30	PBX A user Calls Teams user, Teams user performs Attended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> <li>6. Verify PBX A user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user and PSTN user</li> <li>11. PBX A user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
31	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user 1 to Teams user</li> <li>2. PBX B user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user 2</li> <li>6. Verify PBX B user 1 is placed on hold</li> <li>7. PBX B user 2 answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user 1 and PBX B user 2</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<ol style="list-style-type: none"> <li>11. PBX B user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>			
32	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PBX A user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PBX A user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
33	PBX B user Calls Teams user, Teams user performs Attended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	



External ID	Title	Procedure	Expected Results	Status	Comments
		10. Verify two way audio between PBX B user and PSTN user 11. PBX B user hangs up the call 12. Verify call is cleared successfully			
34	PSTN user Calls Teams user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PSTN user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user and PBX B user 11. PSTN user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
35	PSTN user Calls Teams user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 6. Verify PSTN user is placed on hold 7. PBX A user answers the call	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user and PBX A user 11. PSTN user hangs up the call 12. Verify call is cleared successfully			
36	PSTN 1 user Calls Teams user, Teams user performs Attended Transfer to PSTN 2 user	1. Make a voice call from PSTN user 1 to Teams user 2. PSTN user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 2 6. Verify PSTN user 1 is placed on hold 7. PSTN user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user 1 and PSTN user 2 11. PSTN user 1 hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
37	Teams user Calls PBX A user, Teams user performs Unattended	1. Make a voice call from Teams user to PBX A user 1 2. Teams user hears Ring back Tone 3. PBX A user 1 answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 2	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	Transfer to PBX A user	<ul style="list-style-type: none"> <li>6. PBX A user 2 starts ringing</li> <li>7. PBX A user 1 hears Ring back Tone</li> <li>8. PBX A user 2 answers the call</li> <li>9. Verify two way audio between PBX A user 1 and PBX A user 2</li> <li>10. PBX A user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ul>			
38	Teams user Calls PBX A user, Teams user performs Unattended Transfer to PBX B user	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user</li> <li>6. PBX B user starts ringing</li> <li>7. PBX A user hears Ring back Tone</li> <li>8. PBX B user answers the call</li> <li>9. Verify two way audio between PBX A user and PBX B user</li> <li>10. PBX A user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
39	Teams user Calls PBX A user, Teams user performs Unattended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX A user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX A user and PSTN user</li> <li>10. PBX A user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
40	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user 2</li> <li>6. PBX B user 2 starts ringing</li> <li>7. PBX B user 1 hears Ring back Tone</li> <li>8. PBX B user 2 answers the call</li> <li>9. Verify two way audio between PBX B user 1 and PBX B user 2</li> <li>10. PBX B user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
41	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PBX B user and PBX A user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
42	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX B user and PSTN user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
43	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user</li> <li>6. PBX B user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX B user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX B user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
44	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX A user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
45	Teams user Calls PSTN 1 user, Teams user performs Unattended Transfer to PSTN 2 user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user 2</li> <li>6. PSTN user 2 starts ringing</li> <li>7. PSTN user 1 hears Ring back Tone</li> <li>8. PSTN user 2 answers the call</li> <li>9. Verify two way audio between PSTN user 1 and PSTN user 2</li> <li>10. PSTN user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
46	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user 1 to Teams user</li> <li>2. PBX A user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user 2</li> <li>6. PBX A user 2 starts ringing</li> <li>7. PBX A user 1 hears Ring back Tone</li> <li>8. PBX A user 2 answers the call</li> <li>9. Verify two way audio between PBX A user 1 and PBX A user 2</li> <li>10. PBX A user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
47	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user</li> <li>6. PBX B user starts ringing</li> <li>7. PBX A user hears Ring back Tone</li> <li>8. PBX B user answers the call</li> <li>9. Verify two way audio between PBX A user and PBX B user</li> <li>10. PBX A user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
48	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX A user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX A user and PSTN user</li> <li>10. PBX A user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	



External ID	Title	Procedure	Expected Results	Status	Comments
49	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user 1 to Teams user</li> <li>2. PBX B user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user 2</li> <li>6. PBX B user 2 starts ringing</li> <li>7. PBX B user 1 hears Ring back Tone</li> <li>8. PBX B user 2 answers the call</li> <li>9. Verify two way audio between PBX B user 1 and PBX B user 2</li> <li>10. PBX B user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
50	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PBX B user and PBX A user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
51	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX B user and PSTN user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
52	PSTN user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user</li> <li>6. PBX B user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX B user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX B user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
53	PSTN user Calls Teams user, Teams user performs Unattended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX A user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
54	PSTN 1 user Calls Teams user, Teams user performs Unattended Transfer to PSTN 2 user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user 1 to Teams user</li> <li>2. PSTN user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user 2</li> <li>6. PSTN user 2 starts ringing</li> <li>7. PSTN user 1 hears Ring back Tone</li> <li>8. PSTN user 2 answers the call</li> <li>9. Verify two way audio between PSTN user 1 and PSTN user 2</li> <li>10. PSTN user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
55	PSTN user calls Teams user, Teams user performs Unattended Transfer to second Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user 1</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user 1 transfers the call to Teams user 2</li> <li>6. Teams user 2 starts ringing</li> <li>7. Teams user 2 answers the call</li> <li>8. Verify two way audio between PSTN user and Teams user 2</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
56	Teams user Calls PBX A user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user 2 to the ongoing call</li> <li>6. PBX A user 2 starts ringing</li> <li>7. PBX A user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
57	Teams user user Calls PBX A user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX B user to the ongoing call</li> <li>6. PBX B user starts ringing</li> <li>7. PBX B user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a

External ID	Title	Procedure	Expected Results	Status	Comments
					user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
58	Teams user user Calls PBX A user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user to the ongoing call</li> <li>6. PSTN user starts ringing</li> <li>7. PSTN user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via

External ID	Title	Procedure	Expected Results	Status	Comments
					Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
59	Teams user user Calls PBX B user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX B user 2 to the ongoing call</li> <li>6. PBX B user 2 starts ringing</li> <li>7. PBX B user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
60	Teams user user Calls PBX B user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user to the ongoing call</li> <li>6. PBX A user starts ringing</li> <li>7. PBX A user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone



External ID	Title	Procedure	Expected Results	Status	Comments
					has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
61	Teams user user Calls PBX B user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user to the ongoing call</li> <li>6. PSTN user starts ringing</li> <li>7. PSTN user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to

External ID	Title	Procedure	Expected Results	Status	Comments
					add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
62	Teams user user Calls PSTN user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX B user to the ongoing call</li> <li>6. PBX B user starts ringing</li> <li>7. PBX B user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into

External ID	Title	Procedure	Expected Results	Status	Comments
					conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
63	Teams user user Calls PSTN user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user to the ongoing call</li> <li>6. PBX A user starts ringing</li> <li>7. PBX A user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only

External ID	Title	Procedure	Expected Results	Status	Comments
					with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
64	Teams user user Calls PSTN 1 user, Teams user adds PSTN 2 user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user 2 to the ongoing call</li> <li>6. PSTN user 2 starts ringing</li> <li>7. PSTN user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
65	PBX A user Calls Teams user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user 1 to Teams user</li> <li>2. PBX A user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user 2 to the ongoing call</li> <li>6. PBX A user 2 starts ringing</li> <li>7. PBX A user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
66	PBX A user Calls Teams user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBXB user to the ongoing call</li> <li>6. PBX B user starts ringing</li> <li>7. PBX B user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
67	PBX A user Calls Teams user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user to the ongoing call</li> <li>6. PSTN user starts ringing</li> <li>7. PSTN user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing



External ID	Title	Procedure	Expected Results	Status	Comments
68	PBX B user Calls Teams user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user 1 to Teams user</li> <li>2. PBX B user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX B user 2 to the ongoing call</li> <li>6. PBX B user 2 starts ringing</li> <li>7. PBX B user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
69	PBX B user Calls Teams user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user to the ongoing call</li> <li>6. PBX A user starts ringing</li> <li>7. PBX A user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
70	PBX B user Calls Teams user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user to the ongoing call</li> <li>6. PSTN user starts ringing</li> <li>7. PSTN user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
71	PSTN user Calls Teams user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX B user to the ongoing call</li> <li>6. PBX B user starts ringing</li> <li>7. PBX B user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
72	PSTN user Calls Teams user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PBX A user to the ongoing call</li> <li>6. PBX A user starts ringing</li> <li>7. PBX A user answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
73	PSTN 1 user Calls Teams user, Teams user adds PSTN 2 user to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user 1 to Teams user</li> <li>2. PSTN user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user adds PSTN user 2 to the ongoing call</li> <li>6. PSTN user 2 starts ringing</li> <li>7. PSTN user 2 answers the call</li> <li>9. Verify all three users are able to hear each other</li> <li>10. Teams user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
74	PSTN user Calls Teams user, Teams user adds two or more users to the ongoing call	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user 1</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user 1 adds Teams user 2 to the ongoing call</li> <li>6. Verify Teams user 2 is added successfully to the call</li> <li>7. Teams user 1 adds PBX A user to the ongoing call</li> <li>9. Verify PBX A user is added successfully to the call</li> <li>10. Teams user 1 adds PBX B user to the ongoing call</li> <li>11. Verify PBX B user is added successfully to the call</li> <li>12. Verify all four users are able to hear each other</li> <li>13. All the users hang up and call is cleared successfully for all the users</li> </ol>	<ol style="list-style-type: none"> <li>1. Third user is added to the call successfully</li> <li>2. All three users are able to hear each other</li> </ol>	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
75	PBX A user Calls Teams user, Teams user CFA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user 2</li> <li>2. Make a voice call from PBX A user 1 to Teams user</li> <li>3. PBX A user 2 starts ringing</li> <li>4. PBX A user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
76	PBX A user Calls Teams user, Teams user CFA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
77	PBX A user Calls Teams user, Teams user CFA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	



External ID	Title	Procedure	Expected Results	Status	Comments
78	PBX B user Calls Teams user, Teams user CFA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user 2</li> <li>2. Make a voice call from PBX B user 1 to Teams user</li> <li>3. PBX B user 2 starts ringing</li> <li>4. PBX B user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
79	PBX B user Calls Teams user, Teams user CFA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
80	PBX B user Calls Teams user, Teams user CFA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
81	PSTN user Calls Teams user, Teams user CFA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
82	PSTN user Calls Teams user, Teams user CFA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
83	PSTN 1 user Calls Teams user, Teams user CFA to PSTN 2 user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user 2</li> <li>2. Make a voice call from PSTN user 1 to Teams user</li> <li>3. PSTN user 2 starts ringing</li> <li>4. PSTN user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
84	PBX A user Calls Teams user, Teams user CFNA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX A user 2</li> <li>2. Make a voice call from PBX A user 1 to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX A user 2 starts ringing</li> <li>4. PBX A user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
85	PBX A user Calls Teams user, Teams user CFNA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX B user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
86	PBX A user Calls Teams user, Teams user CFNA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PSTN user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
87	PBX B user Calls Teams user, Teams user CFNA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX B user 2</li> <li>2. Make a voice call from PBX B user 1 to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX B user 2 starts ringing</li> <li>4. PBX B user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
88	PBX B user Calls Teams user, Teams user CFNA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX A user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
89	PBX B user Calls Teams user, Teams user CFNA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PSTN user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
90	PSTN user Calls Teams user, Teams user CFNA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX B user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
91	PSTN user Calls Teams user, Teams user CFNA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PBX A user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
92	PSTN 1 user Calls Teams	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding no answer to PSTN user 2</li> </ol>	1. Teams user is able to forward the	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	user, Teams user CFNA to PSTN 2 user	<ol style="list-style-type: none"> <li>2. Make a voice call from PSTN user 1 to Teams user</li> <li>3. Teams user starts ringing</li> <li>4. Teams user does not answer the call</li> <li>5. Call gets forwarded after the no answer timeout value is reached</li> <li>6. PSTN user 2 starts ringing</li> <li>4. PSTN user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	incoming call successfully on reaching the No answer timeout value		
93	PSTN user calls Teams user, Teams user and users set for simultaneous ringing also rings	<ol style="list-style-type: none"> <li>1. Teams user sets simultaneous ringing to PBX A user and PBX B user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. Teams user, PBX A user and PBX B user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up</li> <li>7. Verify call is cleared successfully</li> <li>8. Repeat steps 2 to 6 where PBX B user answers the call</li> </ol>		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
94	Teams user with restricted Caller ID Calls PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user starts ringing</li> <li>4. Verify caller ID displayed on PBX A user is Unavailable/Private/Anonymous</li> <li>5. PBX A user answers the call</li> <li>6. Verify two way audio</li> <li>7. Teams user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
95	Teams user with restricted Caller ID Calls PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user starts ringing</li> <li>4. Verify caller ID displayed on PBX B user is Unavailable/Private/Anonymous</li> <li>5. PBX B user answers the call</li> <li>6. Verify two way audio</li> <li>7. Teams user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
96	Teams user with restricted Caller ID Calls PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user starts ringing</li> <li>4. Verify caller ID displayed on PSTN user is Unavailable/Private/Anonymous</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	



External ID	Title	Procedure	Expected Results	Status	Comments
		5. PSTN user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully			
97	PBX A user with restricted Caller ID Calls Teams user	1. Make a voice call from PBX A user with restricted caller ID to Teams user 2. PBX A user hears Ring back Tone 3. Teams user starts ringing 4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous 5. Teams user answers the call 6. Verify two way audio 7. PBX A user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to receive an inbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
98	PBX B user with restricted Caller ID Calls Teams user	1. Make a voice call from PBX B user with restricted caller ID to Teams user 2. PBX B user hears Ring back Tone 3. Teams user starts ringing 4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous 5. Teams user answers the call 6. Verify two way audio 7. PBX B user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to receive an inbound call with restricted caller ID 2. Call is successful with two way audio	NOT SUPPORTED	SFB will not send Anonymous in the From header or PAI

External ID	Title	Procedure	Expected Results	Status	Comments
99	PSTN user with restricted Caller ID Calls Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user with restricted caller ID to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user starts ringing</li> <li>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous</li> <li>5. Teams user answers the call</li> <li>6. Verify two way audio</li> <li>7. PSTN user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive an inbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
100	PBX A user Calls Teams user and leaves voicemail	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. Teams user does not answer the call</li> <li>3. Allow the call to get forwarded to voicemail</li> <li>4. PBX A user successfully leaves voicemail</li> <li>5. Teams user receives voicemail notification</li> <li>6. Teams user successfully retrieves voicemail</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive and retrieve voicemail successfully</li> </ol>	PASSED	
101	PBX B user Calls Teams user and	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. Teams user does not answer the call</li> <li>3. Allow the call to get forwarded to</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive and retrieve voicemail successfully</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	leaves voicemail	voicemail 4. PBX B user successfully leaves voicemail 5. Teams user receives voicemail notification 6. Teams user successfully retrieves voicemail			
102	PSTN user Calls Teams user and leaves voicemail	1. Make a voice call from PSTN user to Teams user 2. Teams user does not answer the call 3. Allow the call to get forwarded to voicemail 4. PSTN user successfully leaves voicemail 5. Teams user receives voicemail notification 6. Teams user successfully retrieves voicemail	1. Teams user is able to receive and retrieve voicemail successfully	PASSED	
103	Teams user Calls PBX A user and leaves voicemail	1. Make a voice call from Teams user to PBX A user 2. PBX A user does not answer the call 3. Allow the call to get forwarded to voicemail 4. Teams user successfully leaves voicemail and navigates voicemail menu using DTMF	1. Teams user is able to leave voicemail and navigate voice mail menu using DTMF successfully	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
104	Teams user Calls PBX B user and leaves voicemail	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. PBX B user does not answer the call</li> <li>3. Allow the call to get forwarded to voicemail</li> <li>4. Teams user successfully leaves voicemail and navigates voicemail menu using DTMF</li> </ol>	1. Teams user is able to leave voicemail and navigate voice mail menu using DTMF successfully	PASSED	
105	Teams user Calls PBX A user, PBX A returns call failure response	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. PBX A returns 486 Busy</li> <li>3. Verify Teams user gets appropriate notification or announcement and the call is cleared</li> <li>4. Repeat steps 1 to 3 where PBX A returns 480, 404, 503 SIP responses</li> <li>5. Document the observation on Teams user side</li> </ol>	1. Teams user handles the failure response successfully	PASSED	
106	Teams user Calls PBX A user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user using SIP URI</li> <li>2. PBX A user starts ringing</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	NOT TESTED	SIP URI Not tested for PBX A
107	Teams user Calls PBX B	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user using SIP URI</li> <li>2. PBX B user starts ringing</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	user using SIP URI	<ol style="list-style-type: none"> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	with two way audio successfully		
108	PBX A user Calls Teams user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user using SIP URI</li> <li>2. PBX A user starts ringing</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. PBX A user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	NOT TESTED	SIP URI Not tested for this PBX A
109	PBX B user Calls Teams user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user using SIP URI</li> <li>2. PBX B user starts ringing</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. PBX B user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	PASSED	
110	Teams user calls Skype for Business user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to Skype for Business user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. Skype for Business user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Verify the same scenario where</li> </ol>		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		Skype for Business user is internal and external			
111	Skype for Business user calls Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from Skype for Business user to Teams user</li> <li>2. Skype for Business user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Skype for Business user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Verify the same scenario where Skype for Business user is internal and external</li> </ol>		PASSED	
112	Teams user calls Skype for Business External Mobile user	<ol style="list-style-type: none"> <li>1. Skype for business user is an External Mobile user</li> <li>2. Make a voice call from Teams user to Skype for Business user</li> <li>3. Teams user hears Ring back Tone</li> <li>4. Skype for Business user answers the call</li> <li>5. Verify two way audio</li> <li>6. Teams user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
113	Skype for Business External Mobile user calls Teams user	<ol style="list-style-type: none"> <li>1. Skype for business user is an External Mobile user</li> <li>2. Make a voice call from Skype for Business user to Teams user</li> <li>3. Skype for Business user hears Ring back Tone</li> <li>4. Teams user answers the call</li> <li>5. Verify two way audio</li> <li>6. Skype for Business user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>		PASSED	
114	Teams user call other tenant users	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to another tenant users (Teams desktop client user, Teams mobile user, Skype for Business Online user)</li> <li>2. Verify call is successful</li> <li>3. Make one call to each different user one by one</li> </ol>		PASSED	
115	Teams users joins a meeting scheduled by Skype for business On-premises user	<ol style="list-style-type: none"> <li>1. Skype for business user schedules a meeting and invites Teams user 1 and Teams user 2</li> <li>2. Teams user 1 joins the meeting using the Join button</li> <li>3. Teams user 2 joins the meeting using the dial-in conferencing number</li> <li>4. Verify Teams users are able to join the meeting successfully</li> <li>5. Verify all three users are able to hear</li> </ol>		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<p>each other</p> <p>6. Skype for Business user ends the meeting</p>			
116	Teams user invites Skype for business users for a meeting	<p>1. Teams user schedules a meeting and invites Skype for Business user 1 and Skype for Business user 2</p> <p>2. Skype for Business user 1 joins the meeting using the Meeting link</p> <p>3. Skype for Business user 2 joins the meeting using the dial-in conferencing number</p> <p>4. Verify all three users are able to hear each other</p> <p>5. Teams user ends the meeting</p>		PASSED	