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)

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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 helps 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

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.

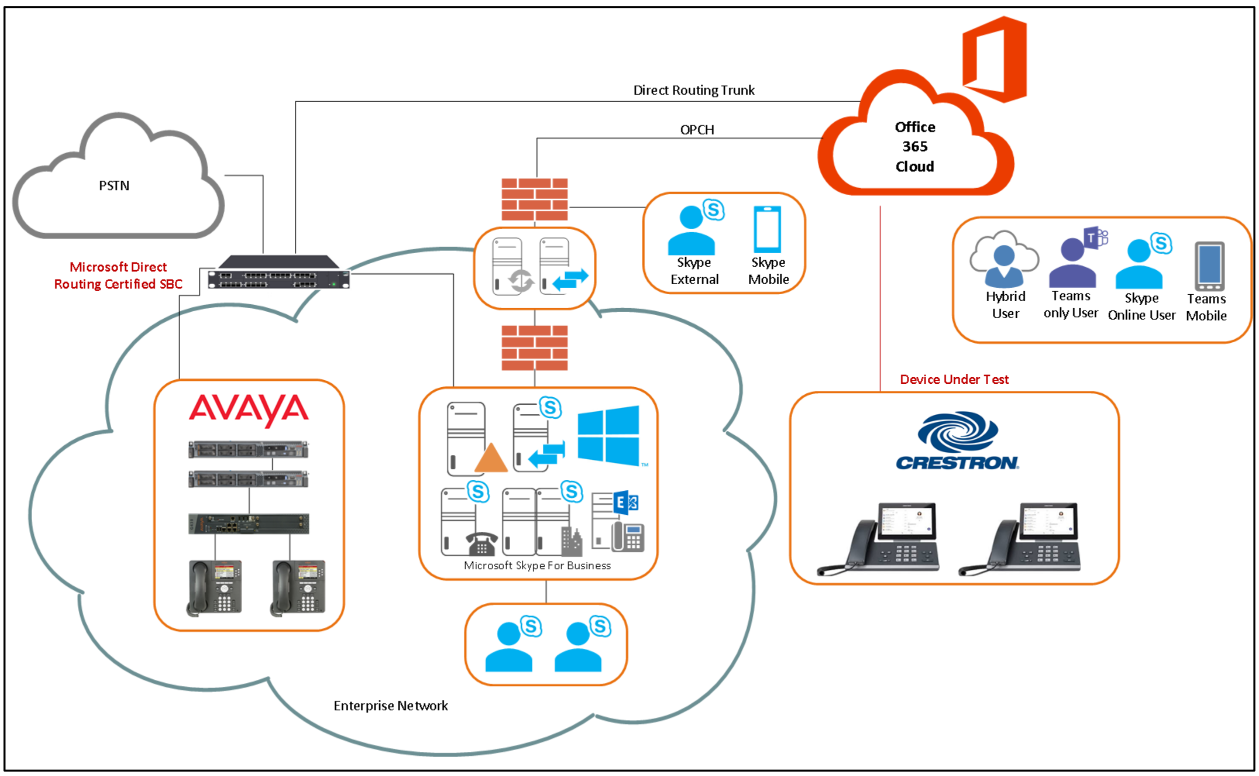


Figure 1 Network Topology

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	Section 4.3
Step 1	Microsoft Teams Configuration	Section 4.3
Step 2	AudioCodes VE SBC Configuration	Section 4.4
Step 3	Avaya Aura Communication Manager	Section 4.5
Step 4	Avaya Aura Session Manager	Section 4.6
Step 5	Avaya SBCE	Section 4.7

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
AudioCodes	
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
Avaya Aura Communication Manager	
IP Address	10.89.26.4 (Signaling)/10.89.26.14 (Media)
Subnet Mask	255.255.255.0
Avaya Aura Session Manager	
LAN IP Address	10.89.26.7
LAN Subnet Mask	255.255.255.0
Avaya SBCE	
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
Skype for Business 2015	
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.

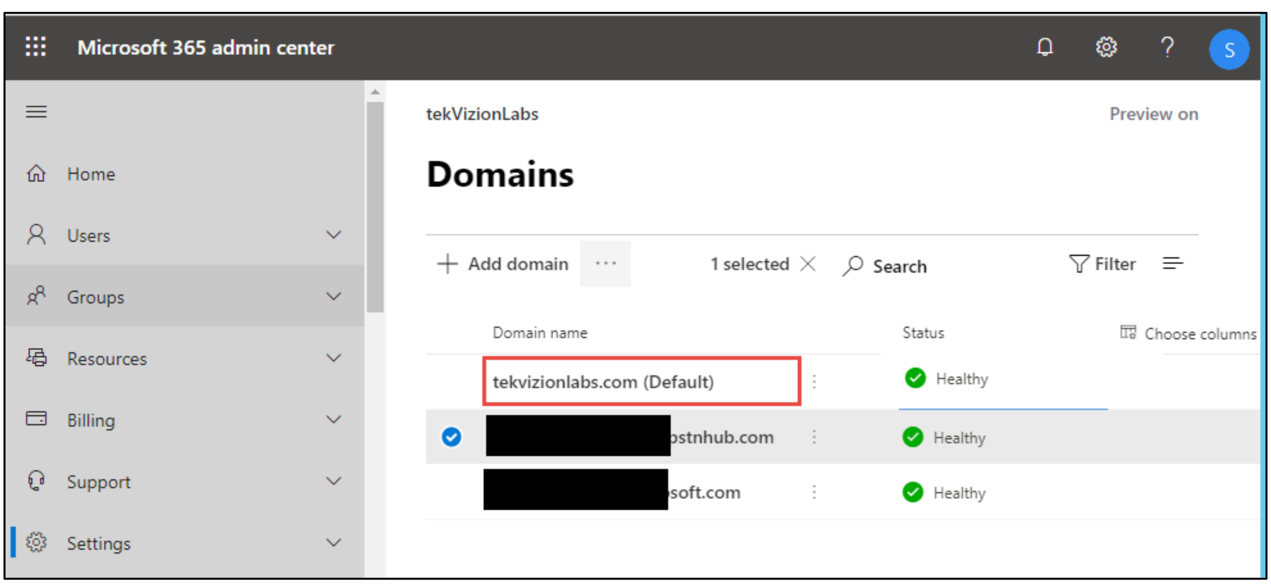


Figure 2 – Add Domain

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.

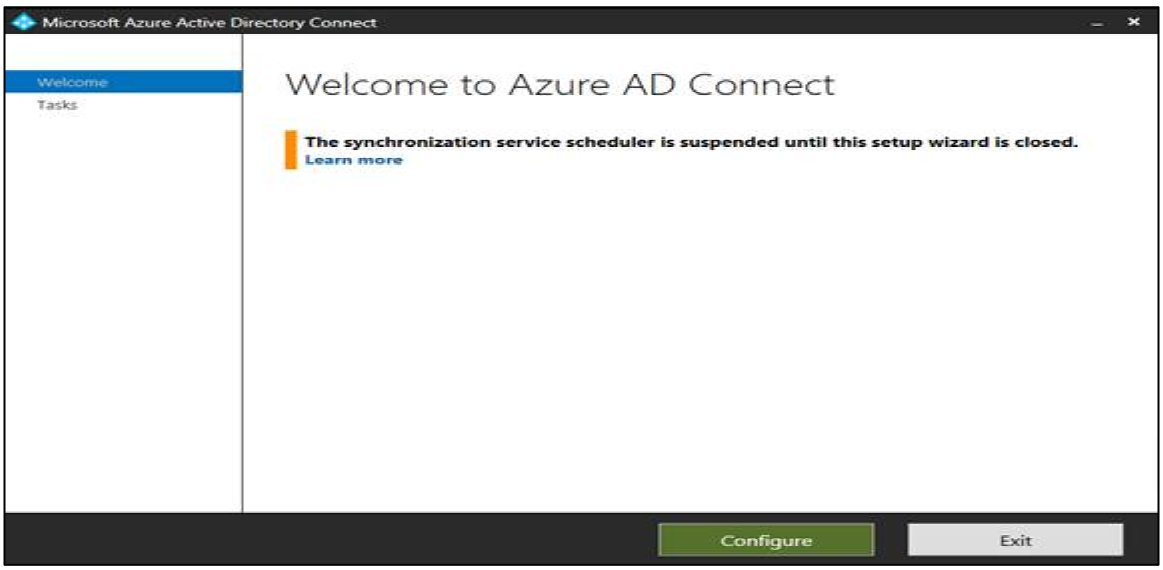


Figure 3 – Azure AD – Sync Process

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.

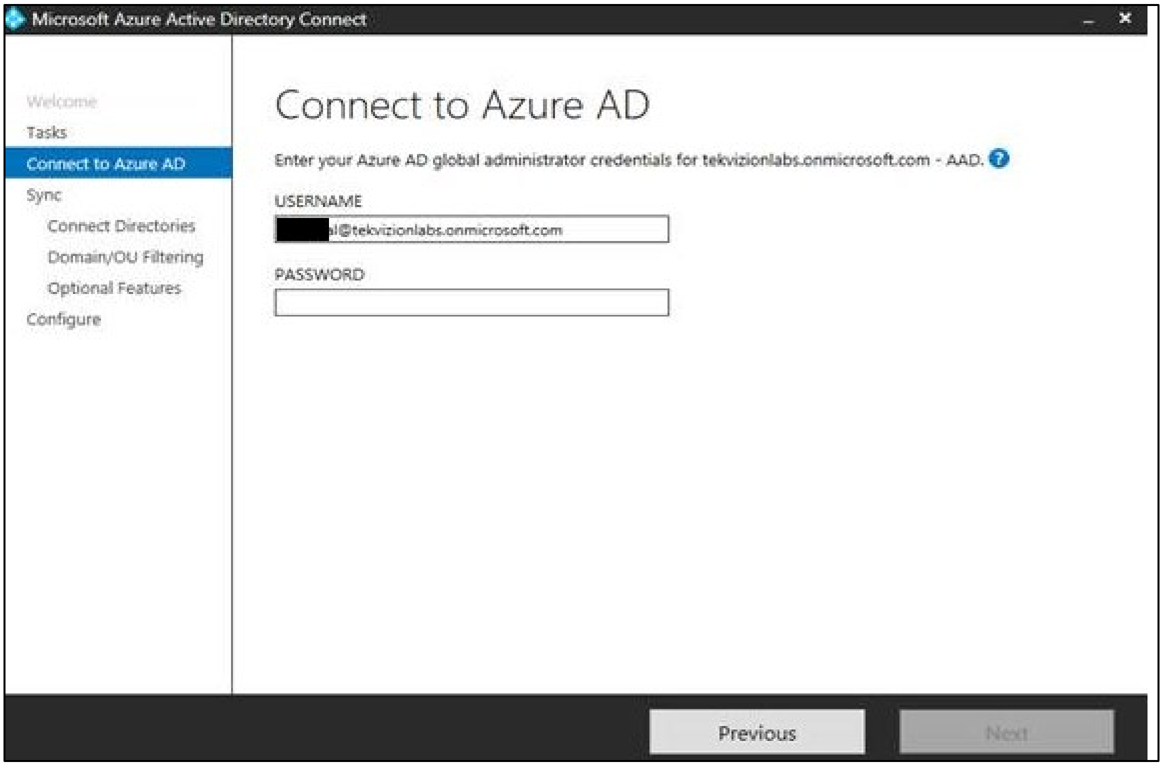


Figure 4 – Azure AD – Sync Process

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

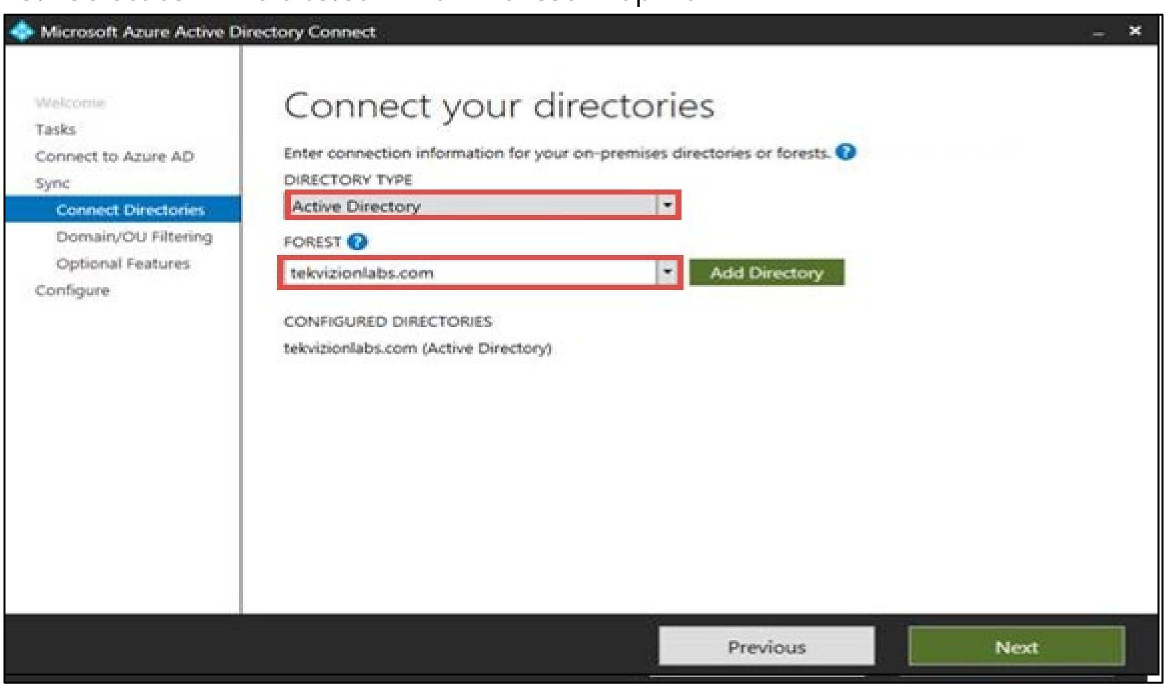


Figure 5 – Azure AD – Sync Process

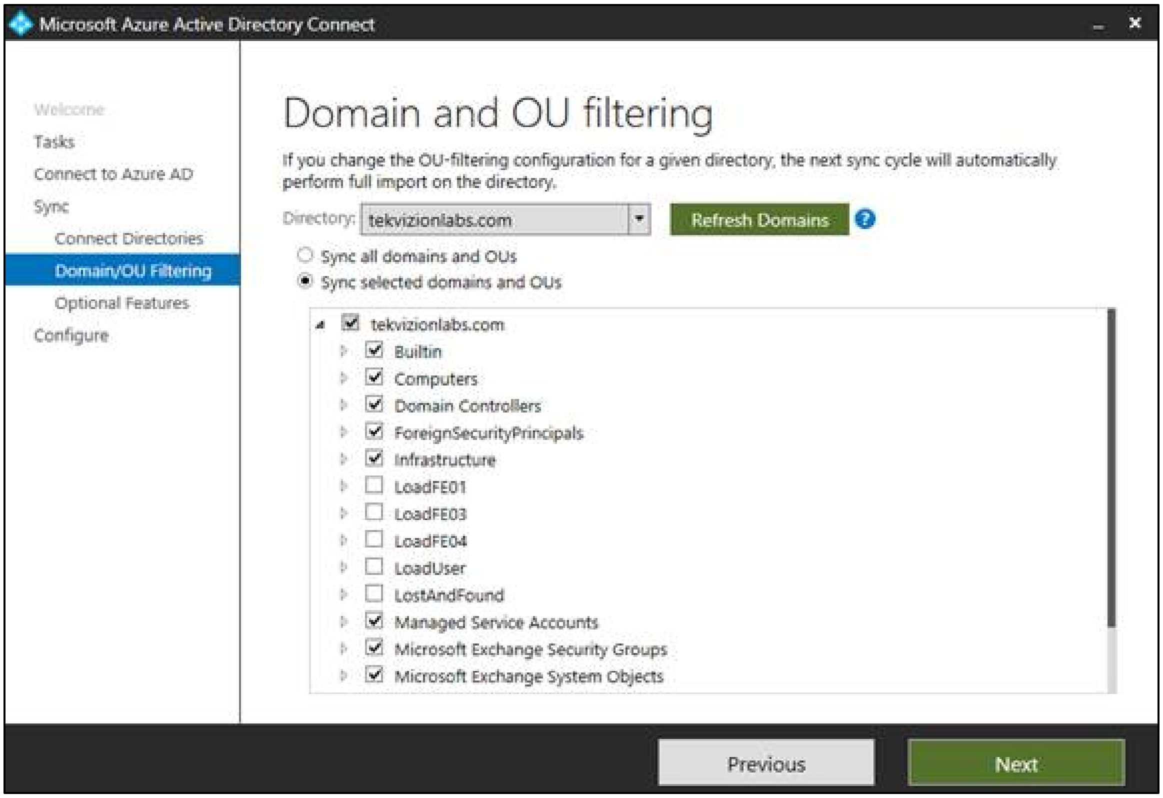


Figure 6 – Azure AD – Sync Process

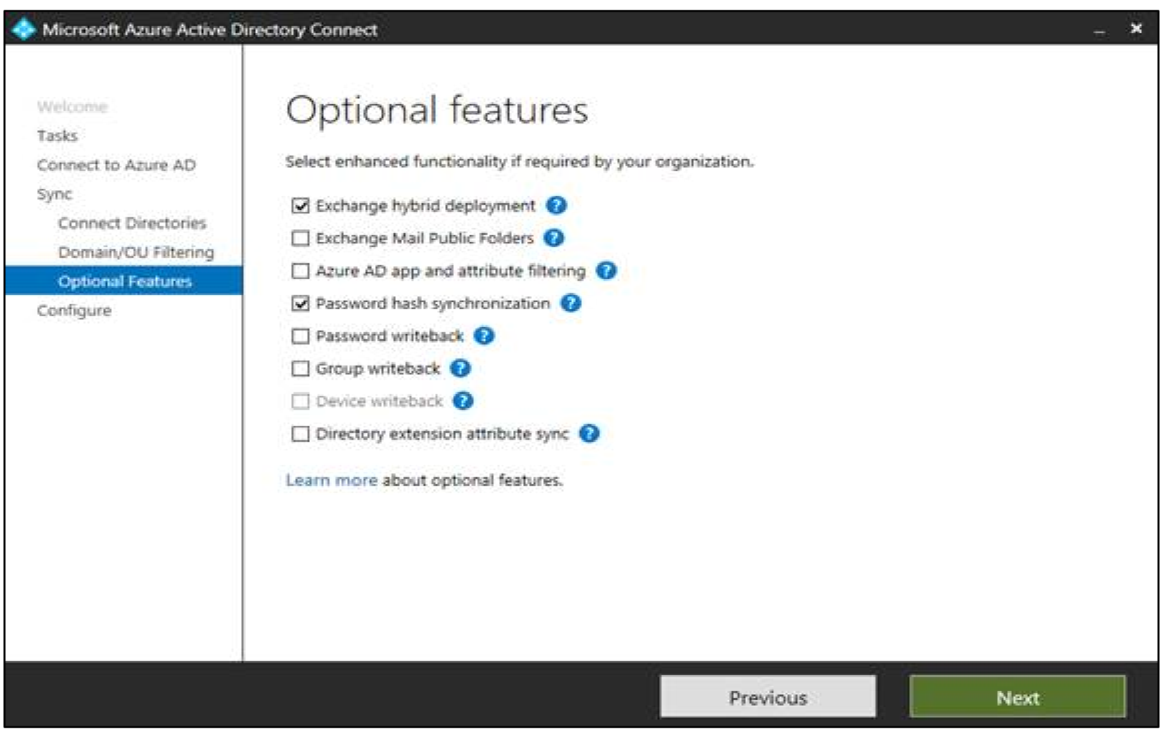


Figure 7 – Azure AD – Sync Process

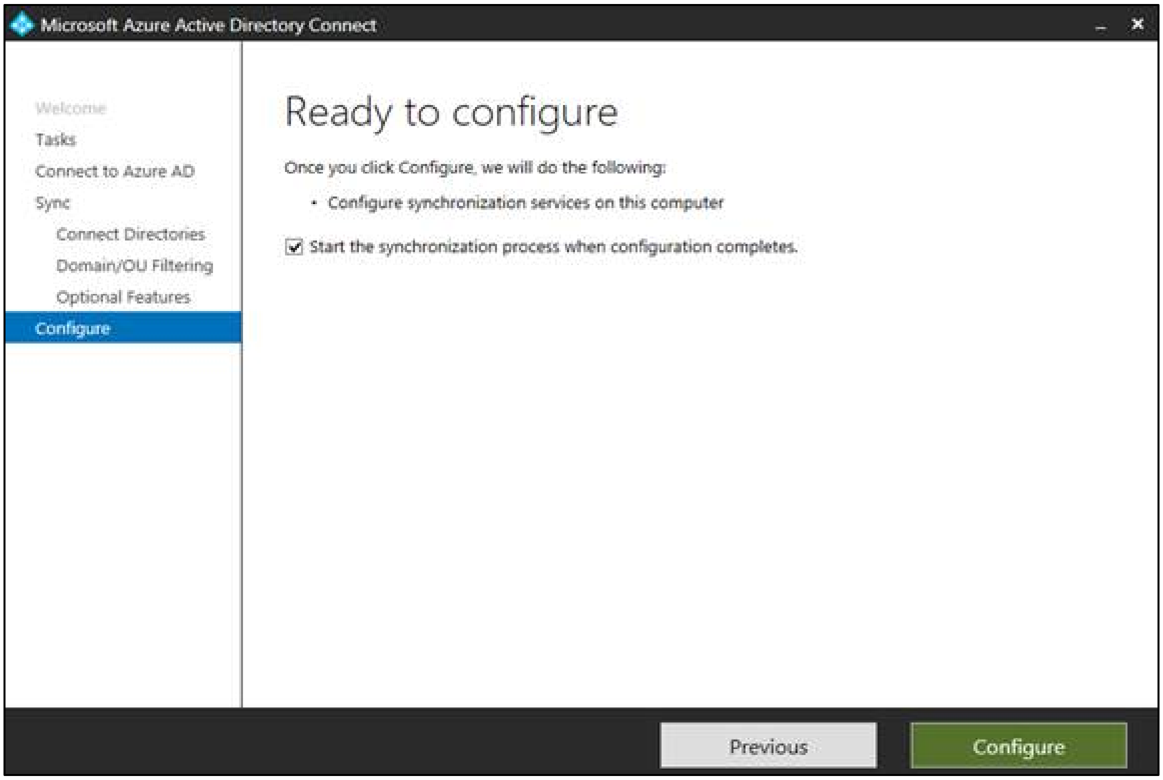


Figure 8 – Azure AD – Sync Process

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

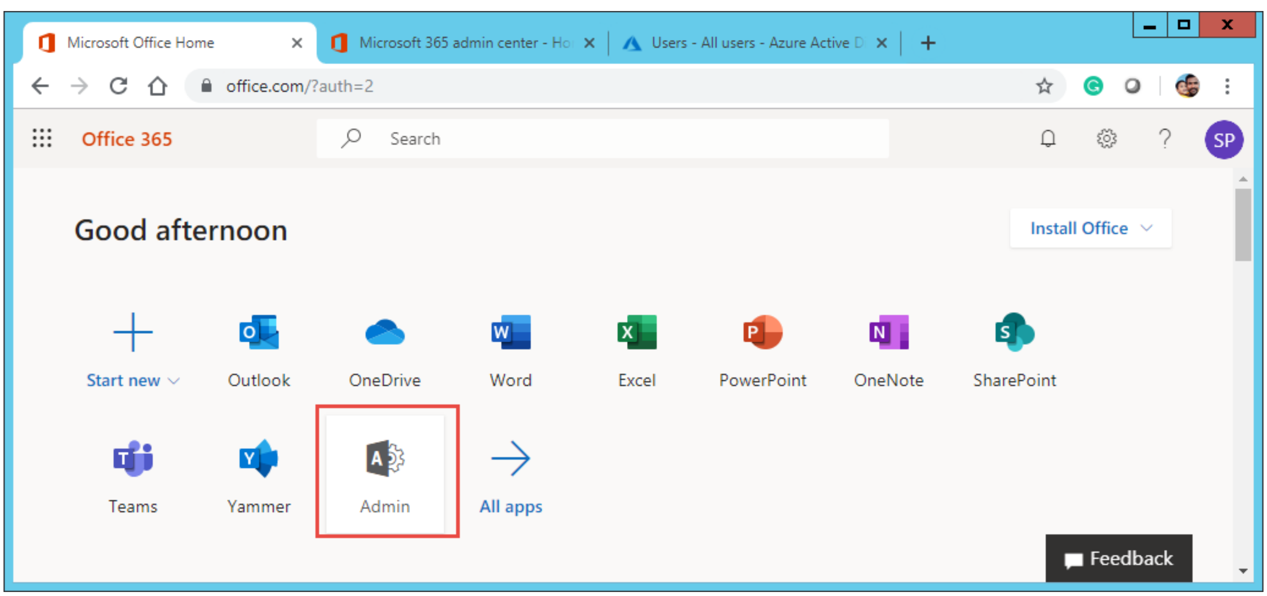


Figure 9 – Azure AD – Sync Process

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

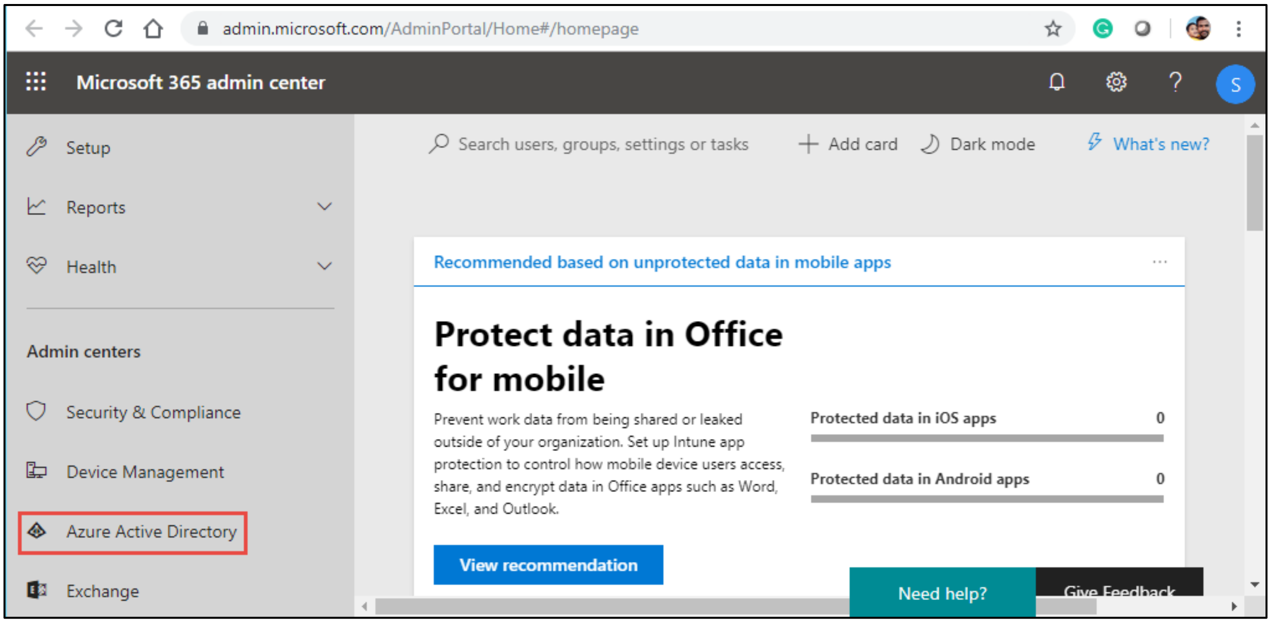


Figure 10 – Azure AD – Sync Process

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

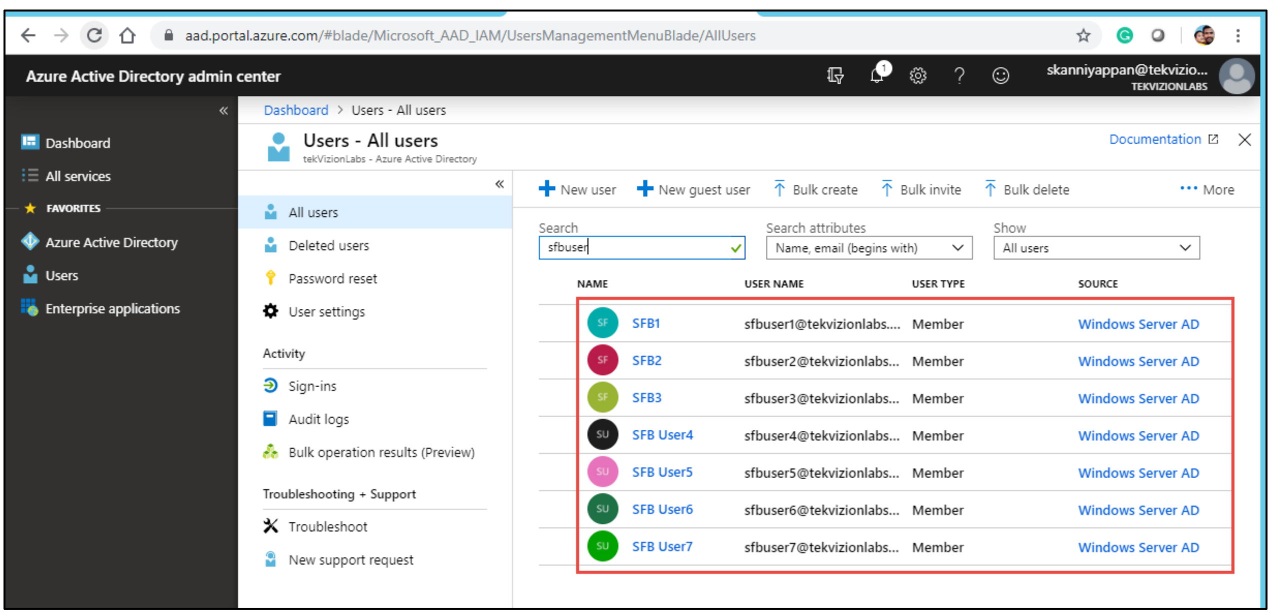


Figure 11 – Azure AD – Sync Process

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

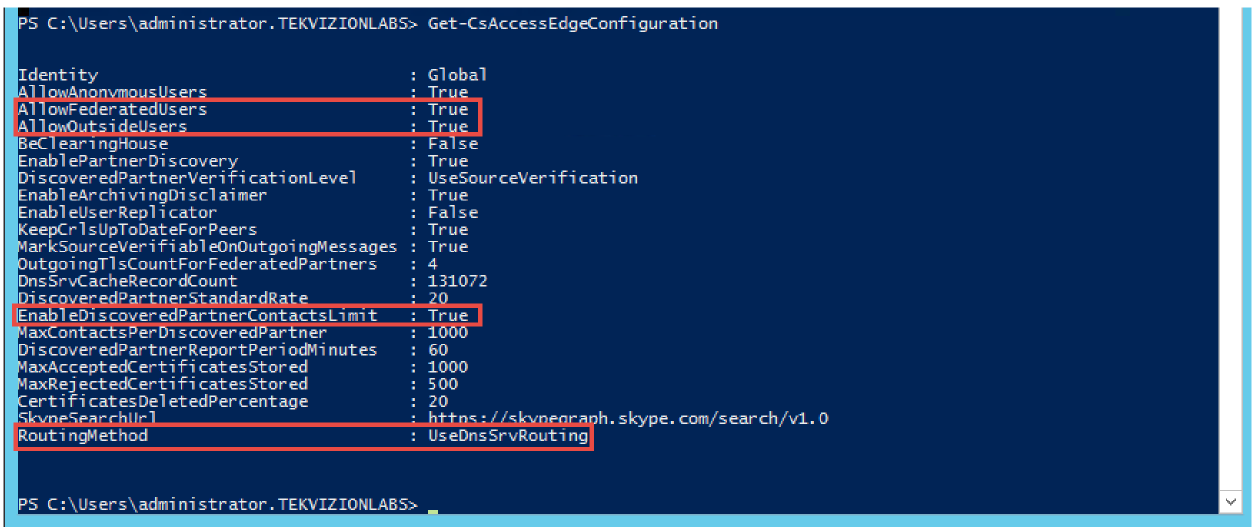


Figure 12 – Edge Federation

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

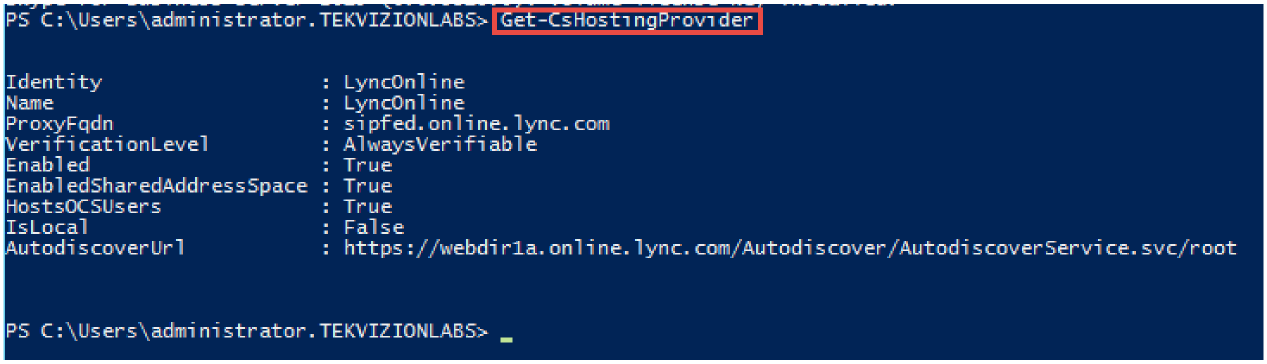


Figure 13 – Hosting Provider

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 enabled 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

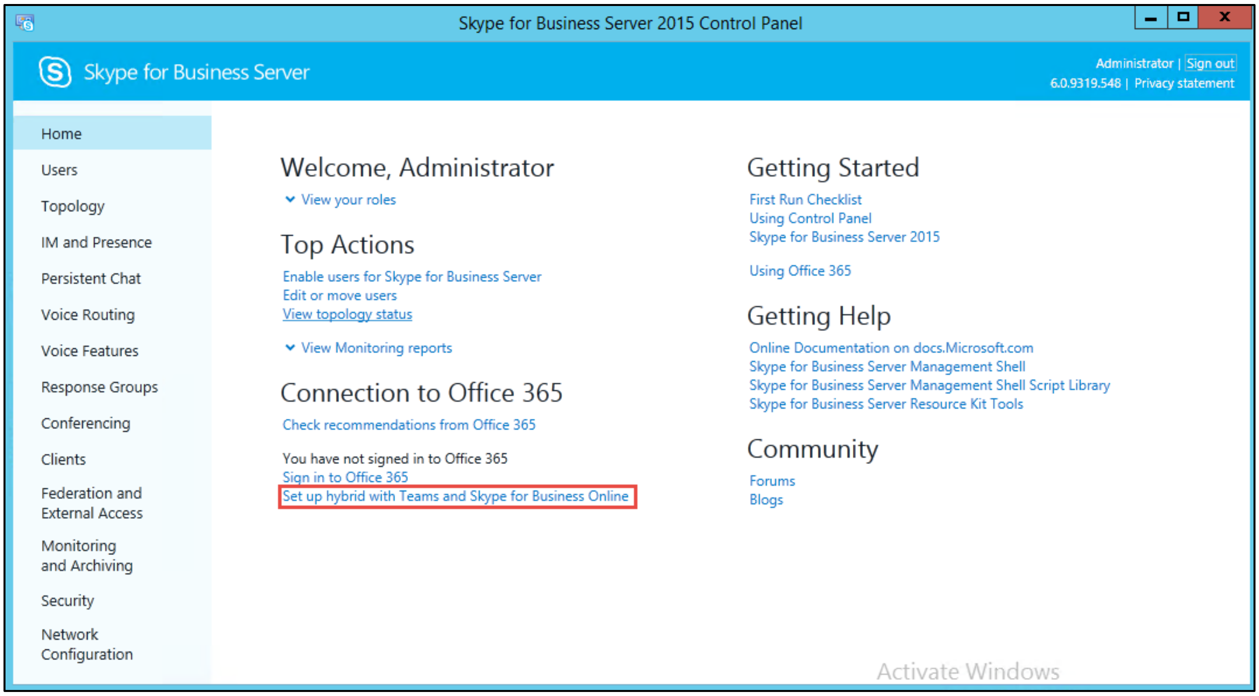


Figure 14 – Hybrid with Teams and SFB

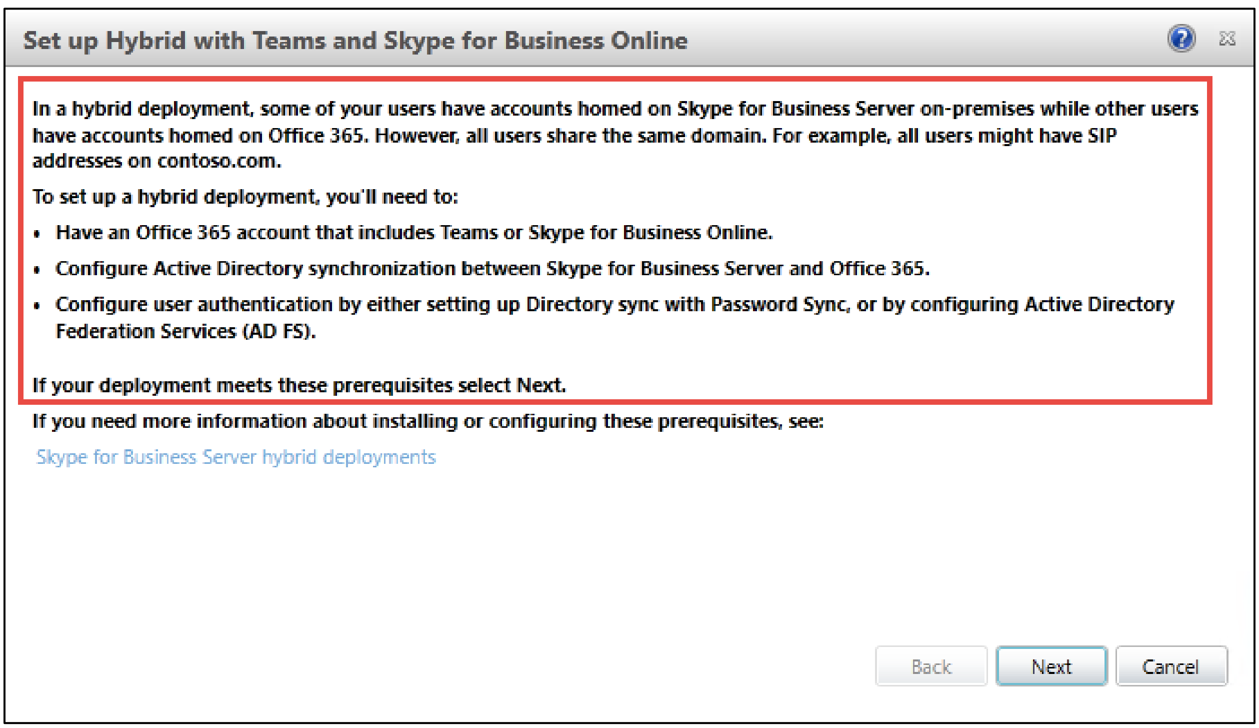


Figure 15 – Hybrid with Teams and SFB Contd.

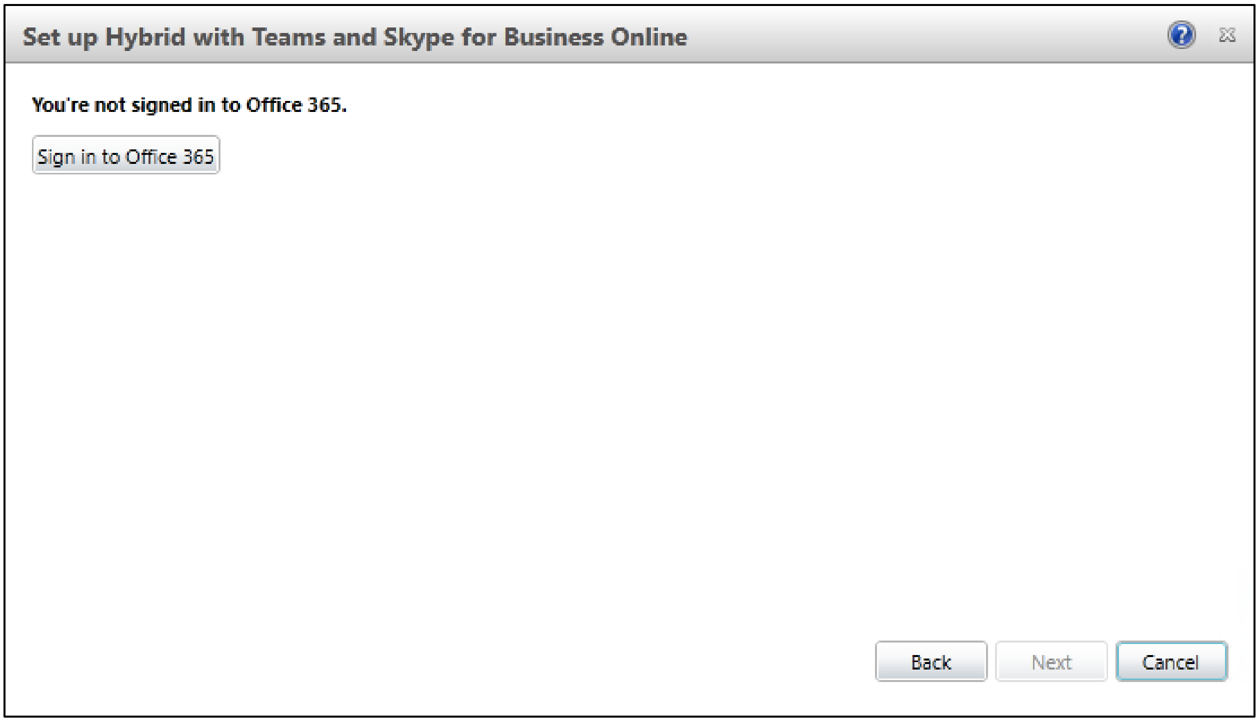


Figure 16 – Hybrid with Teams and SFB Contd.

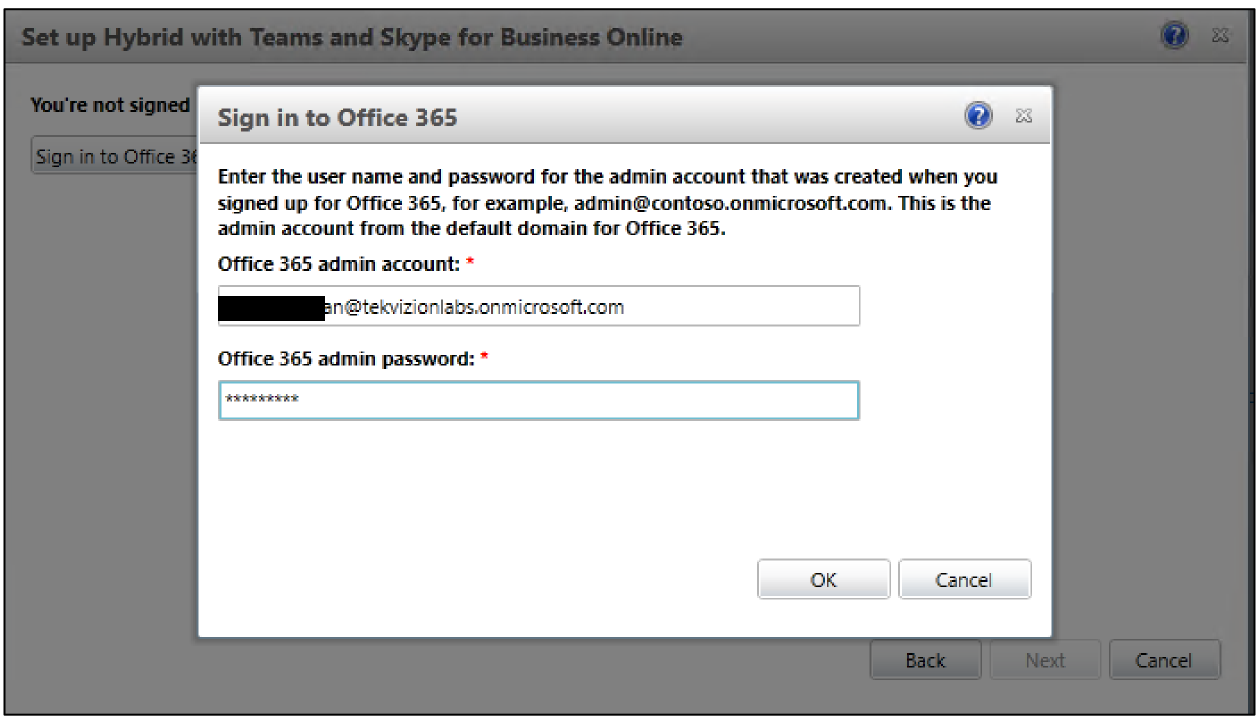


Figure 17 – Hybrid with Teams and SFB Contd.

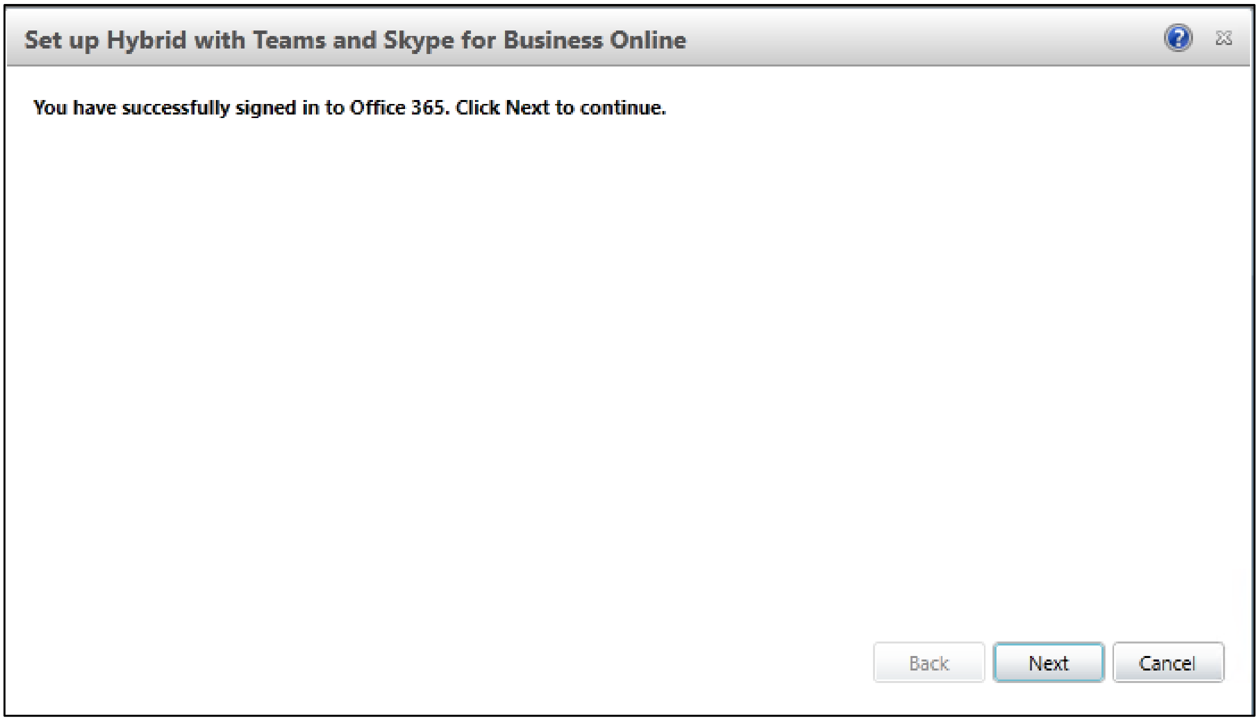


Figure 18 – Hybrid with Teams and SFB Contd.

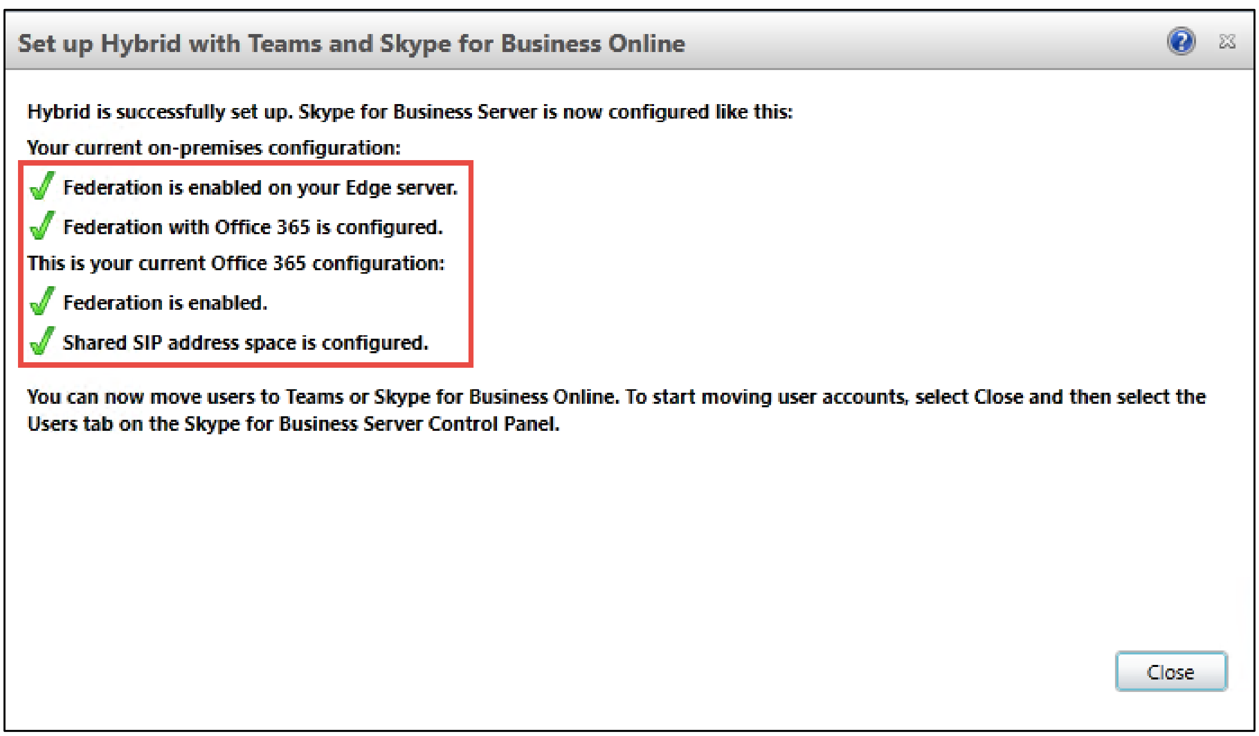


Figure 19 – Hybrid with Teams and SFB Contd.

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.

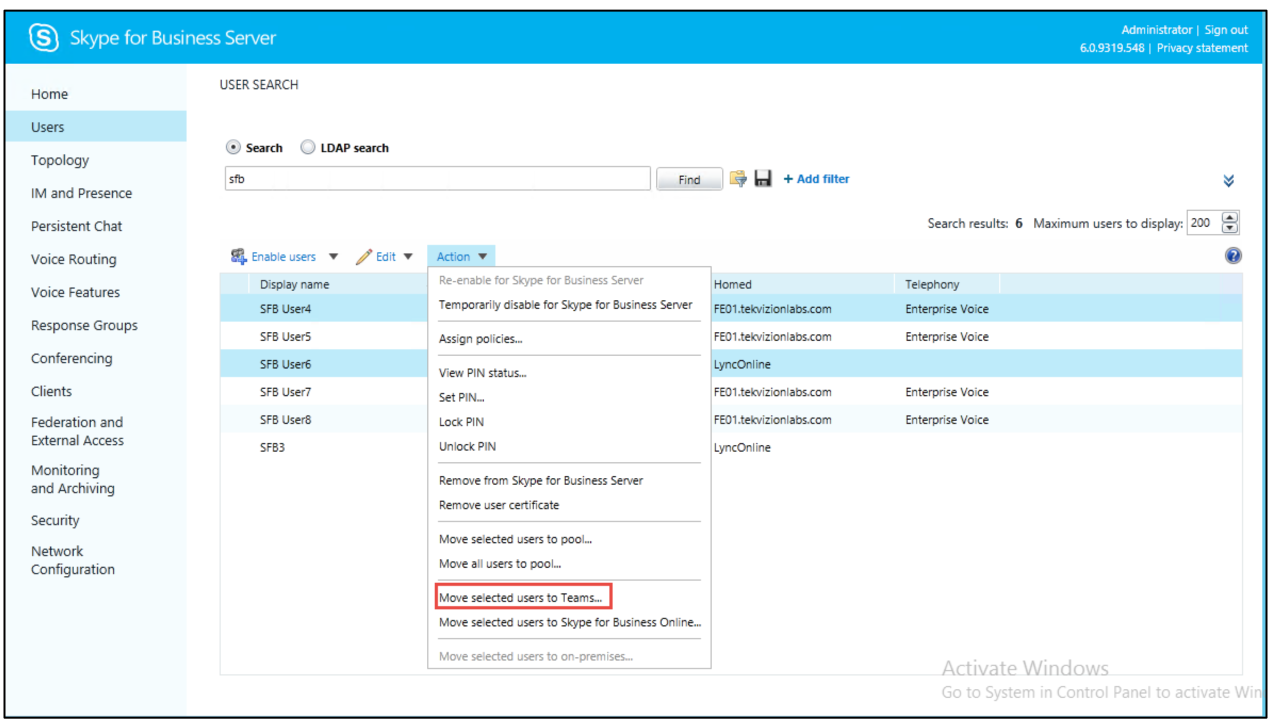


Figure 20 – Move Users to Teams

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

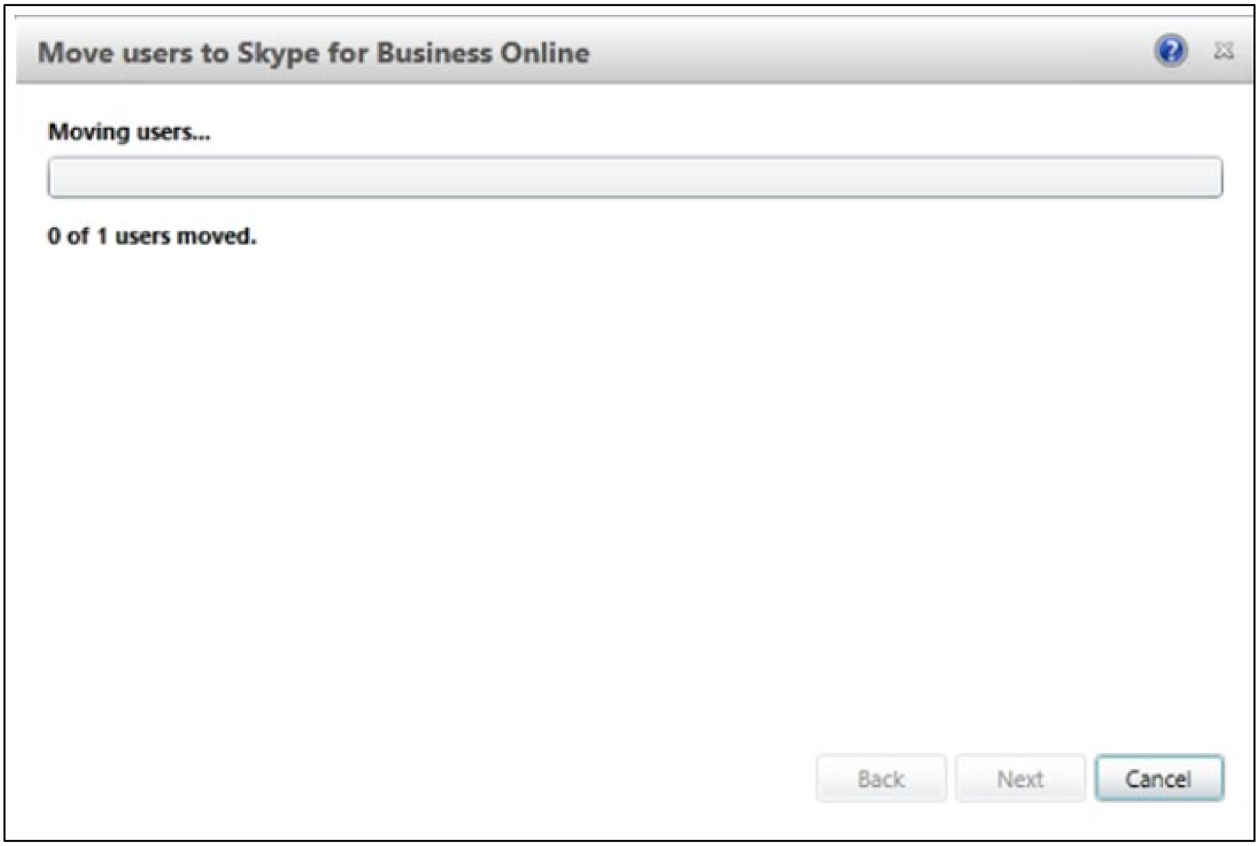


Figure 21 – Move Users to Teams Contd.

The migration of the selected user is complete.

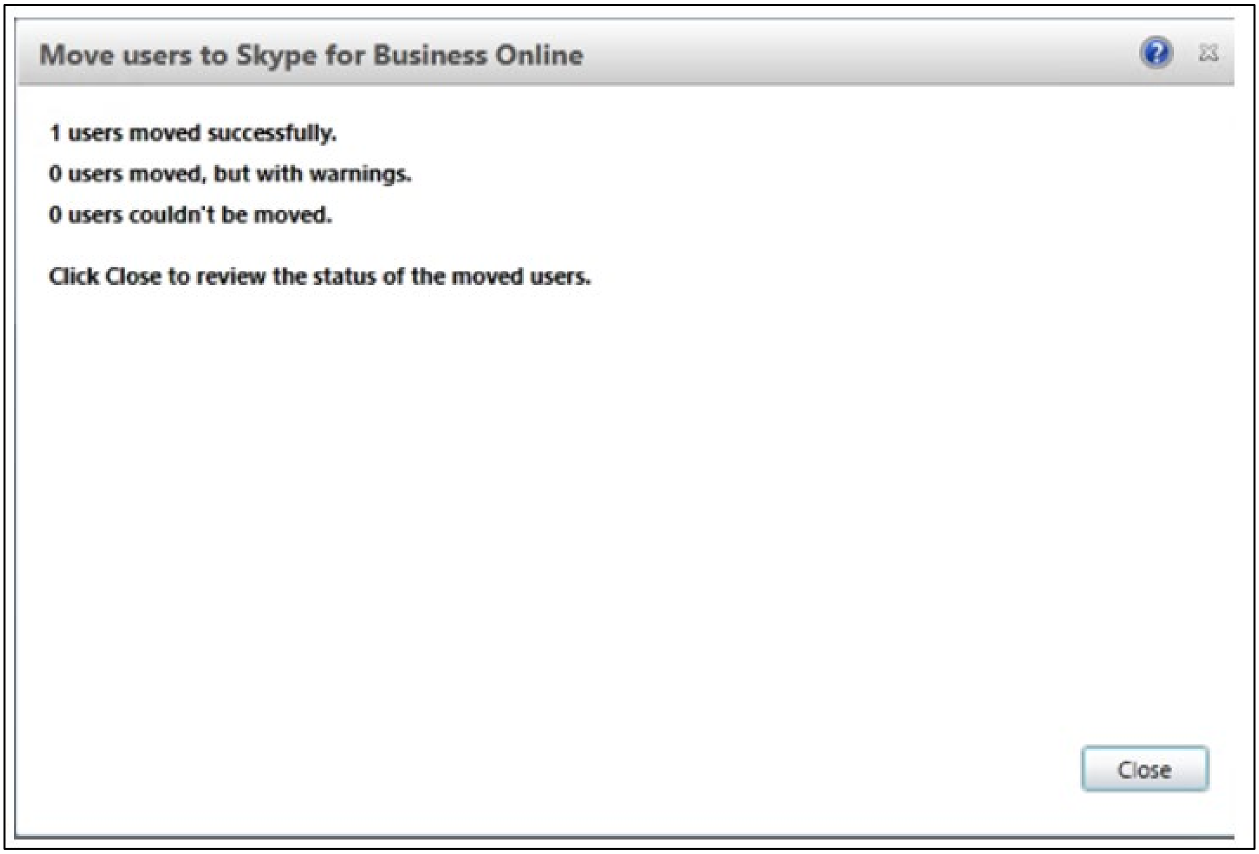


Figure 22 – Move Users to Teams Contd.

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

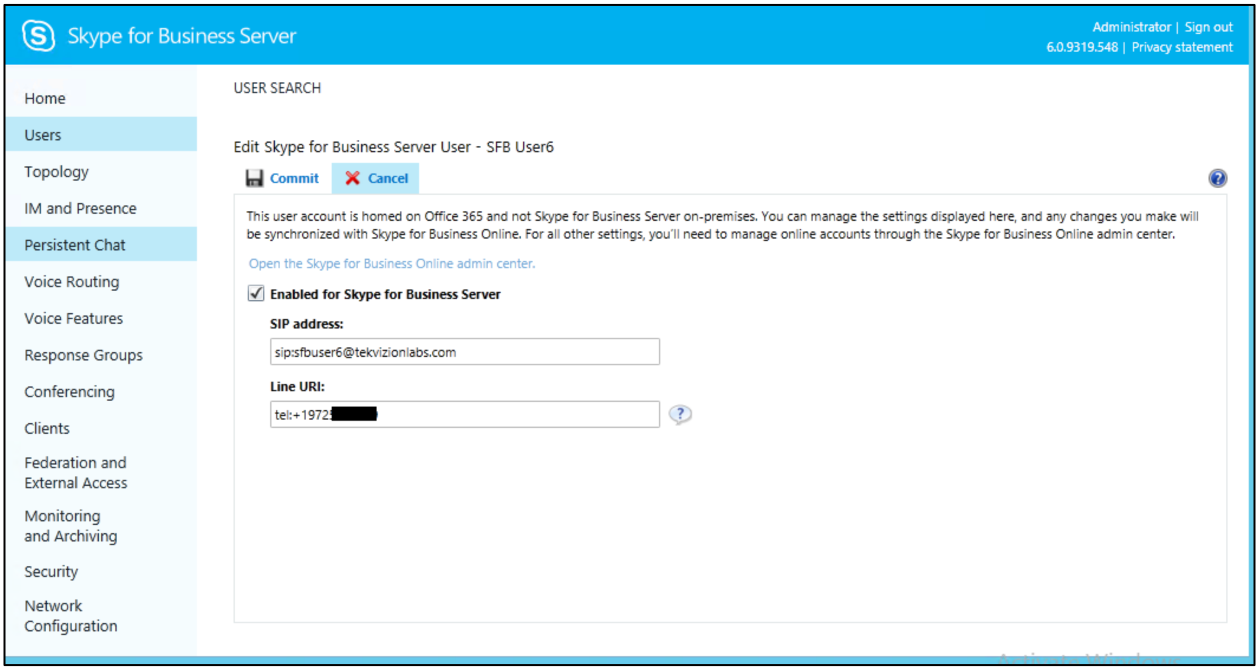


Figure 23 – Move Users to Teams

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.

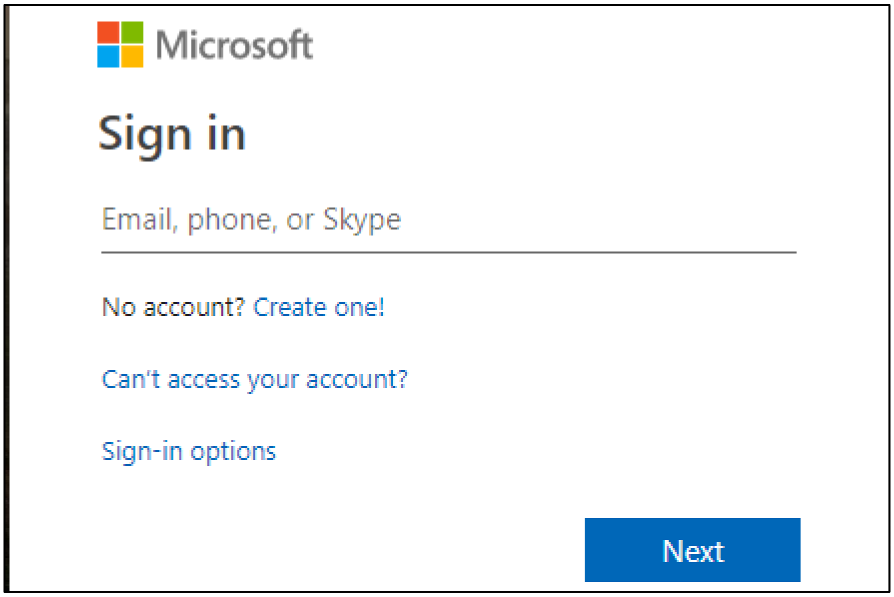


Figure 24: Office 365 Portal Login

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

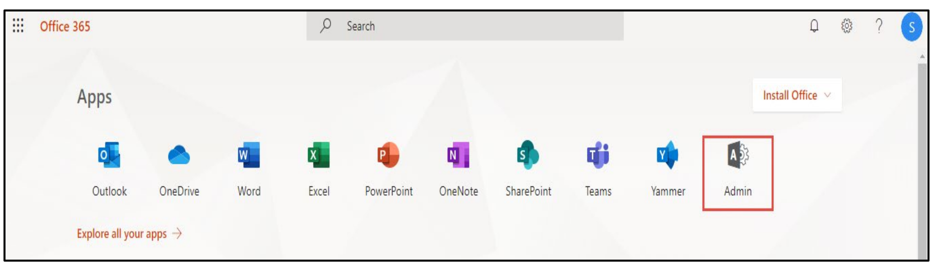


Figure 25: Office 365 Portal Login

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

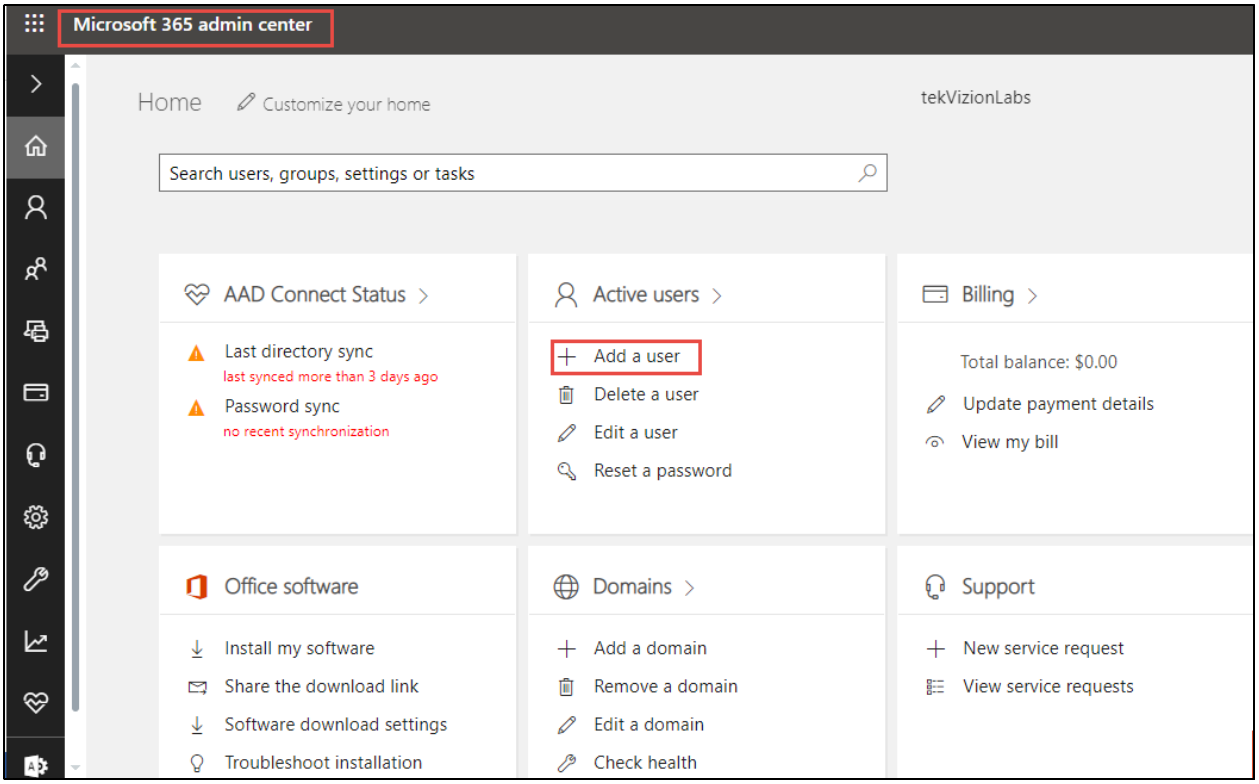


Figure 26: Teams User Creation

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

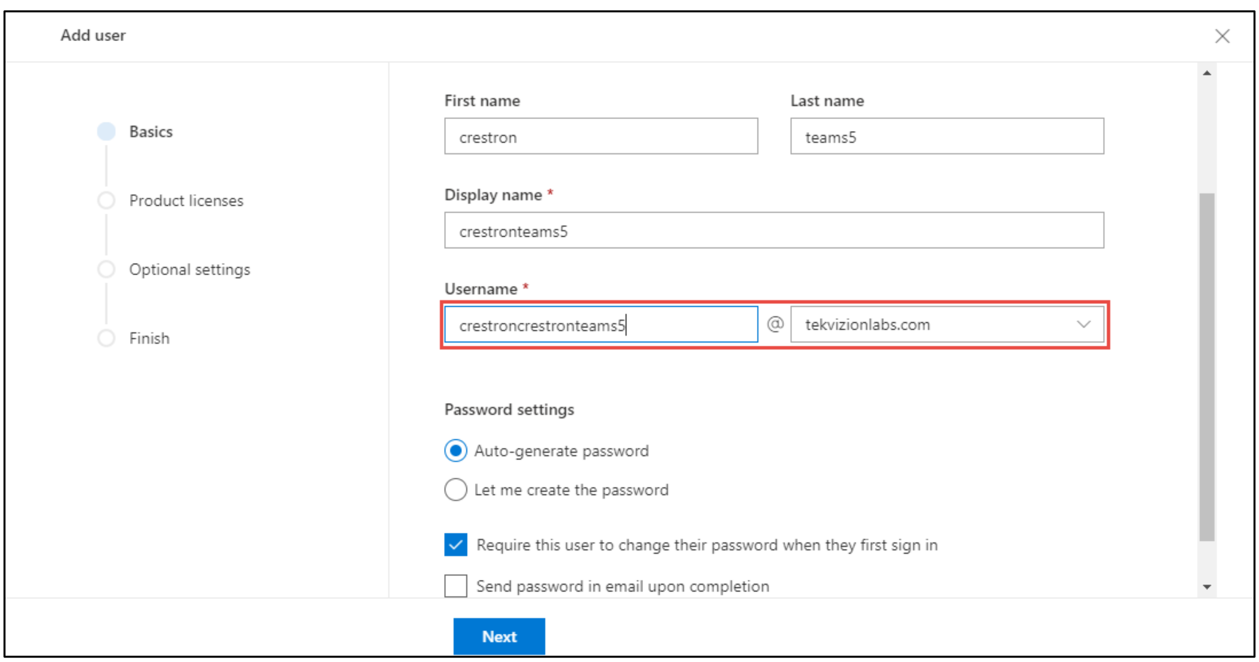


Figure 27: Teams User Creation – Contd.

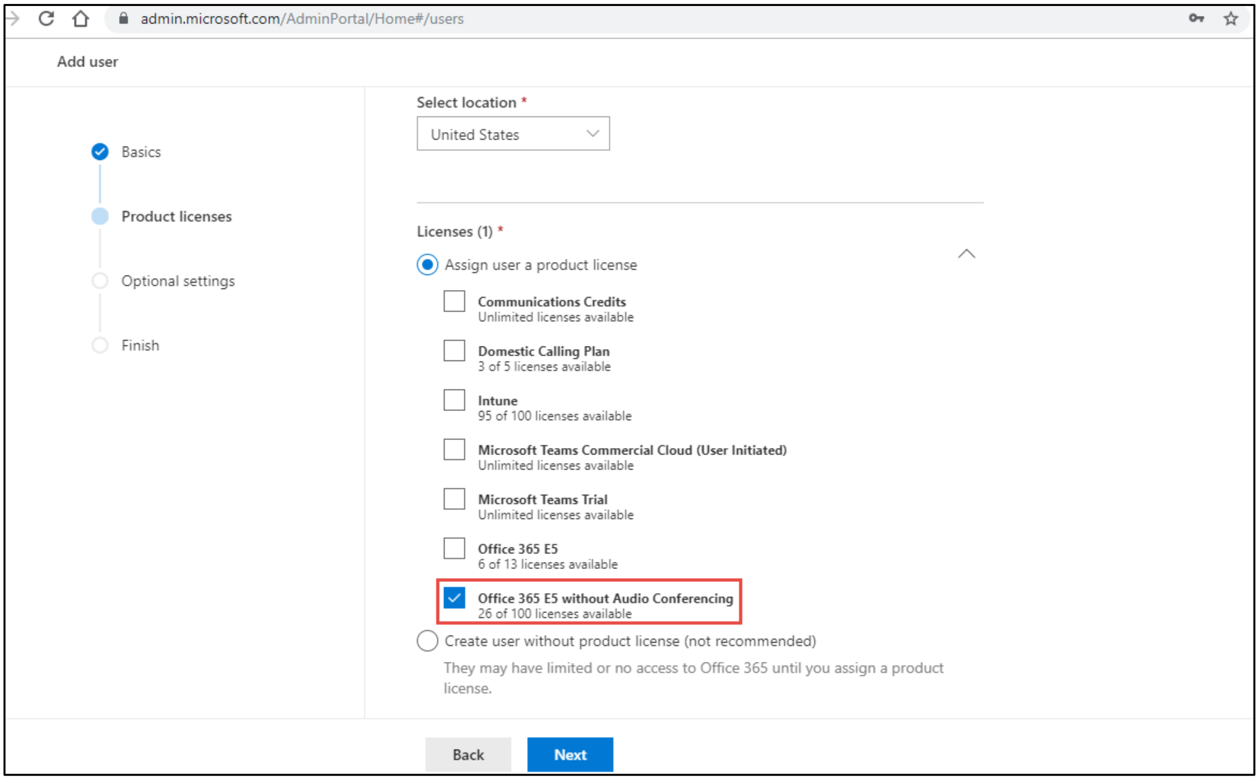


Figure 28: Teams User Creation – Contd.

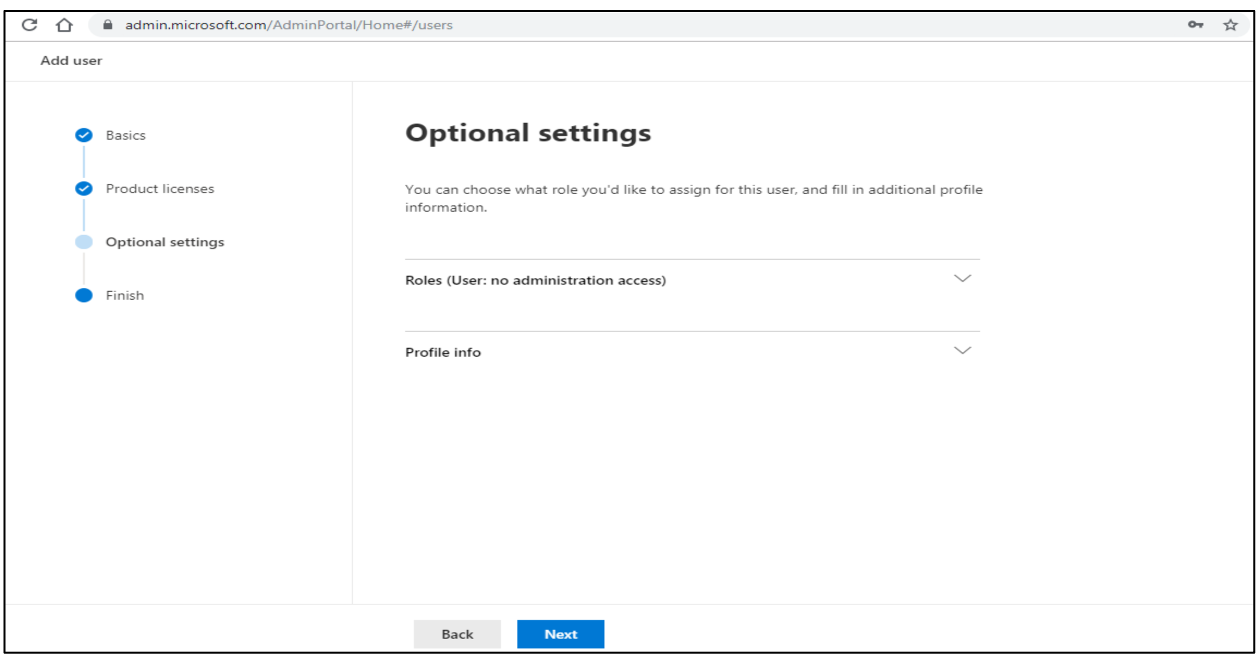


Figure 29: Teams User Creation – Contd.

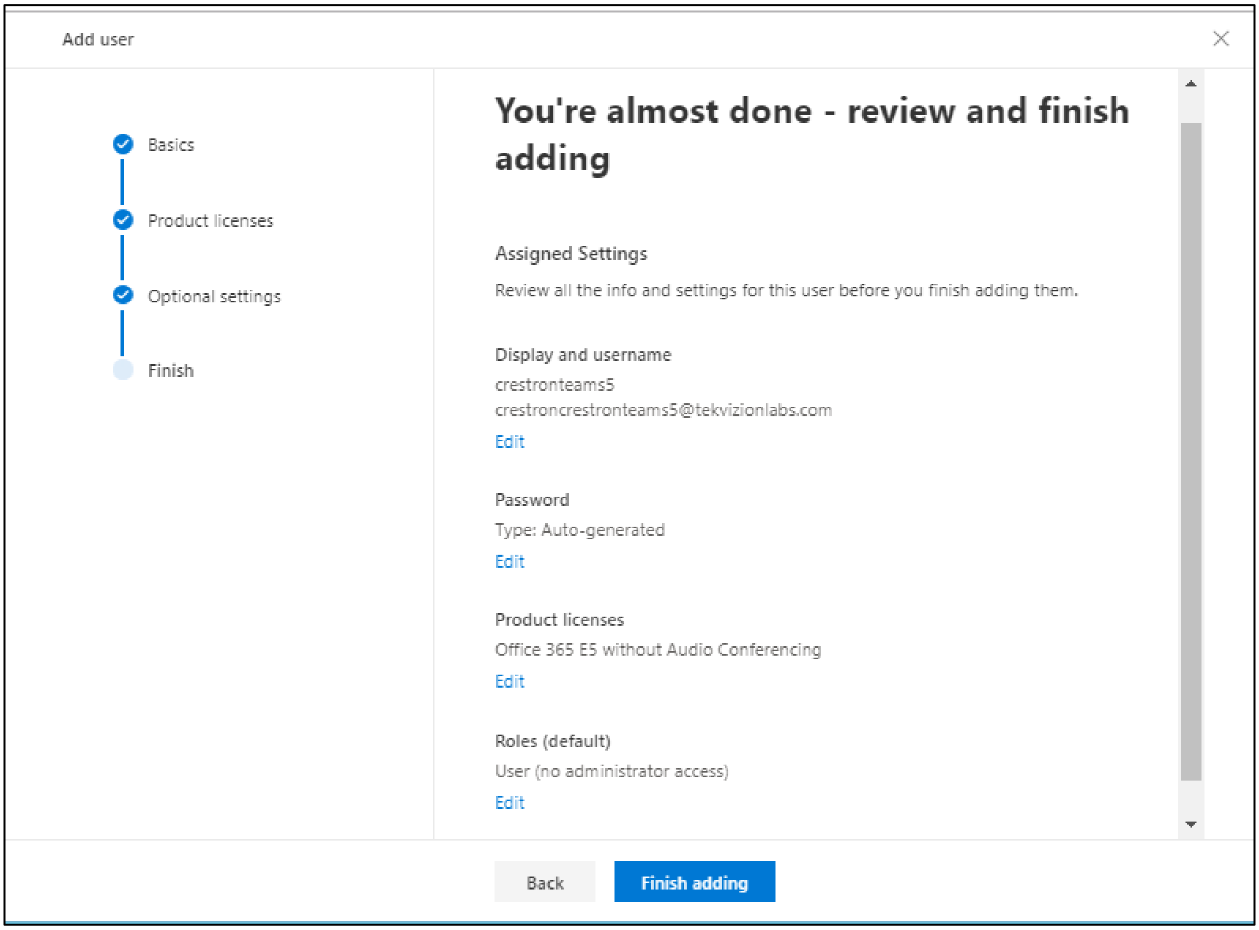


Figure 30: Teams User Creation – Contd.

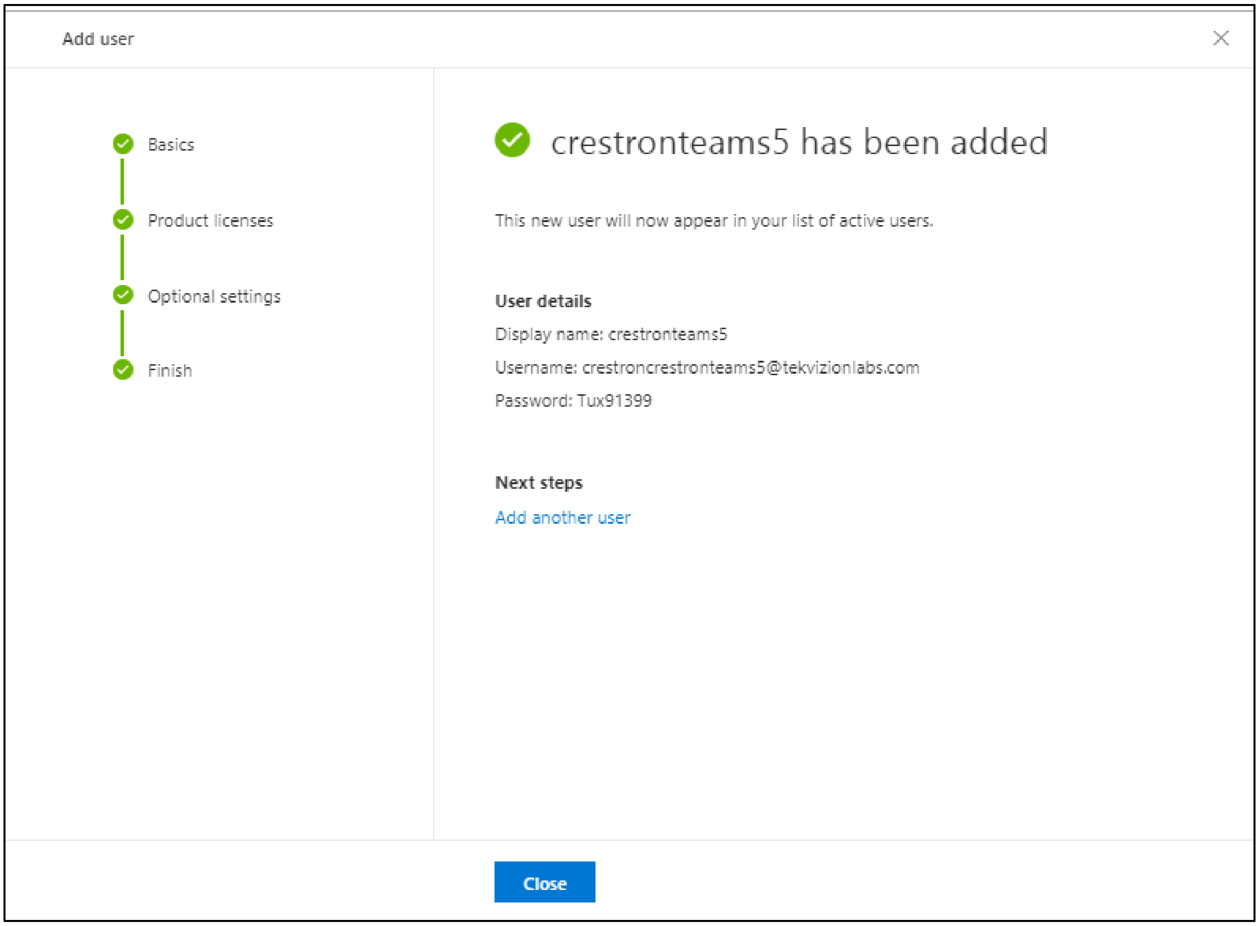


Figure 31: Teams User Creation – Contd.

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

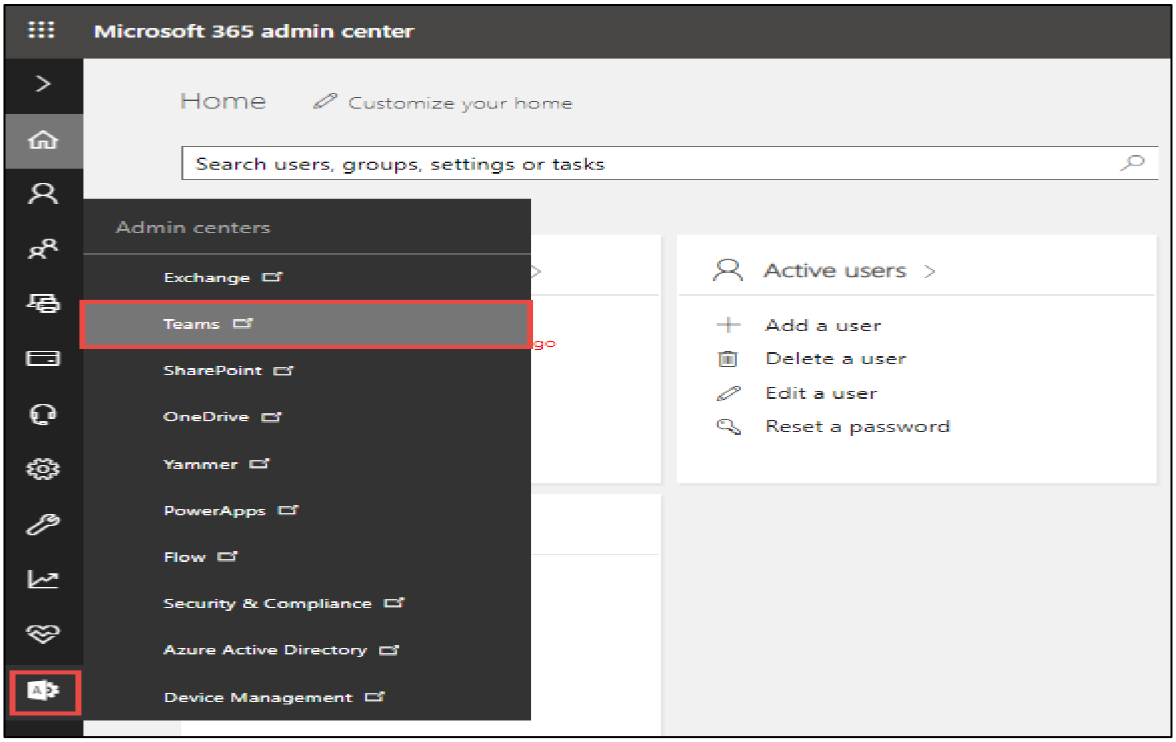


Figure 32: Microsoft O365 admin

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

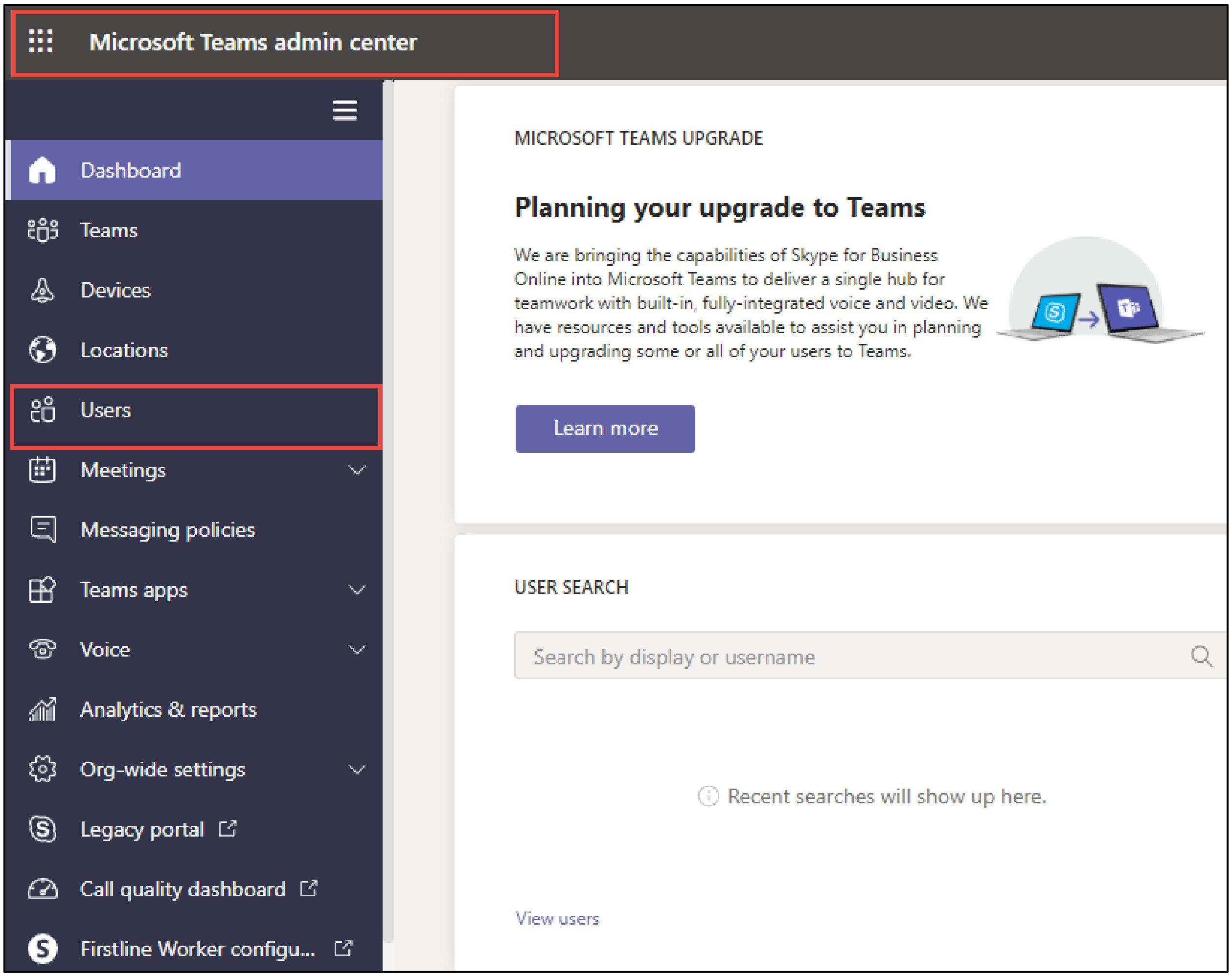


Figure 33: Microsoft O365 admin

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

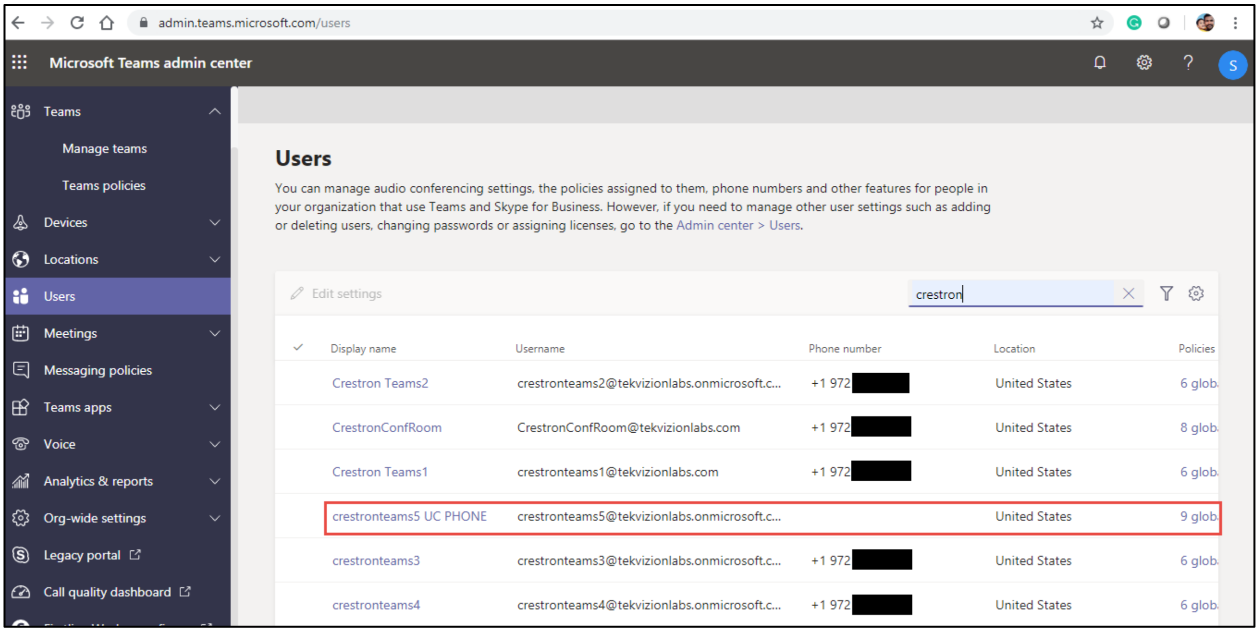


Figure 34: Microsoft O365 admin

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

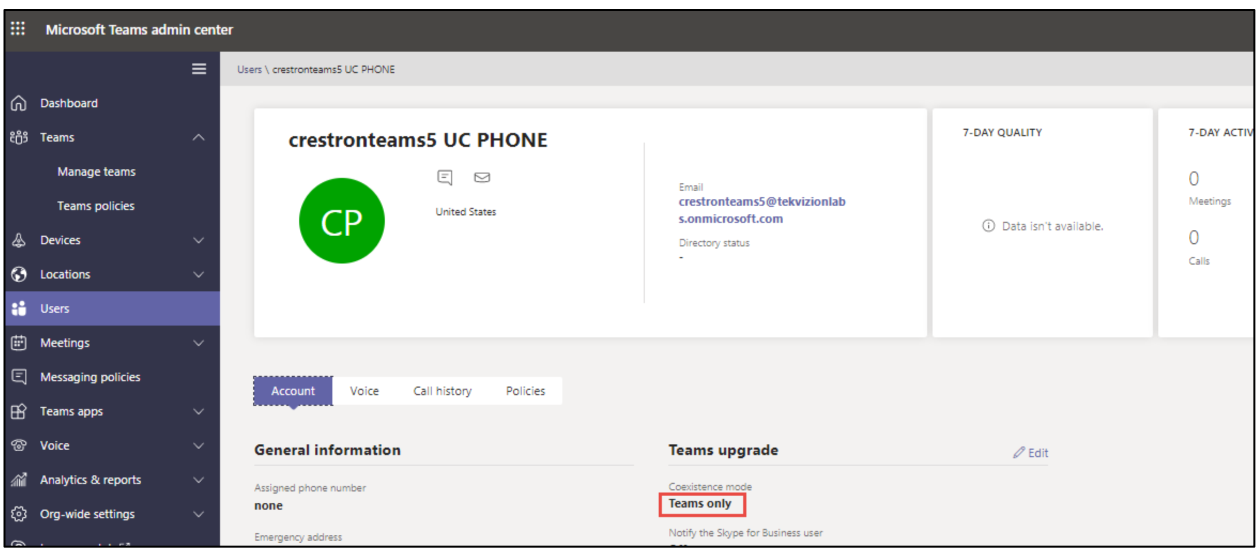


Figure 35: Teams User

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.

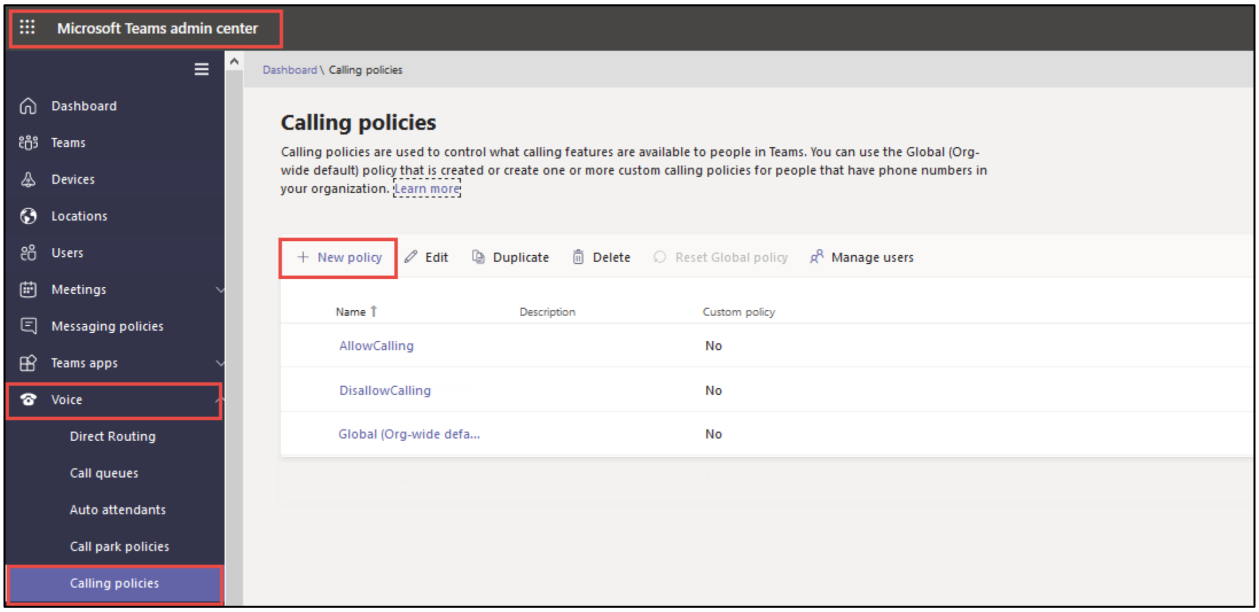


Figure 36 – Calling Policy

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

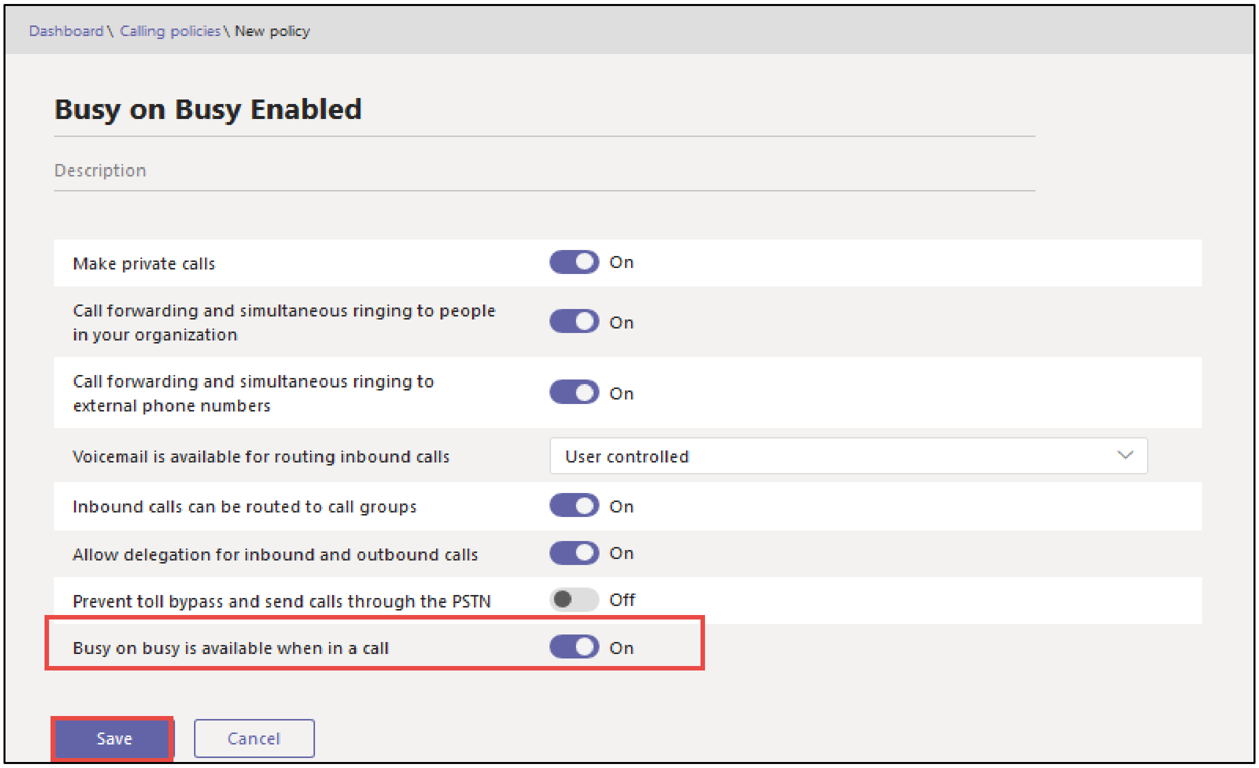


Figure 37 – Calling Policy

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 "crestronteams5@tekvizionlabs.com" –EnterpriseVocieEnabled $true –HostedVoicemail $true

Set-CsUser –identity "crestronteams5@tekvizionlabs.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 "crestronteams5" -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 crestronteams5 -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 "crestronteams5@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.

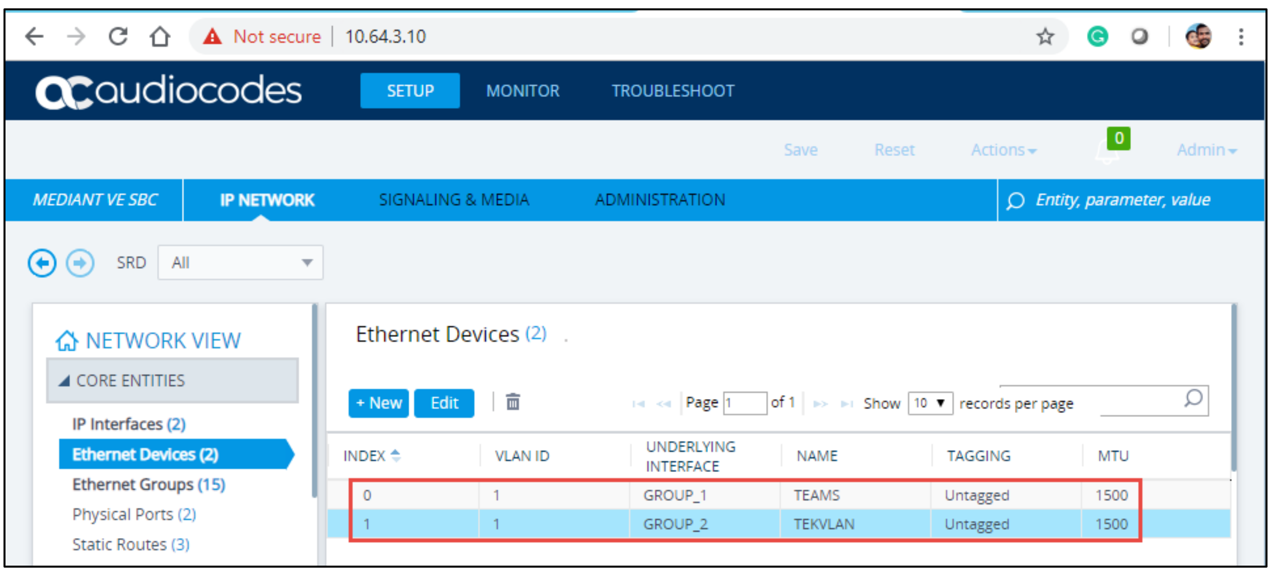


Figure 48 – Ethernet Devices

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:

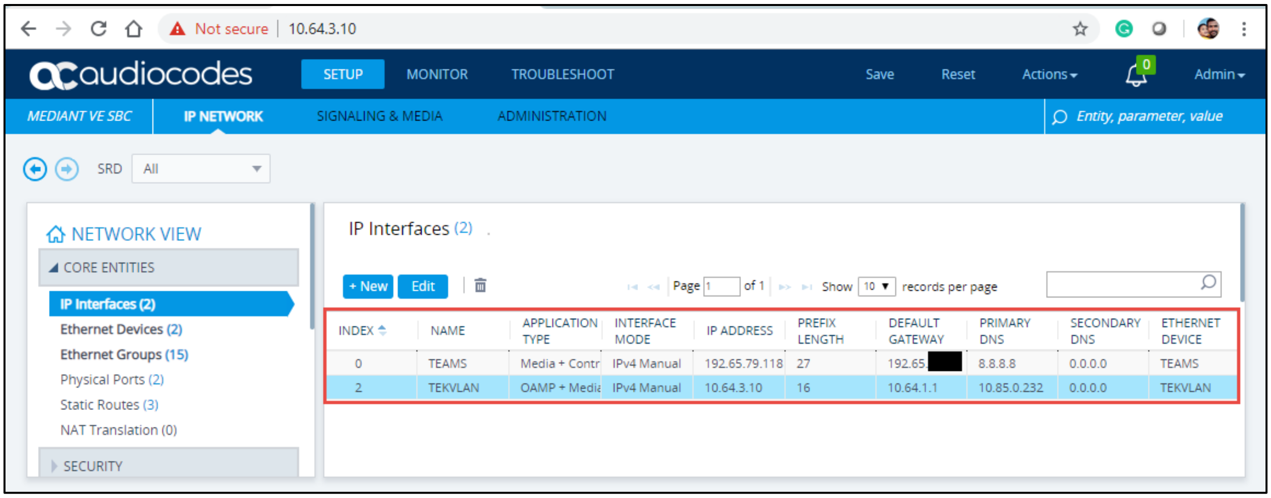


Figure 49 – IP interface Devices

IP interface TEAMS

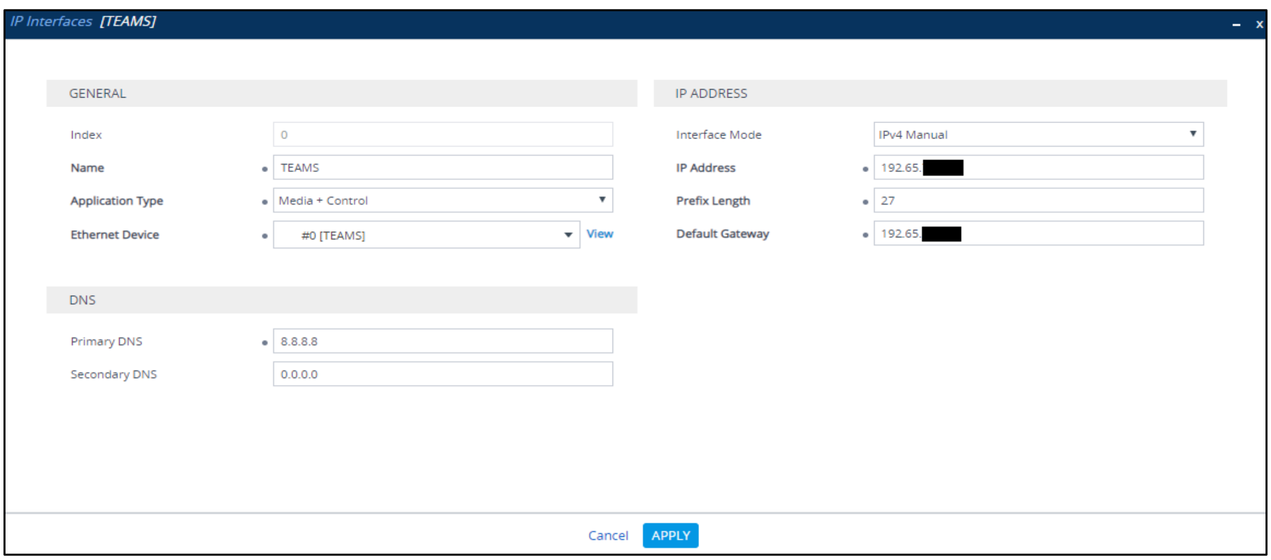


Figure 50 – IP interface Devices

IP Interfaces – TEKVLAN

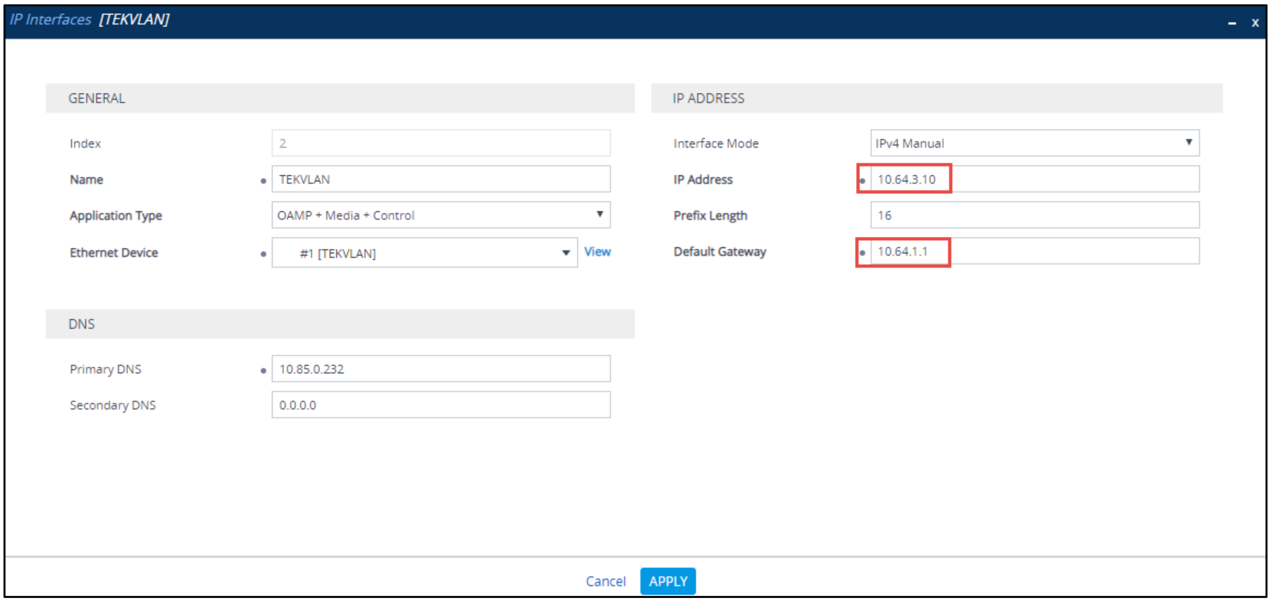


Figure 51 – IP interface Devices

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

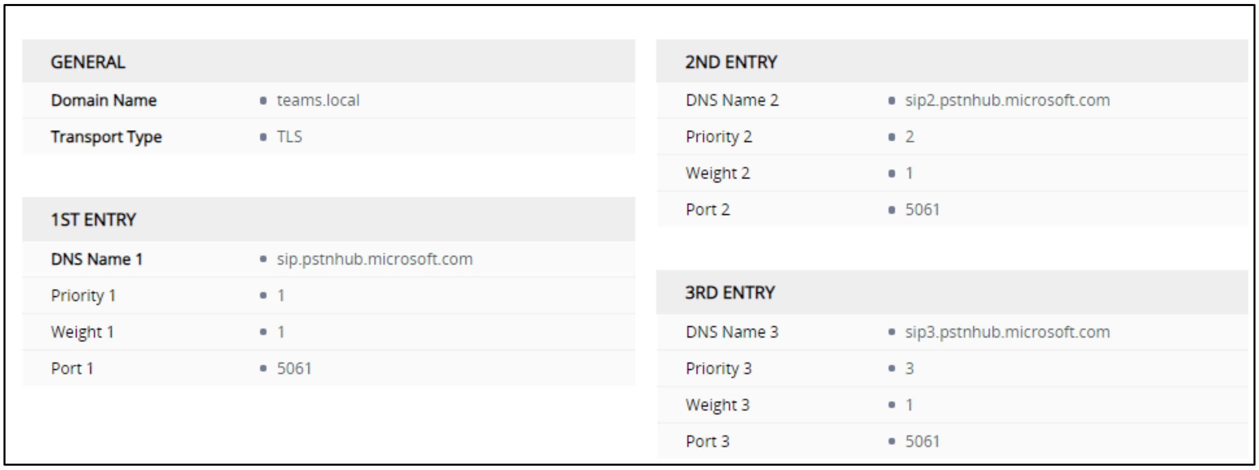


Figure 52 – DNS SRV Records

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

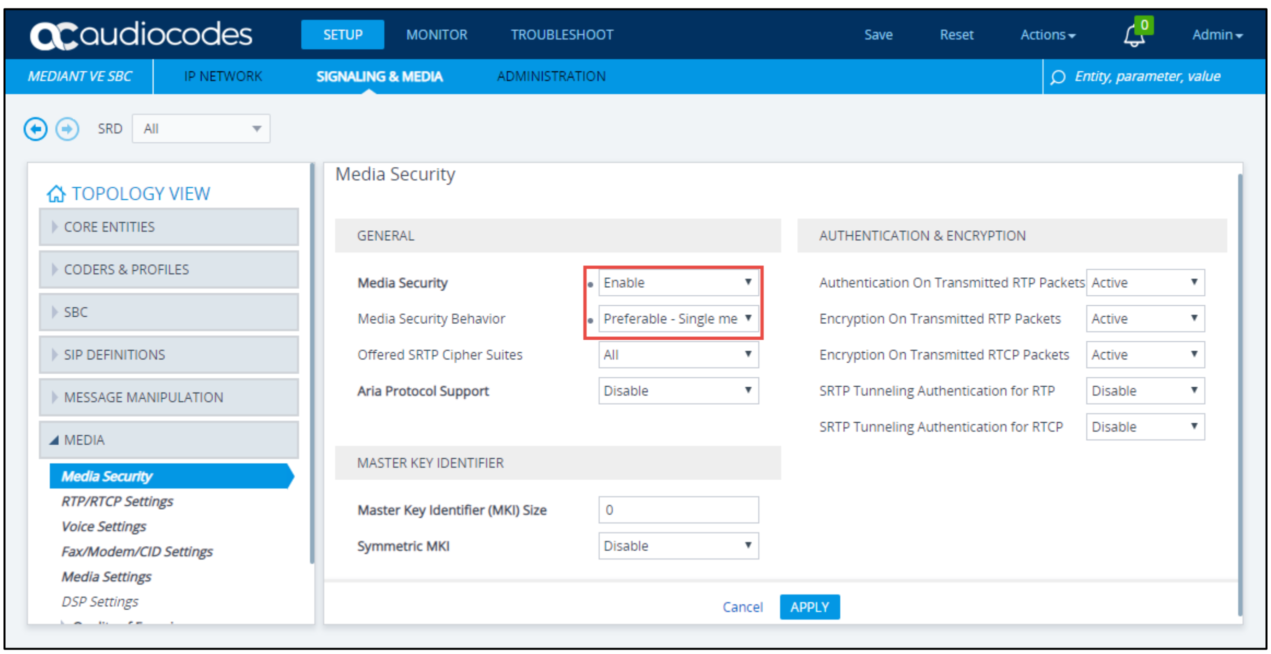


Figure 53 – Media Security

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.

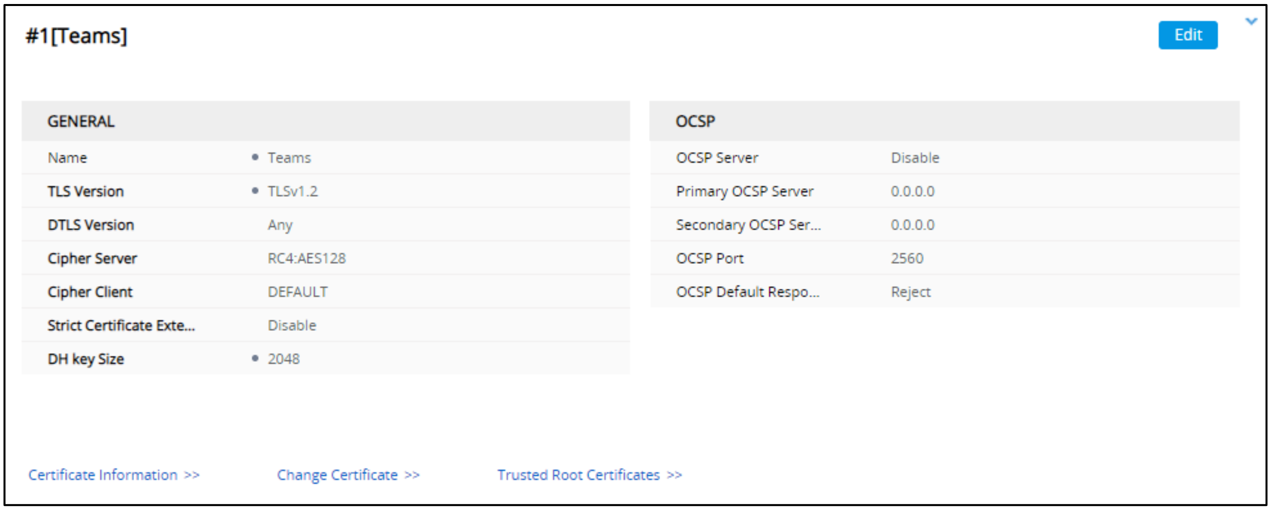


Figure 54 – Teams TLS

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.**

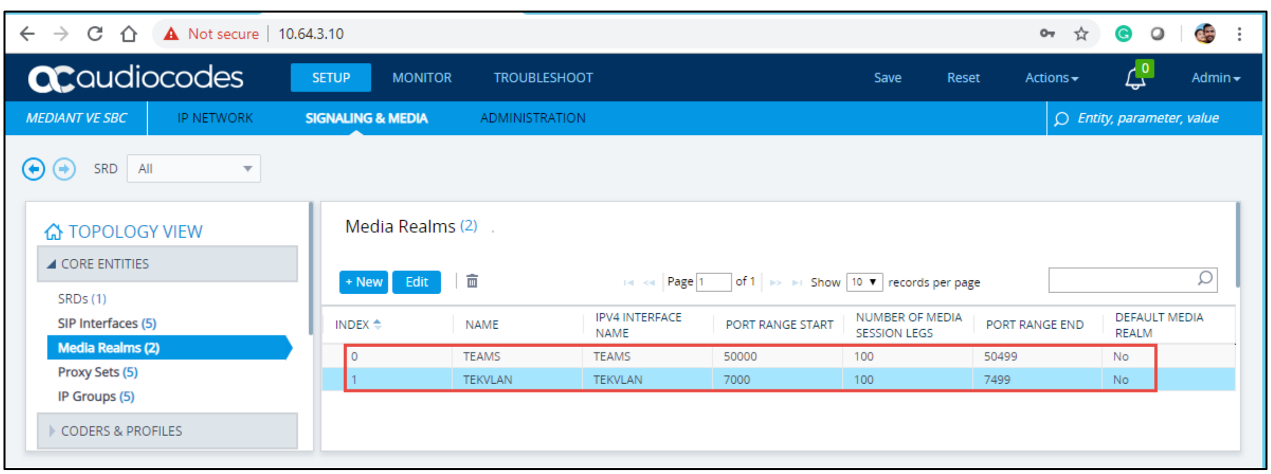


Figure 55 – Media Realms

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

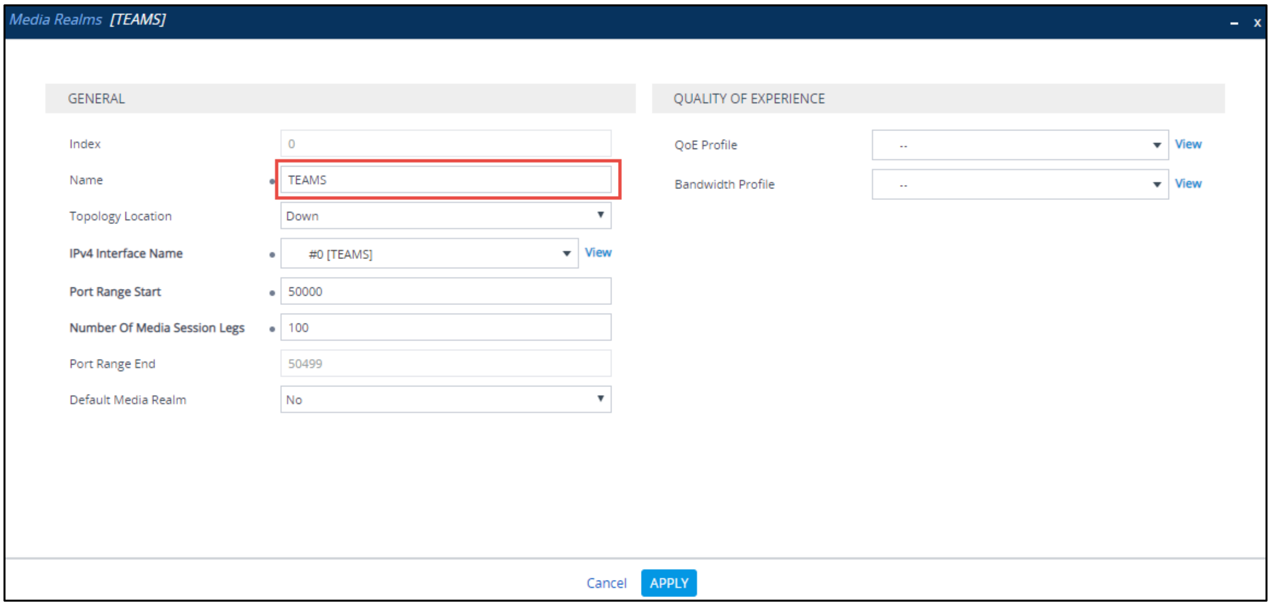


Figure 56 – Teams

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

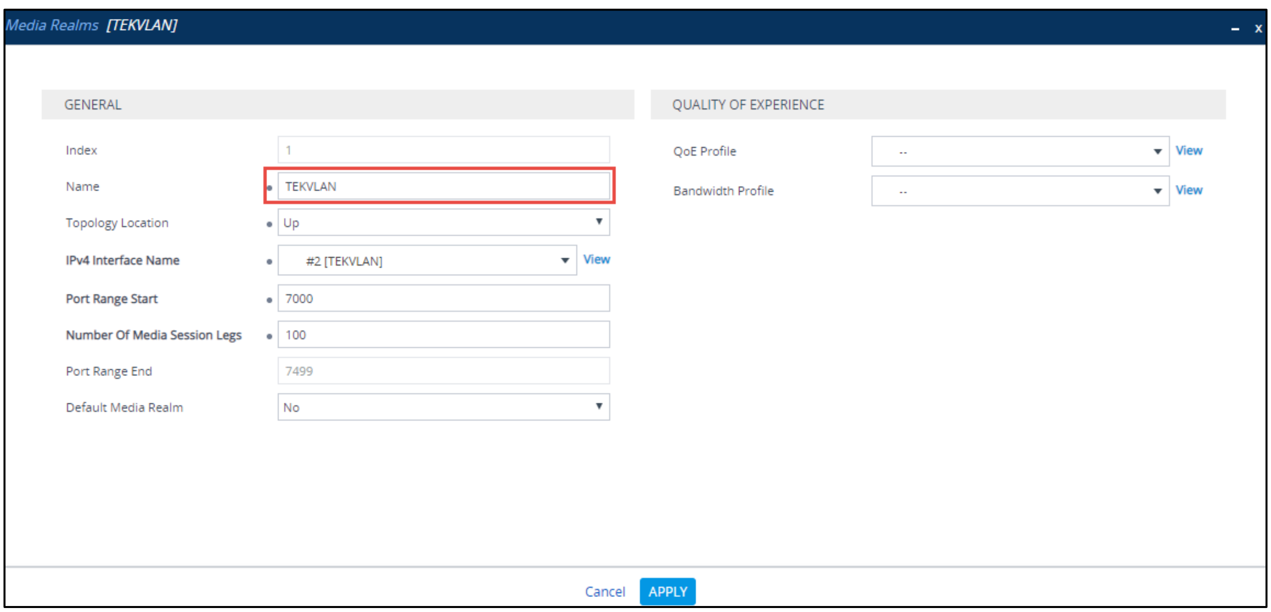


Figure 57 – LAN LAB

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.

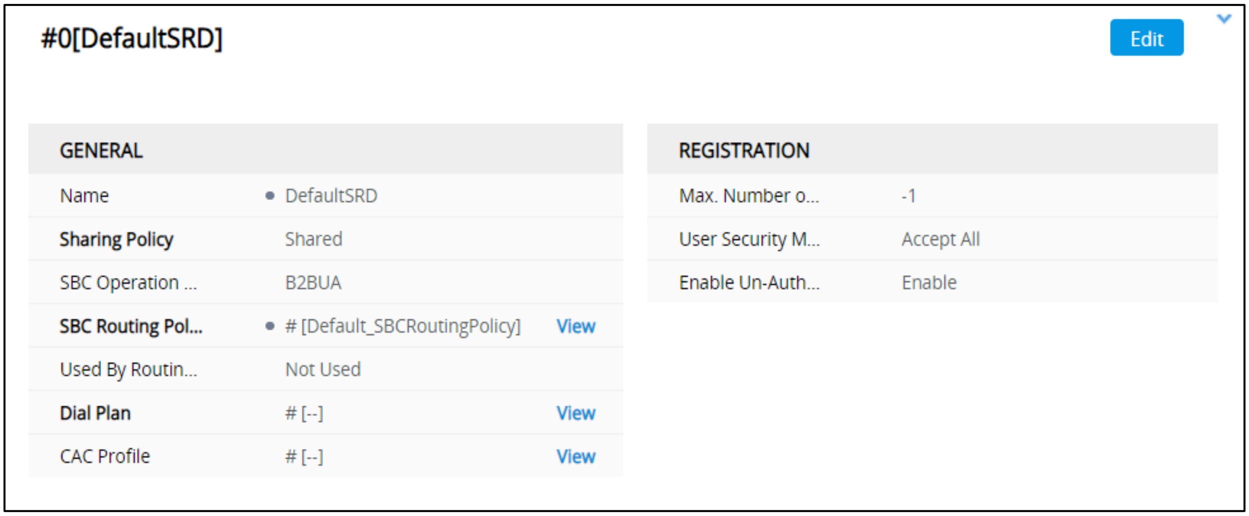


Figure 58 – Default SRD

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.

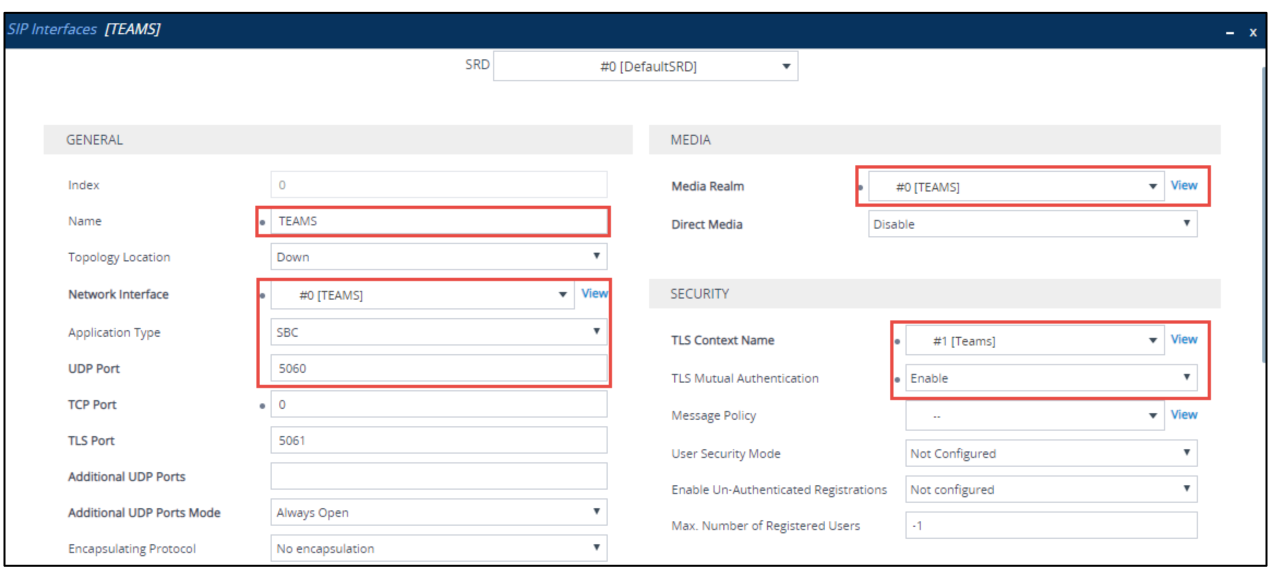


Figure 59 – Teams

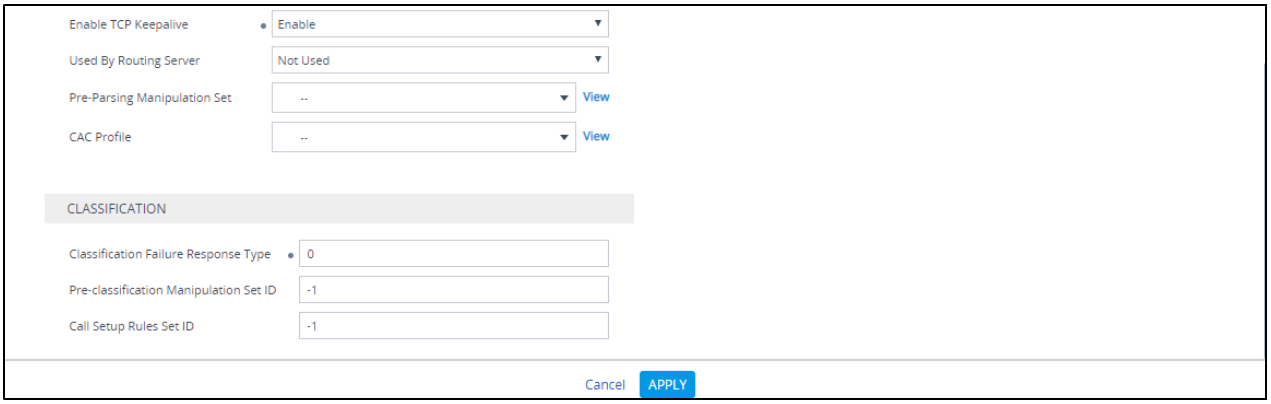


Figure 60 – Teams

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

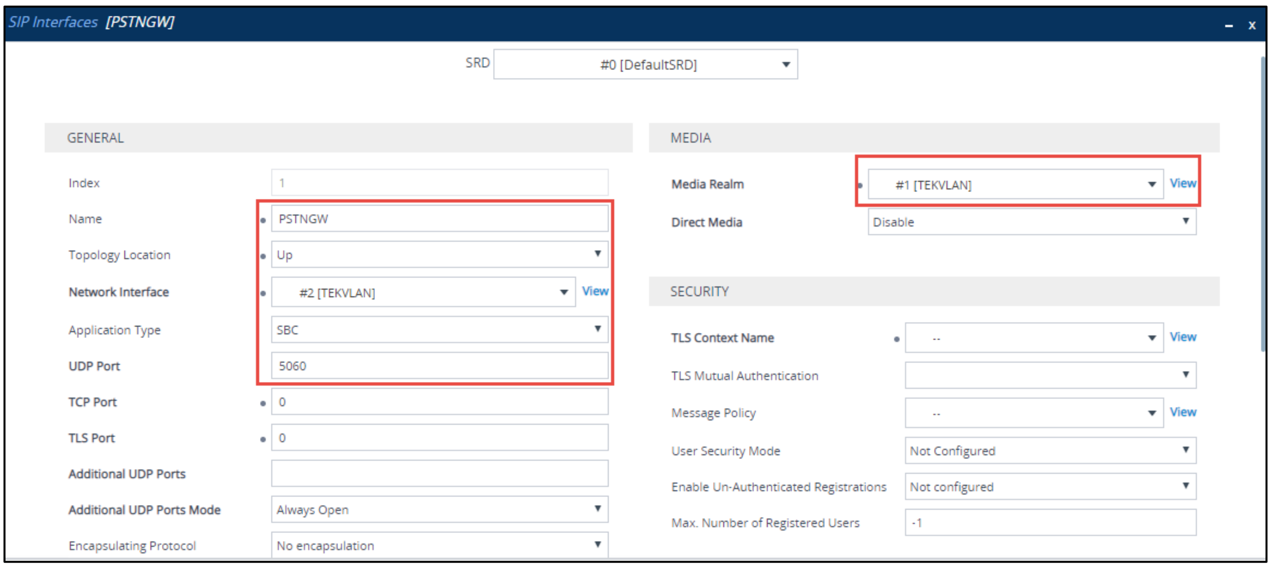


Figure 61 – PSTN

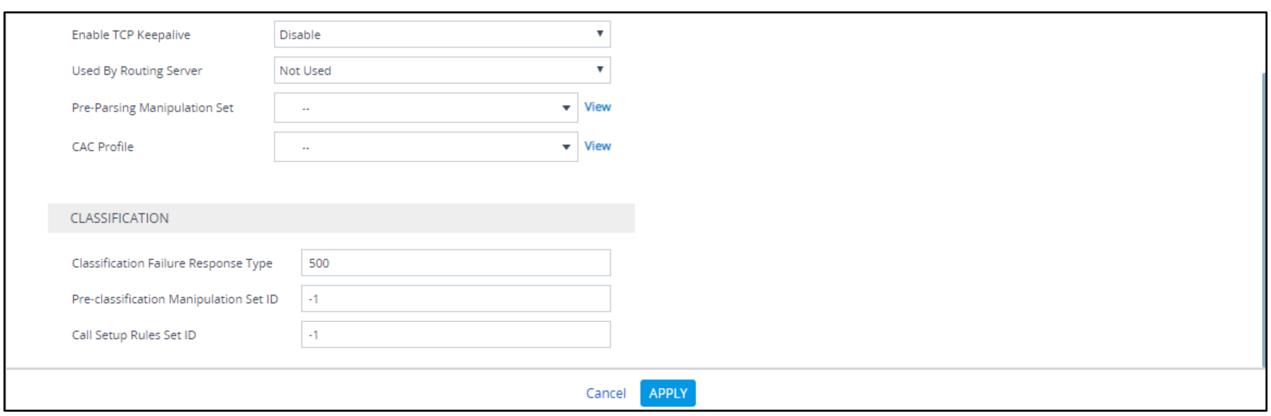


Figure 62 – PSTN

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

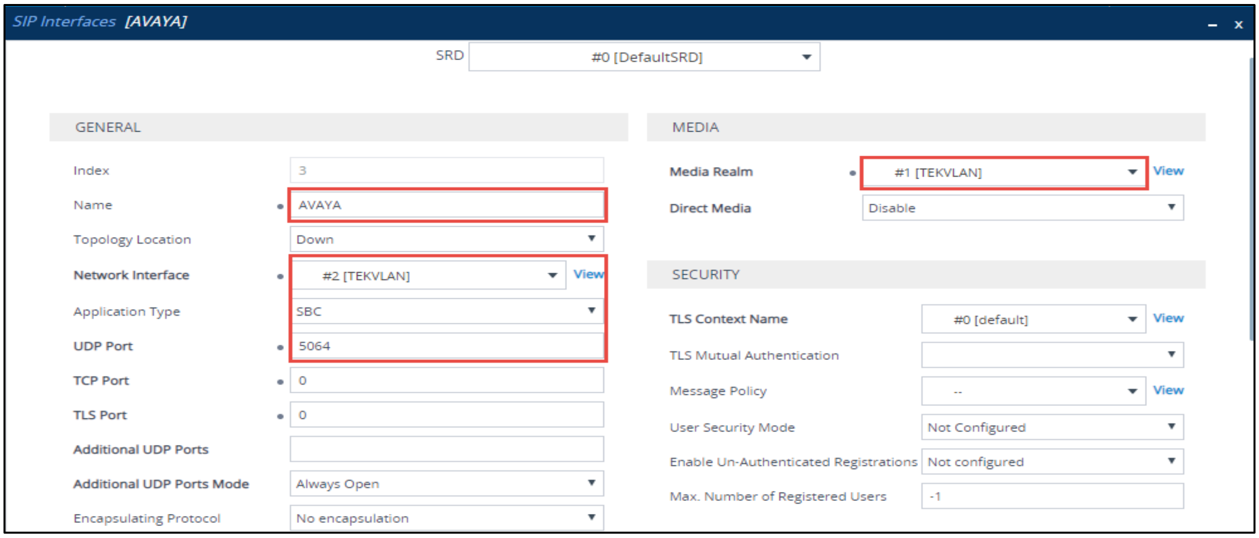


Figure 63 – Avaya

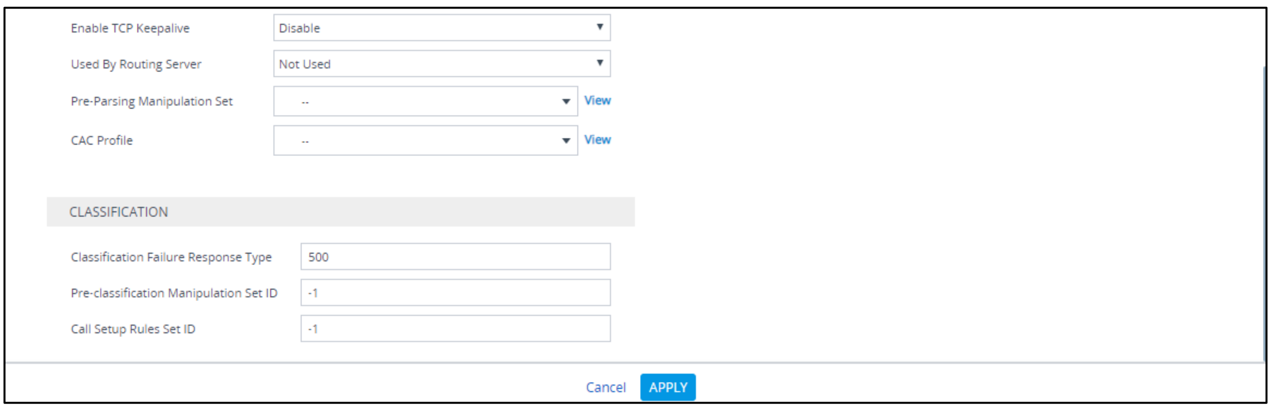


Figure 64 – Avaya

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.

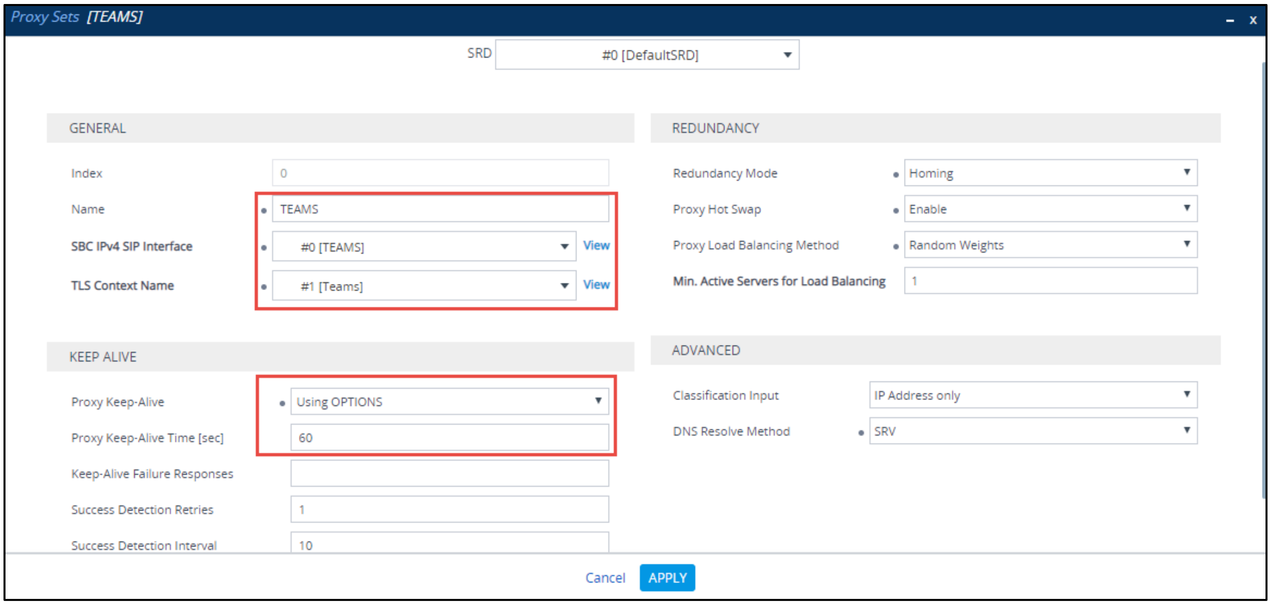


Figure 65 – Teams

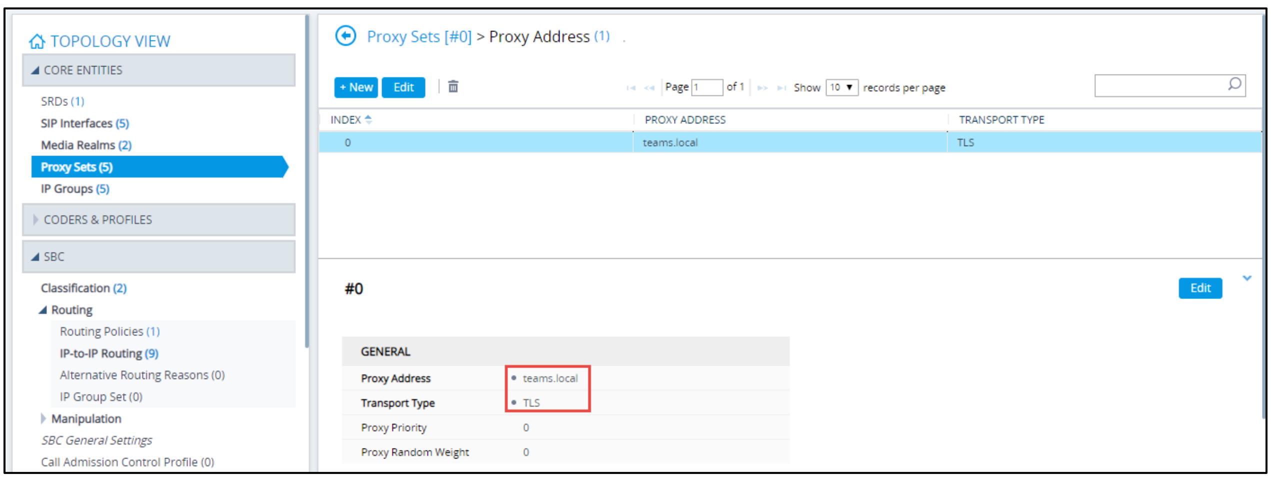


Figure 66 – Teams

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

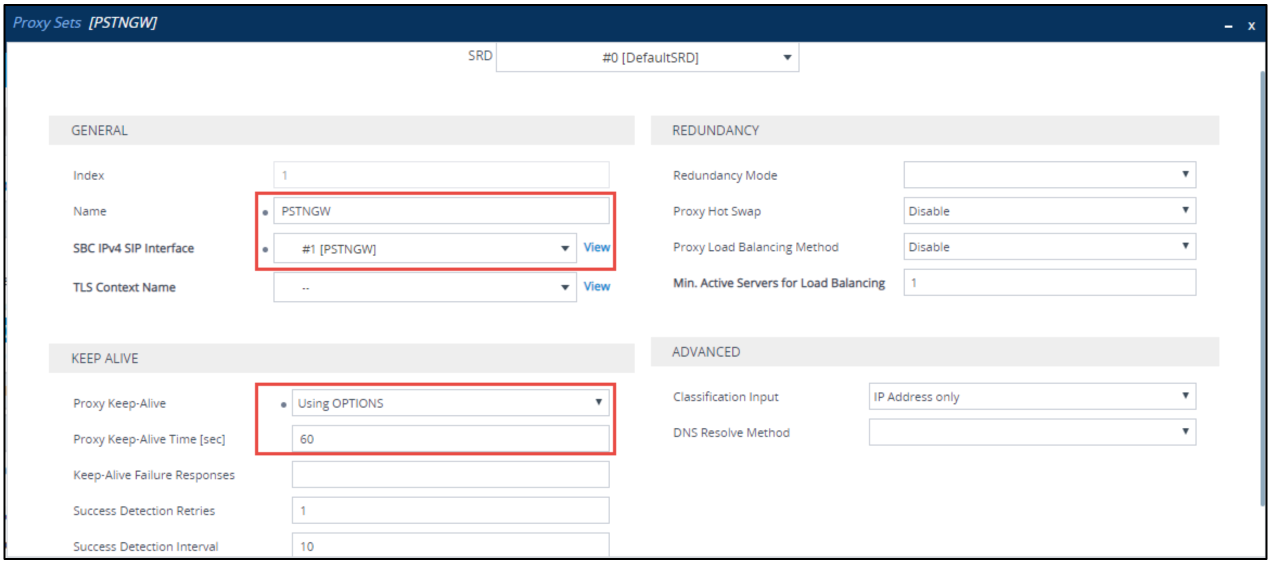


Figure 67 – PSTN Gateway

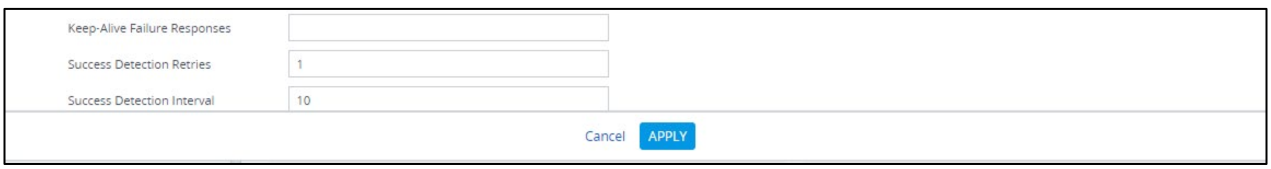


Figure 68 – PSTN Gateway

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

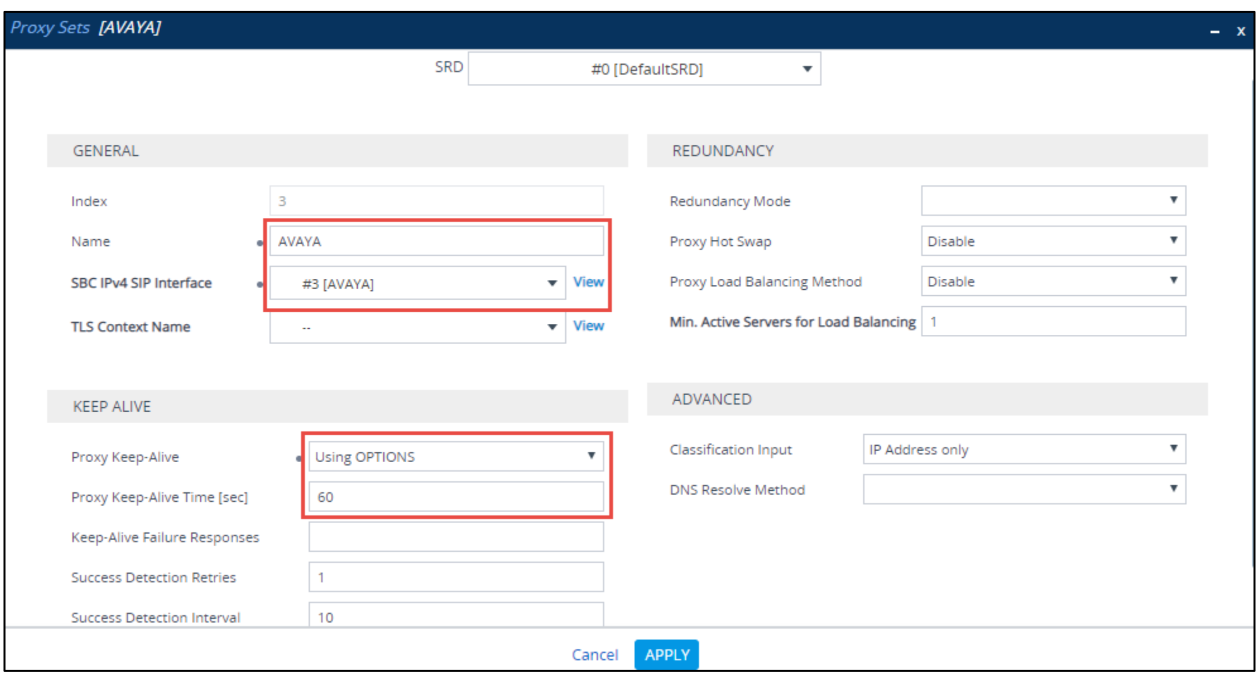


Figure 69 – Avaya

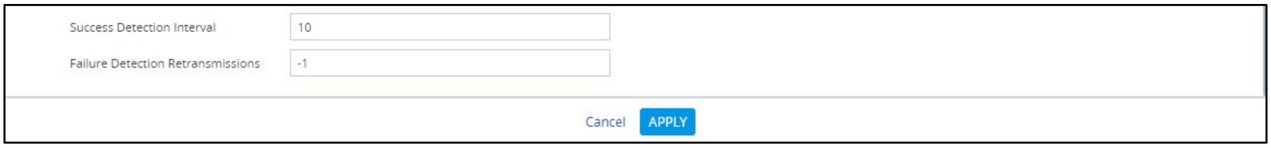


Figure 70 – Avaya

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

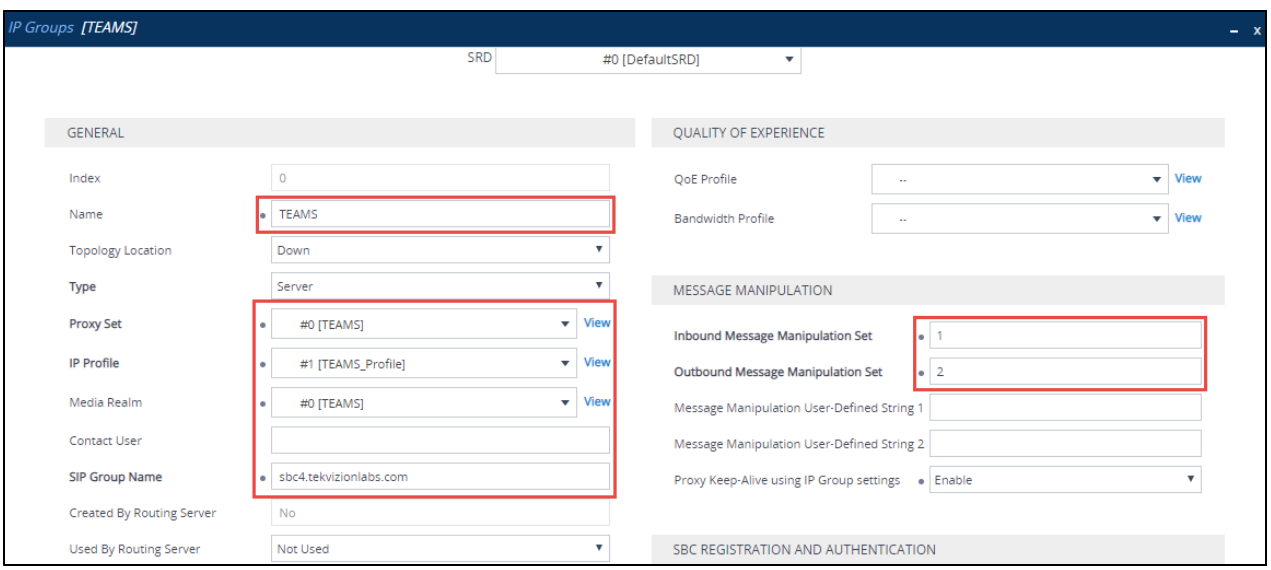


Figure 71 – IP Group – Teams – Contd.

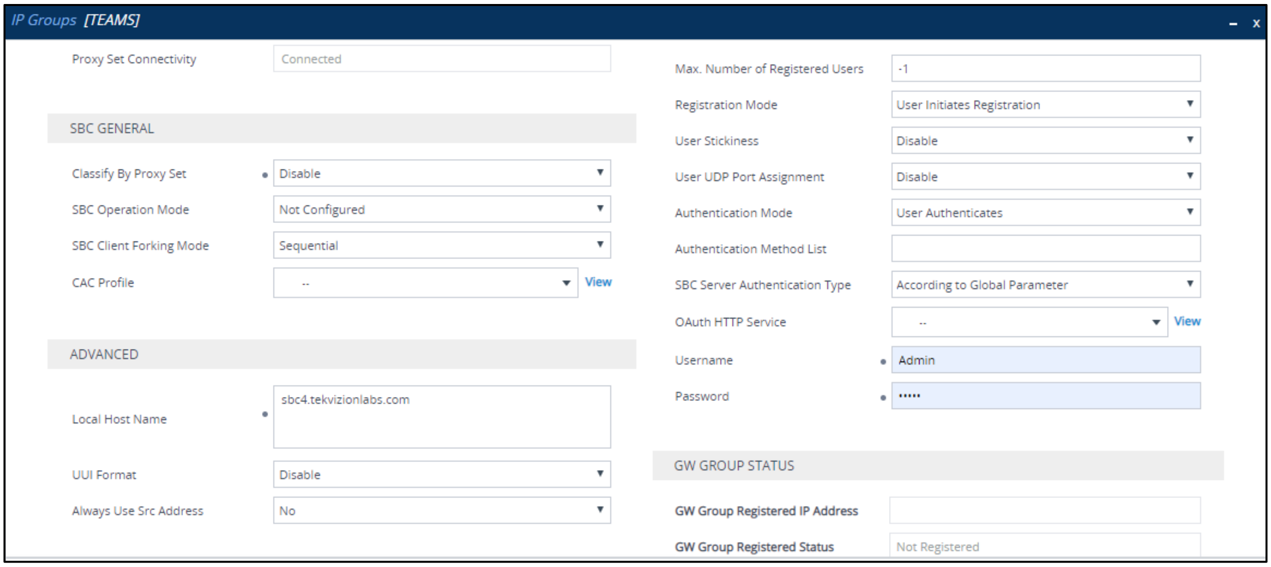


Figure 72 – IP Group – Teams – Contd.

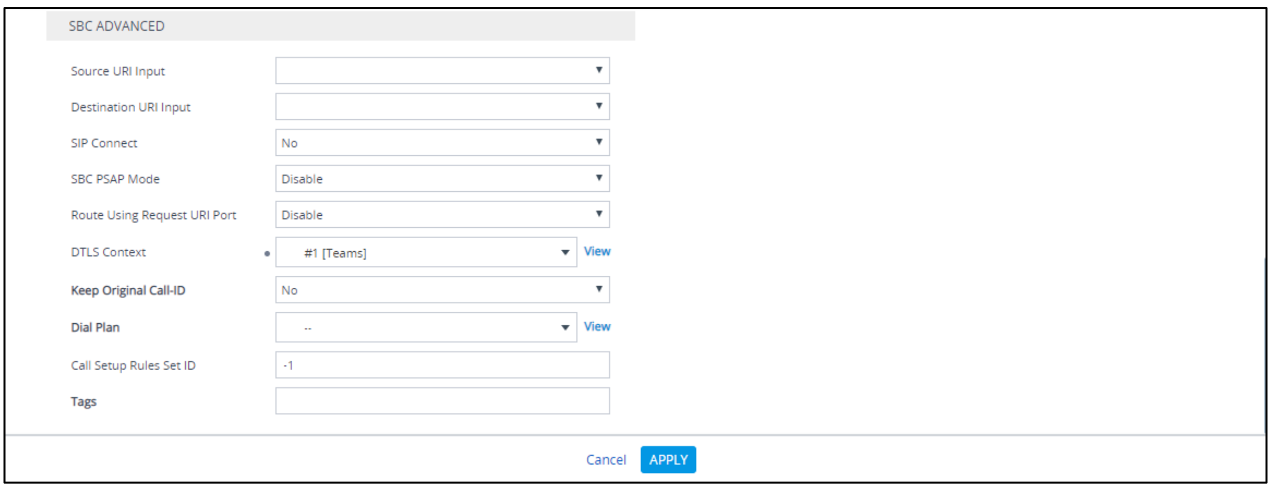


Figure 73 – IP Group – Teams

Configure an IP Group for PSTN Gateway as shown below

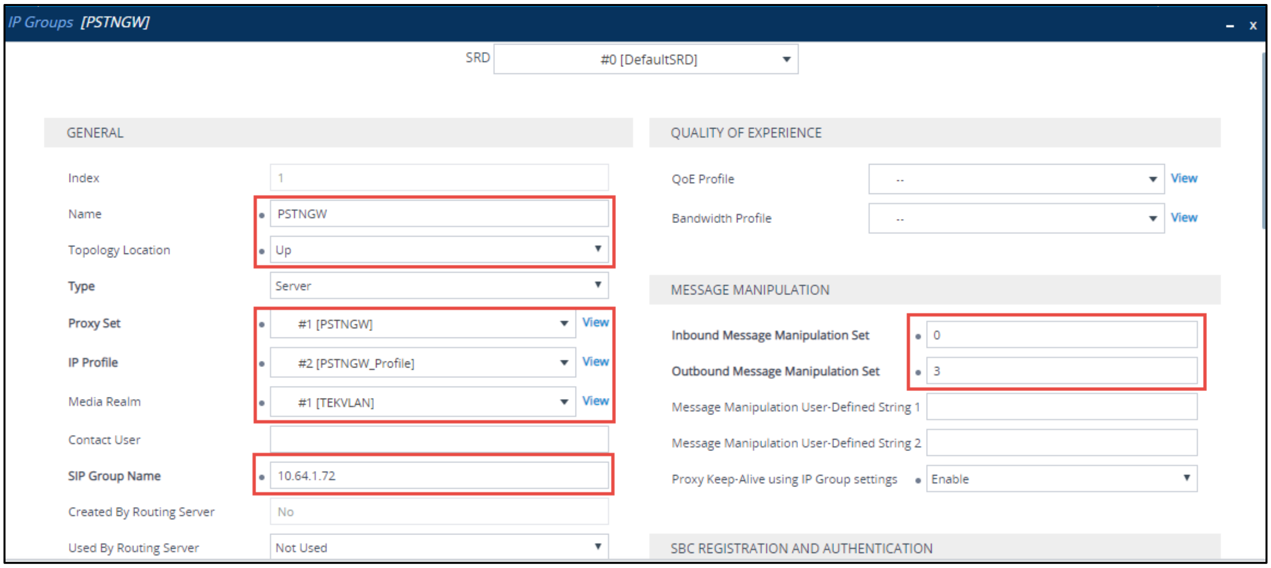


Figure 74 – IP Group – PSTN – Contd.

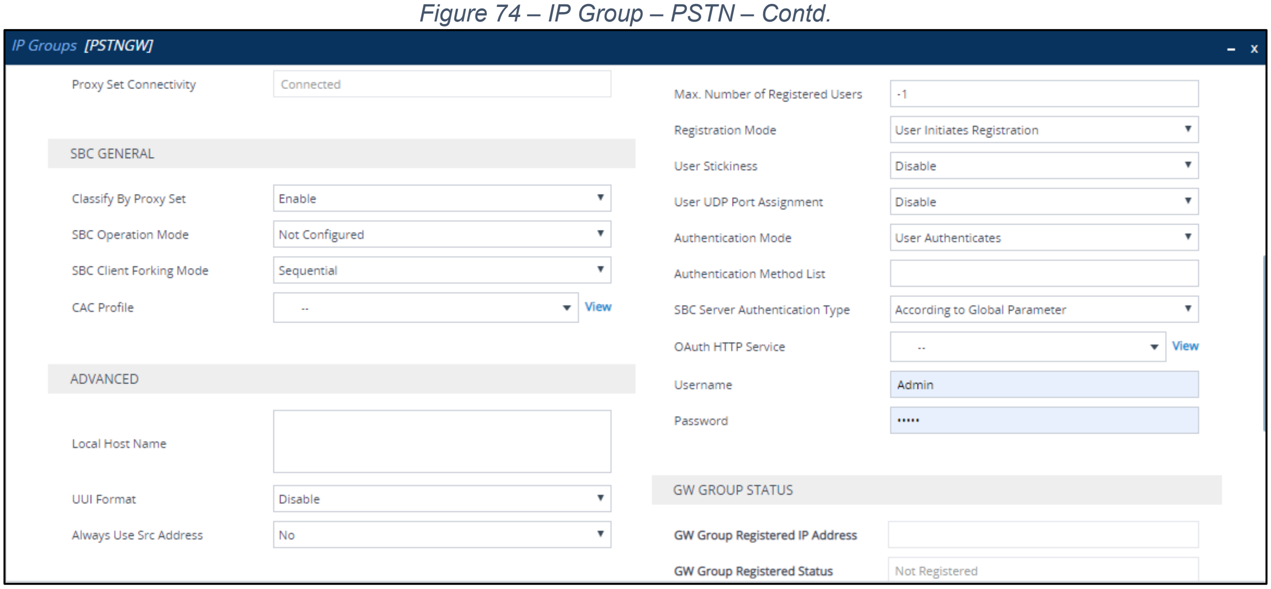


Figure 75 – IP Group – PSTN – Contd.

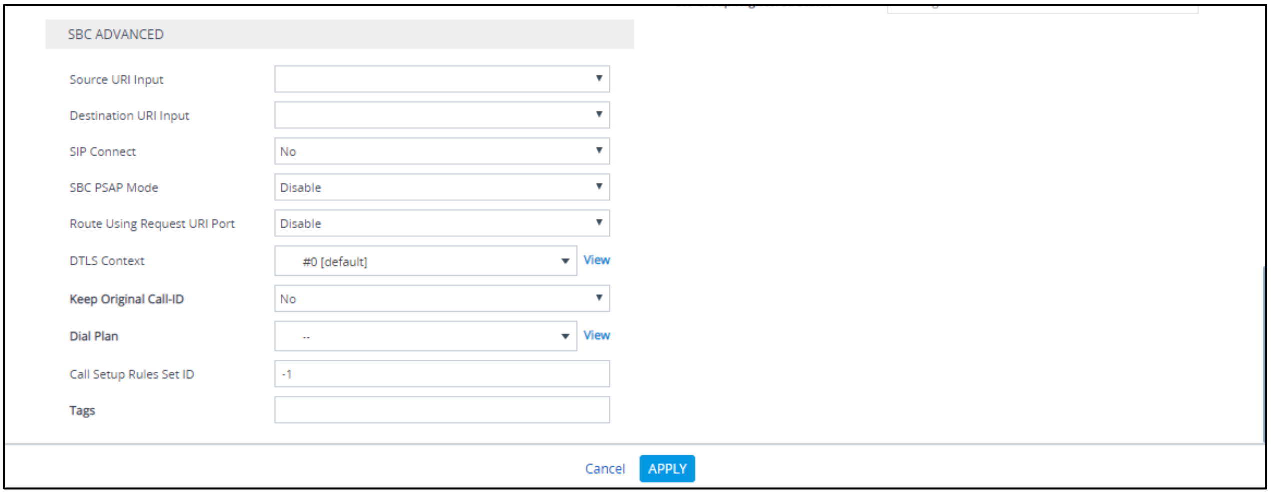


Figure 76 – IP Group

Configure an IP Group for Avaya SBCE as shown below

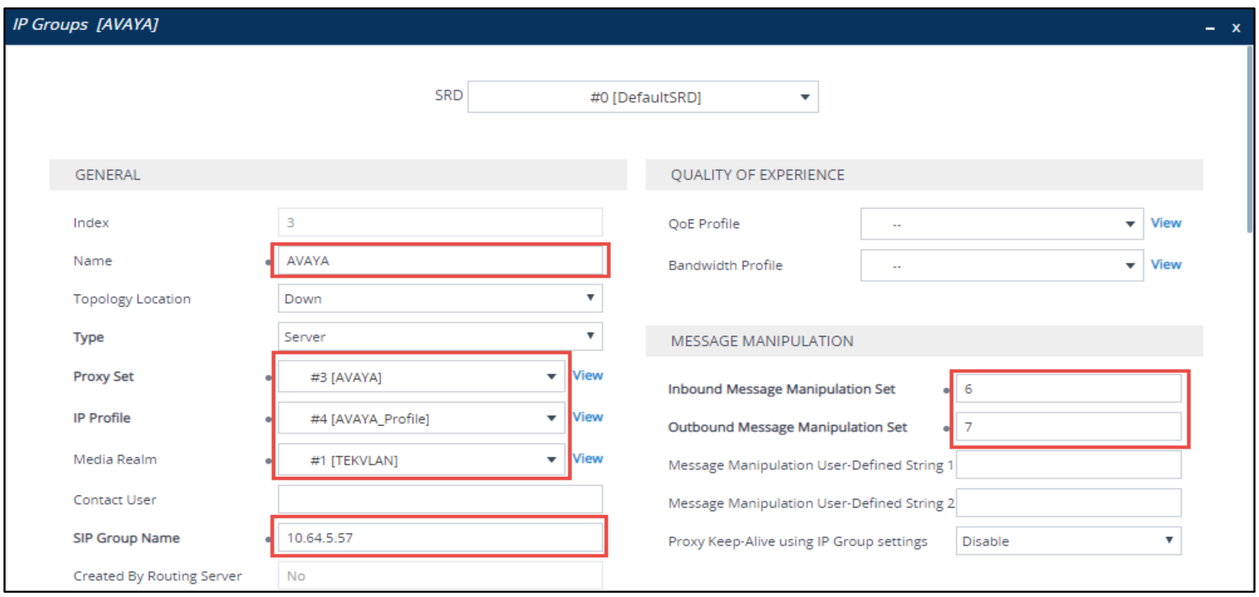


Figure 77 – IP Group – Avaya

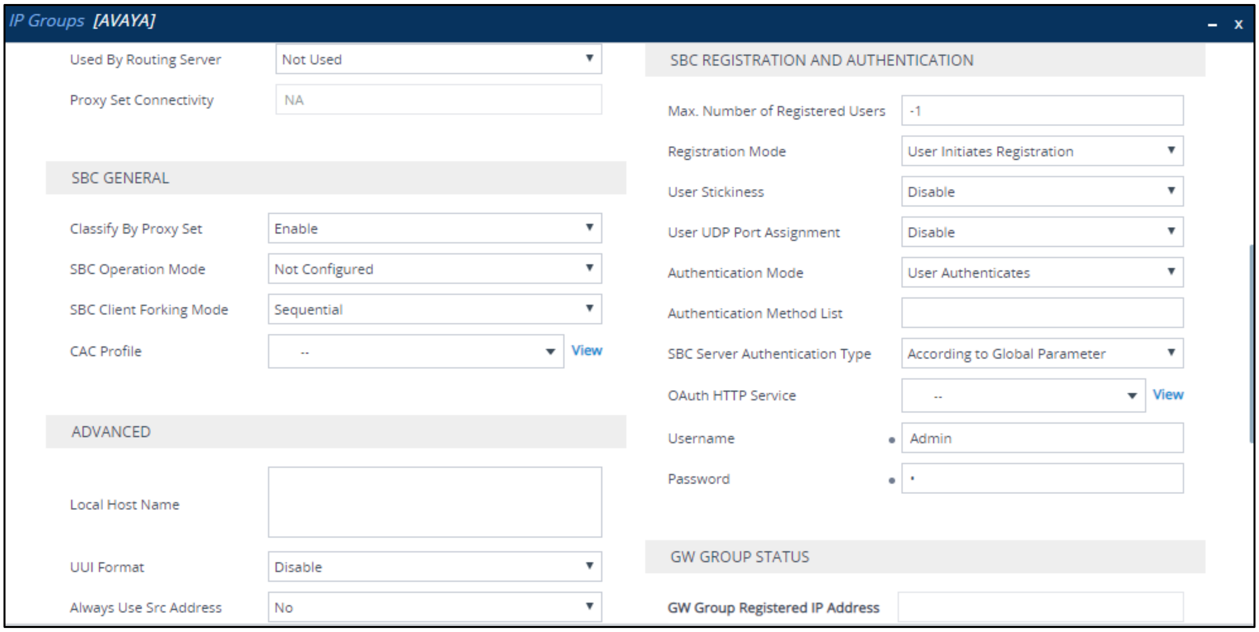


Figure 78 – IP Group – Avaya – Contd.

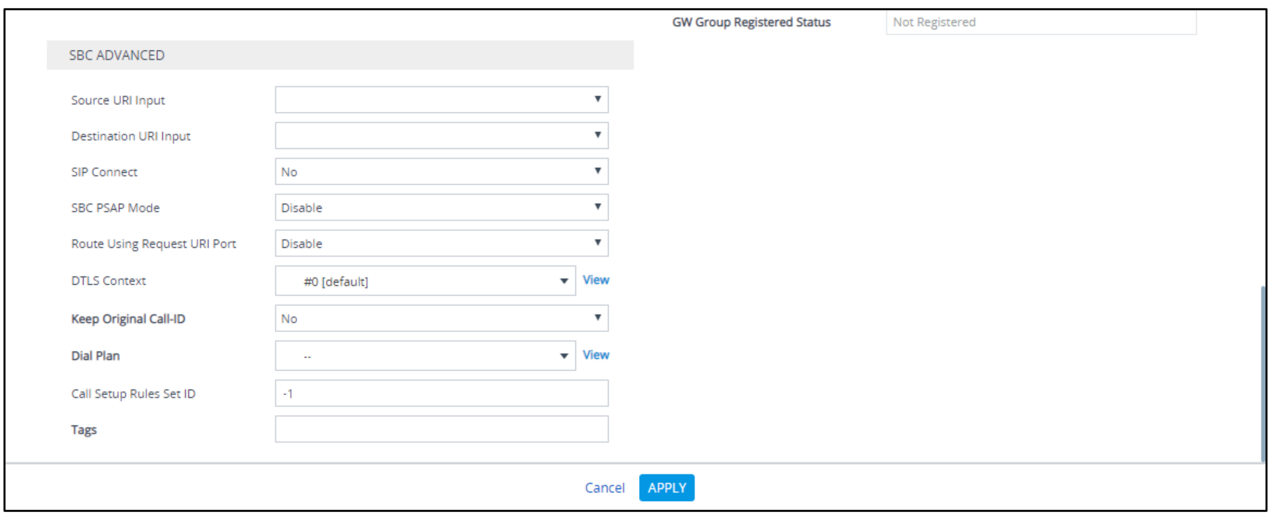


Figure 79 – IP Group – Avaya – Contd

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.

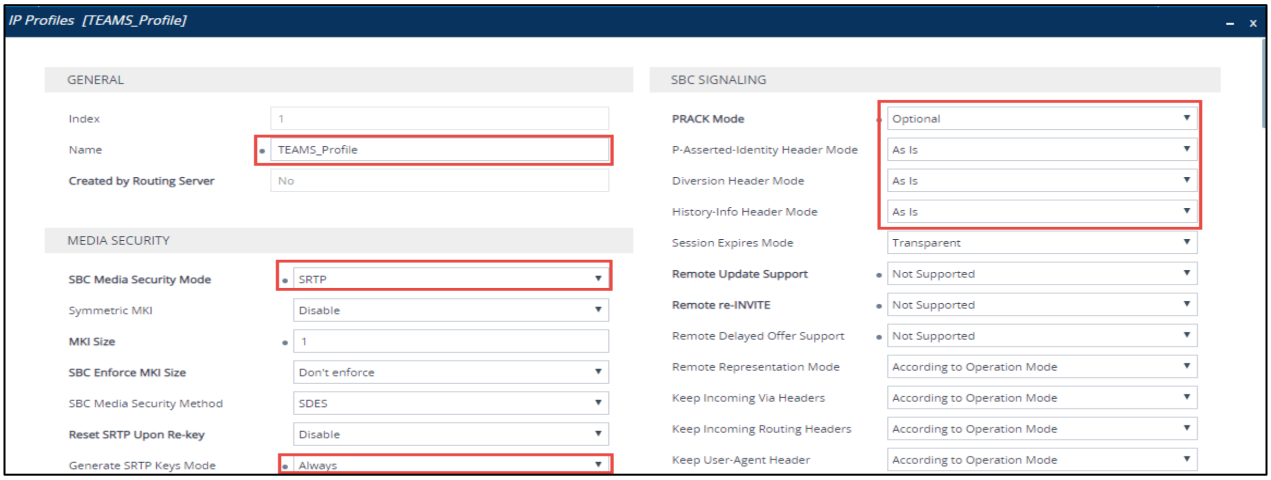


Figure 80 – IP Profile – Teams – Contd.

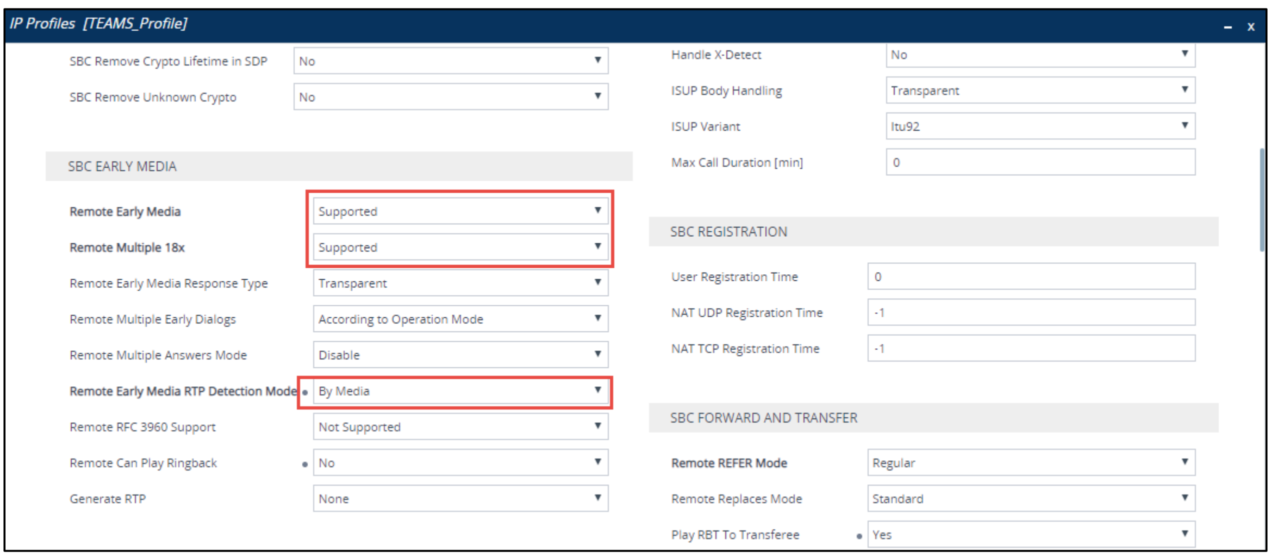


Figure 81 – IP Profile – Teams – Contd.

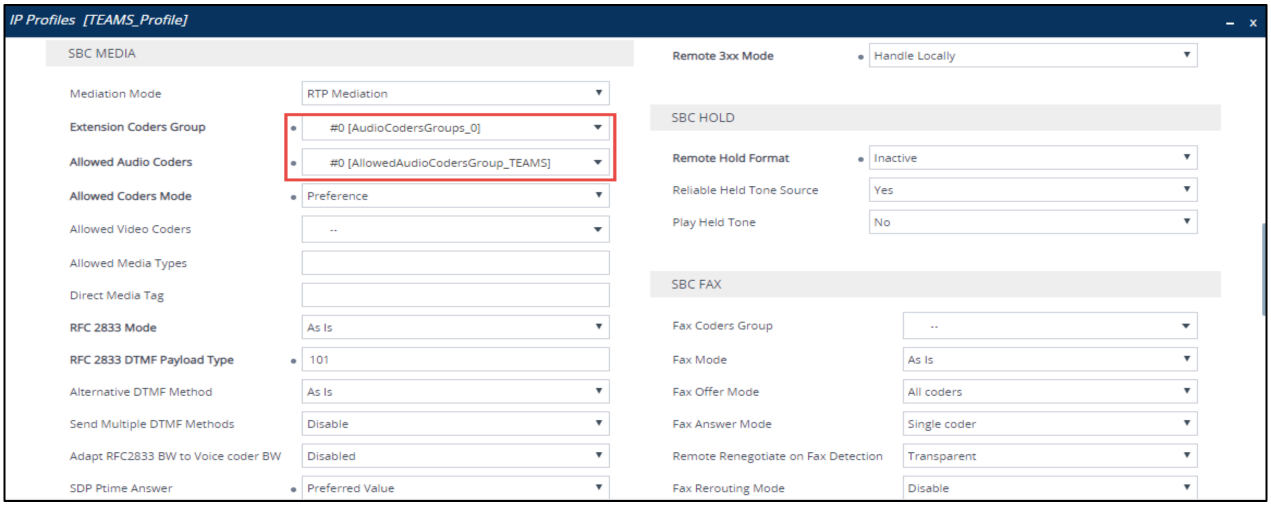


Figure 82 – IP Profile – Teams – Contd.

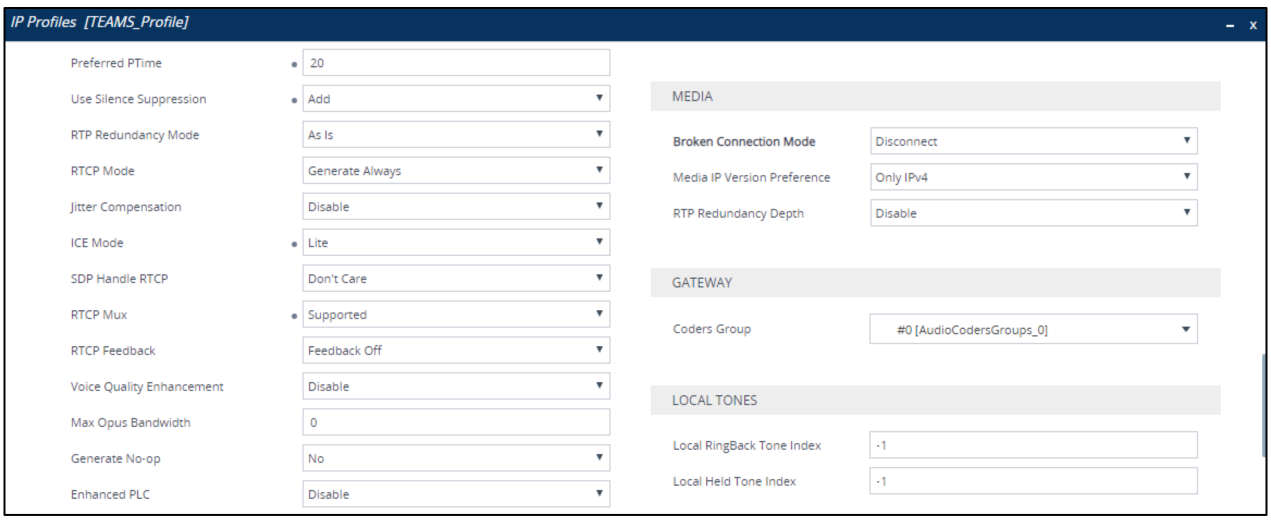


Figure 83 – IP Profile – Teams – Contd.

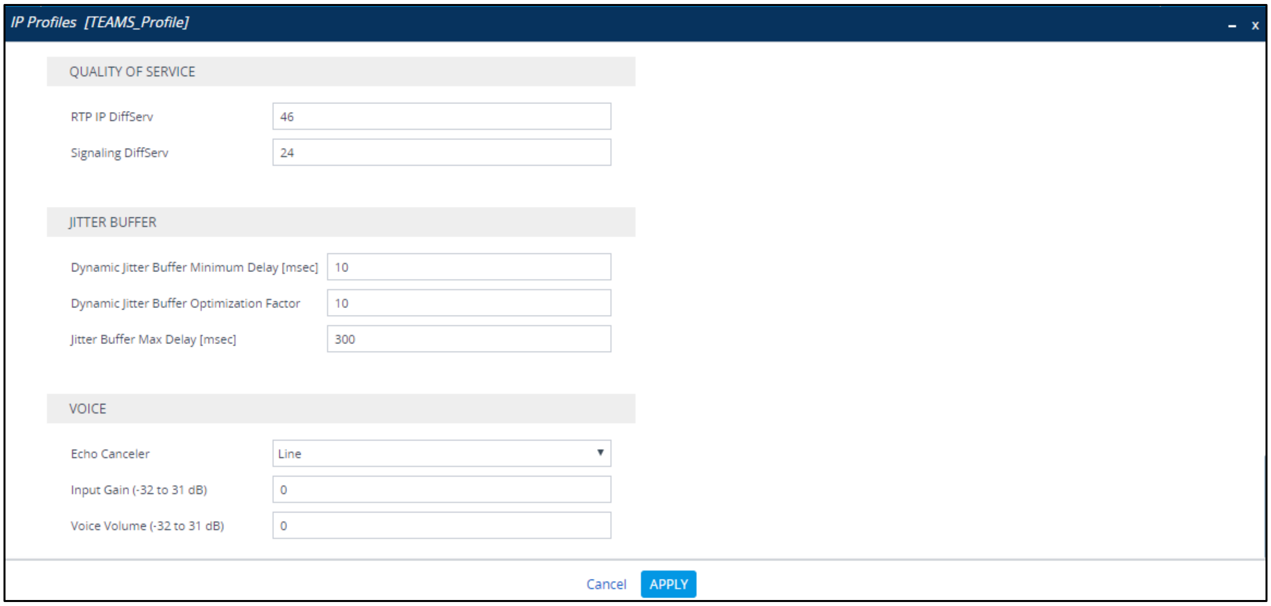


Figure 84 – IP Profile – Teams – Contd.

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

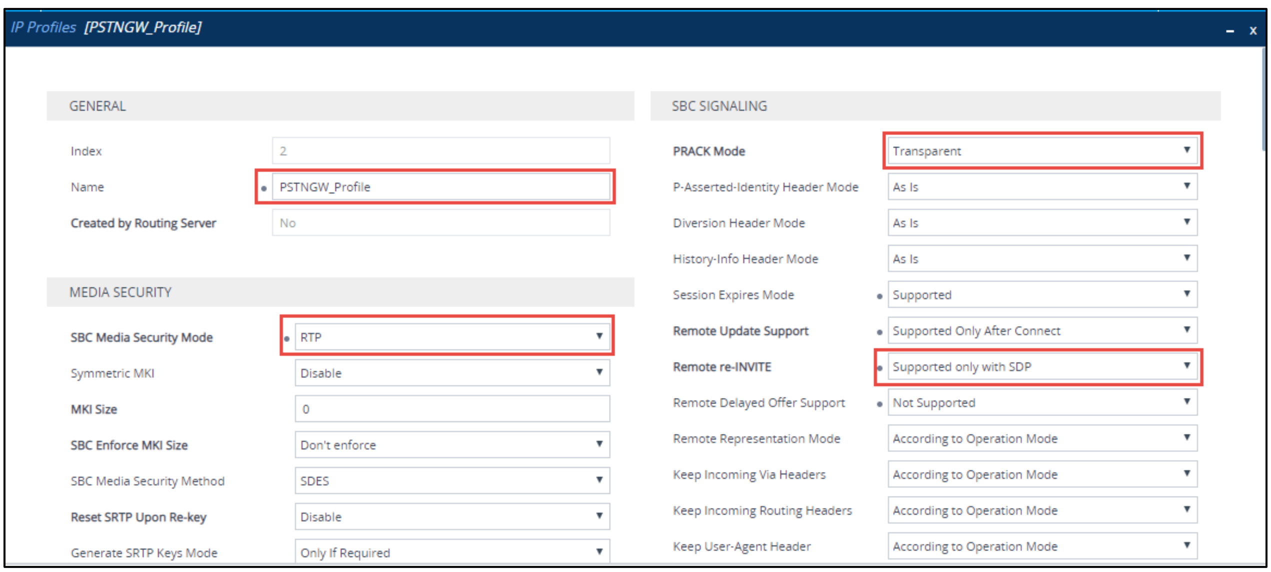


Figure 85 – IP Profile – PSTN Gateway – Contd.

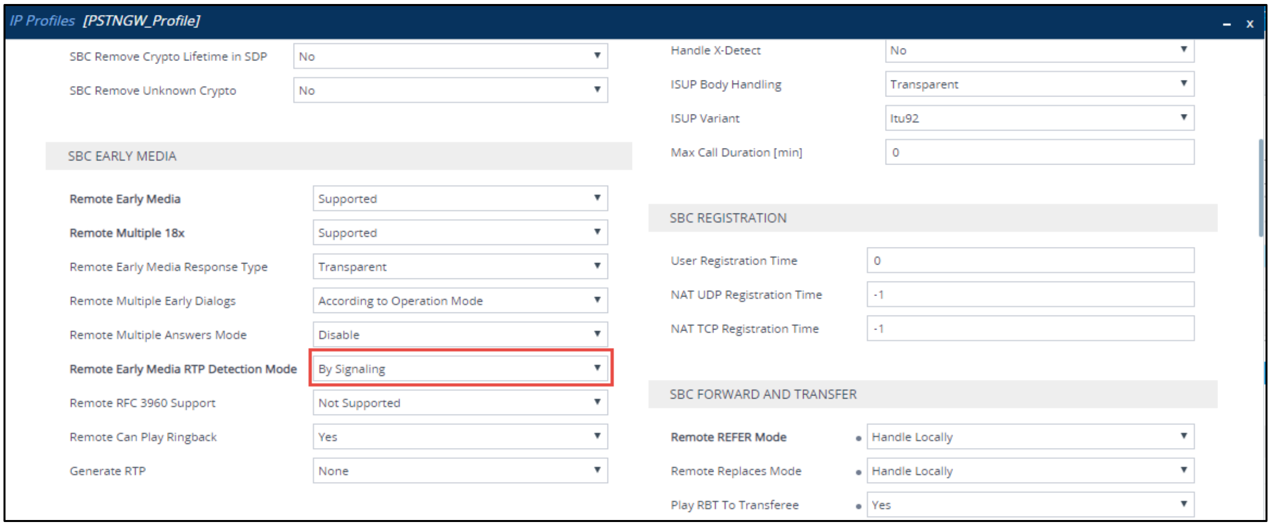


Figure 86 – IP Profile – PSTN Gateway – Contd.

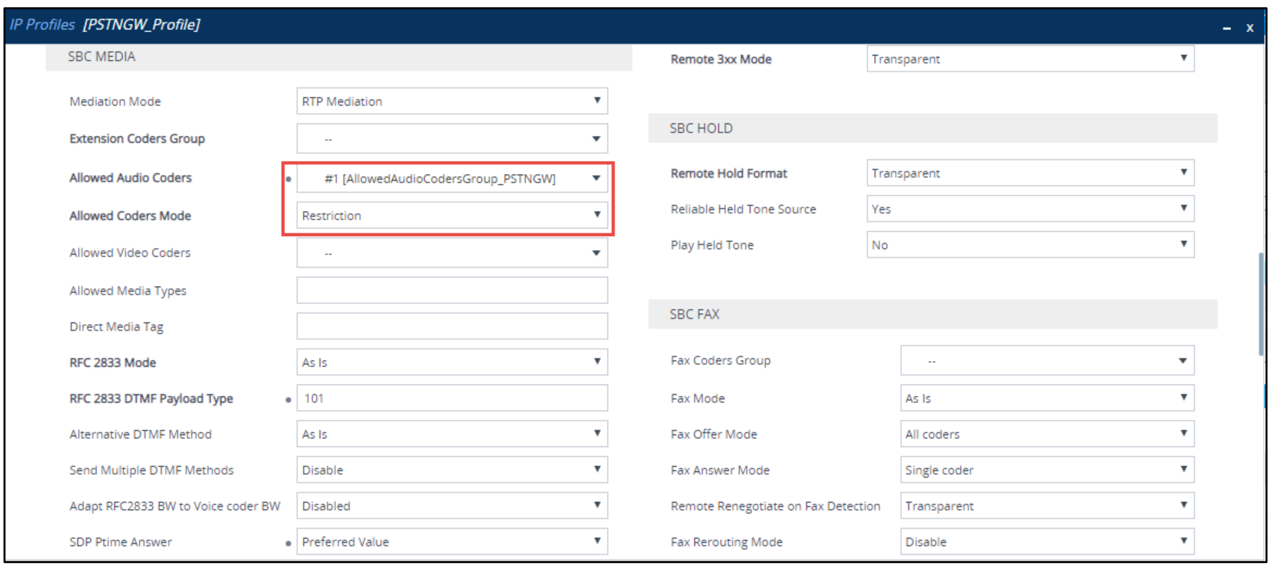


Figure 87 – IP Profile – PSTN Gateway – Contd.

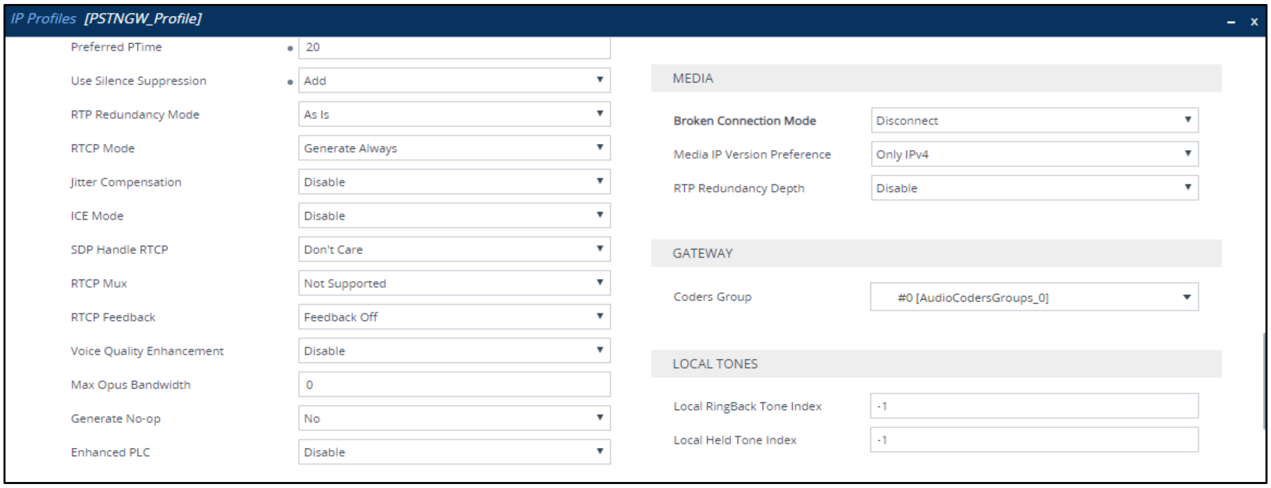


Figure 88 – IP Profile – PSTN Gateway – Contd.

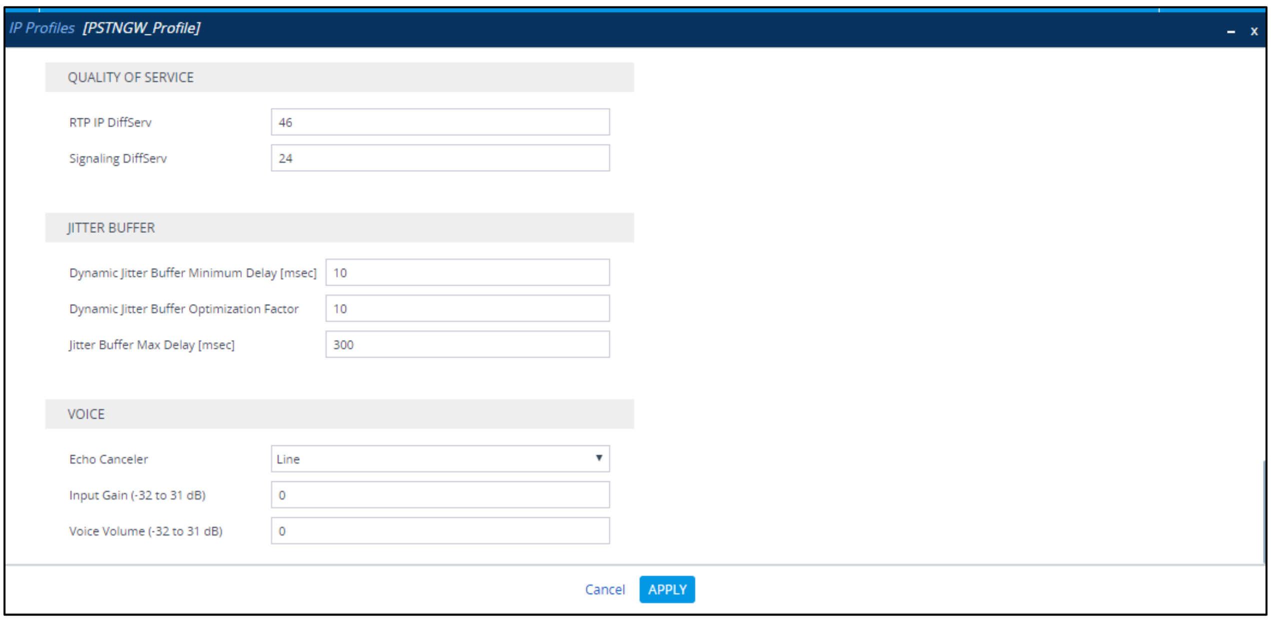


Figure 89 – IP Profile – PSTN Gateway

Configure the IP Profile for the Avaya as shown below.

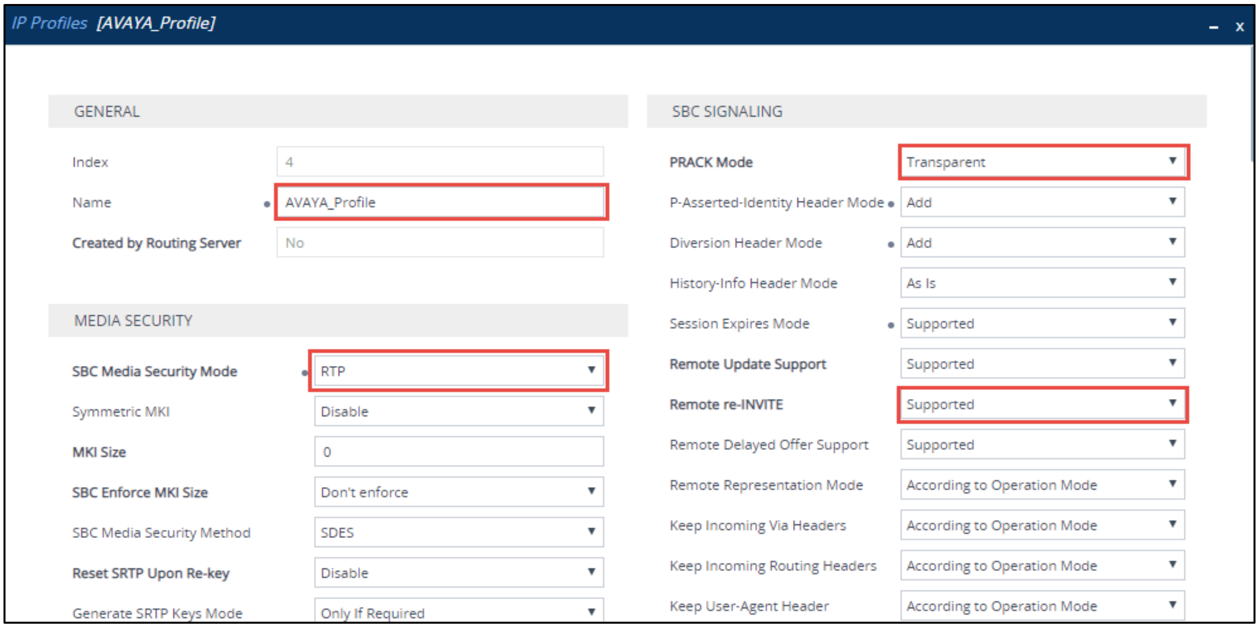


Figure 90 – IP Profile – Avaya – Contd.

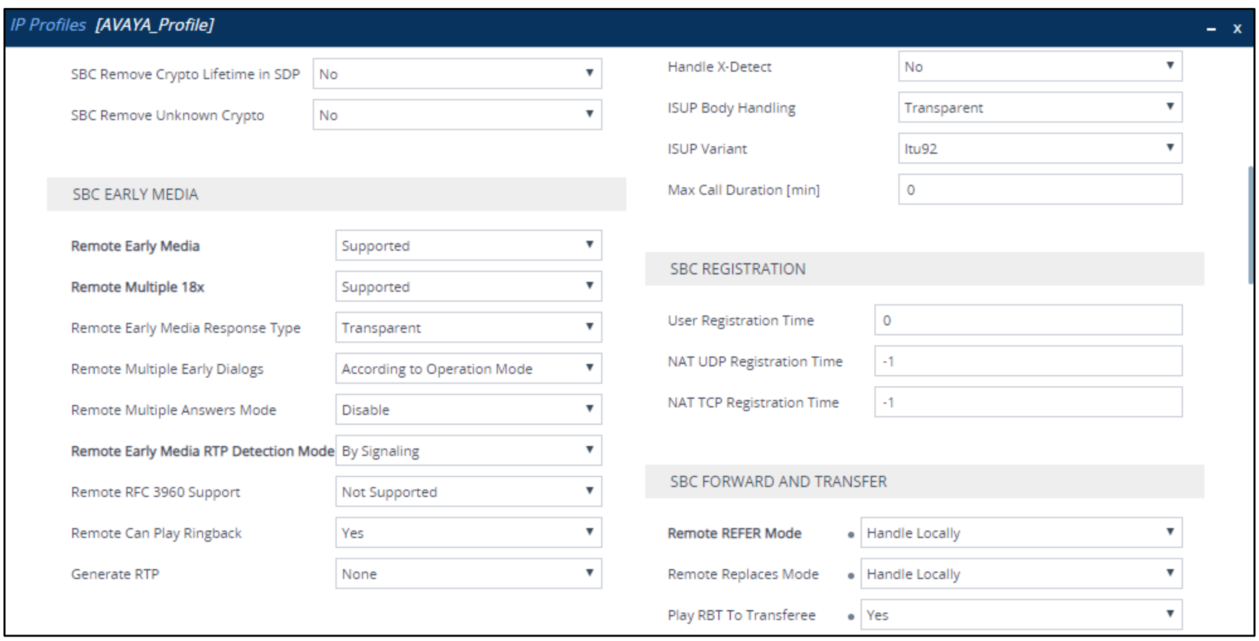


Figure 91 – IP Profile – Avaya – Contd.

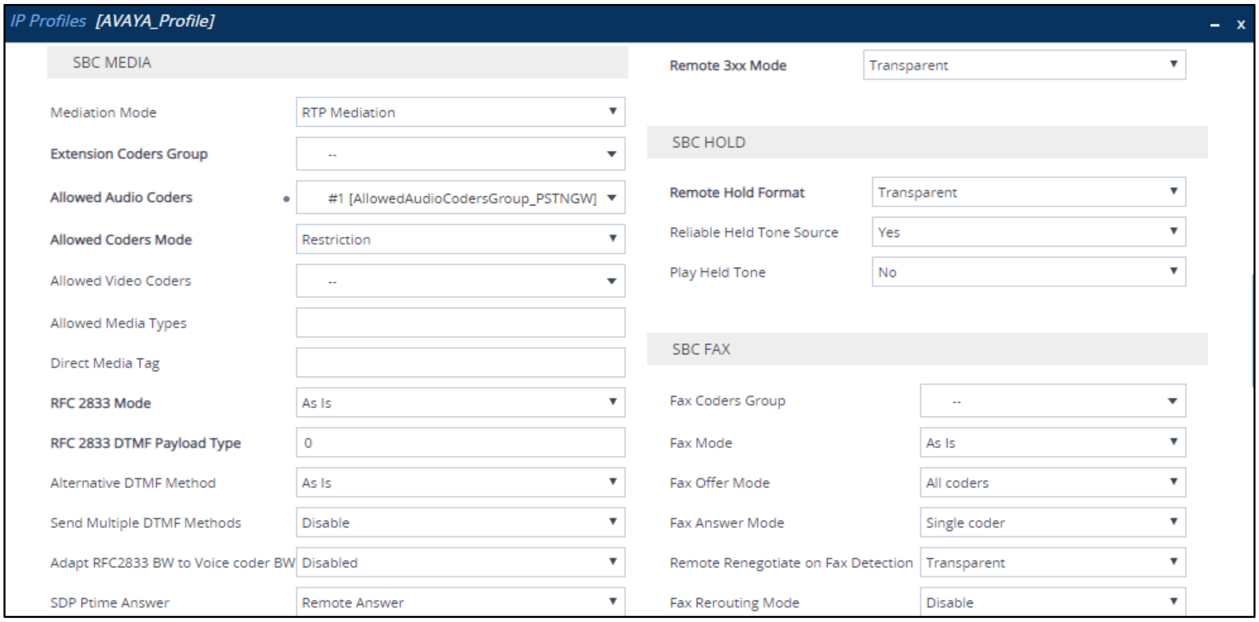


Figure 92 – IP Profile – Avaya – Contd.

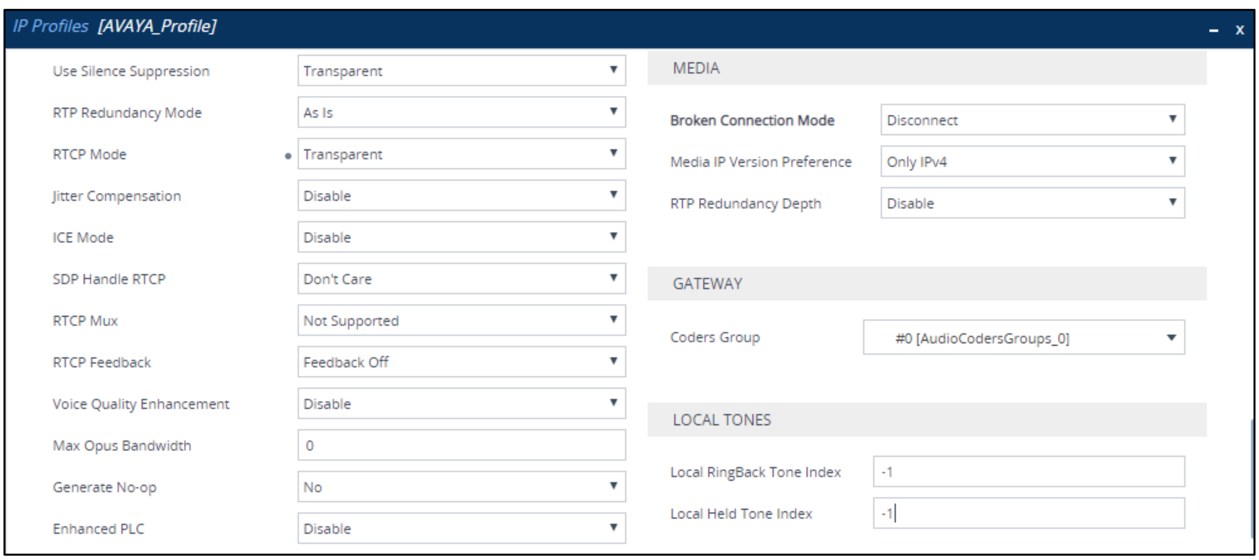


Figure 93 – IP Profile – Avaya – Contd.

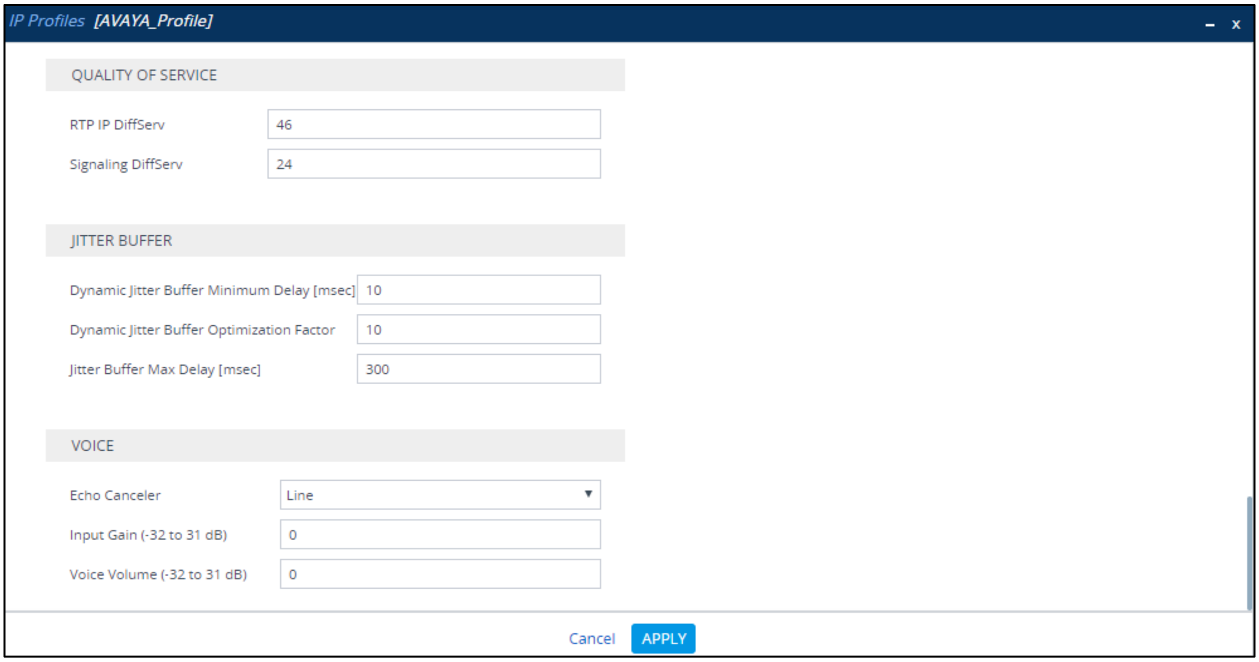


Figure 94 – IP Profile – Avaya

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

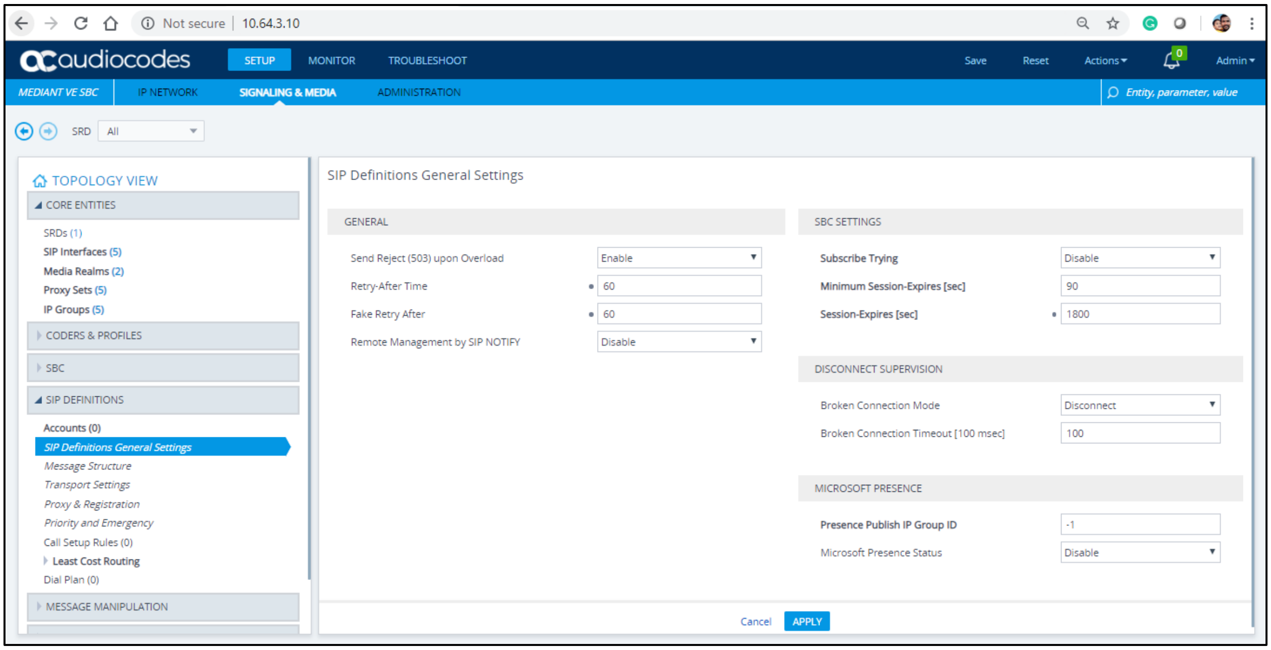


Figure 95 – SIP Definition

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.

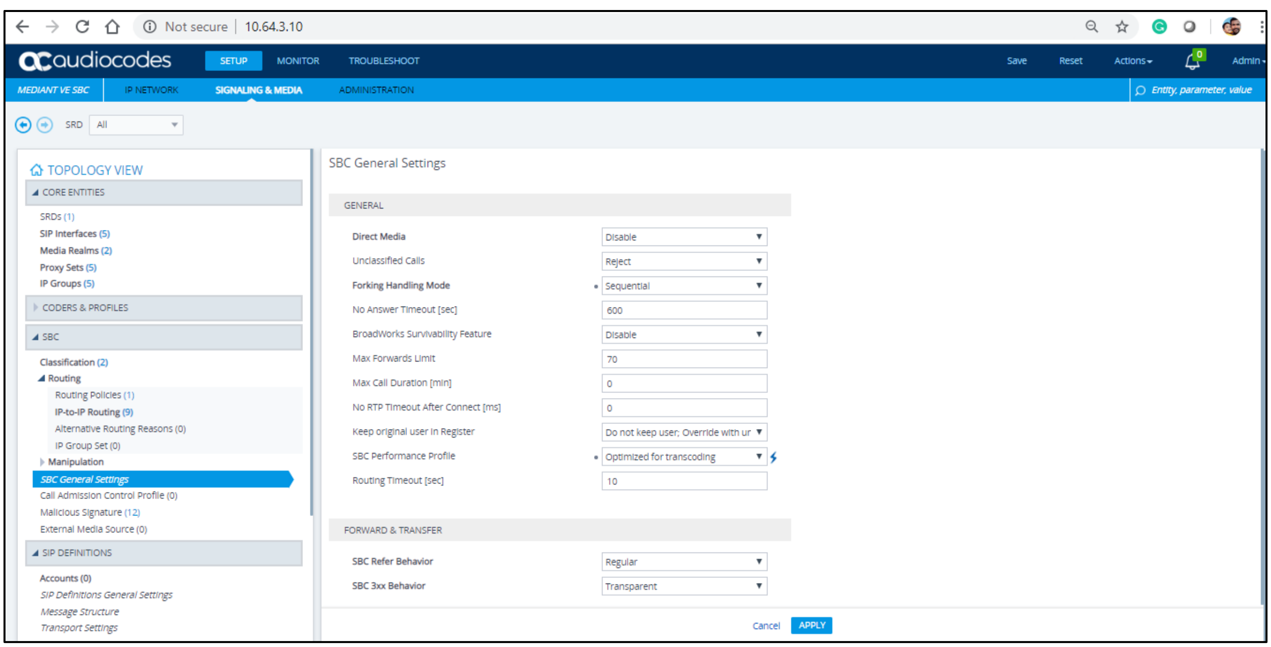


Figure 96 – SBC General Setting – Contd.

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

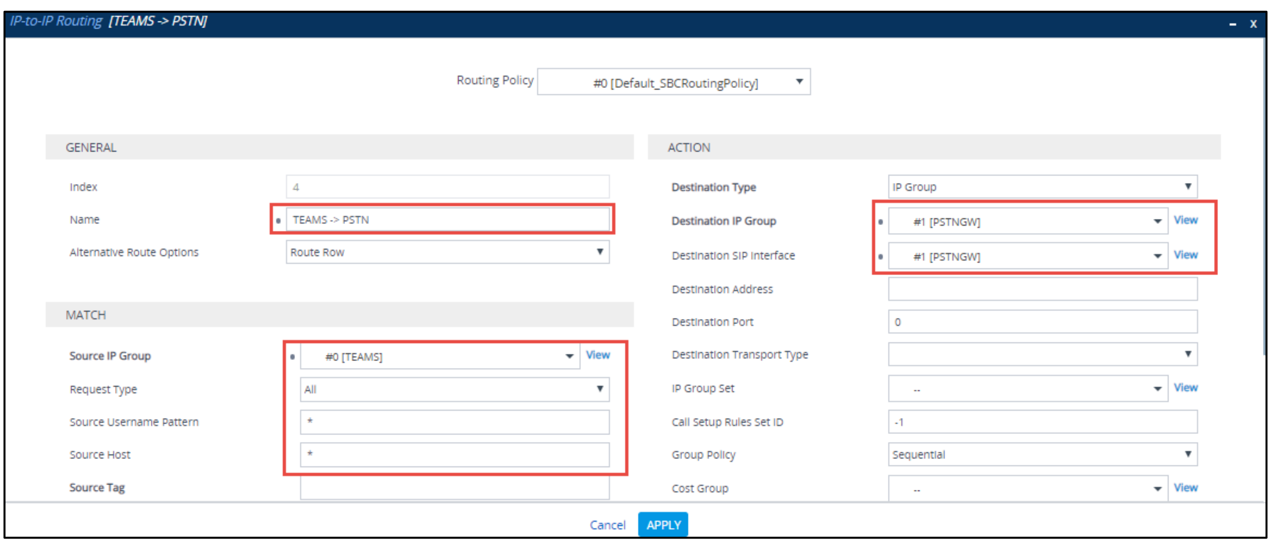


Figure 97 – Teams to PSTN – Contd.

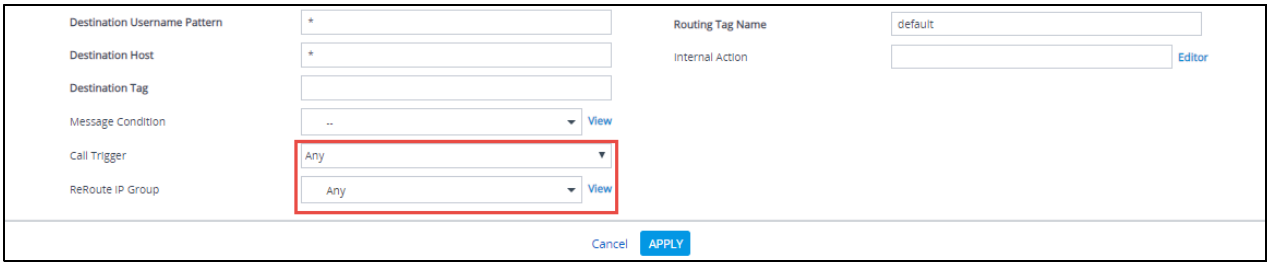


Figure 98 – Teams to PSTN

Calls from PSTN Gateway to Teams

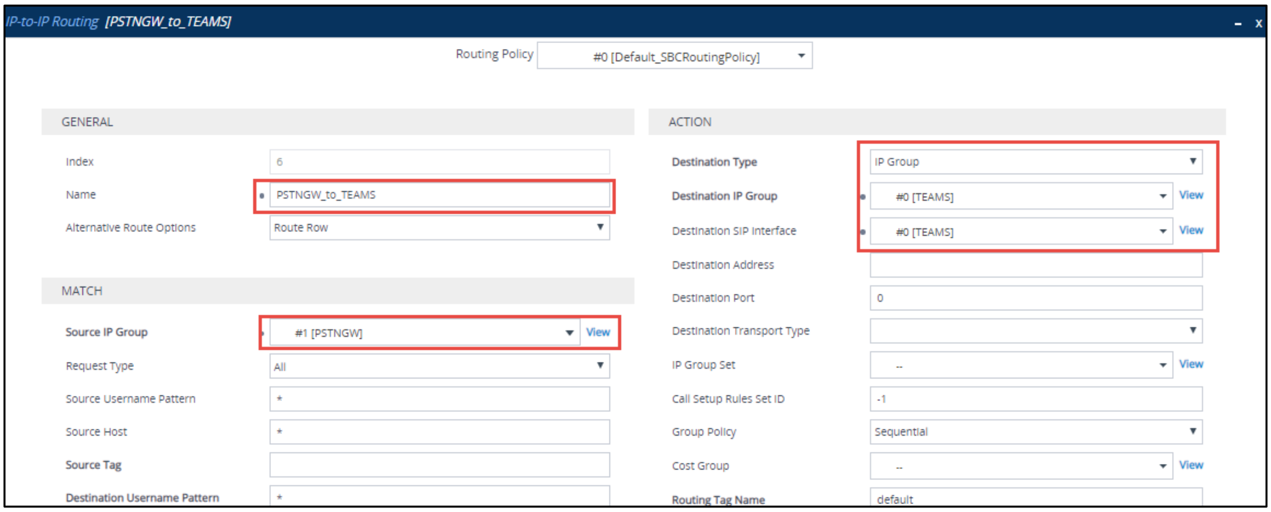


Figure 99 – PSTN to Teams – Contd.

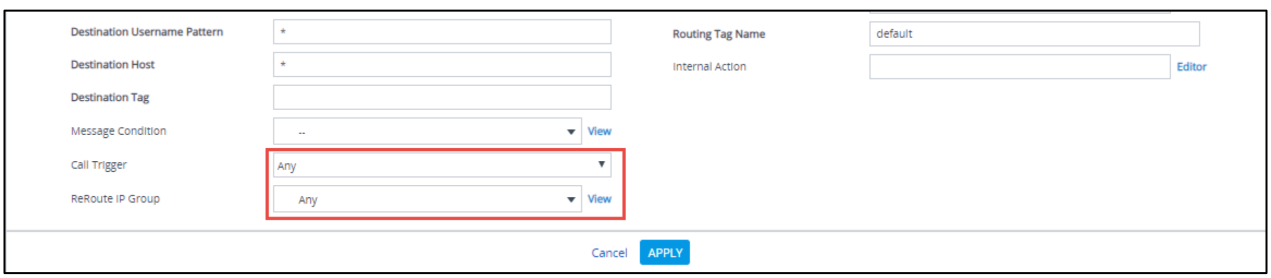


Figure 100 – PSTN to Teams

Calls from Teams to Avaya

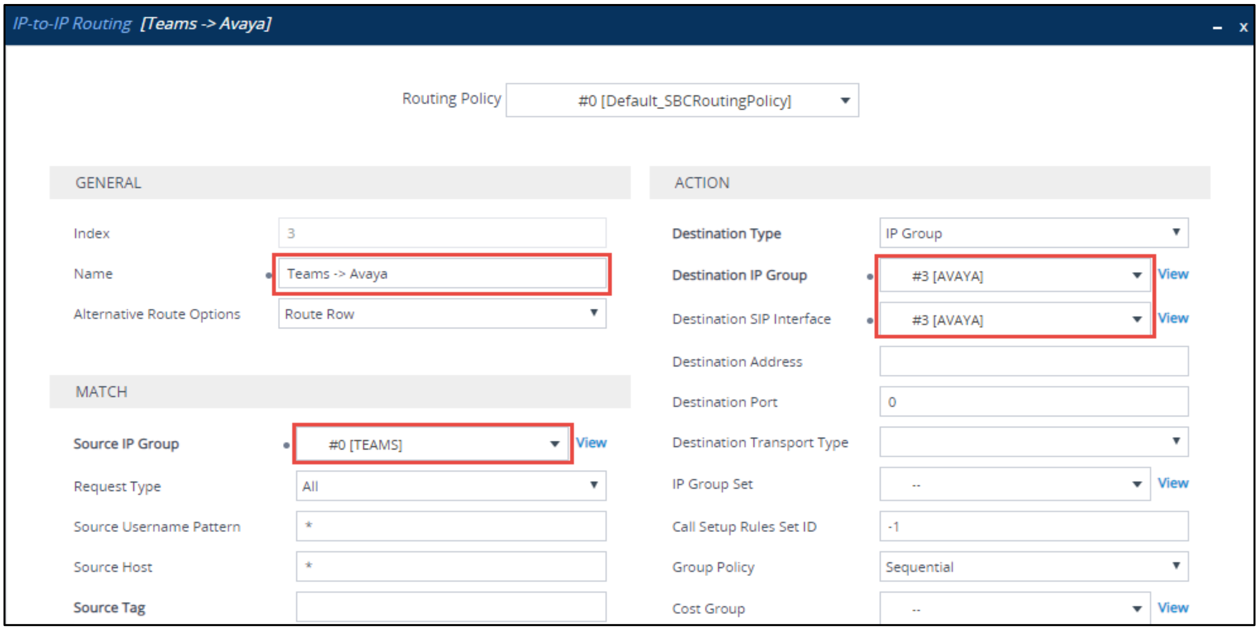


Figure 101 –Teams to Avaya.

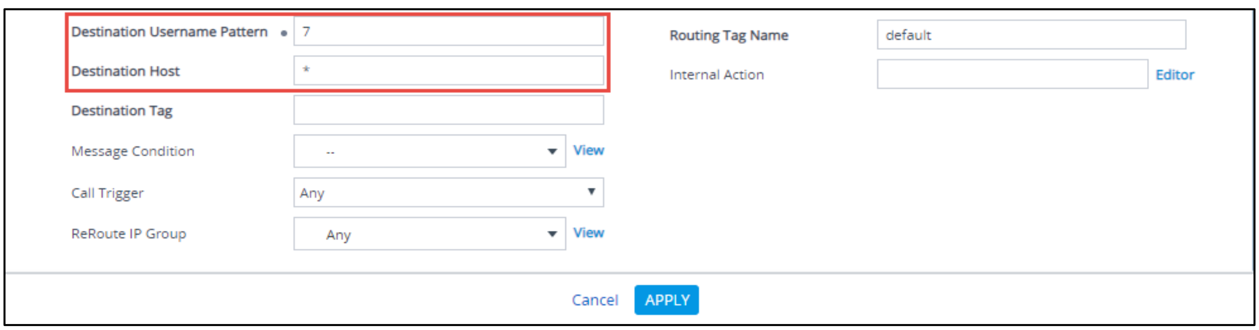


Figure 102 –Teams to Avaya Contd.

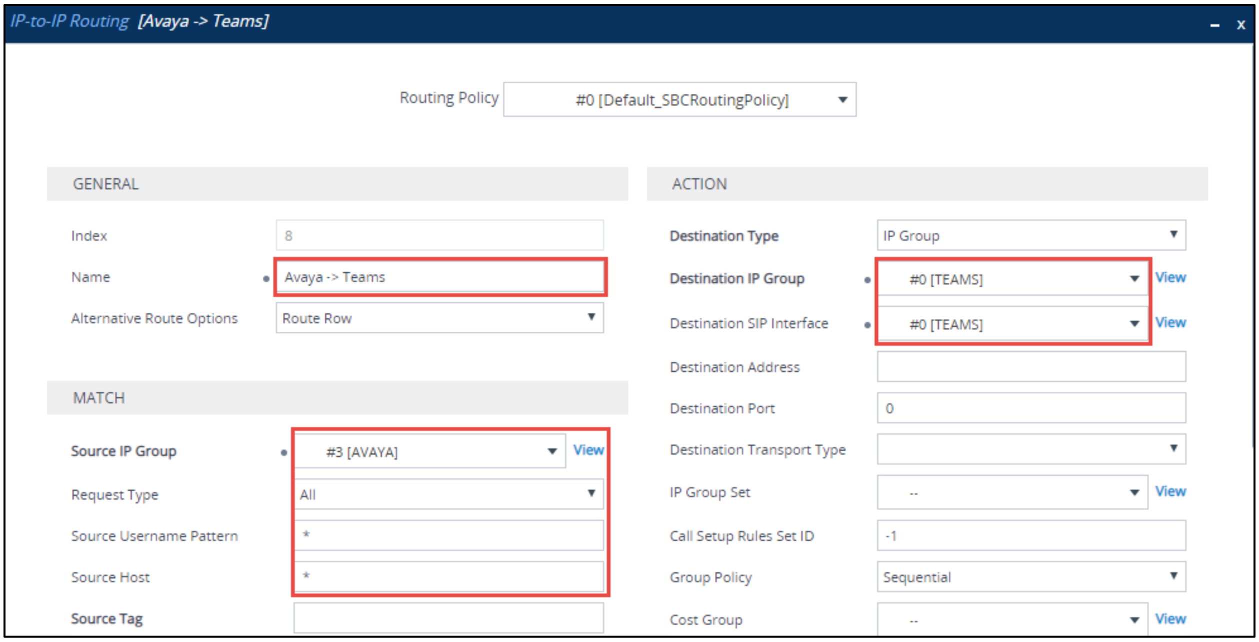


Figure 103 –Avaya to Teams.

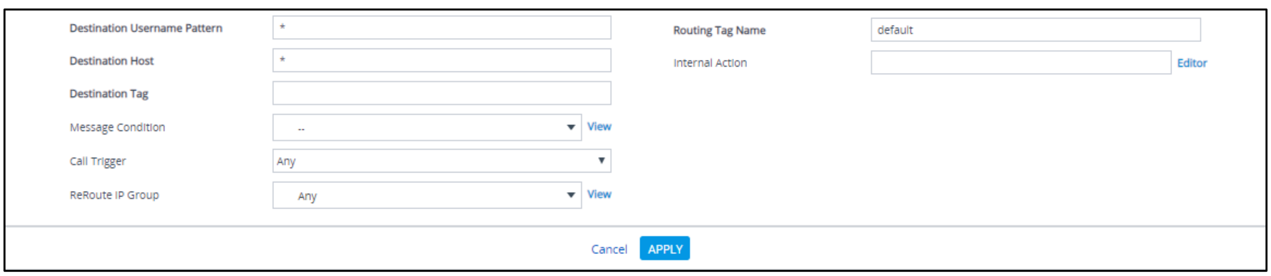


Figure 104 –Avaya to Teams – Contd.

4.5.16 IP Group

IP Group – Teams

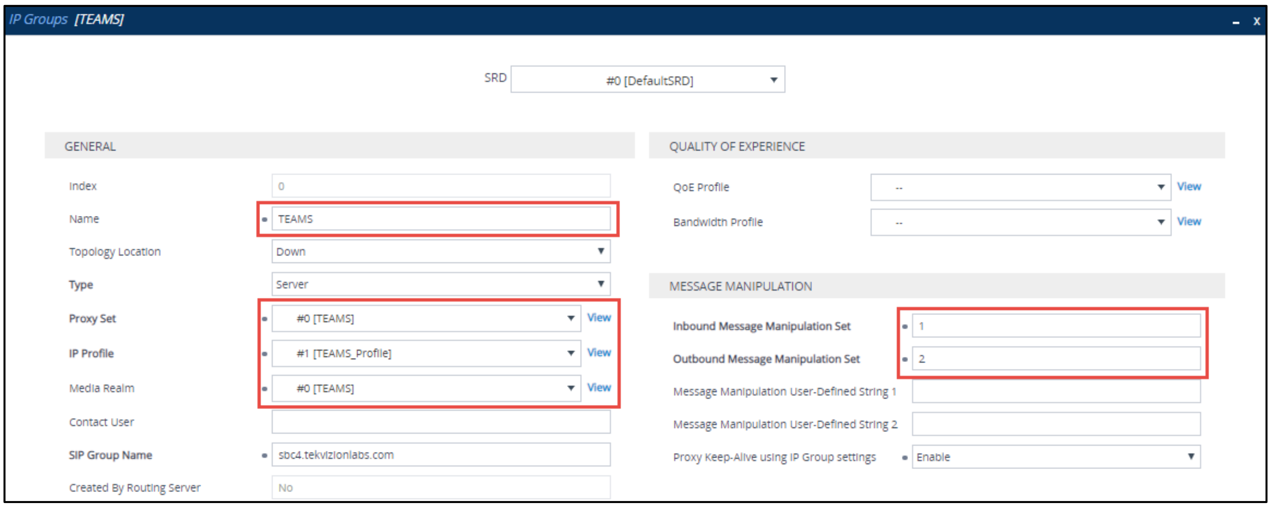


Figure 105 – IP Groups Teams – Contd.

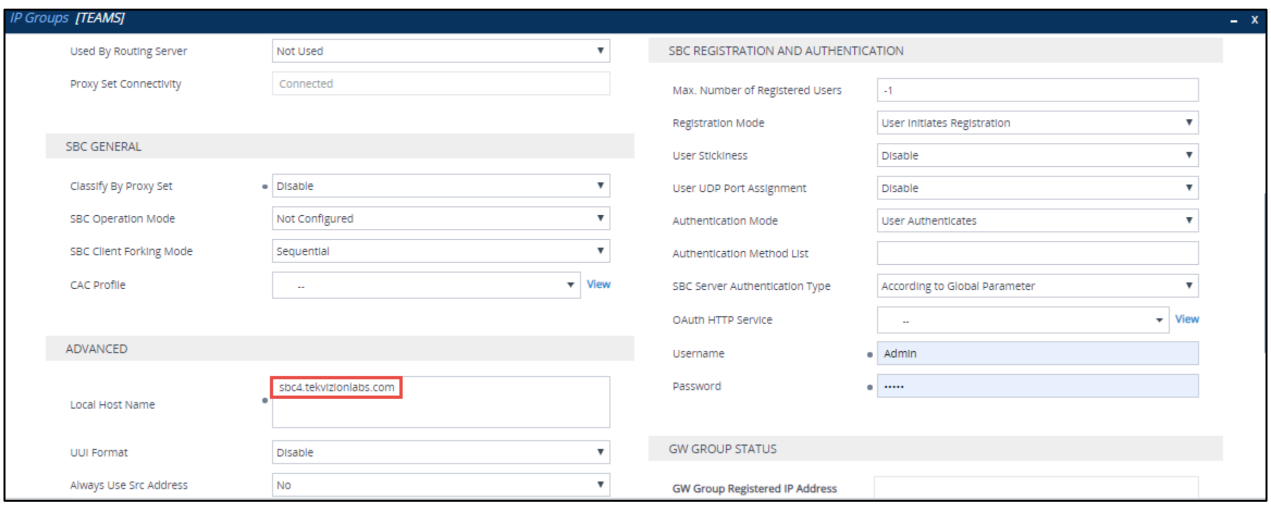


Figure 106 – IP Groups Teams – Contd.

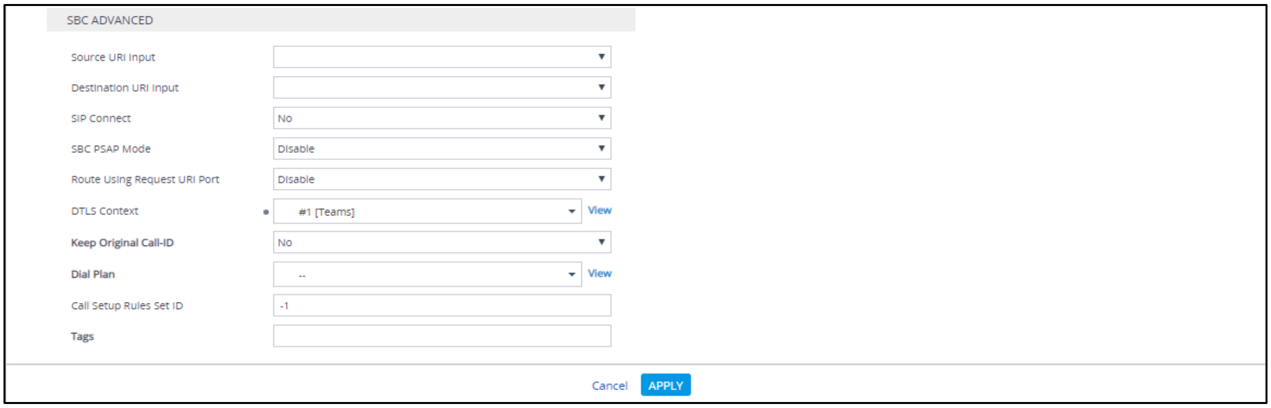


Figure 107 – IP Groups Teams

IP Group – PSTN Gateway

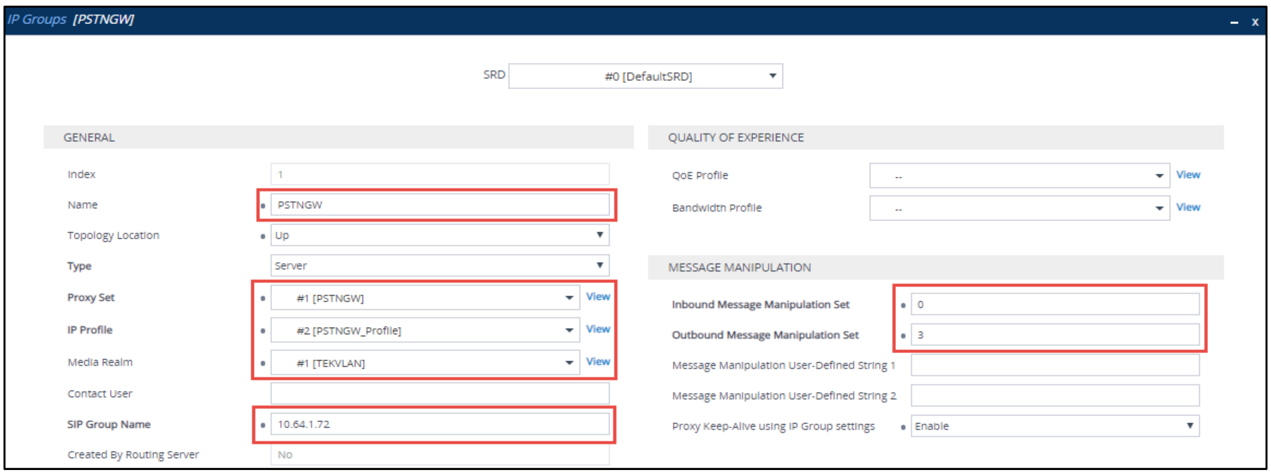


Figure 108 – IP Groups PSTN – Contd.

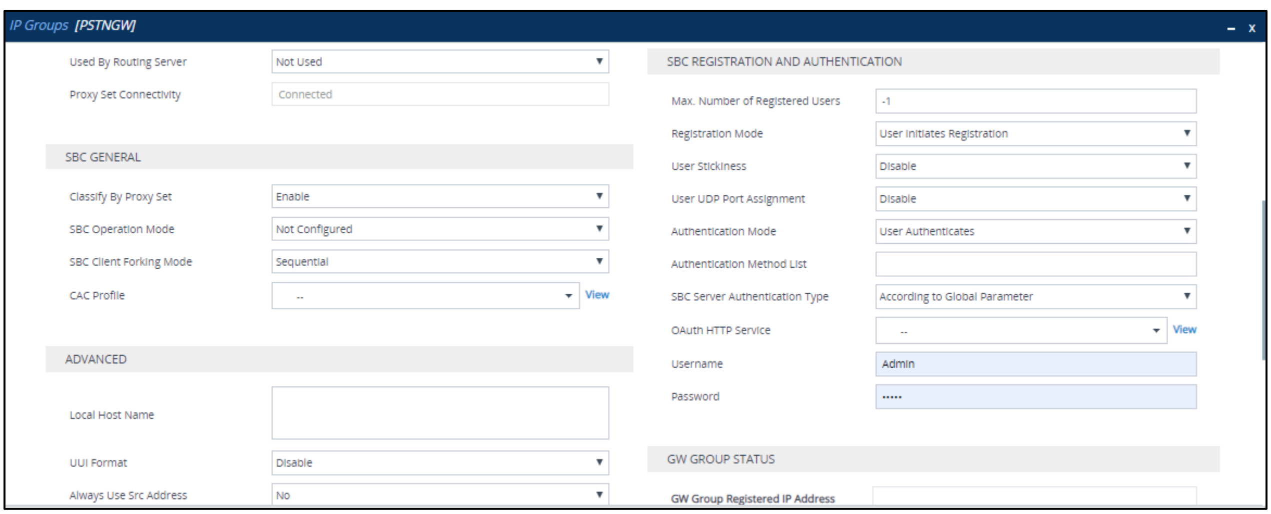


Figure 109 – IP Groups PSTN – Contd.

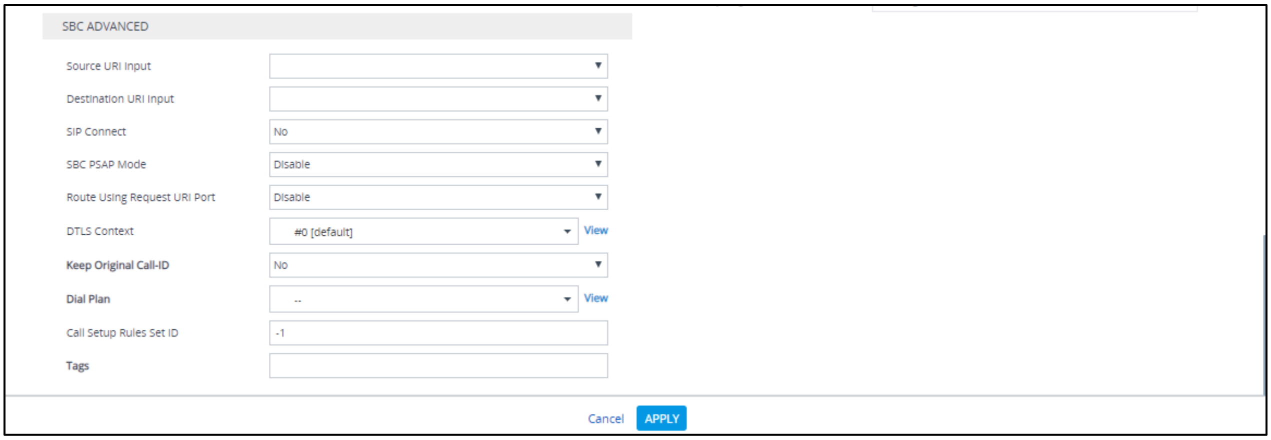


Figure 110 – IP Groups PSTN

IP Group – Avaya

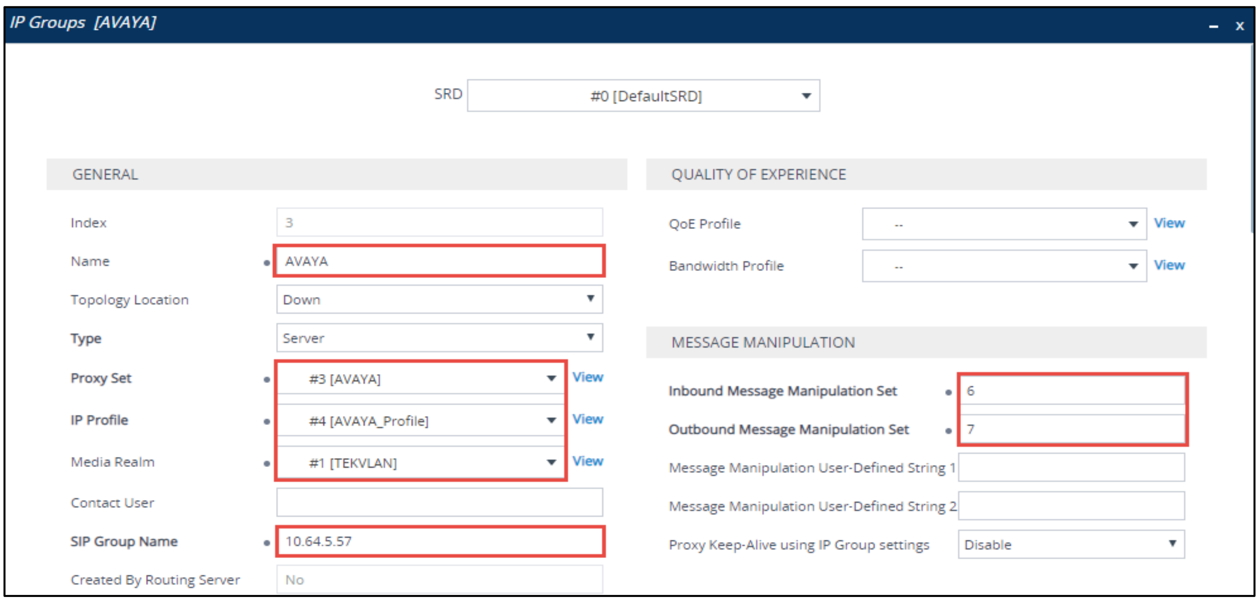


Figure 111 – IP Groups Avaya

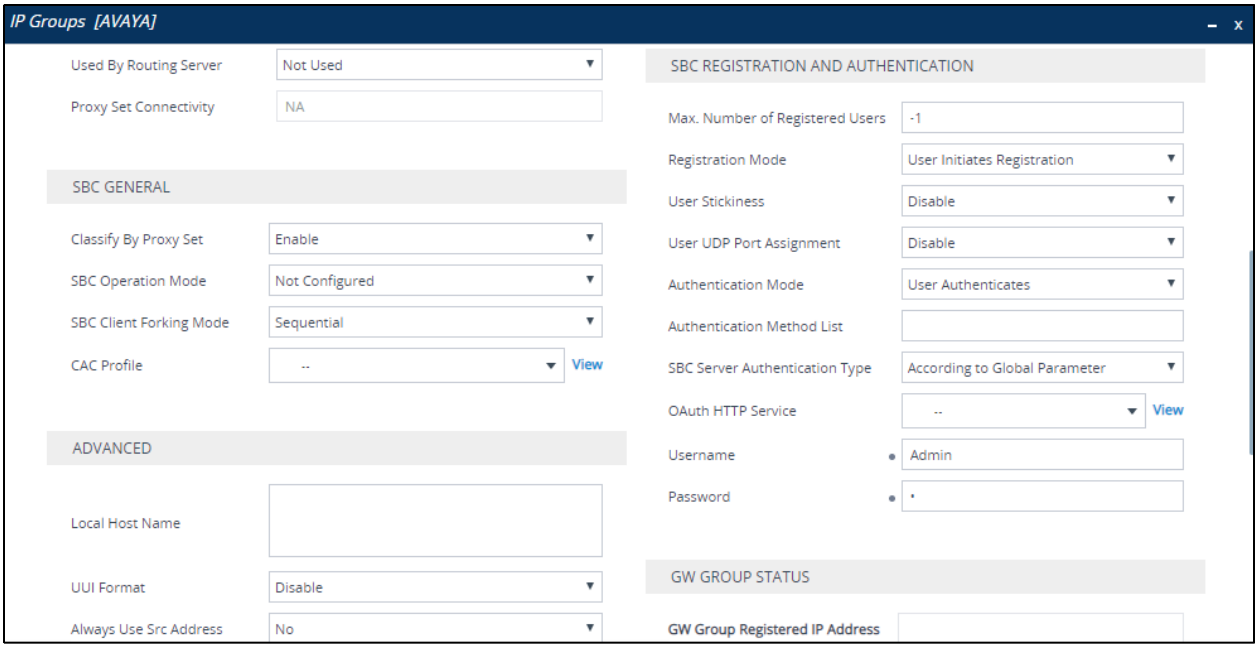


Figure 112 – IP Groups Avaya – Contd.

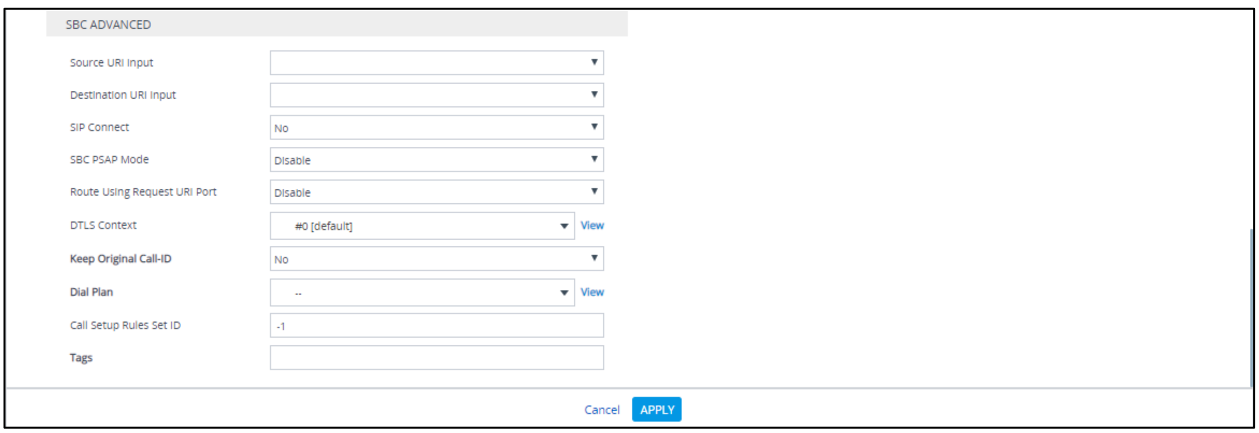


Figure 113 – IP Groups Avaya – Contd.

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:

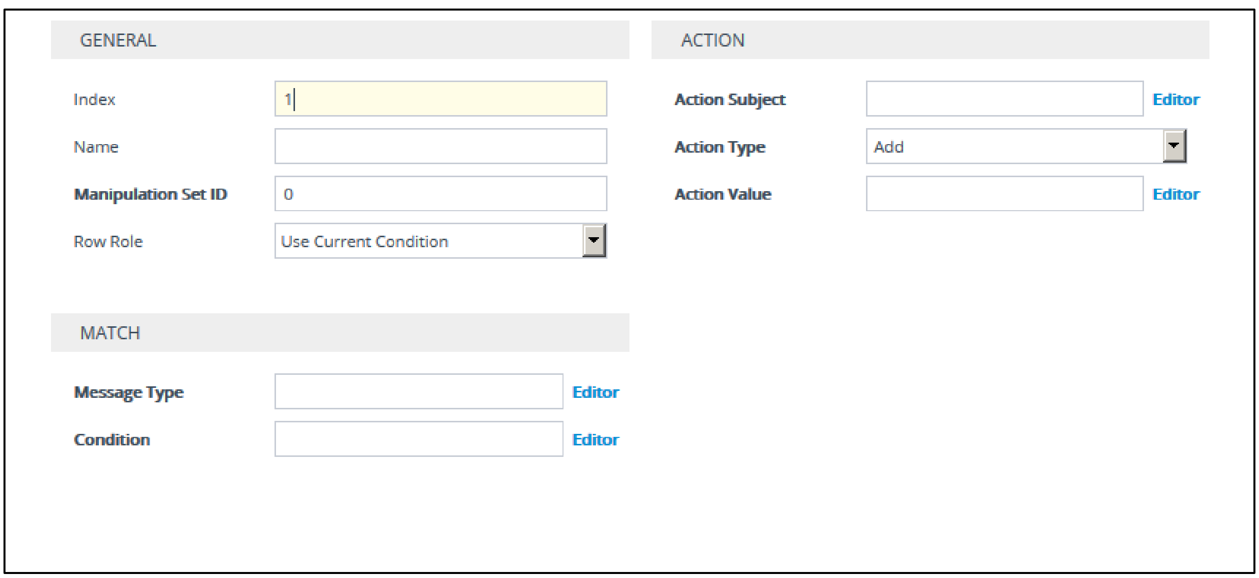


Figure 114 – SIP Message Manipulation

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

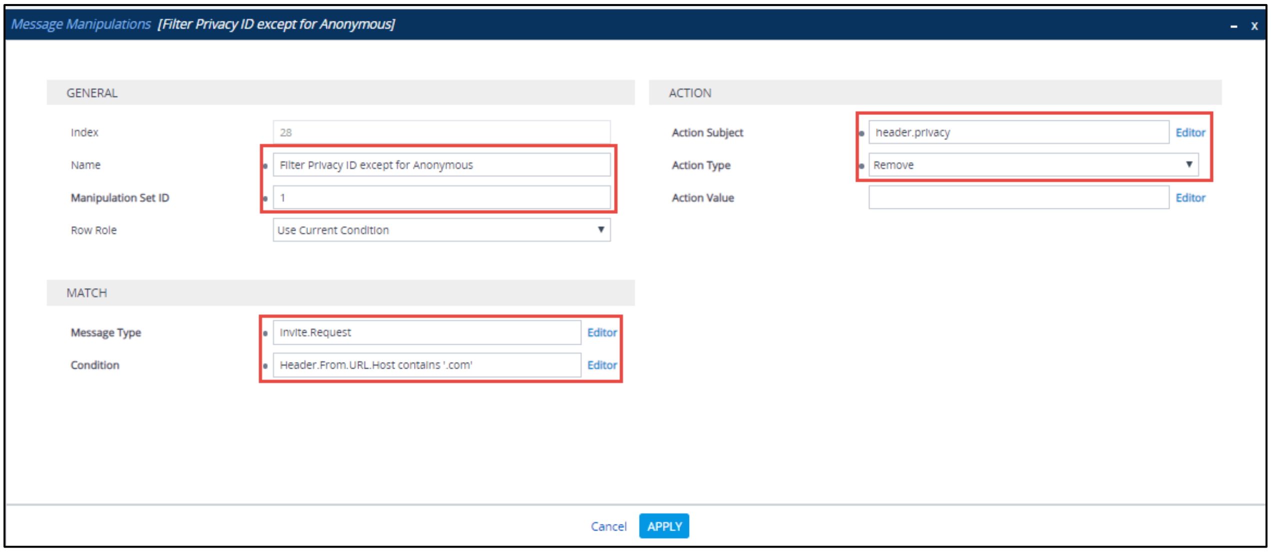


Figure 115 – SIP Message Manipulation - Privacy

Manipulation to Teams

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

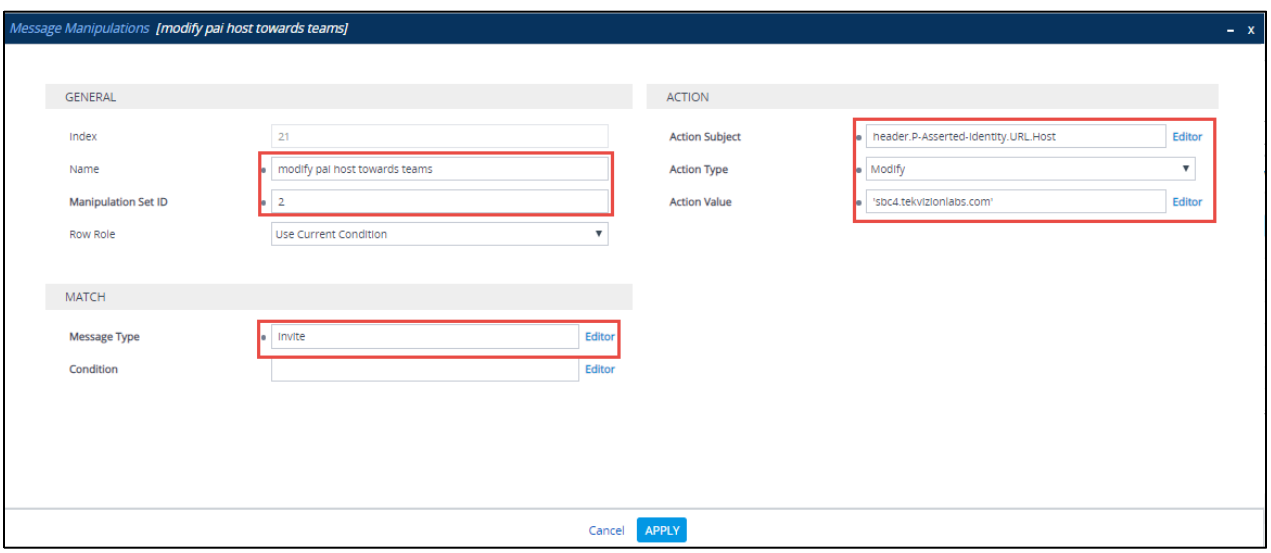


Figure 116 – SIP Message Manipulation - PAI

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

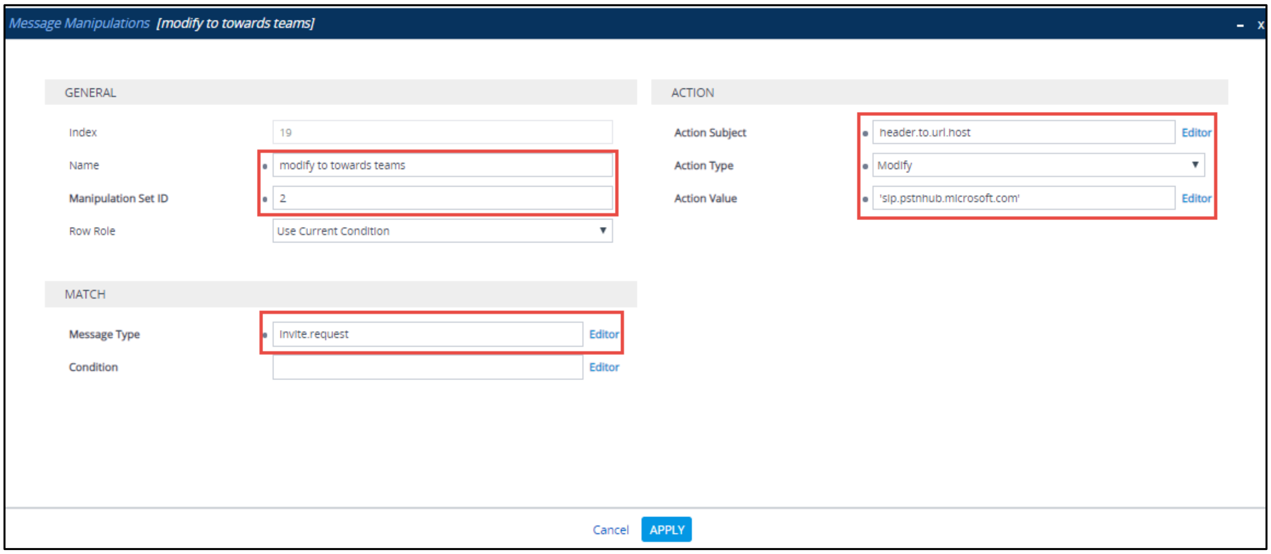


Figure 117 – SIP Message Manipulation - To

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

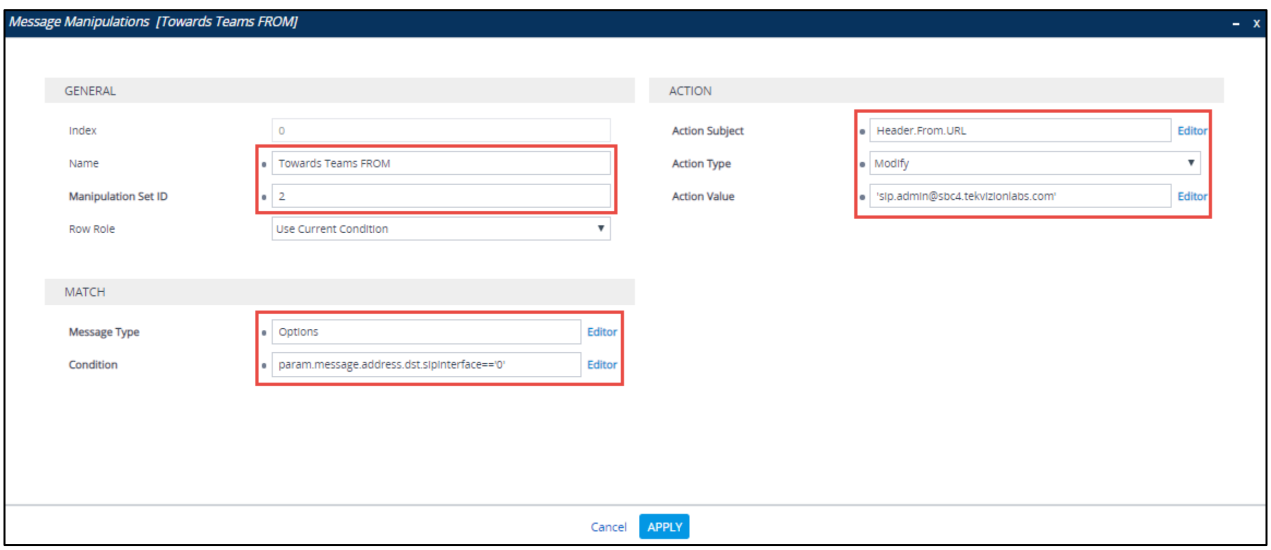


Figure 118 – SIP Message Manipulation - From

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

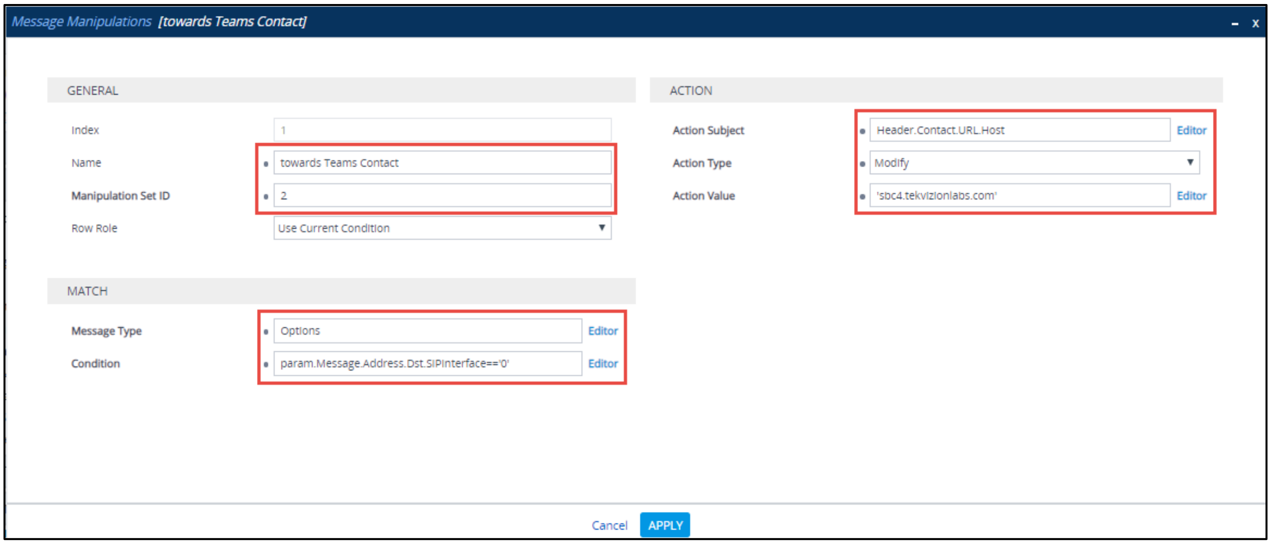


Figure 119 – SIP Message Manipulation - Contact

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

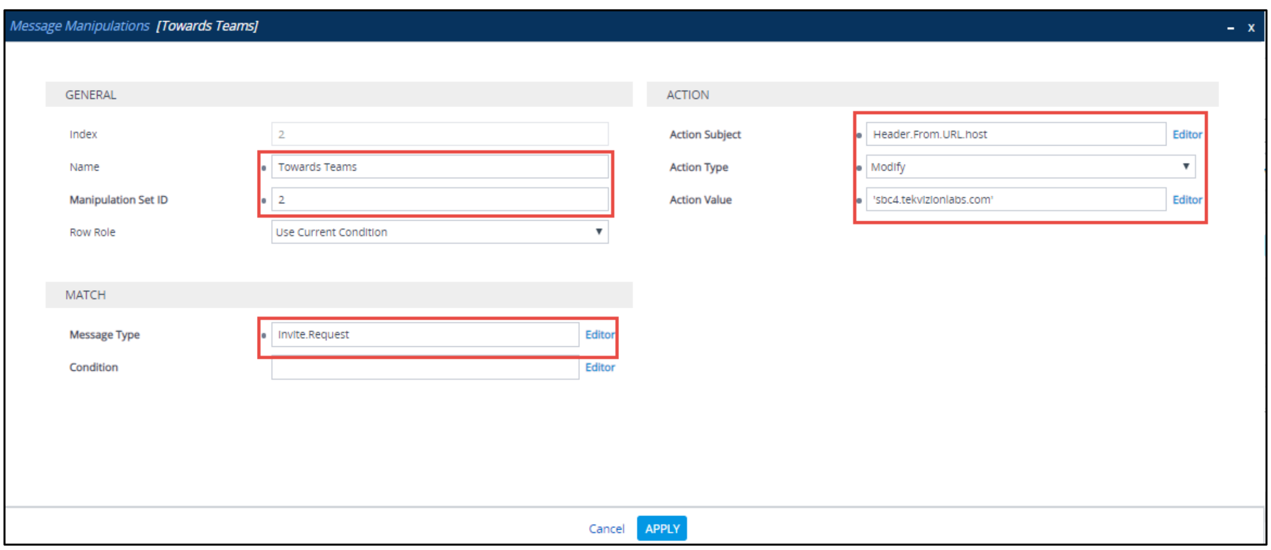


Figure 120 – SIP Message Manipulation - From

Manipulation to PSTN

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

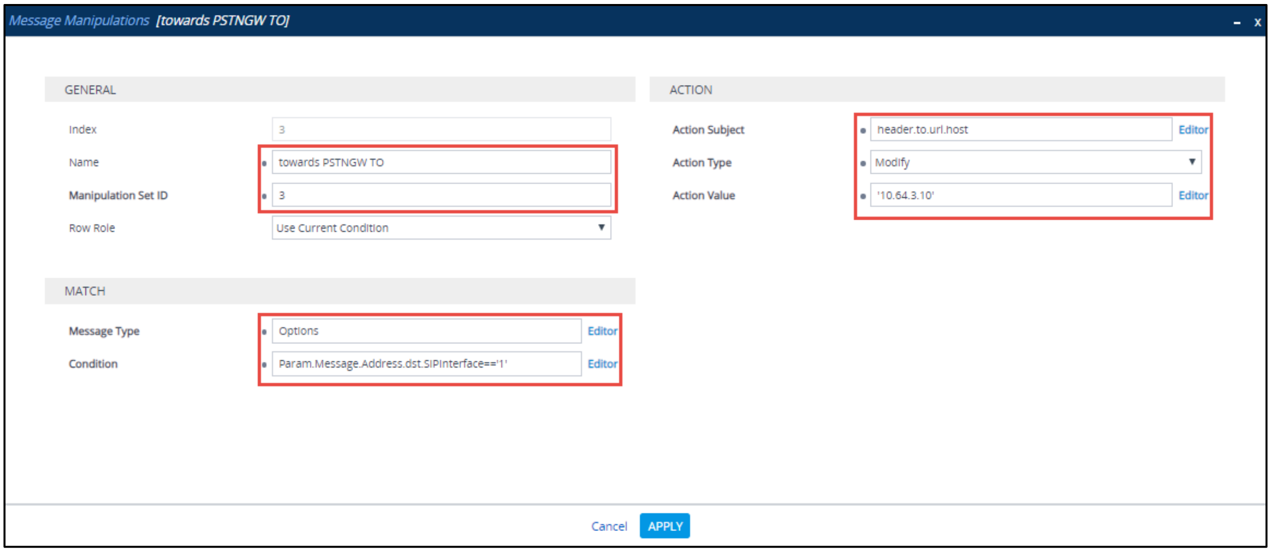


Figure 121 – SIP Message Manipulation – To

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

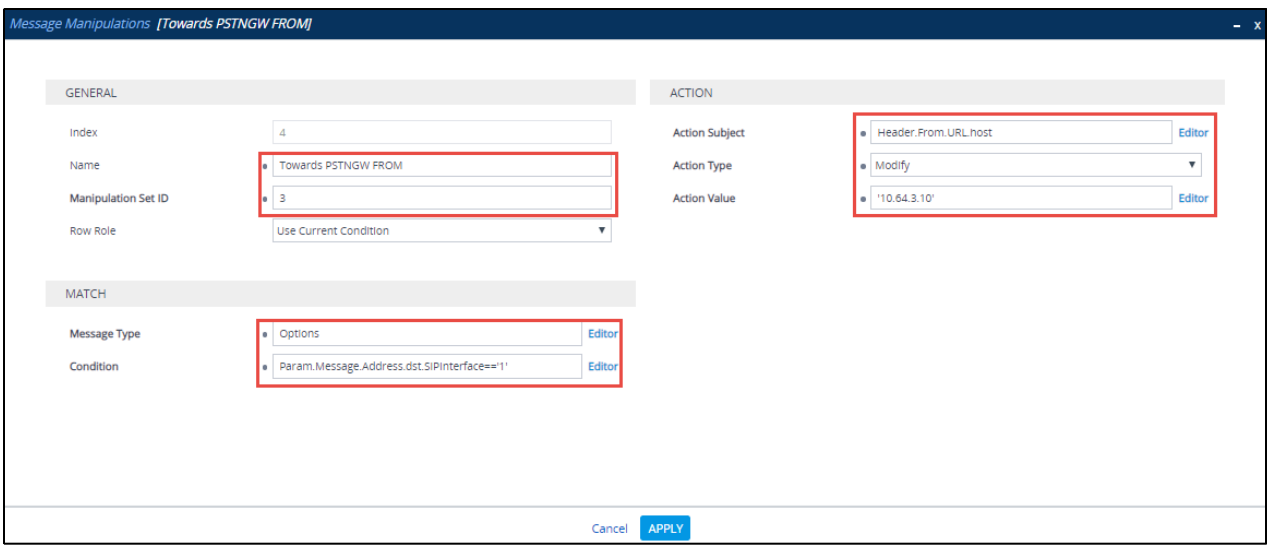


Figure 122 – SIP Message Manipulation – From

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

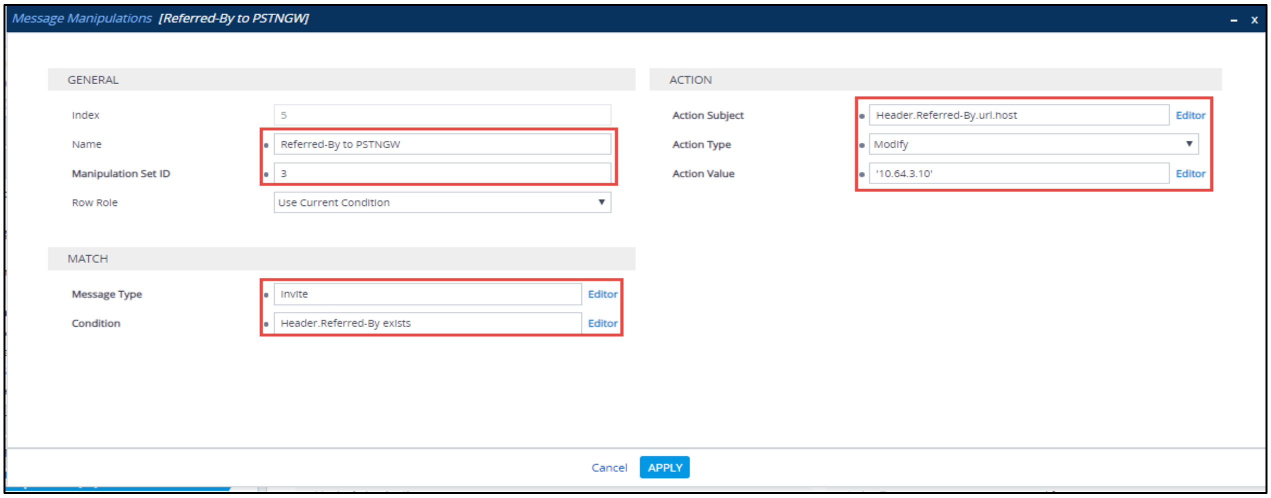


Figure 123 – SIP Message Manipulation – Referred - By

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

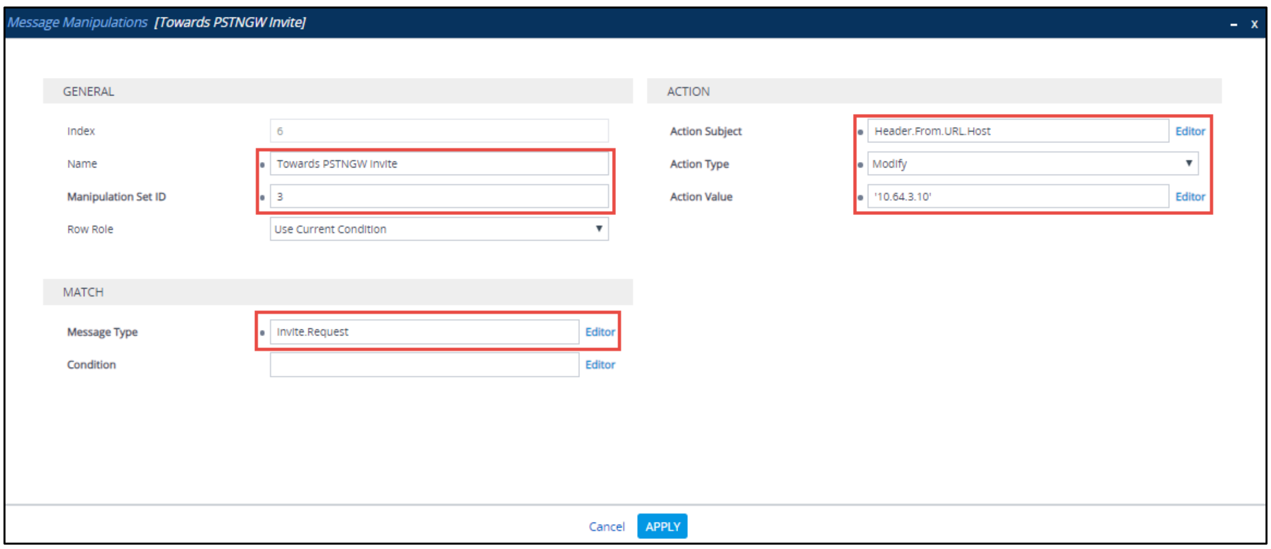


Figure 124 – SIP Message Manipulation – From

Manipulation to Avaya

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

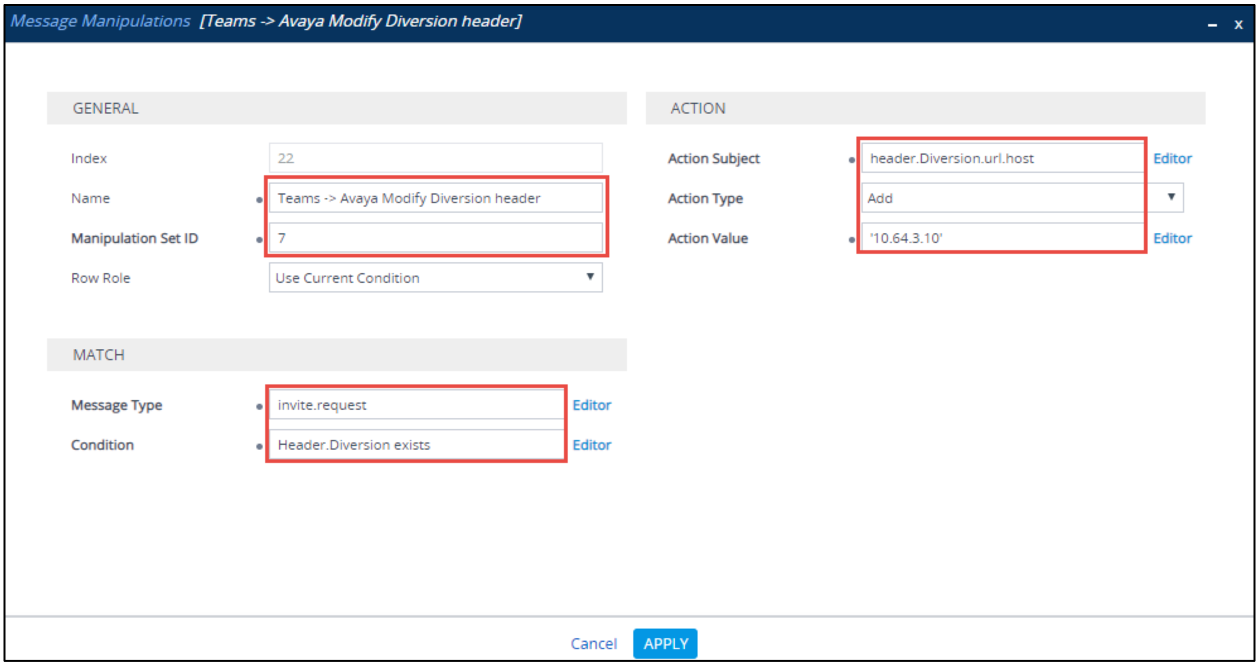


Figure 125 – SIP Message Manipulation – Diversion

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

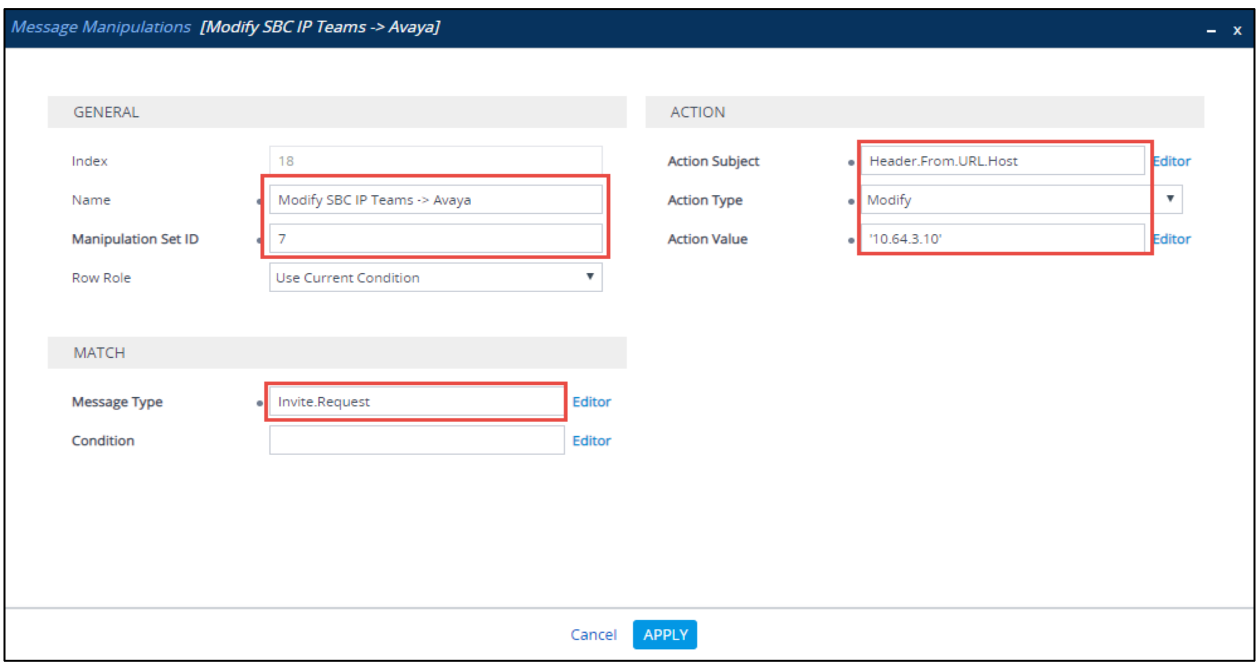


Figure 126 – SIP Message Manipulation – From

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

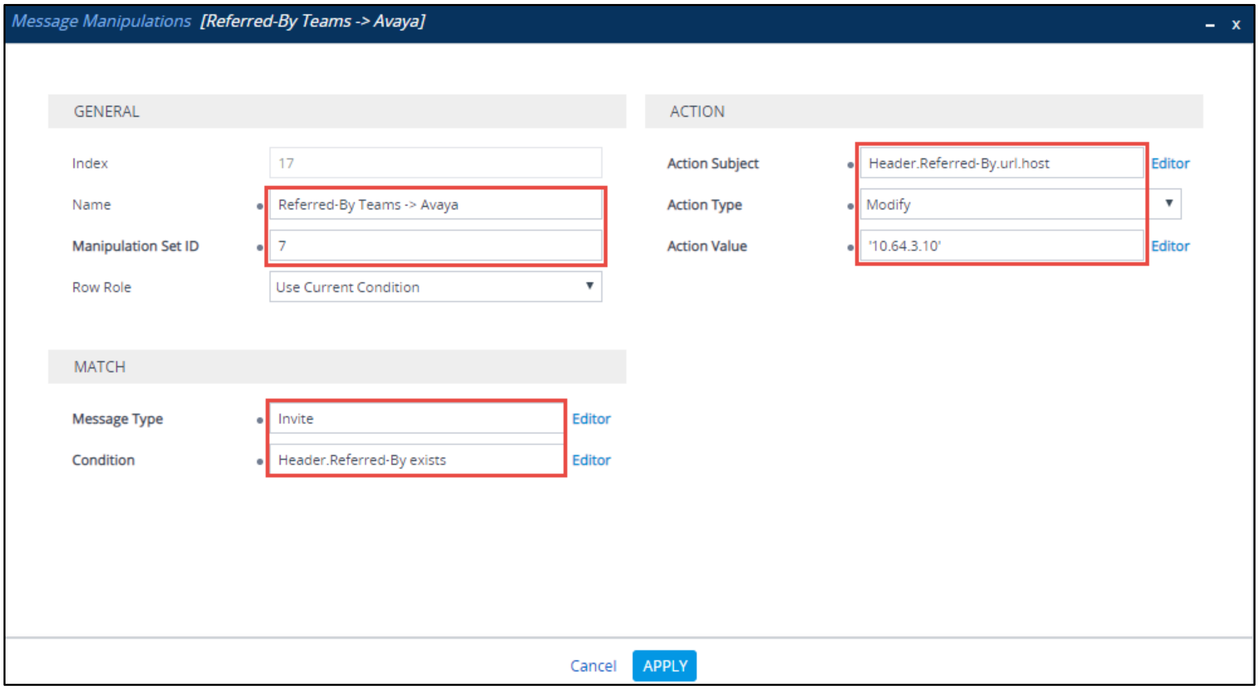


Figure 127 – SIP Message Manipulation – Referred By

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

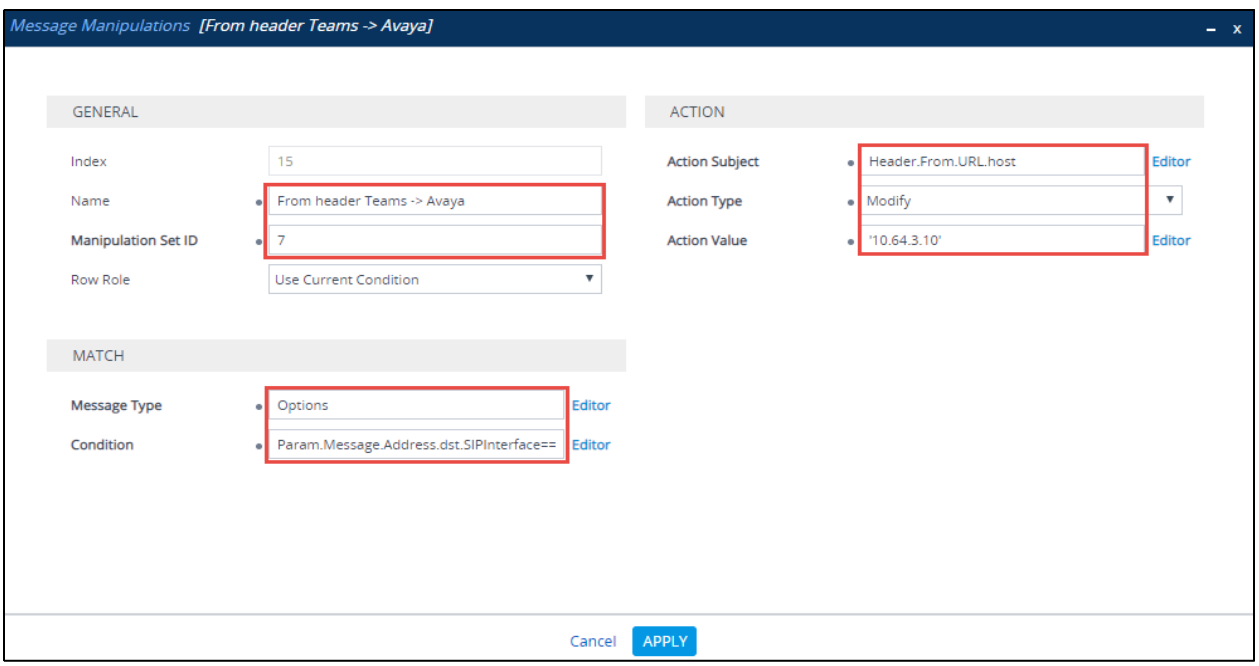


Figure 128 – SIP Message Manipulation – From

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

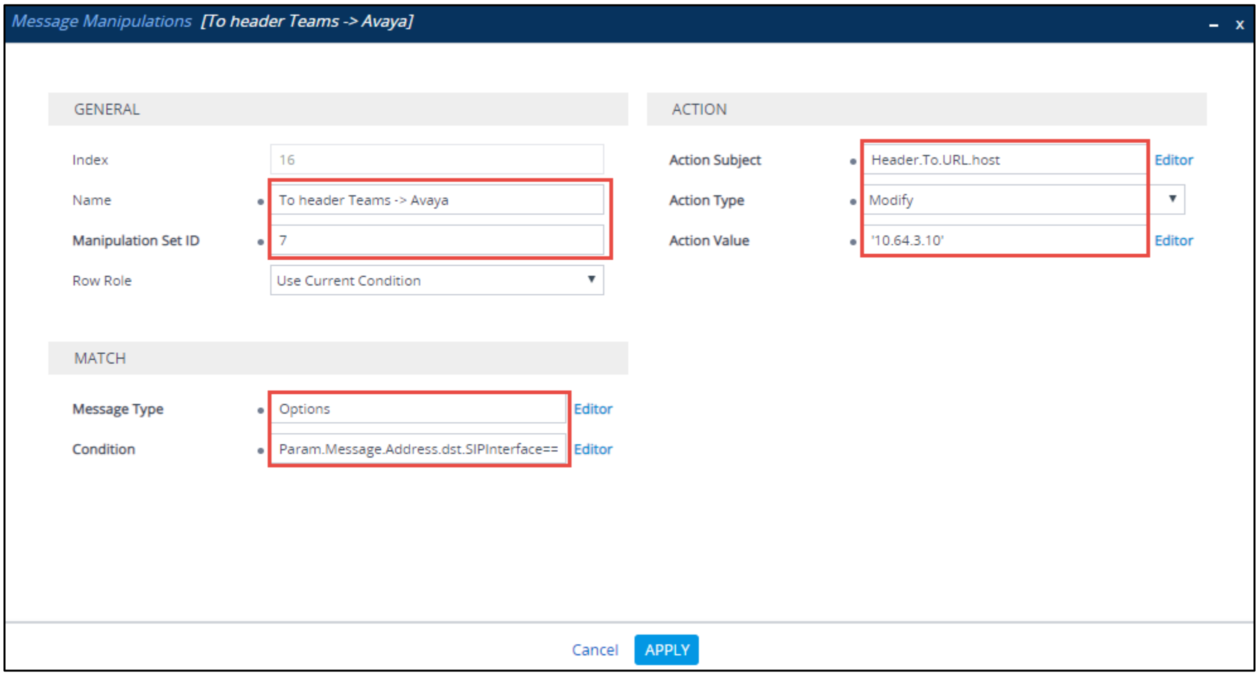


Figure 129 – SIP Message Manipulation – to

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.

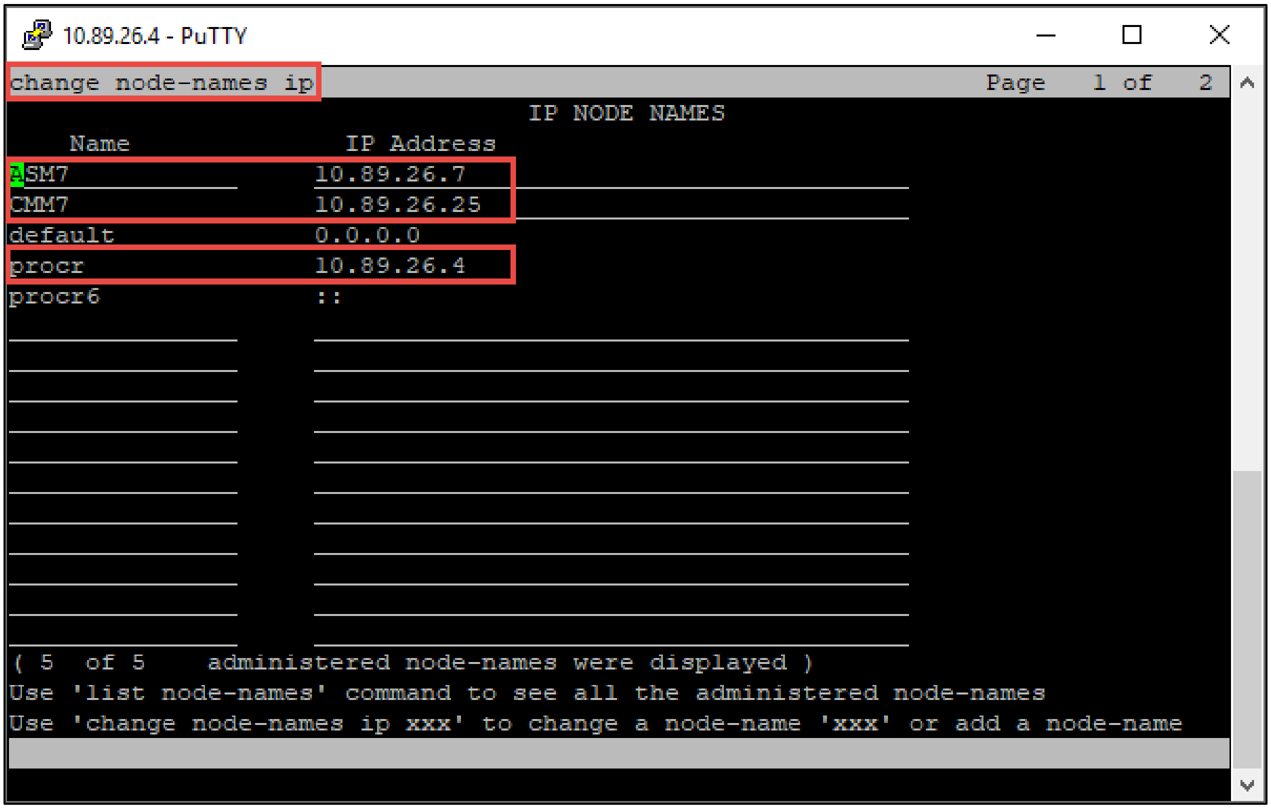


Figure 131 - IP Node Name

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

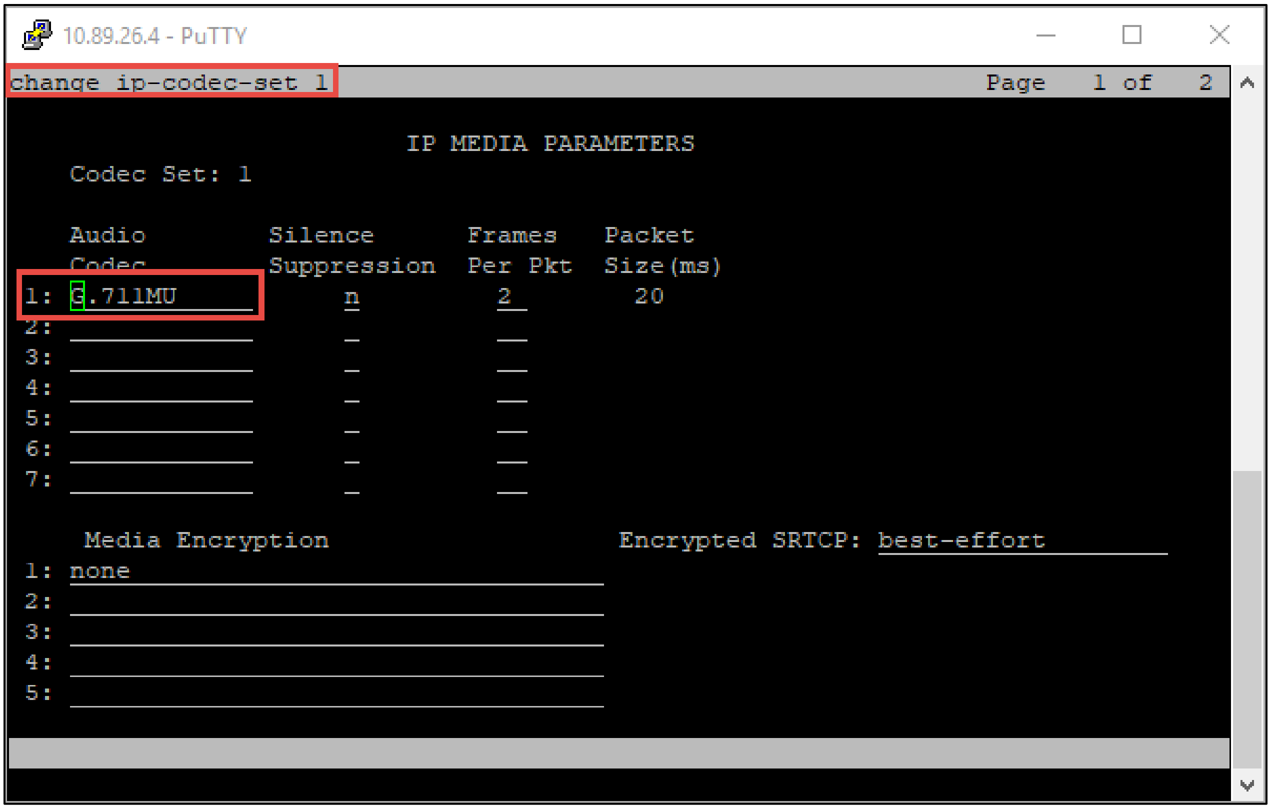


Figure 132 - IP Codec Set

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

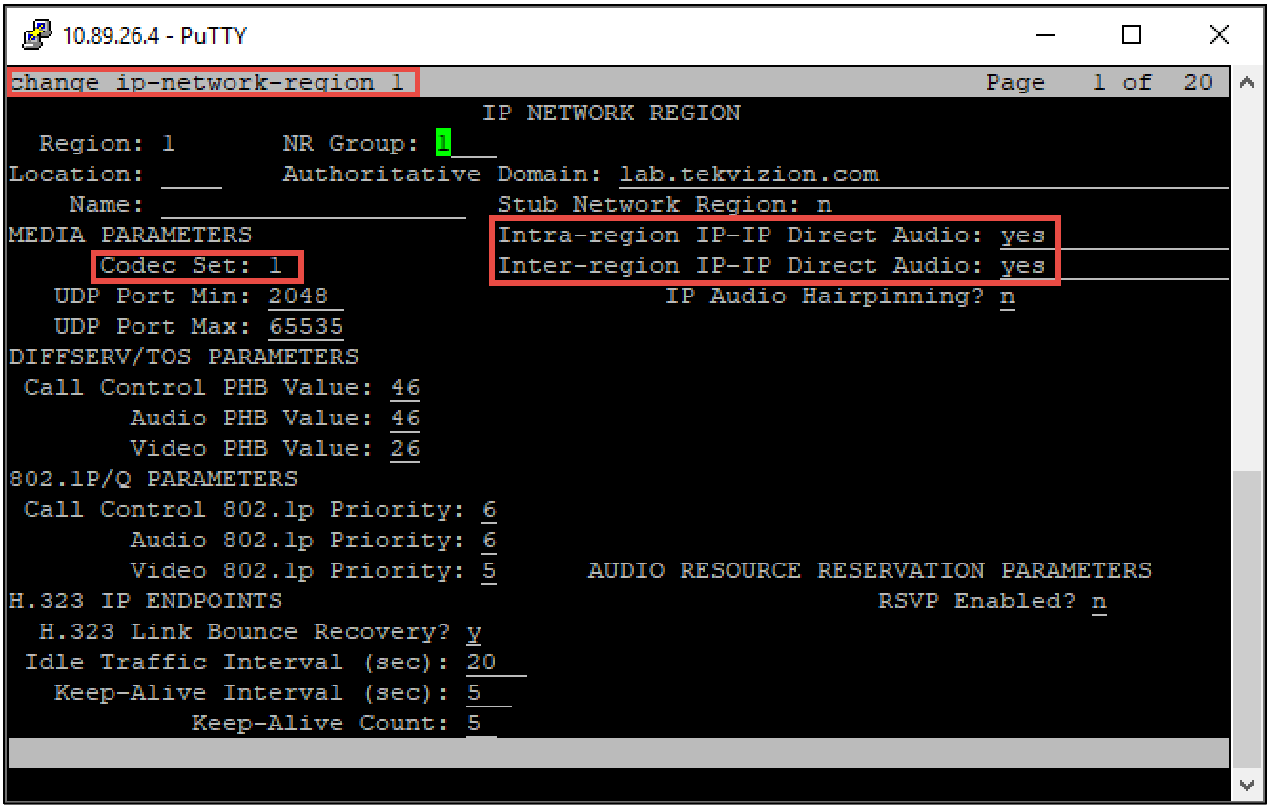


Figure 133 - IP Network Region

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

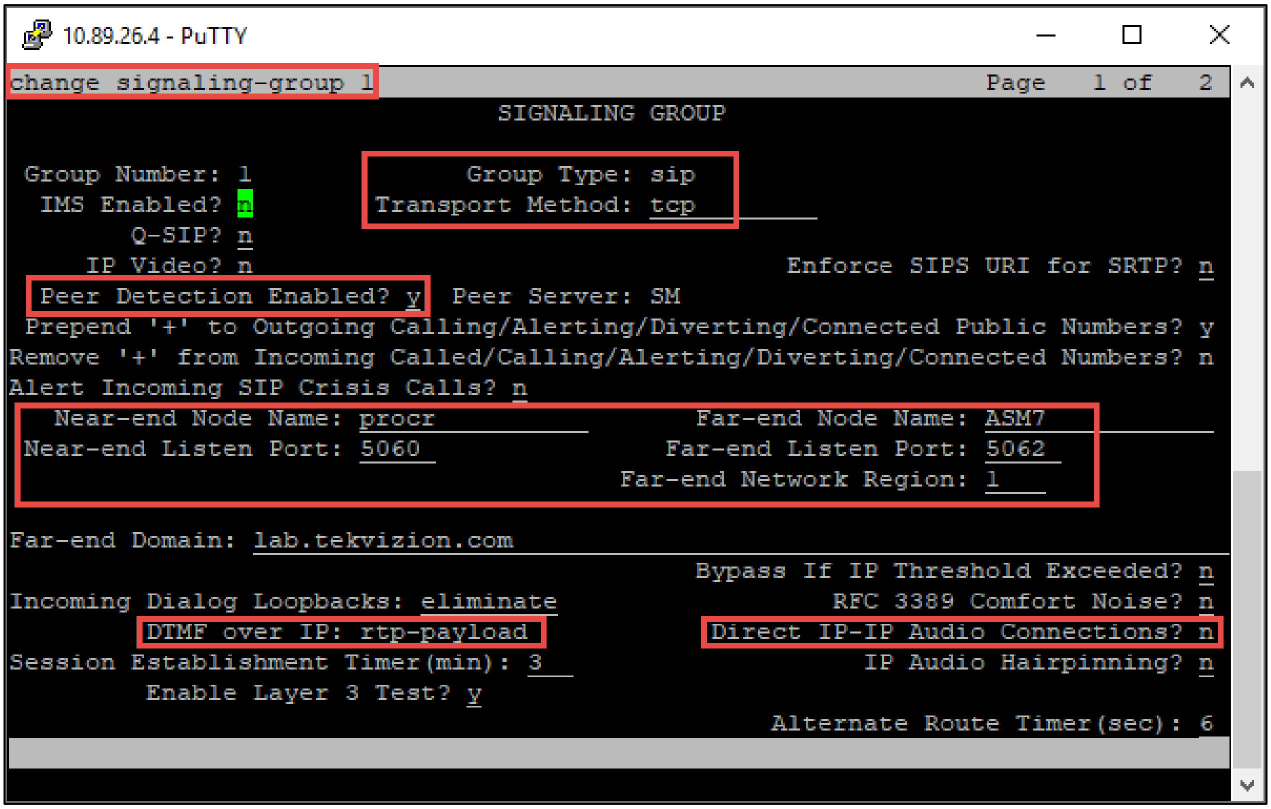


Figure 134 - Signaling Group

4.6.6 Trunk Groups

Similar to Signaling Group, Trunk Group is reated 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

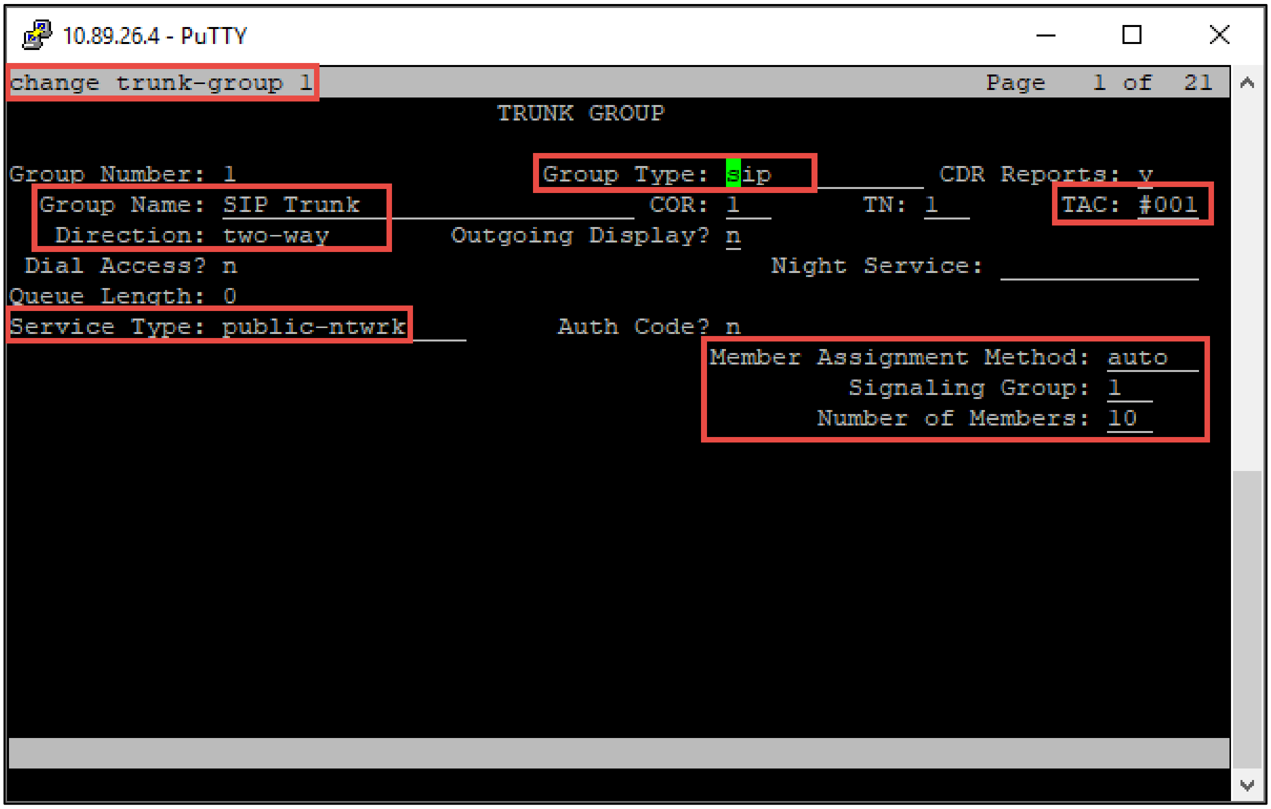


Figure 135 - Trunk Group

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

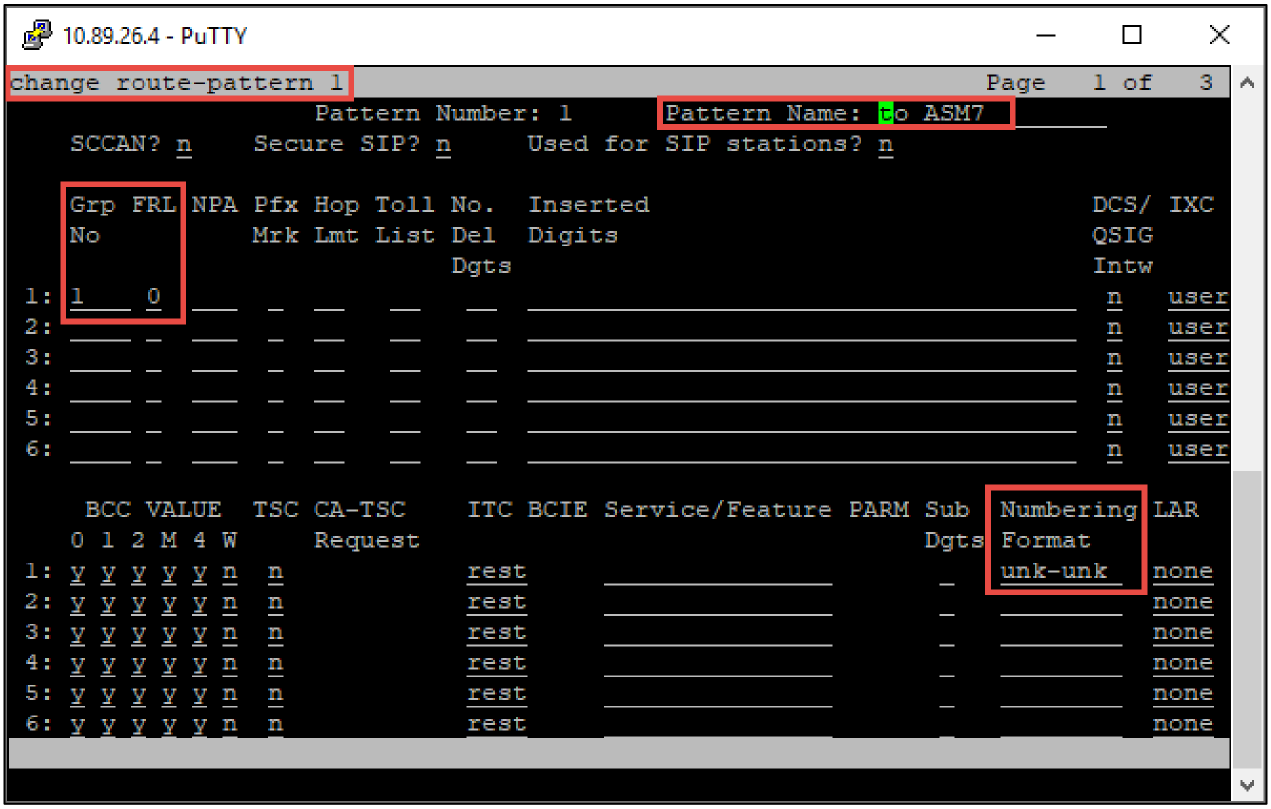
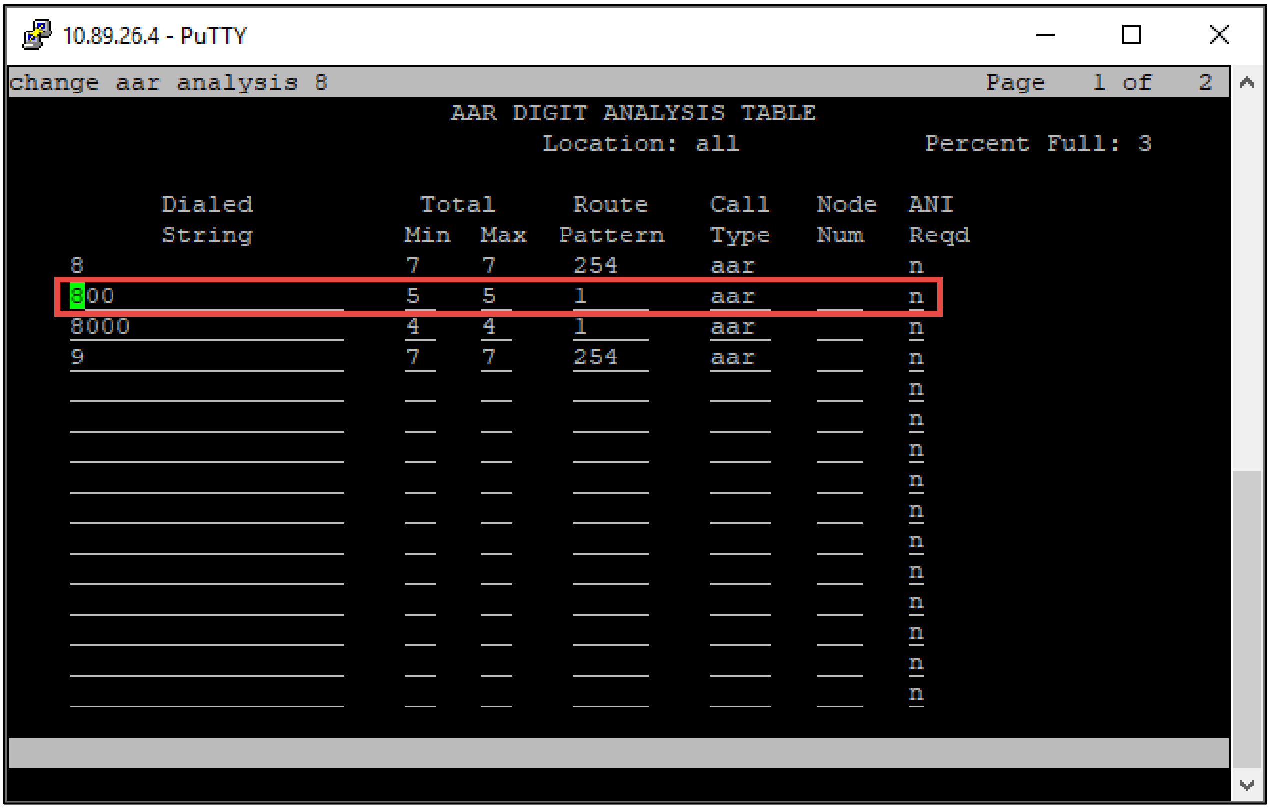


Figure 136 - Route Pattern

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
						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

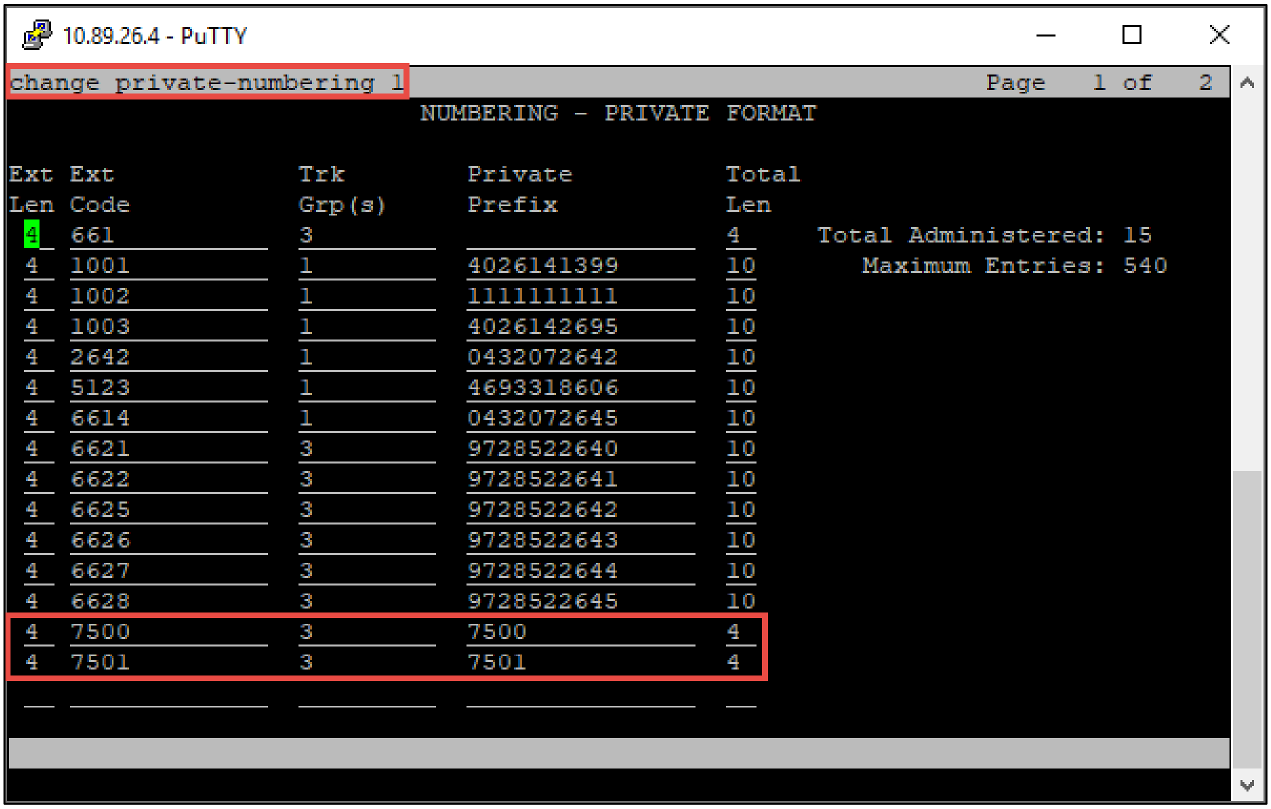


Figure 138 - Inbound Call Routing

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

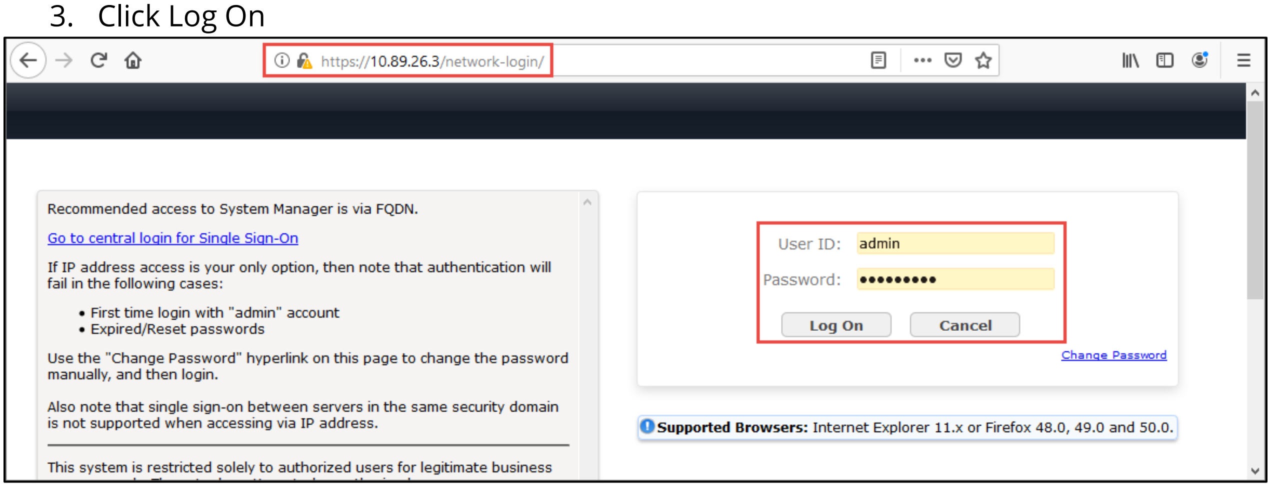


Figure 139 - Log into Avaya Aura System Manager

Navigate to Elements → Routing

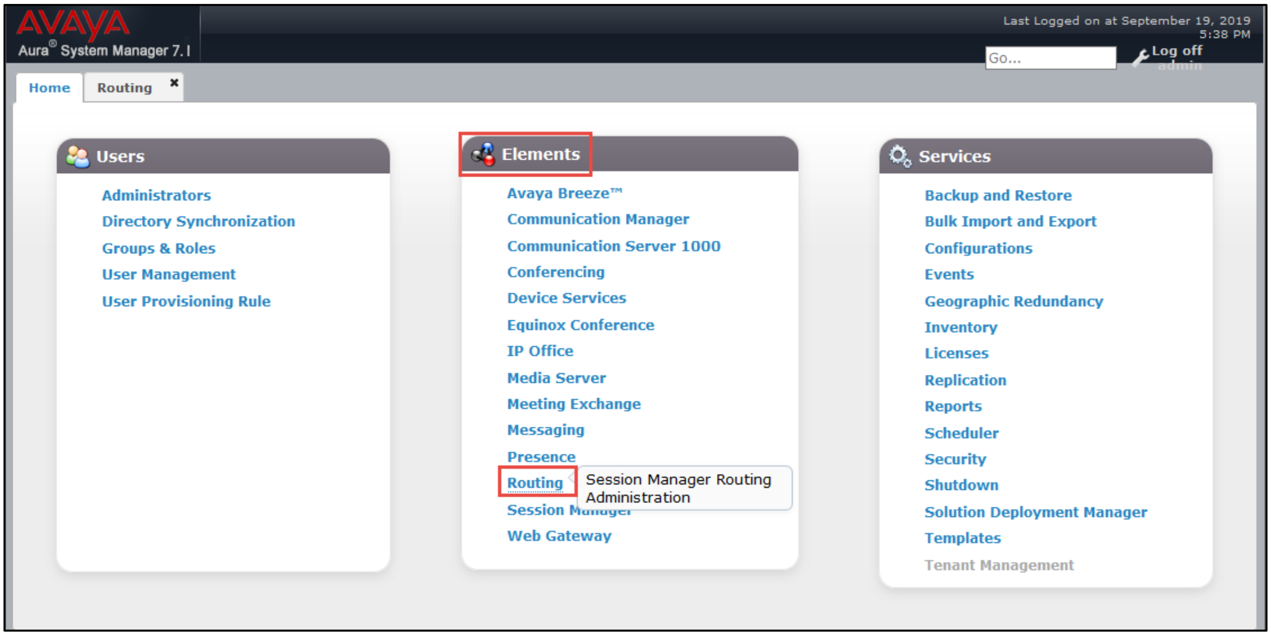


Figure 140 - Routing

4.7.1 Version

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

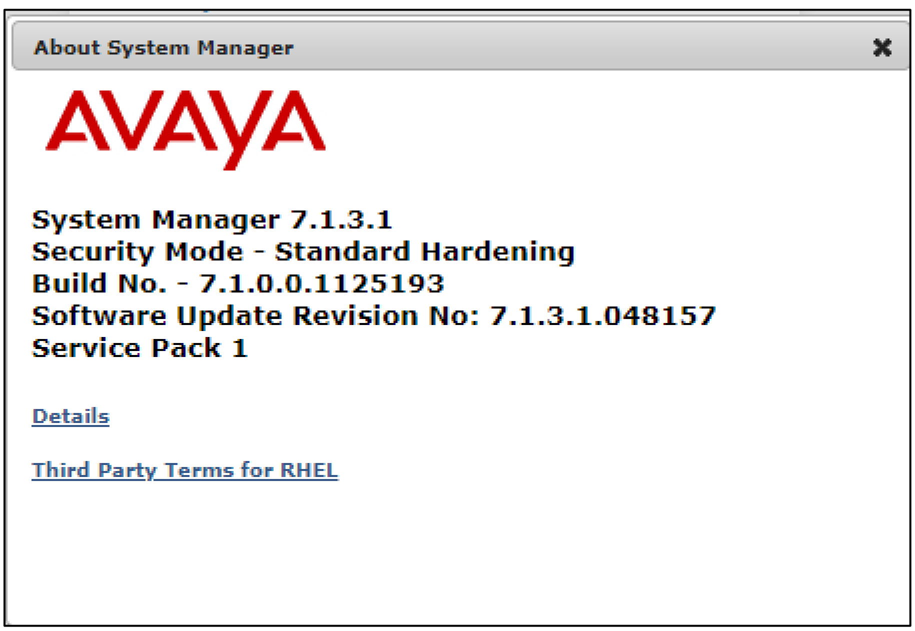


Figure 141 – Version

4.7.2 Domains

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

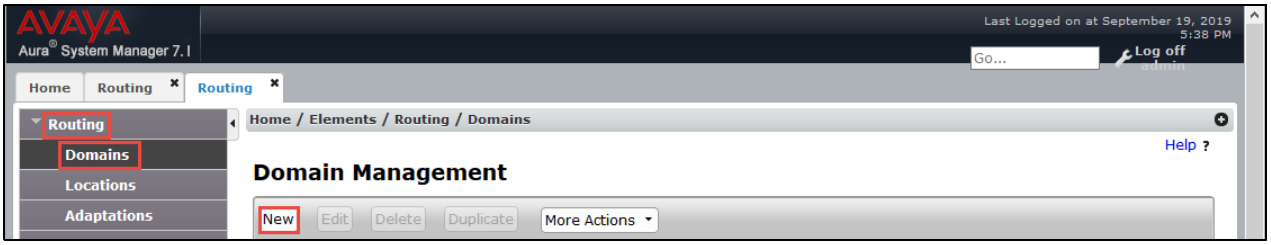


Figure 142 – Add Domain

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**

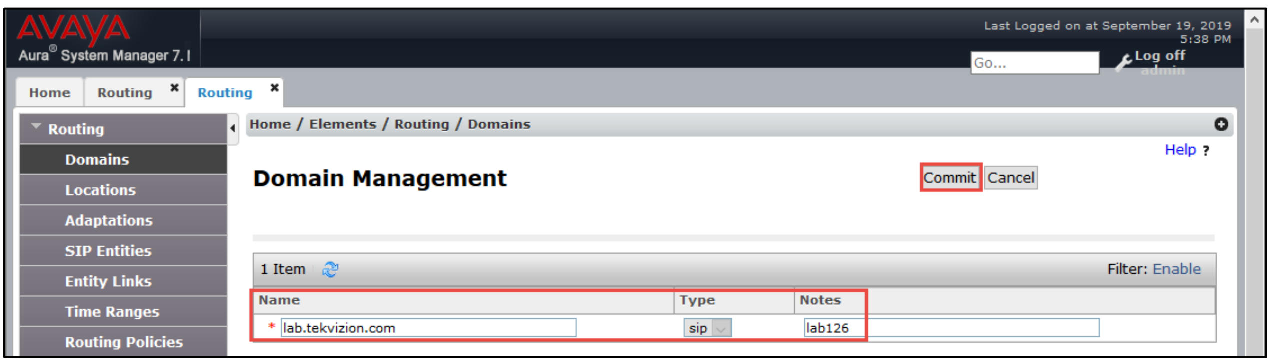


Figure 143 - Domain

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**

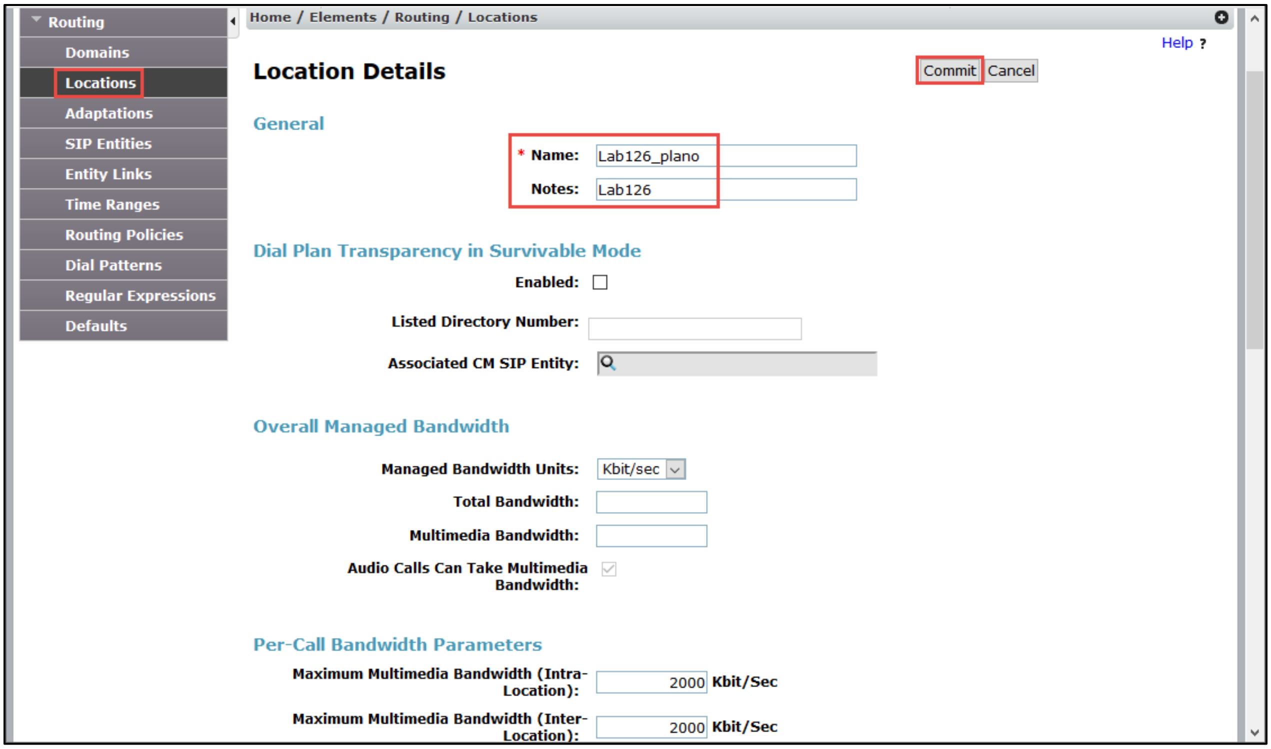


Figure 144- Add Location

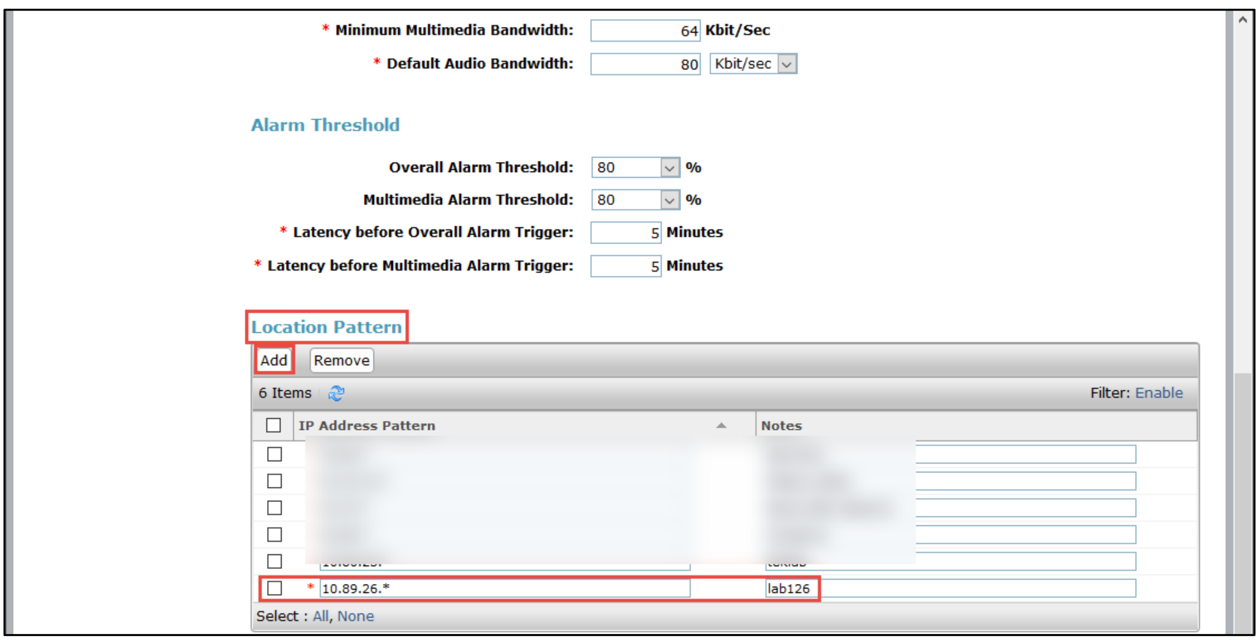


Figure 145 - Add Location

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**

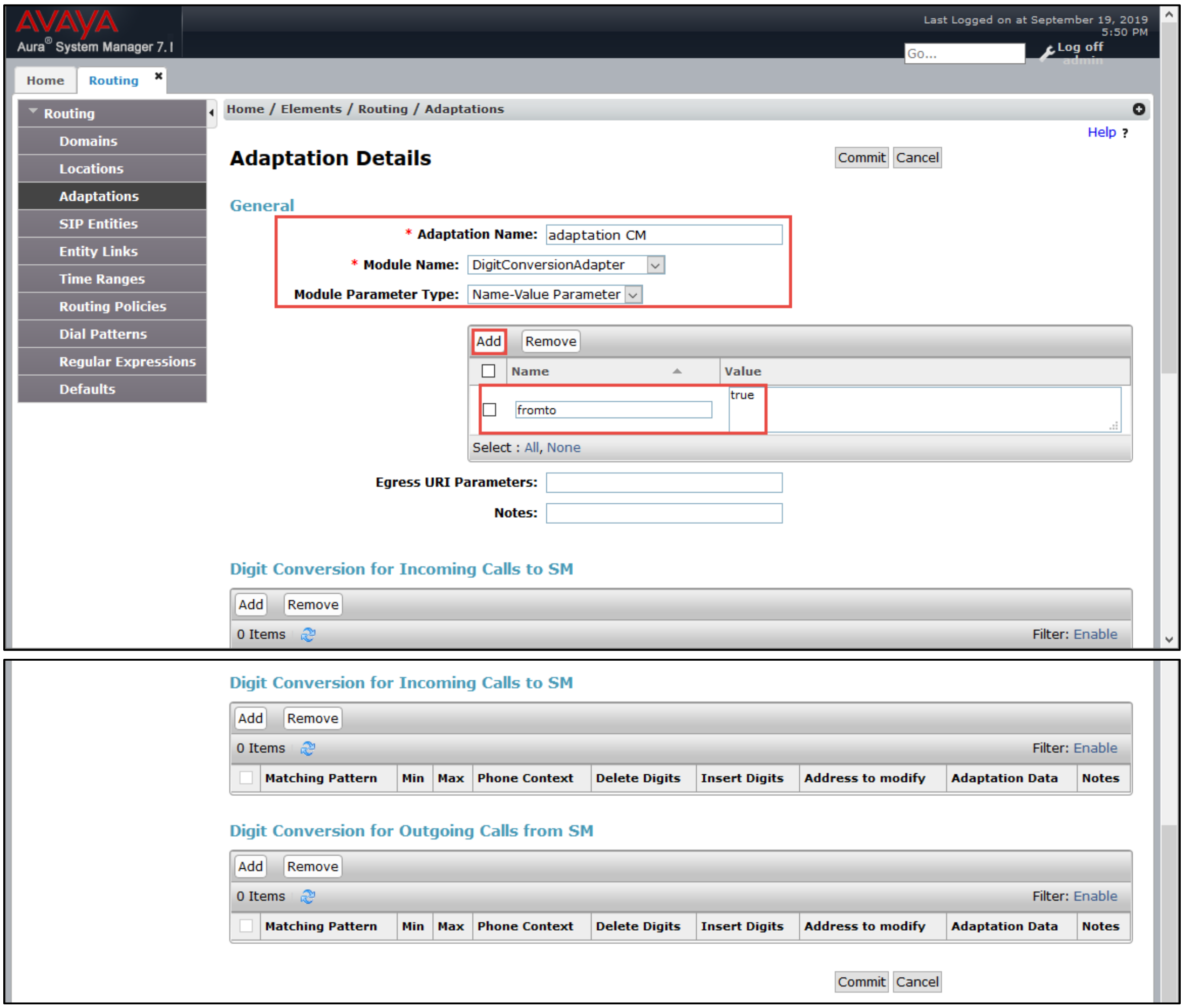


Figure 146 - Add Adaptation

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**

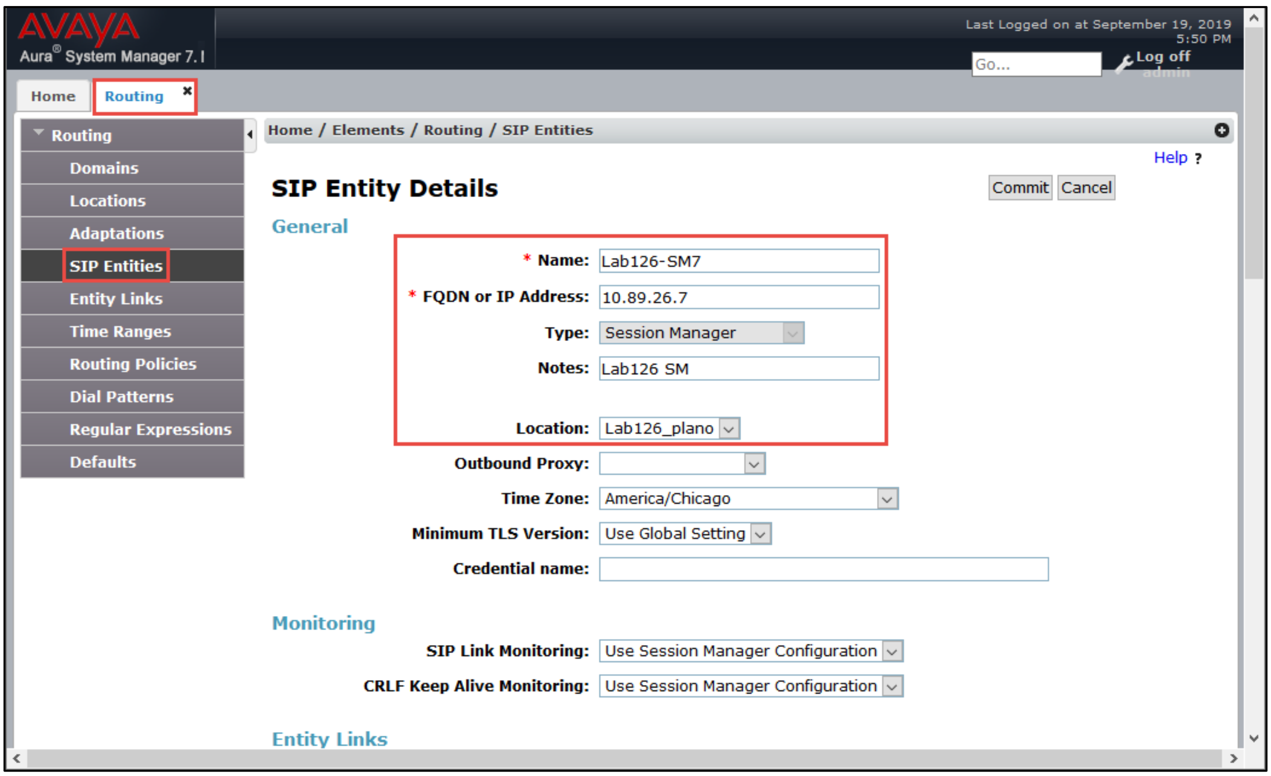


Figure 147 - SIP Entity: Avaya Aura Session Manager

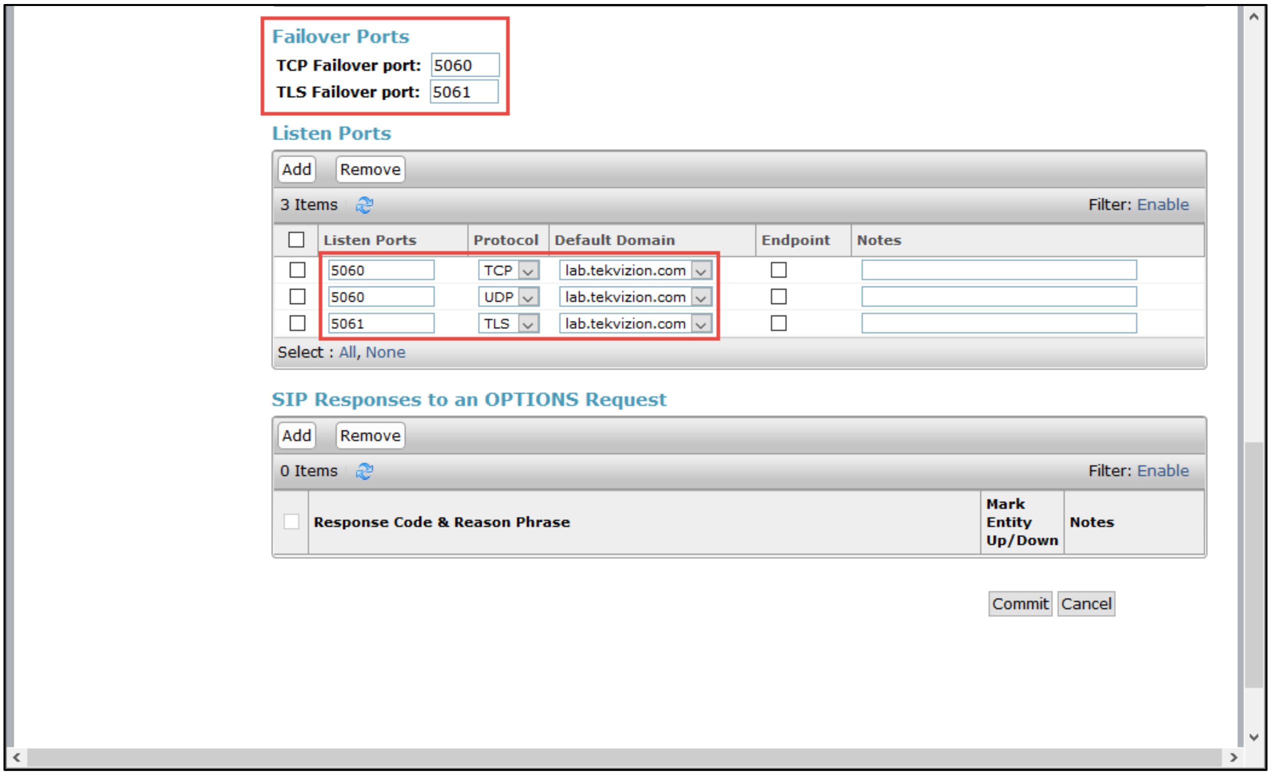


Figure 148 - SIP Entity: Avaya Aura Session Manager

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**

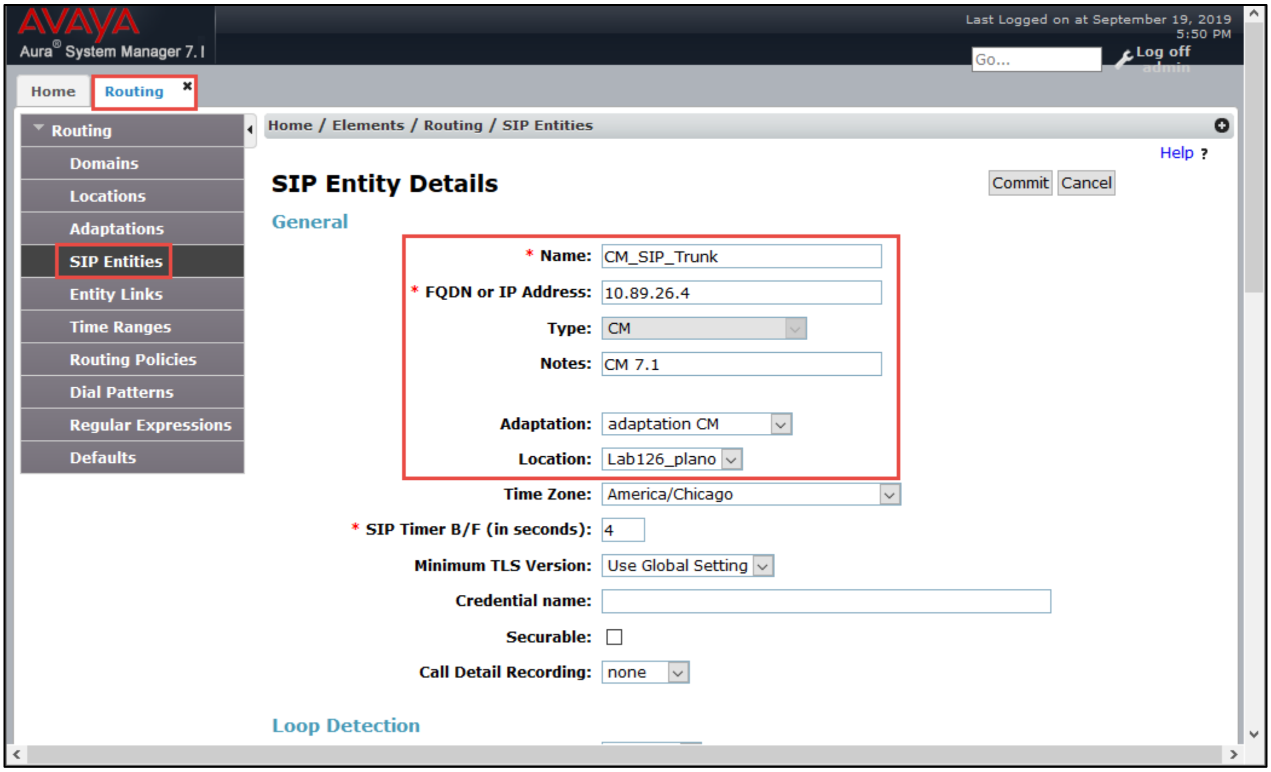


Figure 149- SIP Entity: Avaya Aura Communication Manager for SIP Trunk

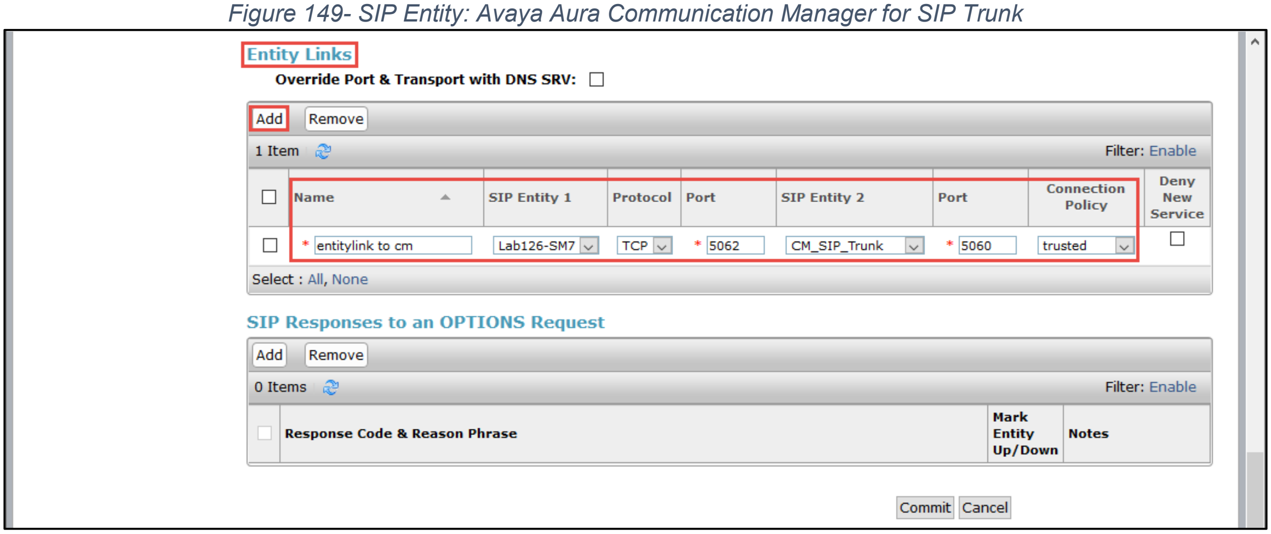


Figure 150 - SIP Entity: Avaya Aura Communication Manager for SIP Trunk

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 ABSC_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**

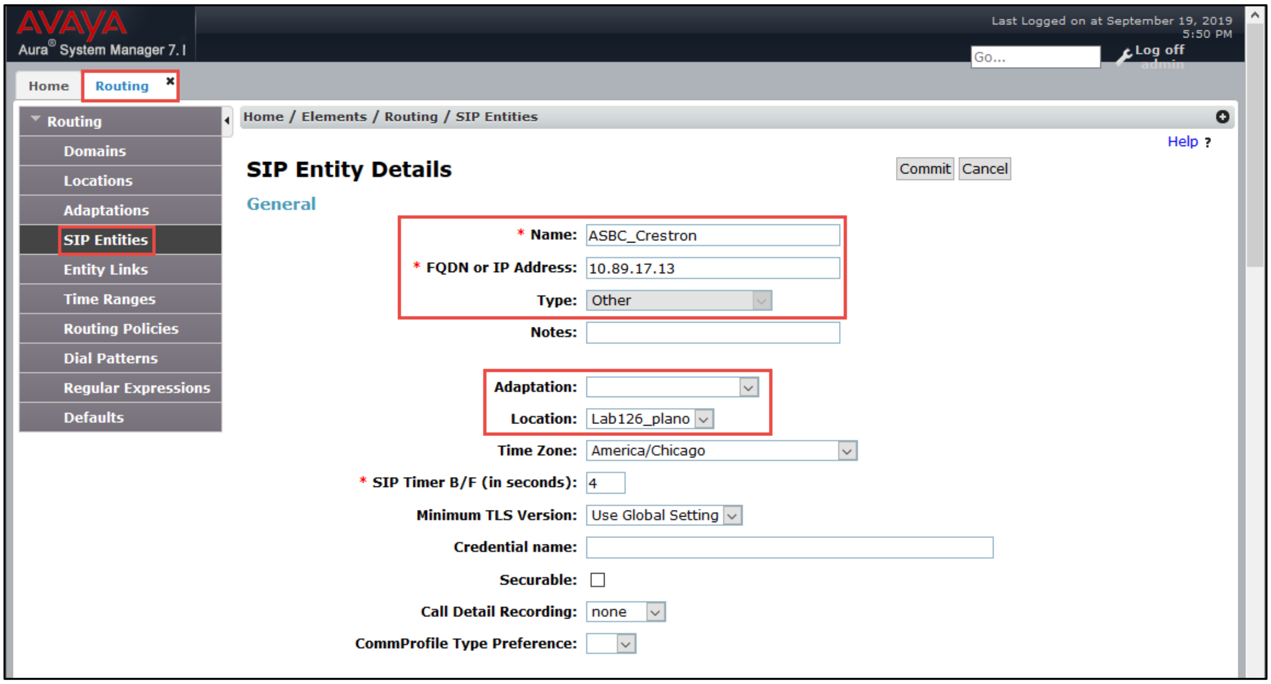


Figure 151 - SIP Entity: Avaya SBCE

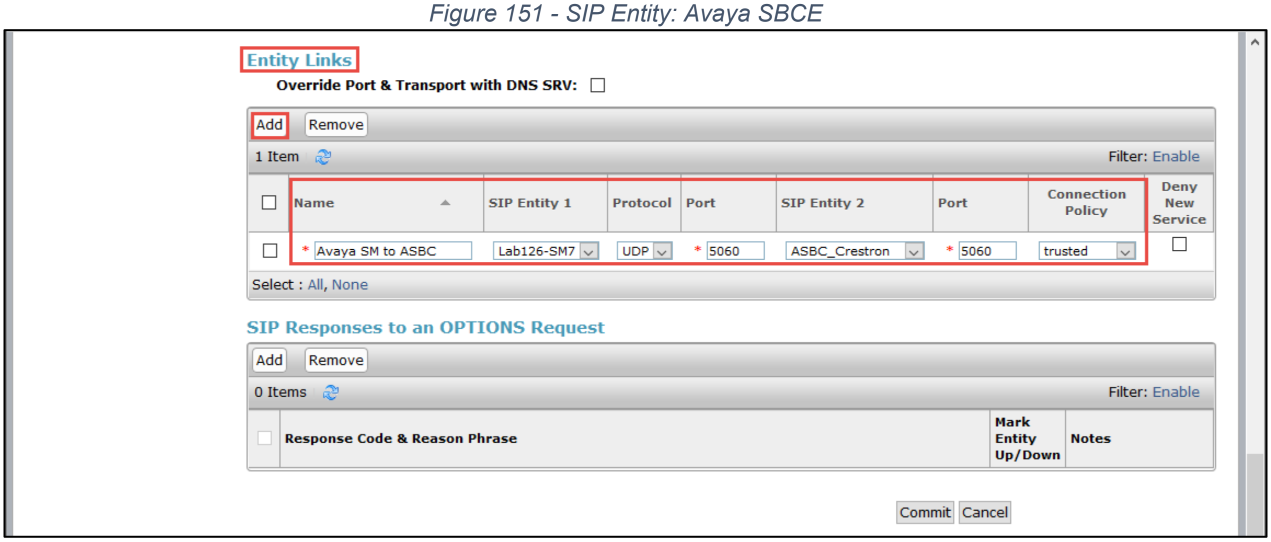


Figure 152 - SIP Entity: Avaya SBCE

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**

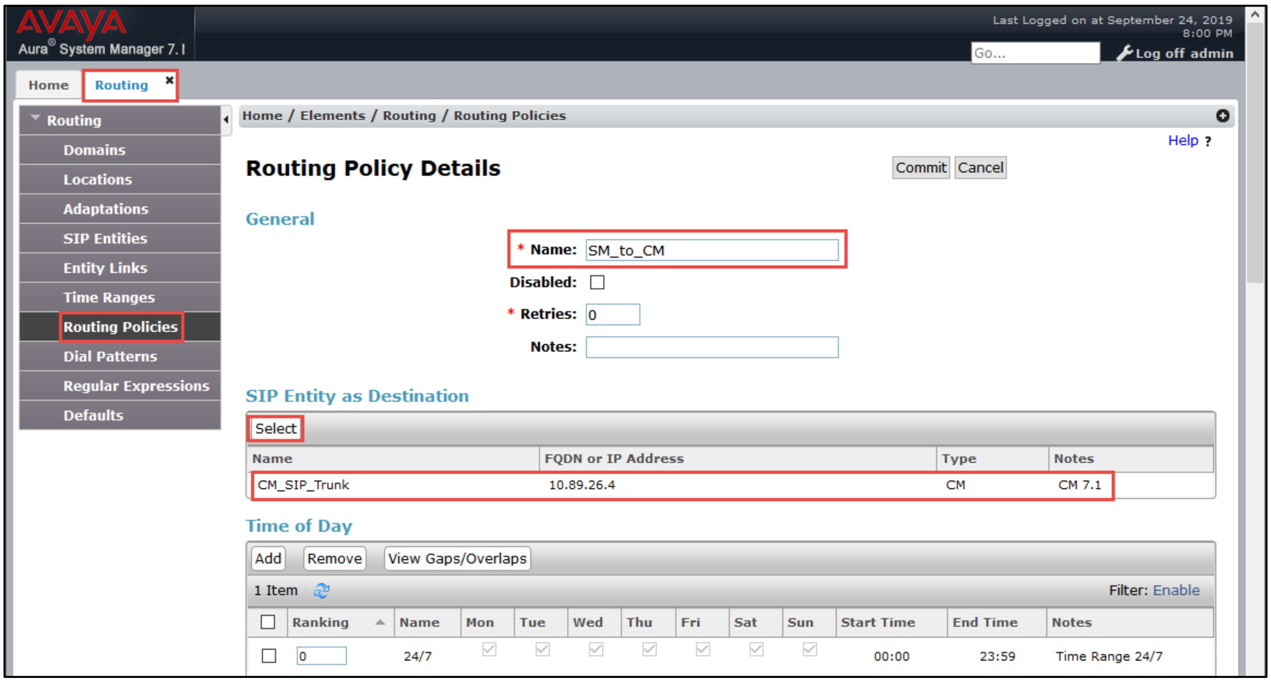


Figure 153 - Routing Policy to Avaya Aura Communication Manager

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**

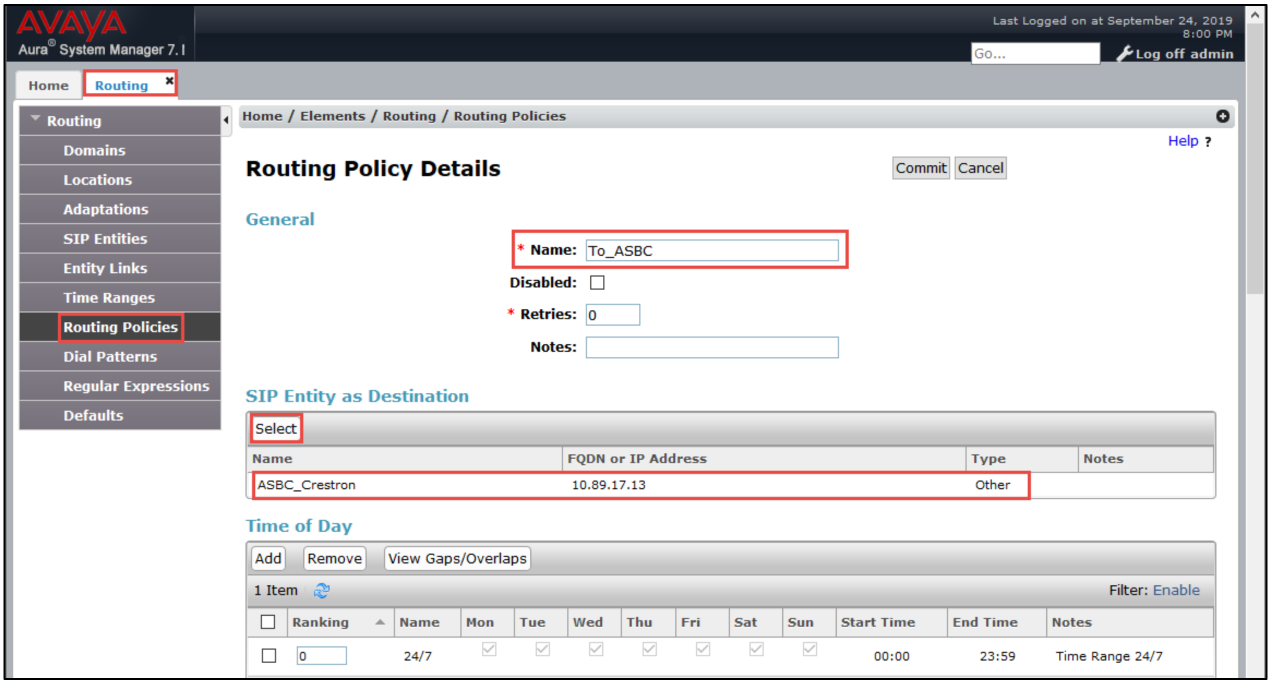


Figure 154 - Routing Policy to Avaya SBCE

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**

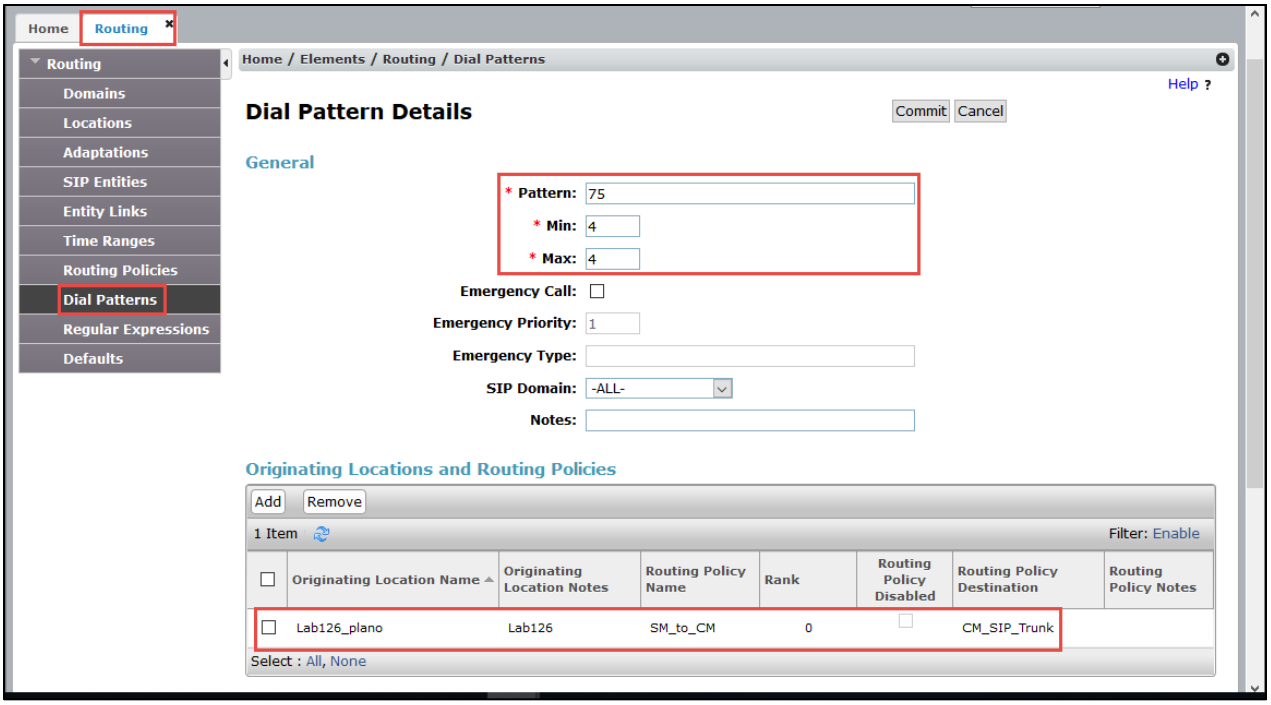


Figure 155 - Dial Pattern to Avaya Aura Communication Manager

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**

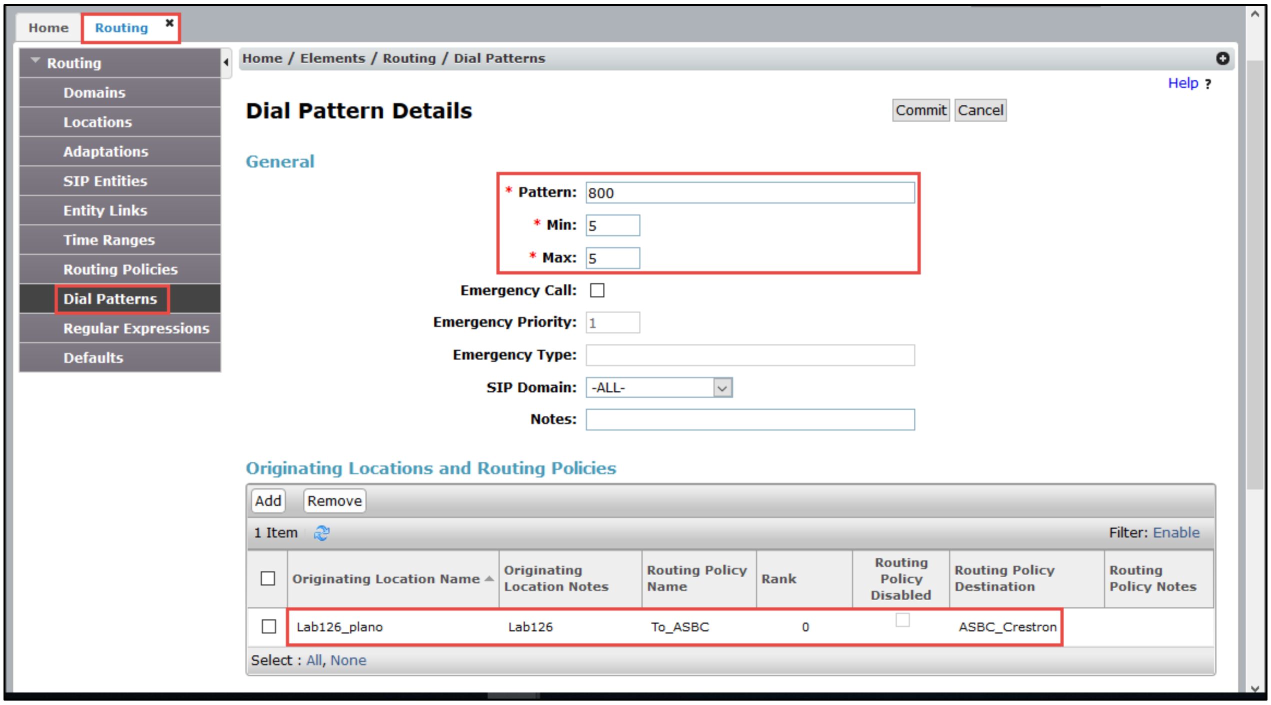


Figure 156 - Dial Pattern to Avaya SBCE

4.8 Avaya SBCE Configuration

4.8.1 Version

The following version of Avaya SBCE is used for this testing

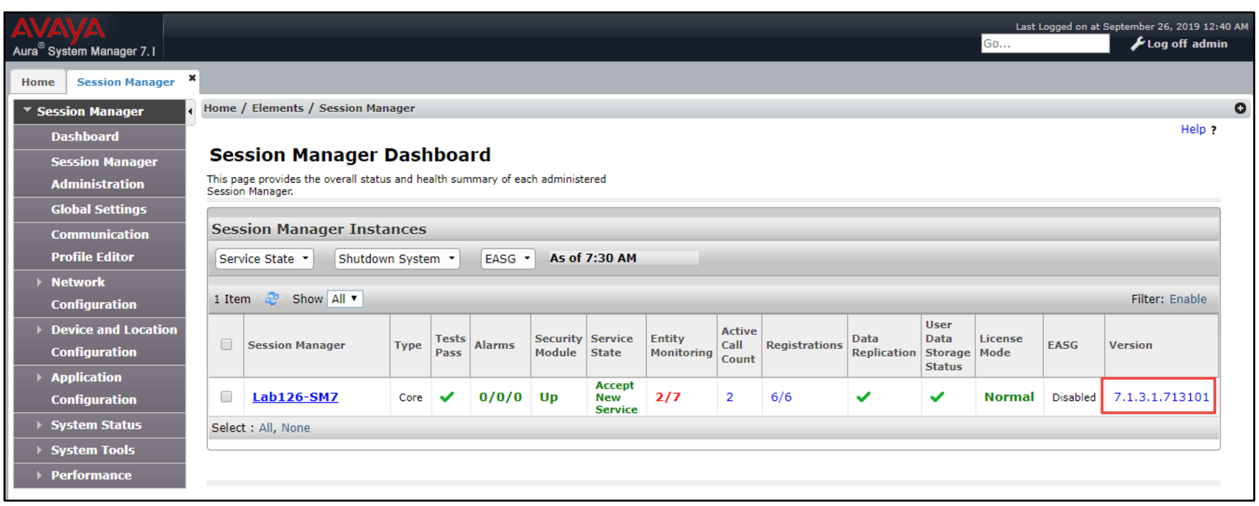


Figure 157 – Version

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

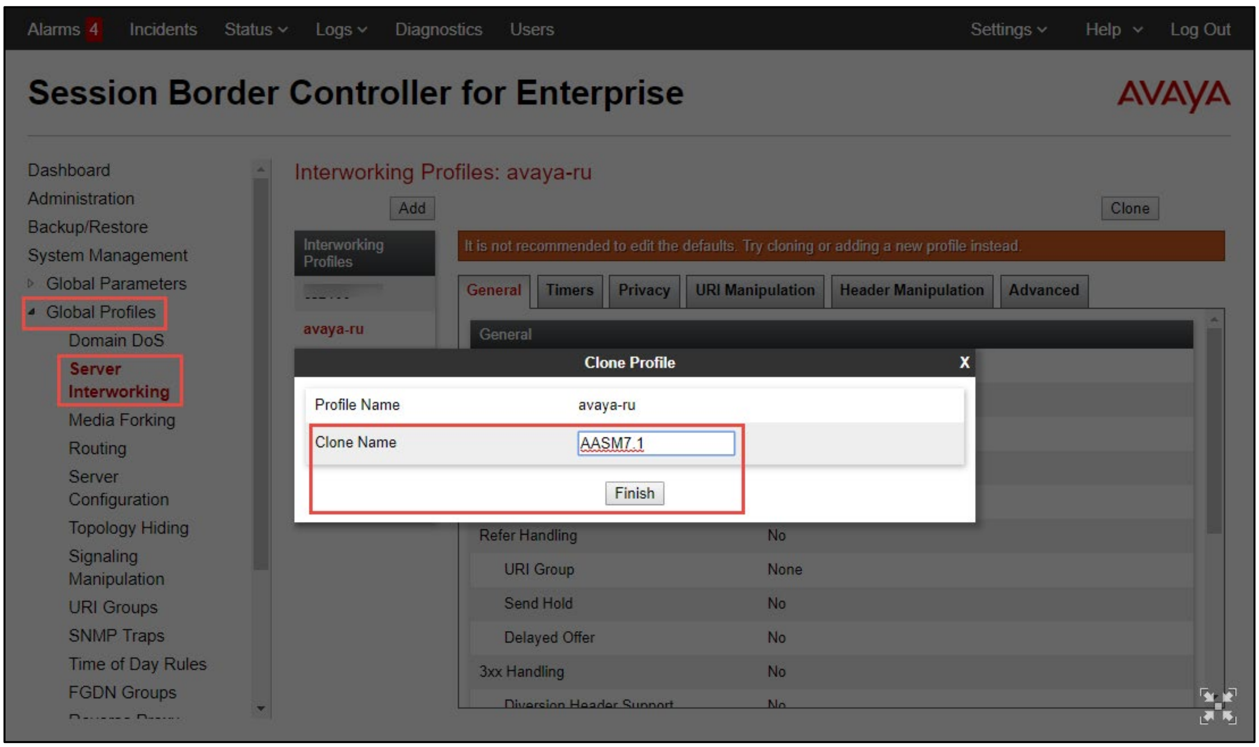


Figure 158 - Server Interworking for Avaya

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

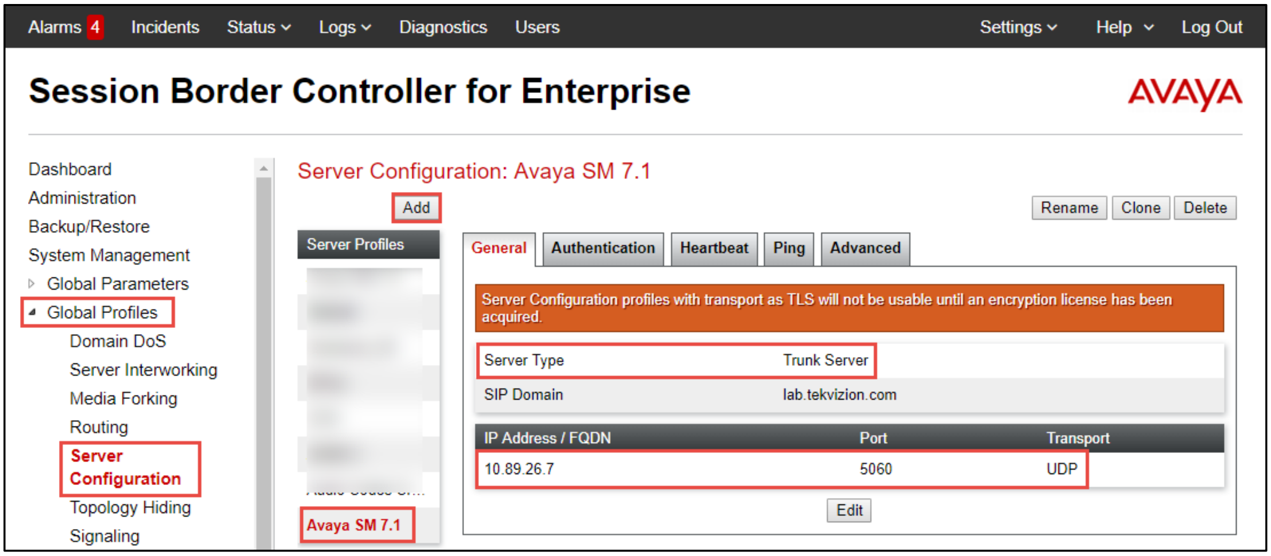


Figure 159- Add SIP Server – Avaya SM

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

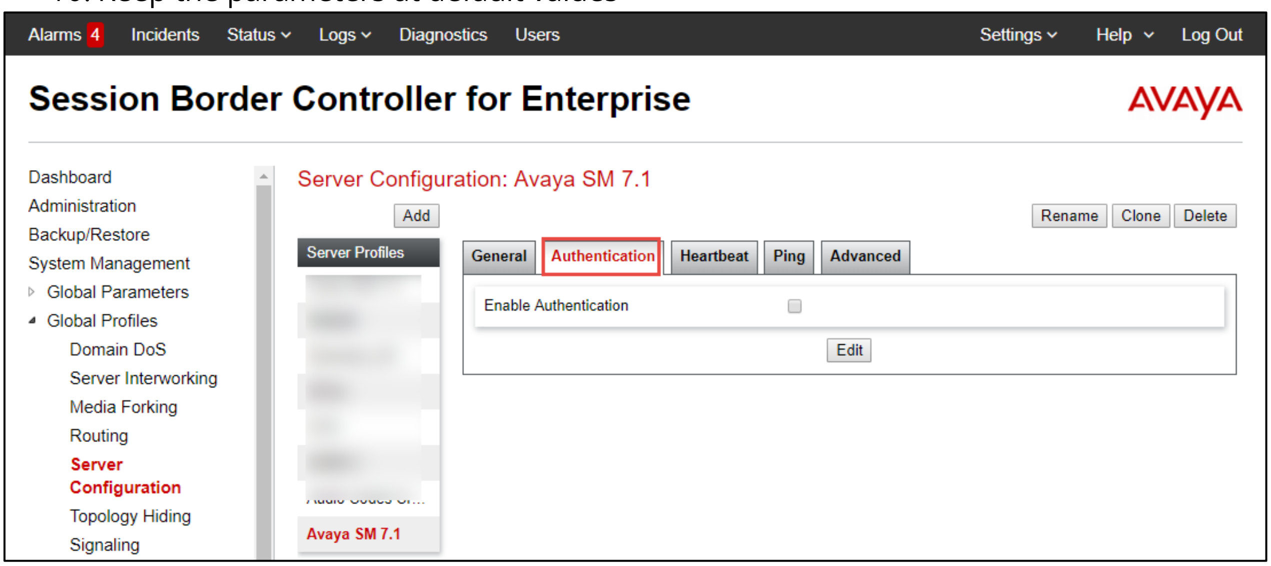


Figure 160 - Add SIP Server – Avaya SM

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

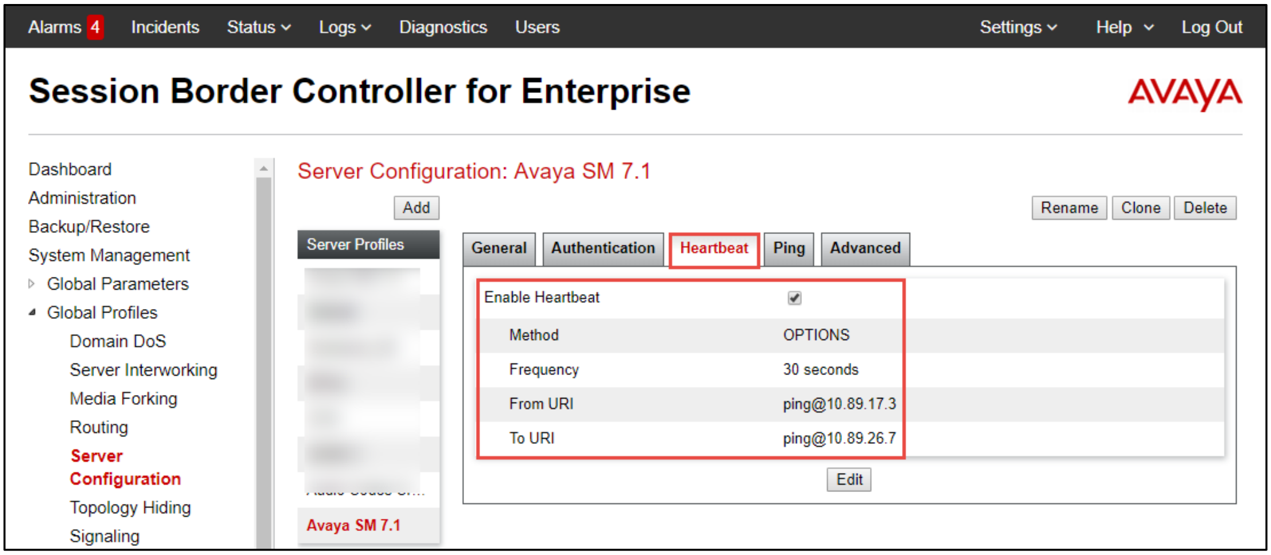


Figure 161 - Add SIP Server – Avaya SM

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

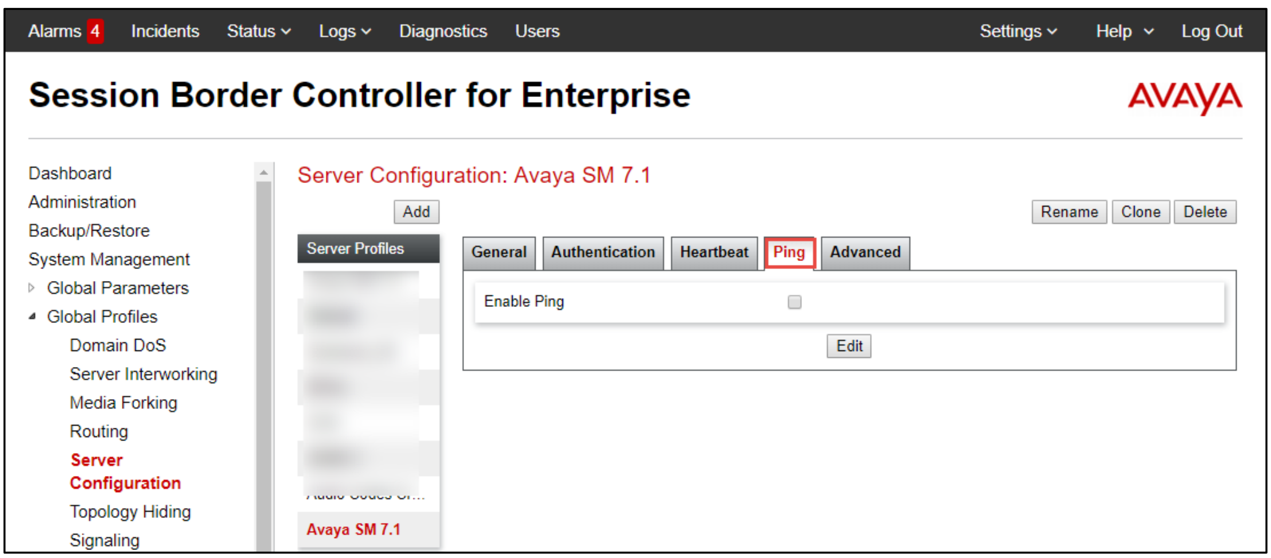


Figure 162 - Add SIP Server – Avaya SM

17. Select **Advanced**
18. Keep the parameters at default values

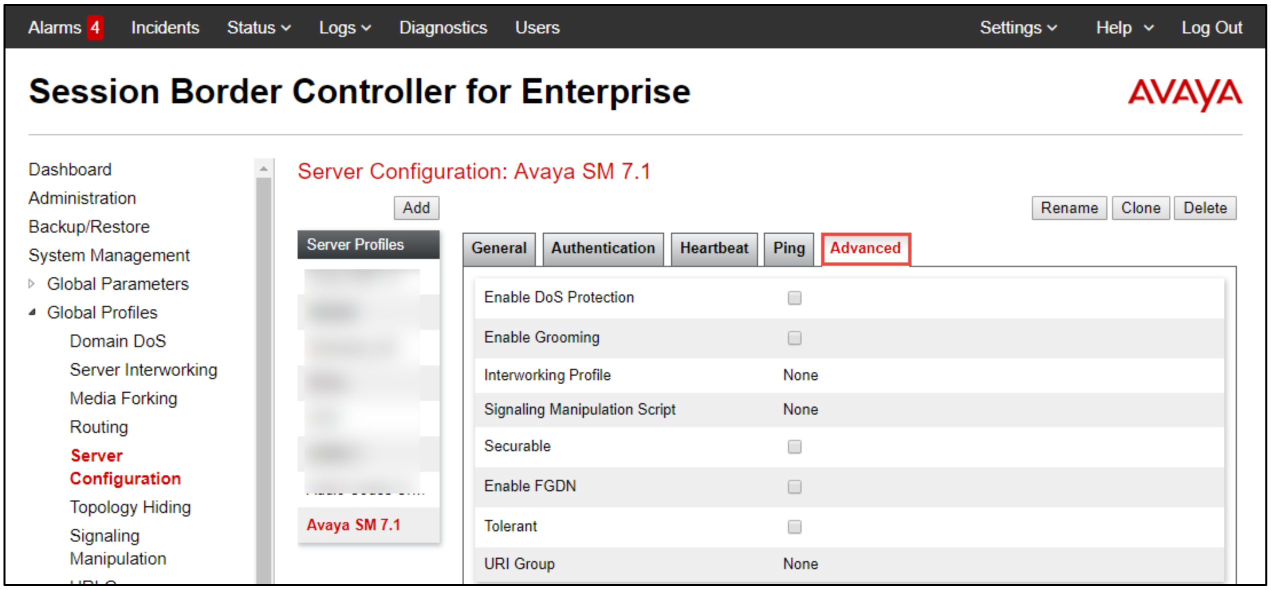


Figure 163 - Add SIP Server – Avaya SM

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

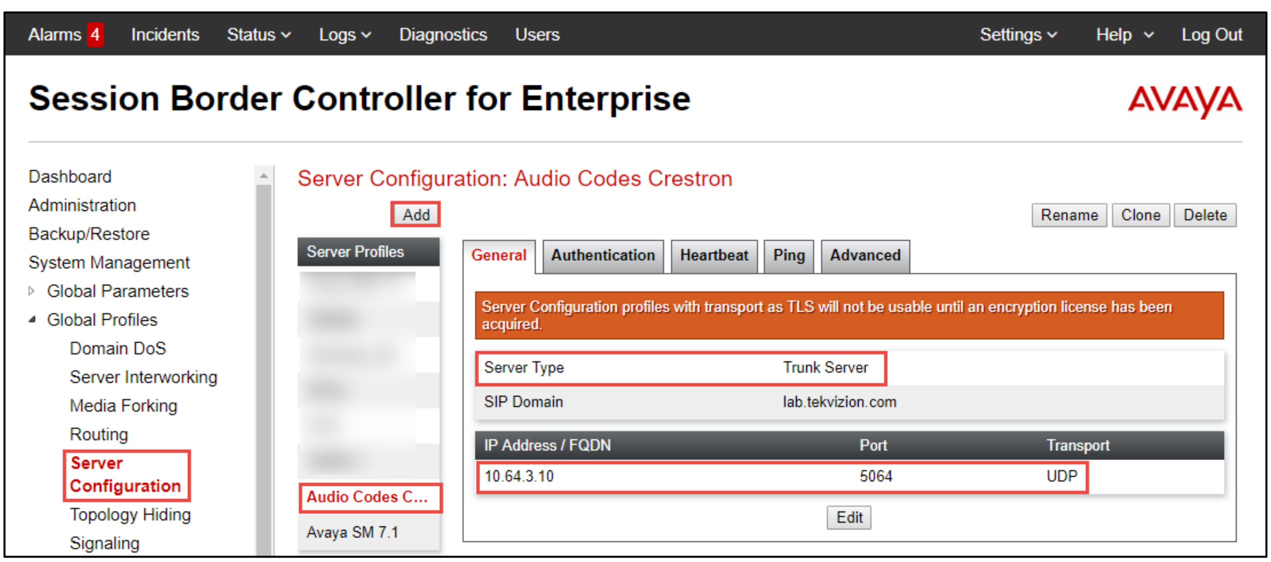


Figure 164 - Add SIP Server – AudioCodes

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

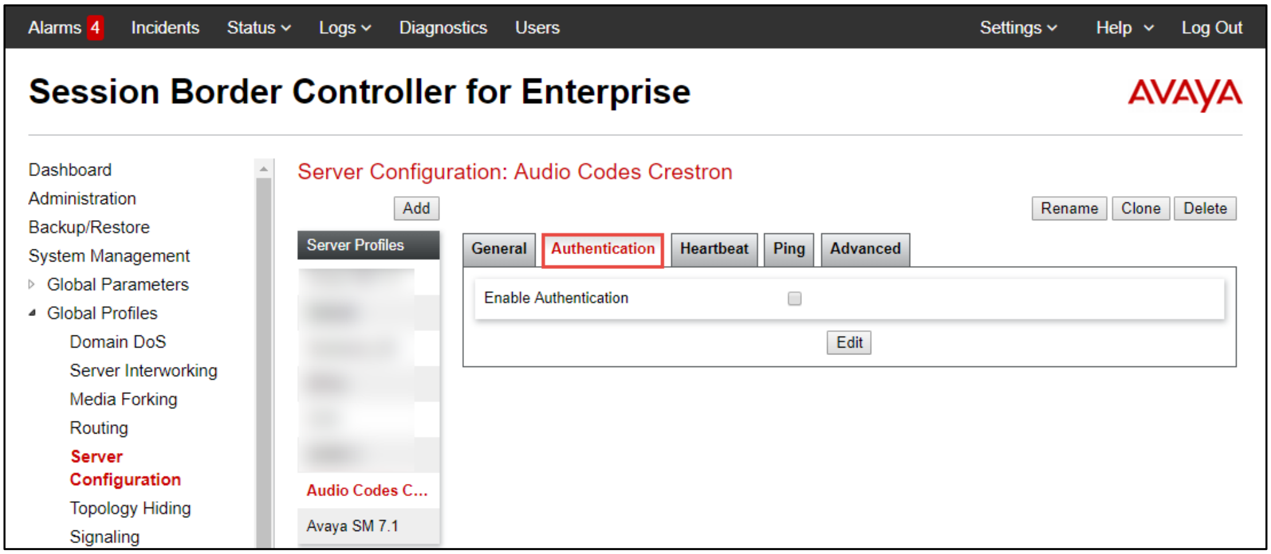


Figure 165 - Add SIP Server – AudioCodes

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

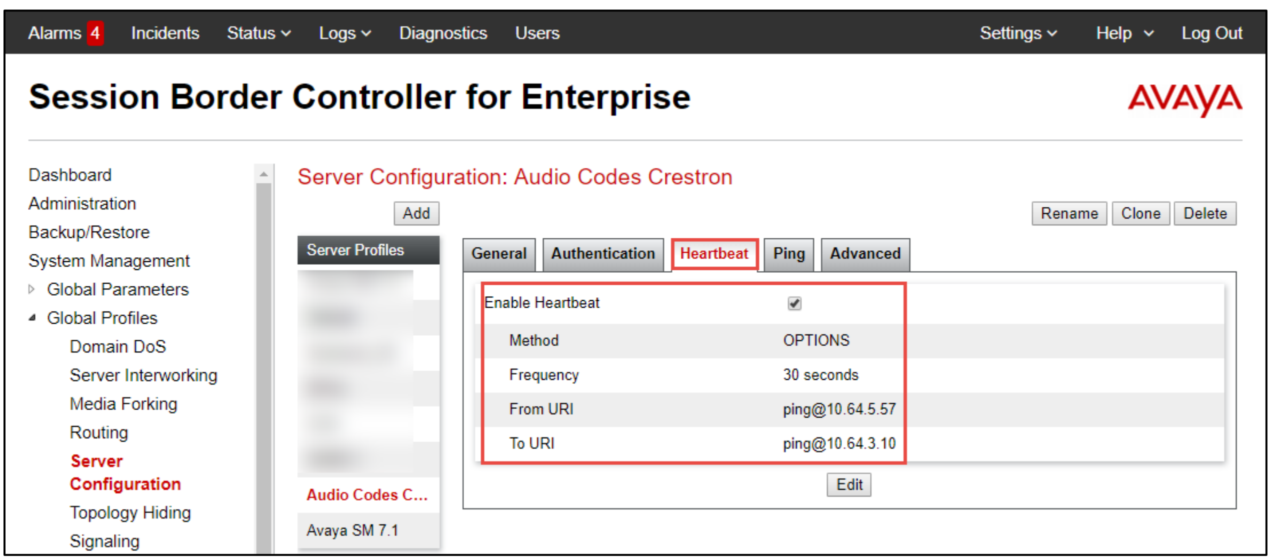


Figure 166 - Add SIP Server – AudioCodes

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

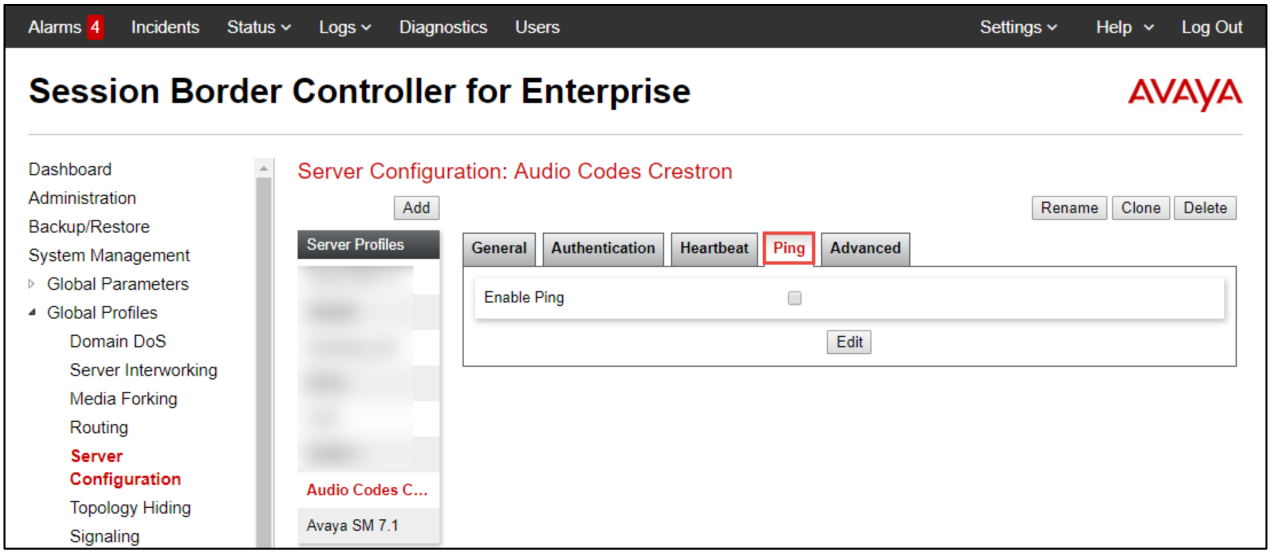


Figure 167 - Add SIP Server – AudioCodes

17. Select **Advanced**

18. Keep the parameters at default values

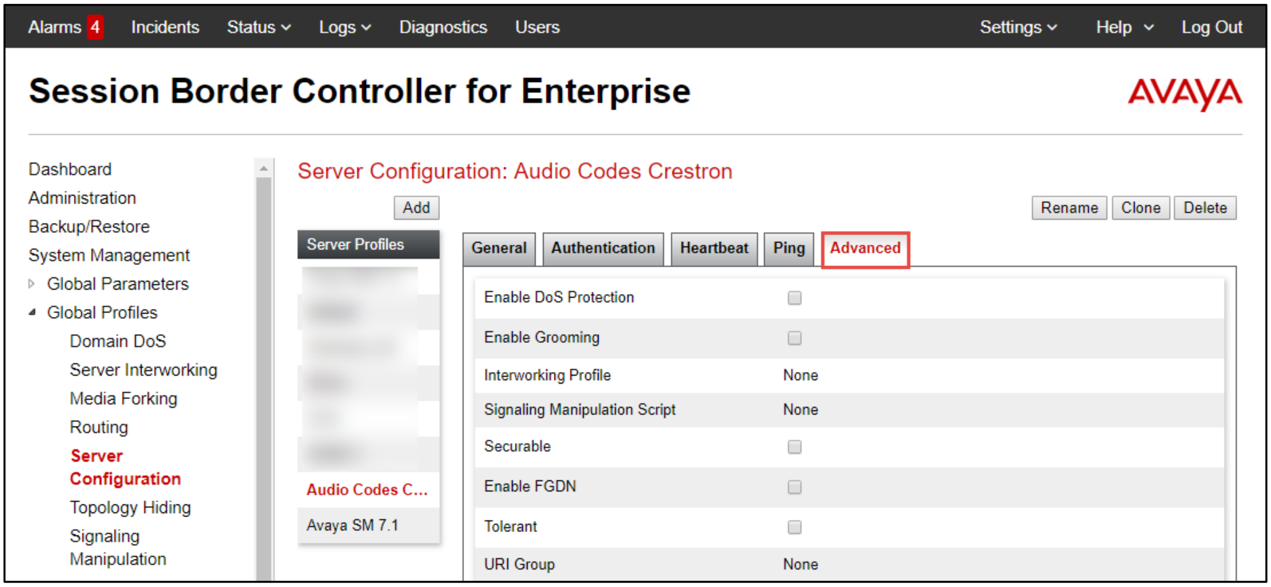


Figure 168 - Add SIP Server – AudioCodes

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.

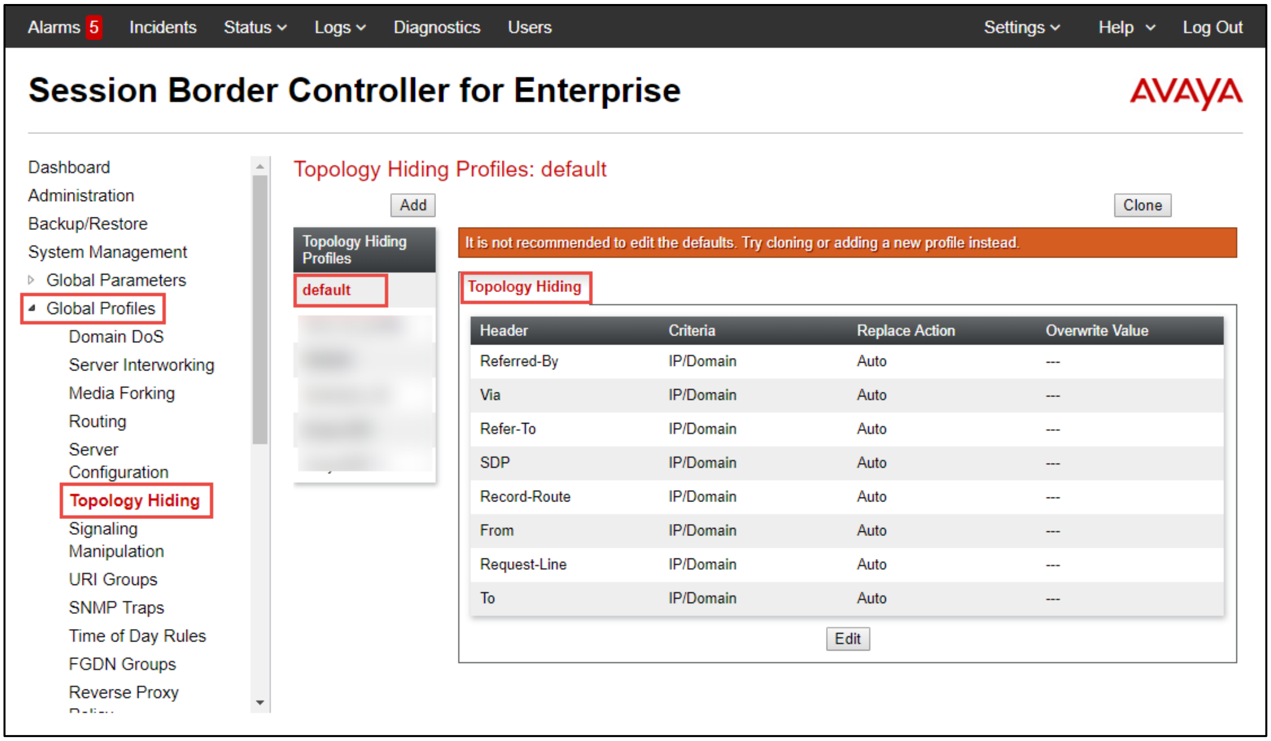


Figure 169 - Topology Hiding

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**

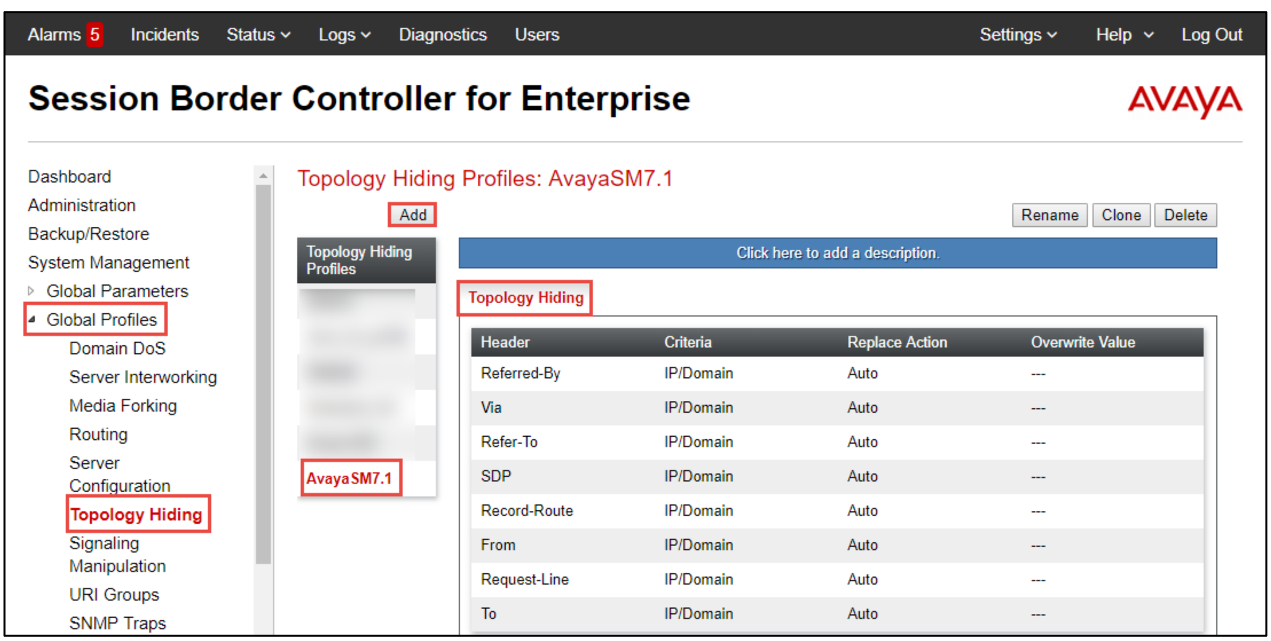


Figure 170 - Topology Hiding

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**

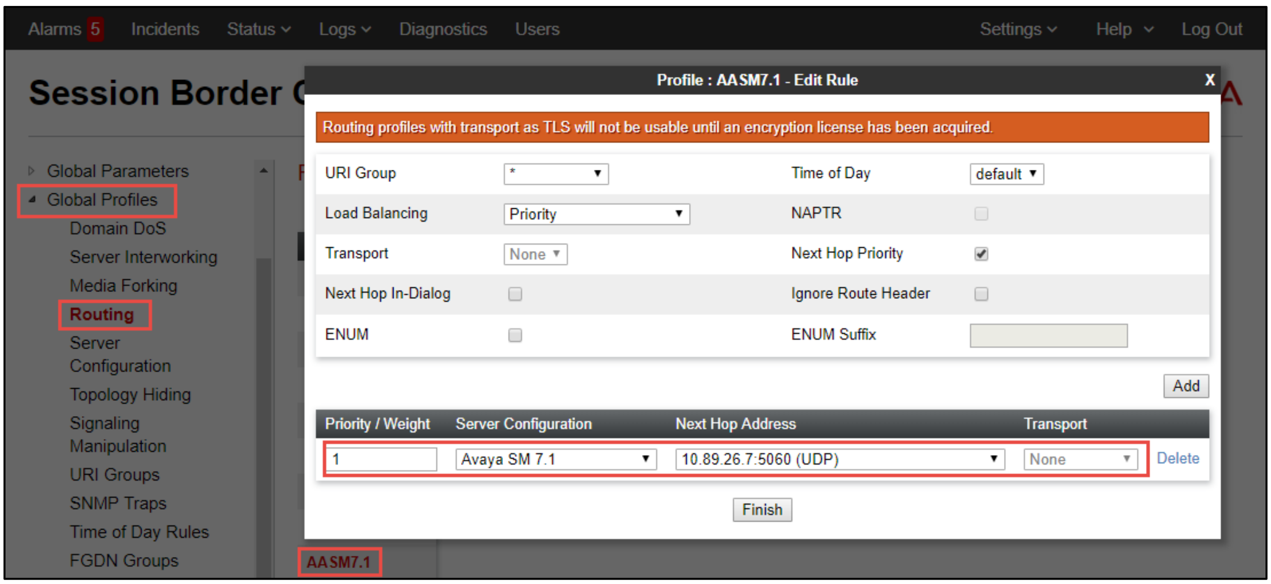


Figure 171 - Routing Profile – Avaya SM

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

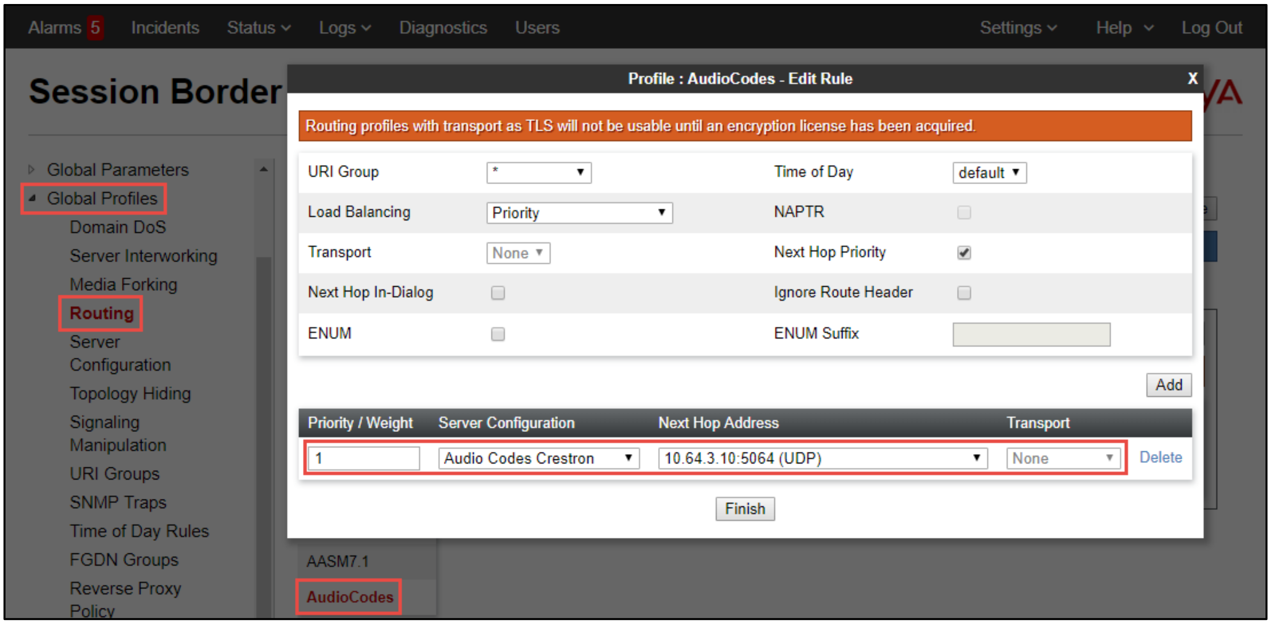


Figure 172 - Routing Profile – AudioCodes

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**

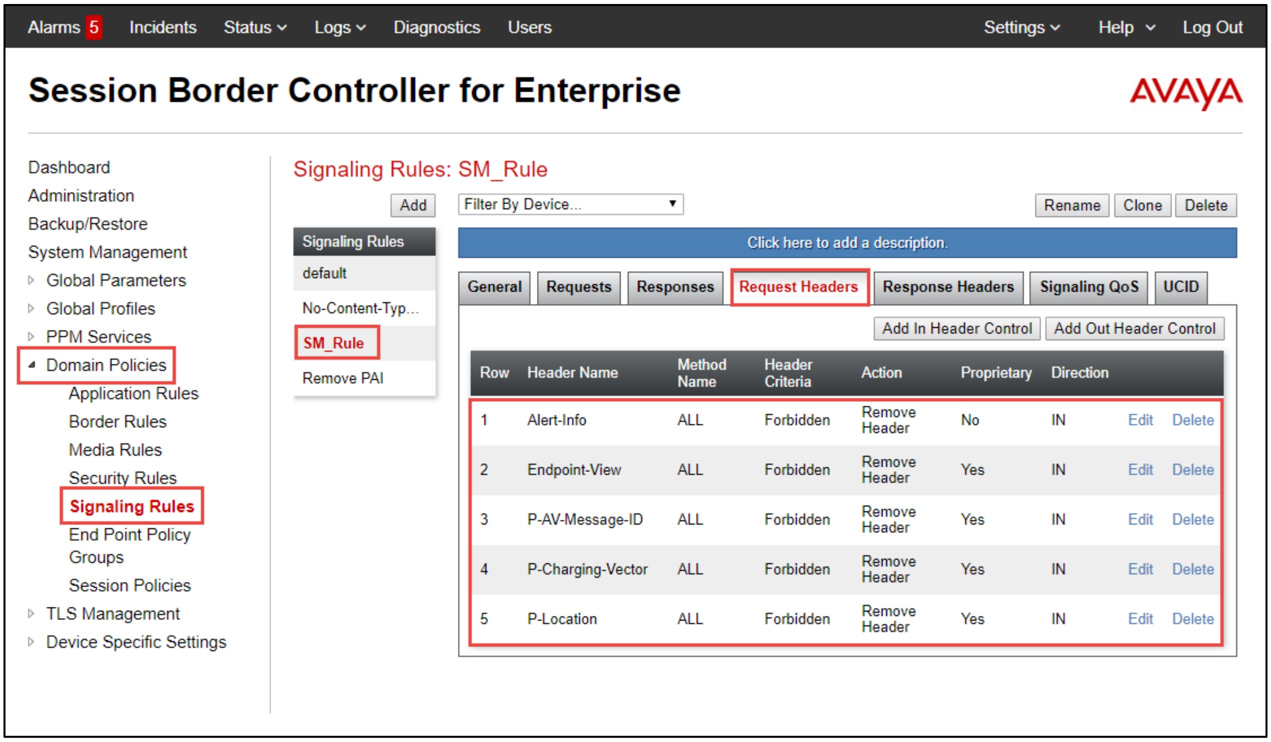


Figure 173 - Signaling Rule – Avaya SM

7. Repeat the same for Response Headers also

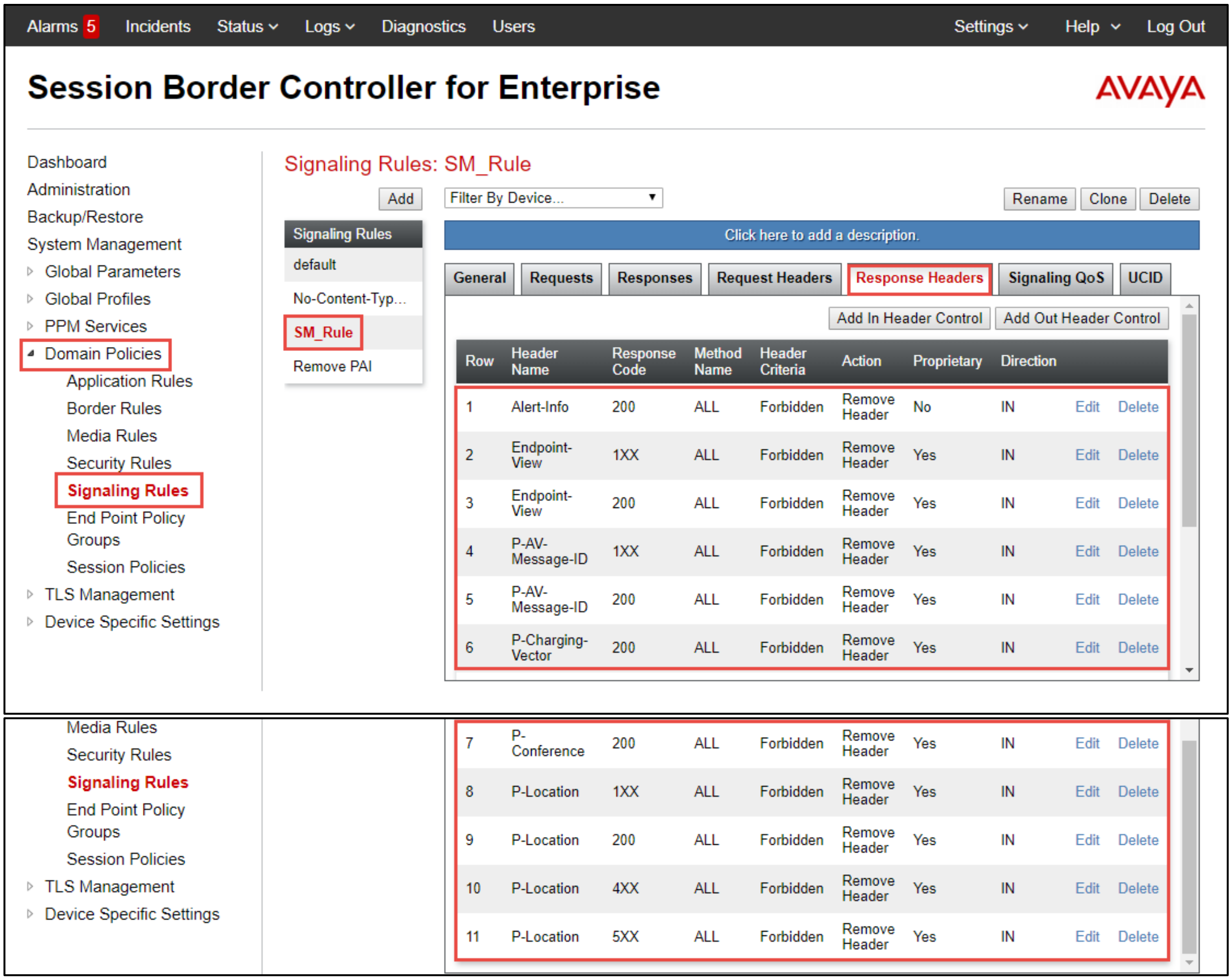


Figure 174- Signaling Rule – Avaya SM

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.

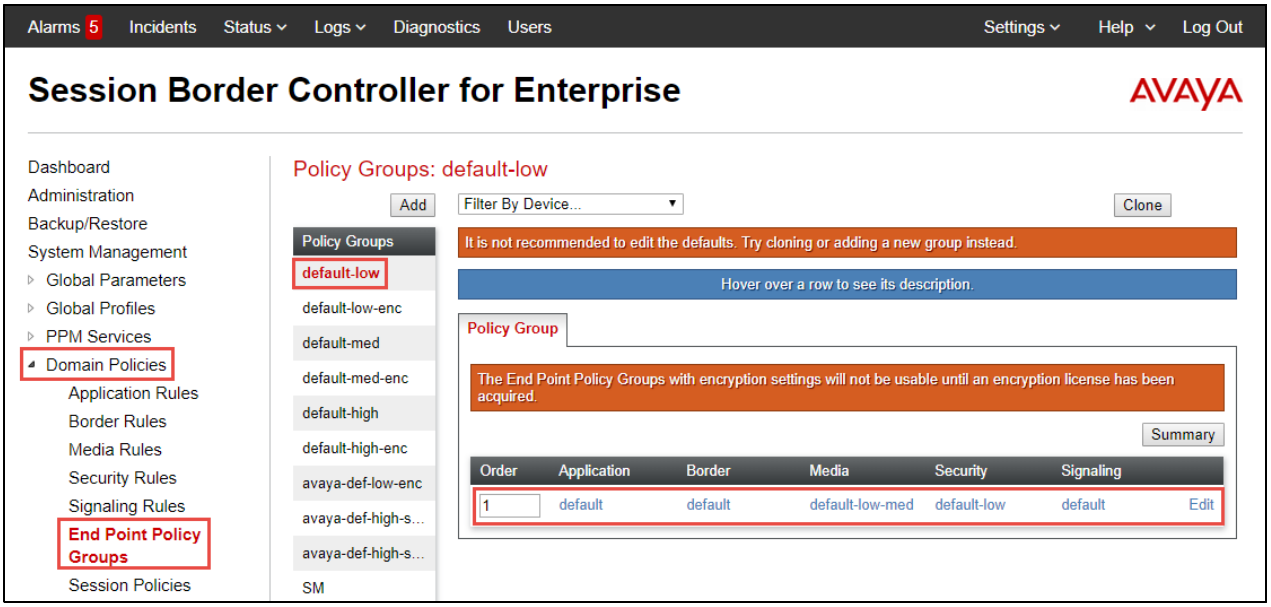


Figure 175- End Point Policy Group – Avaya SM

3. Select **default-low** under Policy Groups
4. Click **Clone**
5. Set Clone Name: **PG_ASM7.1** is given
6. Click **Finish**

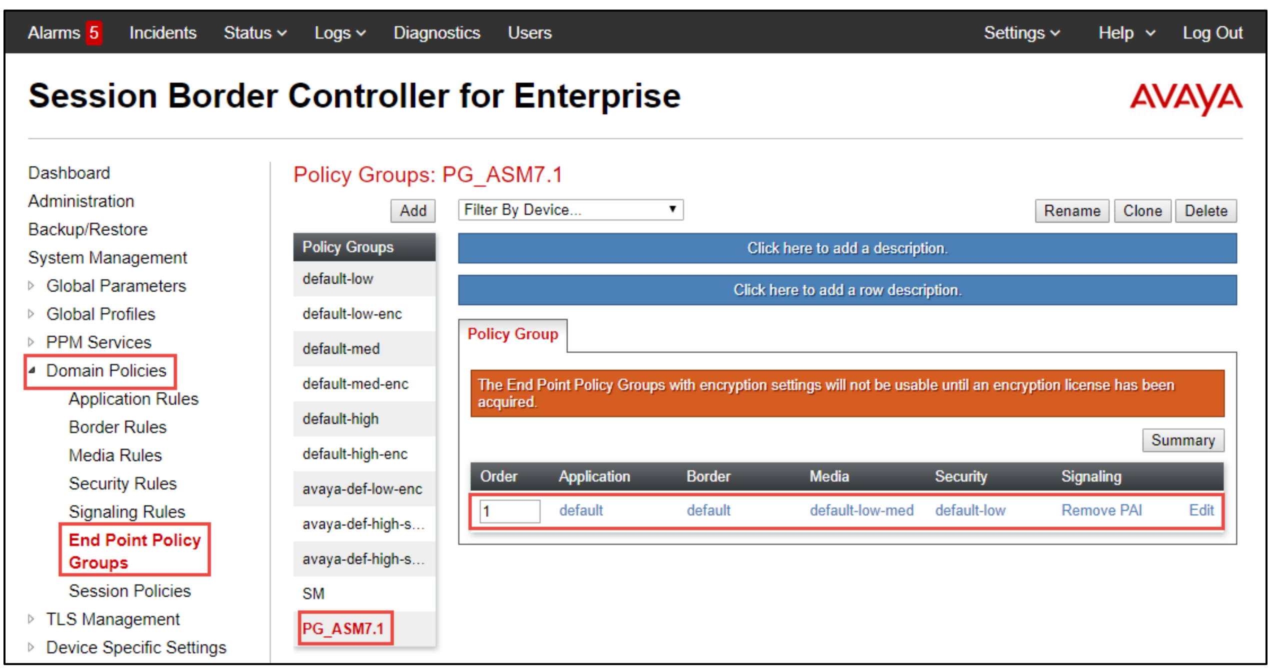


Figure 176 - End Point Policy Group – Avaya SM

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

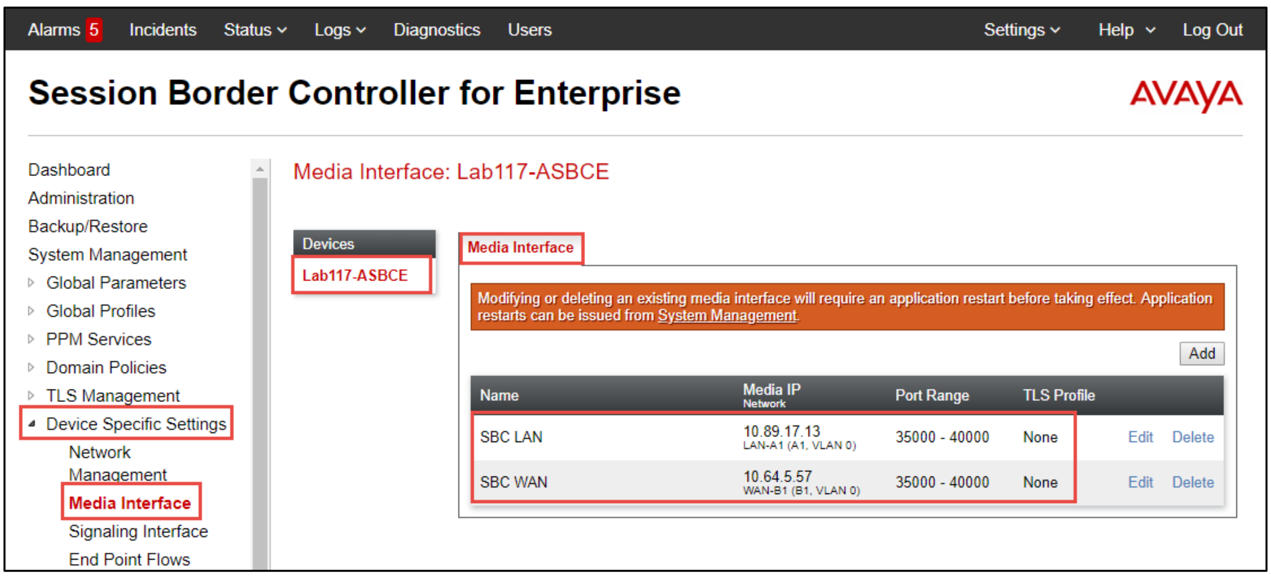


Figure 177- Media Interface

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.

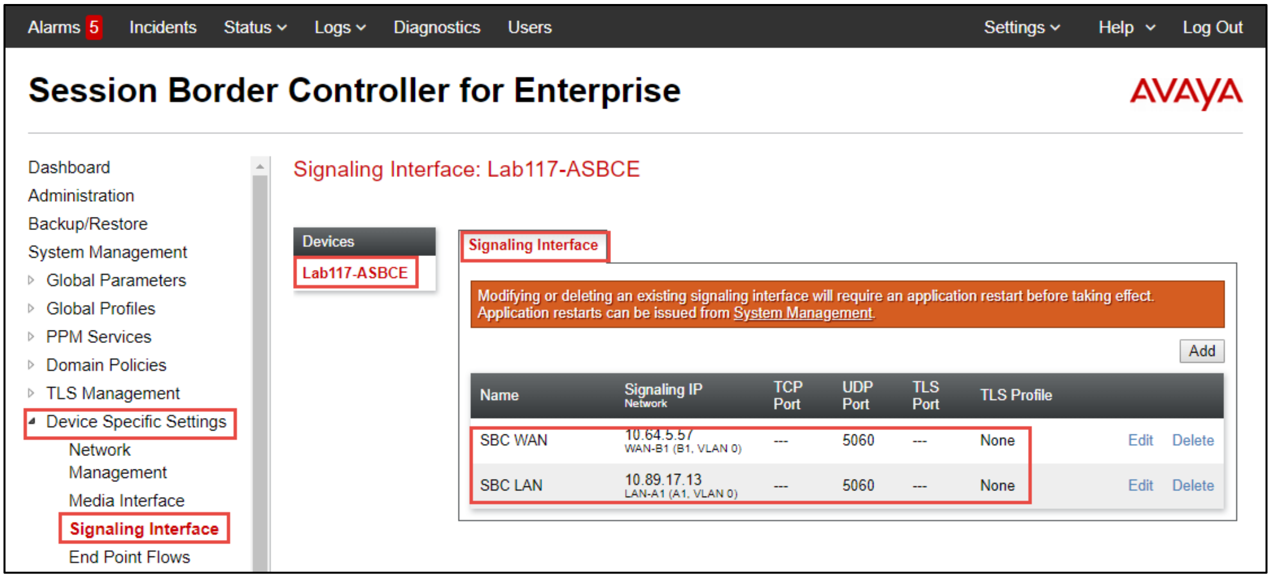


Figure 178 - Signaling Interface

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**

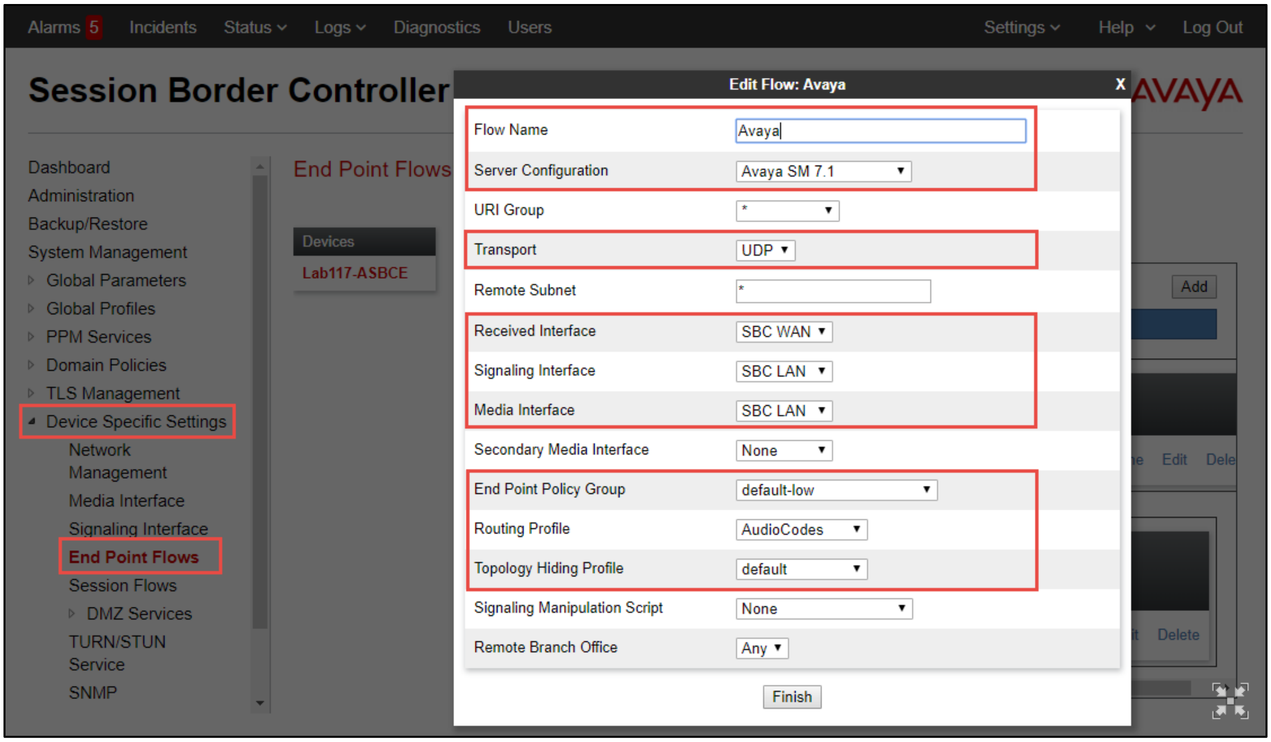


Figure 179 - Server Flow

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

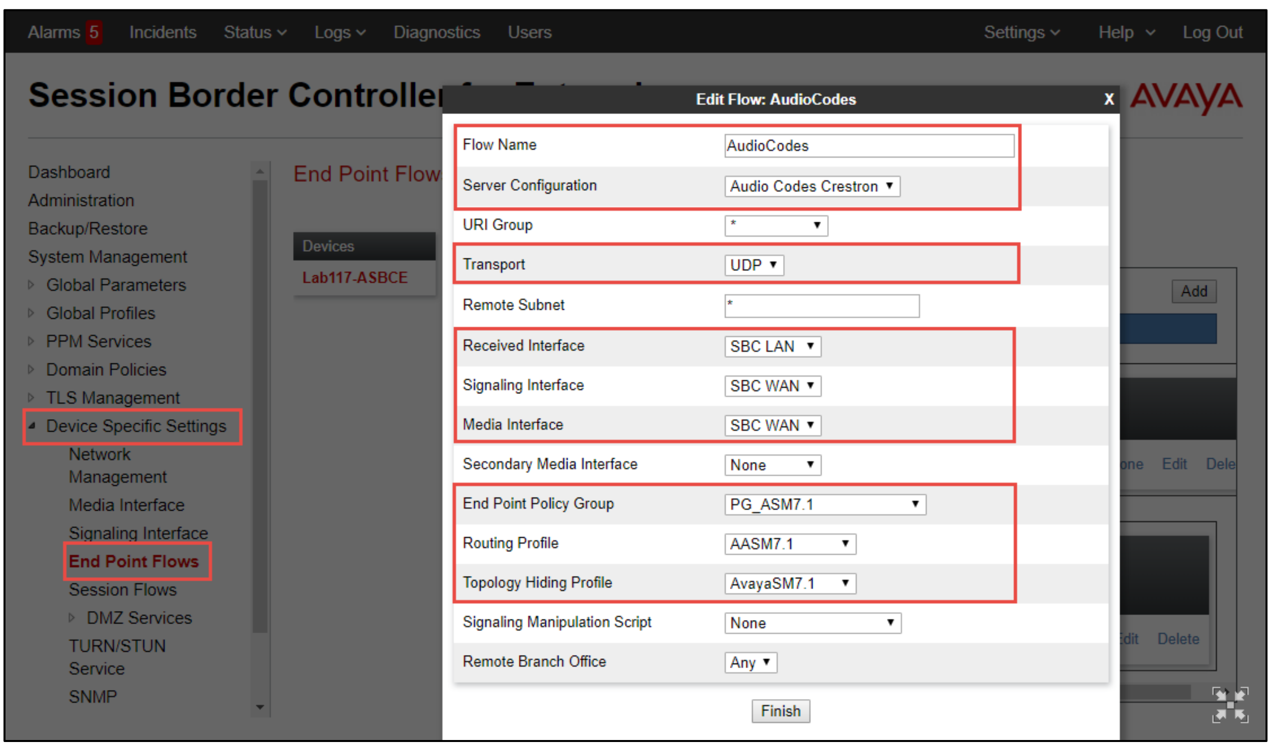


Figure 180 - Server Flow

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	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 hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PBX A 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	
2	Teams user Calls PBX B user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PBX B 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	
3	Teams user Calls PSTN user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio	1. Call is connected with bi-directional audio, voice is clear, no echo	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	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. PSTN 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	
10	PBX A user Calls Teams user and hangs up before answer	1. Make a voice call from PBX A user to Teams user 2. Teams user starts ringing 3. PBX A user hears Ring back Tone 4. PBX A user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
11	PBX B user Calls Teams user and hangs up before answer	1. Make a voice call from PBX B user to Teams user 2. Teams user starts ringing 3. PBX B user hears Ring back Tone 4. PBX B user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
12	PSTN user Calls Teams user and hangs up before answer	1. Make a voice call from PSTN user to Teams user 2. Teams user starts ringing 3. PSTN user hears Ring back Tone 4. PSTN user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
13	Teams user Calls PBX A user and performs hold/resume	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 initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. Teams user hangs up the call 10. Verify call is cleared successfully	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	PASSED	
14	Teams user Calls PBX B user and performs hold/resume	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. Teams user hangs up the call 10. Verify call is cleared successfully	present after call is resumed		
15	Teams user Calls PSTN user and performs hold/resume	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. Teams user hangs up the call 10. Verify call is cleared successfully	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	PASSED	
16	PBX A user Calls Teams user and Teams user performs hold/resume	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. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	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
		8. Verify two way audio is re-established between the two end points 9. PBX A user hangs up the call 10. Verify call is cleared successfully			receiver or speaker button.
17	PBX B user Calls Teams user and Teams user performs hold/resume	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. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. PBX B user hangs up the call 10. Verify call is cleared successfully	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	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	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 initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	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
		8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user 1 and PBX B user 2 11. PBX B user 1 hangs up the call 12. Verify call is cleared successfully			
23	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 6. Verify PBX B 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 PBX B user and PBX A user 11. PBX B 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	
24	Teams user Calls PBX B user, Teams user performs Attended	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user	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 PSTN user	6. Verify PBX B 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 B user and PSTN user 11. PBX B user hangs up the call 12. Verify call is cleared successfully			
25	Teams user Calls PSTN user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN 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	
26	Teams user Calls PSTN user, Teams user performs	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio	1. Call is transferred successfully 2. Two way audio present after call is transferred	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	3. Teams user 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	present after call is transferred		
29	PBX A user Calls Teams user, Teams user performs Attended Transfer to PBX B 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. 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 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	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
39	Teams user Calls PBX A user, Teams user performs Unattended 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 transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX A user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX A user and PSTN user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
40	Teams user Calls PBX B user, Teams user performs Unattended 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 transfers the call to PBX B user 2 6.PBX B user 2 starts ringing 7. PBX B user 1 hears Ring back Tone 8. PBX B user 2 answers the call 9. Verify two way audio between PBX B user 1 and PBX B user 2 10. PBX B user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	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	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PBX B user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PBX B user and PBX A user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
42	Teams userCalls PBX B user, Teams user performs Unattended Transfer to PSTN user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX B user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX B user and PSTN user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
43	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PSTN user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PSTN user and PBX B user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
44	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX A user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PSTN user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PSTN user and PBX A user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	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	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 transfers the call to PSTN user 2 6.PSTN user 2 starts ringing 7. PSTN user 1 hears Ring back Tone 8. PSTN user 2 answers the call 9. Verify two way audio between PSTN user 1 and PSTN user 2 10. PSTN user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
46	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PBX A user	1. Make a voice call from PBX A user 1 to Teams user 2. PBX A user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 2 6.PBX A user 2 starts ringing 7. PBX A user 1 hears Ring back Tone 8. PBX A user 2 answers the call 9. Verify two way audio between PBX A user 1 and PBX A user 2 10. PBX A user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	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	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. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PBX A user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PBX A user and PBX B user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
48	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PSTN 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. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX A user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX A user and PSTN user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	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	1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 2 6.PBX B user 2 starts ringing 7. PBX B user 1 hears Ring back Tone 8. PBX B user 2 answers the call 9. Verify two way audio between PBX B user 1 and PBX B user 2 10. PBX B user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
50	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PBX A 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. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PBX B user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PBX B user and PBX A user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
51	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PSTN 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. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX B user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX B user and PSTN user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
52	PSTN user Calls Teams user, Teams user performs Unattended 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 transfers the call to PBX B user 6.PBX B user starts ringing 7. PSTN user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PSTN user and PBX B user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
53	PSTN user Calls Teams user, Teams user performs Unattended 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 transfers the call to PBX A user 6.PBX A user starts ringing 7. PSTN user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PSTN user and PBX A user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
54	PSTN 1 user Calls Teams user, Teams user performs Unattended 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 transfers the call to PSTN user 2 6.PSTN user 2 starts ringing 7. PSTN user 1 hears Ring back Tone 8. PSTN user 2 answers the call 9. Verify two way audio between PSTN user 1 and PSTN user 2 10. PSTN user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
55	PSTN user calls Teams user, Teams user performs Unattended Transfer to second Teams user	1. Make a voice call from PSTN user to Teams user 1 2. PSTN user hears Ring back Tone 3. Teams user 1 answers the call 4. Verify two way audio 5. Teams user 1 transfers the call to Teams user 2 6.Teams user 2 starts ringing 7. Teams user 2 answers the call 8. Verify two way audio between PSTN user and Teams user 2 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
56	Teams user Calls PBX A user, Teams user adds PBX A user to the ongoing call	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 adds PBX A user 2 to the ongoing call 6.PBX A user 2 starts ringing 7. PBX A user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PBX B user to the ongoing call 6.PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PSTN user to the ongoing call 6.PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PBX B user 2 to the ongoing call 6.PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6.PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PSTN user 2 to the ongoing call 6. PSTN user 2 starts ringing 7. PSTN user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from PBX A user 1 to Teams user 2. PBX A user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX A user 2 to the ongoing call 6. PBX A user 2 starts ringing 7. PBX A user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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. Teams user adds PBXB user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX B user 2 to the ongoing call 6. PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	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 adds PSTN user 2 to the ongoing call 6. PSTN user 2 starts ringing 7. PSTN user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Make a voice call from PSTN user to Teams user 1 2. PSTN user hears Ring back Tone 3. Teams user 1 answers the call 4. Verify two way audio 5. Teams user 1 adds Teams user 2 to the ongoing call 6. Verify Teams user 2 is added successfully to the call 7. Teams user 1 adds PBX A user to the ongoing call 9. Verify PBX A user is added successfully to the call 10. Teams user 1 adds PBX B user to the ongoing call 11. Verify PBX B user is added successfully to the call 12. Verify all four users are able to hear each other 13. All the users hang up and call is cleared successfully for all the users	1. Third user is added to the call successfully 2. All three users are able to hear each other	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	1. Teams user sets call forwarding all to PBX A user 2 2. Make a voice call from PBX A user 1 to Teams user 3. PBX A user 2 starts ringing 4. PBX A user 2 answers the call 5. Verify two way audio 6. PBX A user 1 hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PBX B user 2. Make a voice call from PBX A user to Teams user 3. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PSTN user 2. Make a voice call from PBX A user to Teams user 3. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PBX B user 2 2. Make a voice call from PBX B user 1 to Teams user 3. PBX B user 2 starts ringing 4. PBX B user 2 answers the call 5. Verify two way audio 6. PBX B user 1 hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PBX A user 2. Make a voice call from PBX B user to Teams user 3. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PSTN user 2. Make a voice call from PBX B user to Teams user 3. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PBX B user 2. Make a voice call from PSTN user to Teams user 3. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PBX A user 2. Make a voice call from PSTN user to Teams user 3. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding all to PSTN user 2 2. Make a voice call from PSTN user 1 to Teams user 3. PSTN user 2 starts ringing 4. PSTN user 2 answers the call 5. Verify two way audio 6. PSTN user 1 hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX A user 2 2. Make a voice call from PBX A user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user 2 starts ringing 4. PBX A user 2 answers the call 5. Verify two way audio 6. PBX A user 1 hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX B user 2. Make a voice call from PBX A user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PSTN user 2. Make a voice call from PBX A user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX B user 2 2. Make a voice call from PBX B user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user 2 starts ringing 4. PBX B user 2 answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX A user 2. Make a voice call from PBX B user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PSTN user 2. Make a voice call from PBX B user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX B user 2. Make a voice call from PSTN user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PBX A user 2. Make a voice call from PSTN user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets call forwarding no answer to PSTN user 2	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	2. Make a voice call from PSTN user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user 2 starts ringing 4. PSTN user 2 answers the call 5. Verify two way audio 6. PSTN user 1 hangs up the call 7. Verify call is cleared successfully	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	1. Teams user sets simultaneous ringing to PBX A user and PBX B user 2. Make a voice call from PSTN user to Teams user 3. Teams user, PBX A user and PBX B user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up 7. Verify call is cleared successfully 8. Repeat steps 2 to 6 where PBX B user answers the call		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
94	Teams user with restricted Caller ID Calls PBX A user	1. Make a voice call from Teams user with restricted caller ID to PBX A user 2.Teams user hears Ring back Tone 3. PBX A user starts ringing 4. Verify caller ID displayed on PBX A user is Unavailable/Private/Anonymous 5. PBX A user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
95	Teams user with restricted Caller ID Calls PBX B user	1. Make a voice call from Teams user with restricted caller ID to PBX B user 2.Teams user hears Ring back Tone 3. PBX B user starts ringing 4. Verify caller ID displayed on PBX B user is Unavailable/Private/Anonymous 5. PBX B user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
96	Teams user with restricted Caller ID Calls PSTN user	1. Make a voice call from Teams user with restricted caller ID to PSTN user 2.Teams user hears Ring back Tone 3. PSTN user starts ringing 4. Verify caller ID displayed on PSTN user is Unavailable/Private/Anonymous	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	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	1. Make a voice call from PSTN user with restricted caller ID to Teams user 2. PSTN 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. PSTN 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	
100	PBX A user Calls Teams user and leaves voicemail	1. Make a voice call from PBX A user to Teams user 2. Teams user does not answer the call 3. Allow the call to get forwarded to voicemail 4. PBX A 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	
101	PBX B user Calls Teams user and	1. Make a voice call from PBX B user to Teams user 2. Teams user does not answer the call 3. Allow the call to get forwarded to	1. Teams user is able to receive and retrieve voicemail successfully	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	1. Make a voice call from Teams user to PBX B user 2. PBX B 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	
105	Teams user Calls PBX A user, PBX A returns call failure response	1. Make a voice call from Teams user to PBX A user 2. PBX A returns 486 Busy 3. Verify Teams user gets appropriate notification or announcement and the call is cleared 4. Repeat steps 1 to 3 where PBX A returns 480, 404, 503 SIP responses 5. Document the observation on Teams user side	1. Teams user handles the failure response successfully	PASSED	
106	Teams user Calls PBX A user using SIP URI	1. Make a voice call from Teams user to PBX A user using SIP URI 2. PBX A user starts ringing 3. PBX A user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully	1. Teams user is able to call using SIP URI 2. Call is connected with two way audio successfully	NOT TESTED	SIP URI Not tested for PBX A
107	Teams user Calls PBX B	1. Make a voice call from Teams user to PBX B user using SIP URI 2. PBX B user starts ringing	1. Teams user is able to call using SIP URI 2. Call is connected	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	user using SIP URI	3. PBX B user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully	with two way audio successfully		
108	PBX A user Calls Teams user using SIP URI	1. Make a voice call from PBX A user to Teams user using SIP URI 2. PBX A user starts ringing 3. PBX A user answers the call 4. Verify two way audio 5. PBX A user hangs up the call 6. Verify call is cleared successfully	1. Teams user is able to call using SIP URI 2. Call is connected with two way audio successfully	NOT TESTED	SIP URI Not tested for this PBX A
109	PBX B user Calls Teams user using SIP URI	1. Make a voice call from PBX B user to Teams user using SIP URI 2. PBX B user starts ringing 3. PBX B user answers the call 4. Verify two way audio 5. PBX B user hangs up the call 6. Verify call is cleared successfully	1. Teams user is able to call using SIP URI 2. Call is connected with two way audio successfully	PASSED	
110	Teams user calls Skype for Business user	1. Make a voice call from Teams user to Skype for Business user 2. Teams user hears Ring back Tone 3. Skype for Business user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Verify the same scenario where		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	1. Make a voice call from Skype for Business user to Teams user 2. Skype for Business user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Skype for Business user hangs up the call 6. Verify call is cleared successfully 7. Verify the same scenario where Skype for Business user is internal and external		PASSED	
112	Teams user calls Skype for Business External Mobile user	1. Skype for business user is an External Mobile user 2. Make a voice call from Teams user to Skype for Business user 3. Teams user hears Ring back Tone 4. Skype for Business user answers the call 5. Verify two way audio 6. Teams user hangs up the call 7. Verify call is cleared successfully		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
113	Skype for Business External Mobile user calls Teams user	1. Skype for business user is an External Mobile user 2. Make a voice call from Skype for Business user to Teams user 3. Skype for Business user hears Ring back Tone 4. Teams user answers the call 5. Verify two way audio 6. Skype for Business user hangs up the call 7. Verify call is cleared successfully		PASSED	
114	Teams user call other tenant users	1. Make a voice call from Teams user to another tenant users (Teams desktop client user, Teams mobile user, Skype for Business Online user) 2. Verify call is successful 3. Make one call to each different user one by one		PASSED	
115	Teams users joins a meeting scheduled by Skype for business On-premises user	1. Skype for business user schedules a meeting and invites Teams user 1 and Teams user 2 2. Teams user 1 joins the meeting using the Join button 3. Teams user 2 joins the meeting using the dial-in conferencing number 4. Verify Teams users are able to join the meeting successfully 5. Verify all three users are able to hear		PASSED	

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