# Crestron UC-PHONE and UC-PHONE-PLUS

# Connecting Microsoft Teams Direct Routing using AudioCodes Mediant Virtual Edition (VE), Cisco UCM 11.5 and Skype for Business 2015(Hybrid)

September 2019

## Document History

| Rev. No. | Date | Document Owner | Description |
|---|---|---|---|
| 1.0 | Sep-15-2019 | tekVizion | Draft - Configuration Guide |
| 1.1 | Sep-23-2019 | tekVizion | Updated after Internal Review |

# Table of Contents

# 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, Cisco UCM 11.5 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](http://www.tekVizion.com)*

# 2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representation of Crestron UC-PHONE and UC-PHONE-PLUS connected O365 Cloud with Microsoft Teams Direct Routing to Cisco UCM 11.5 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.

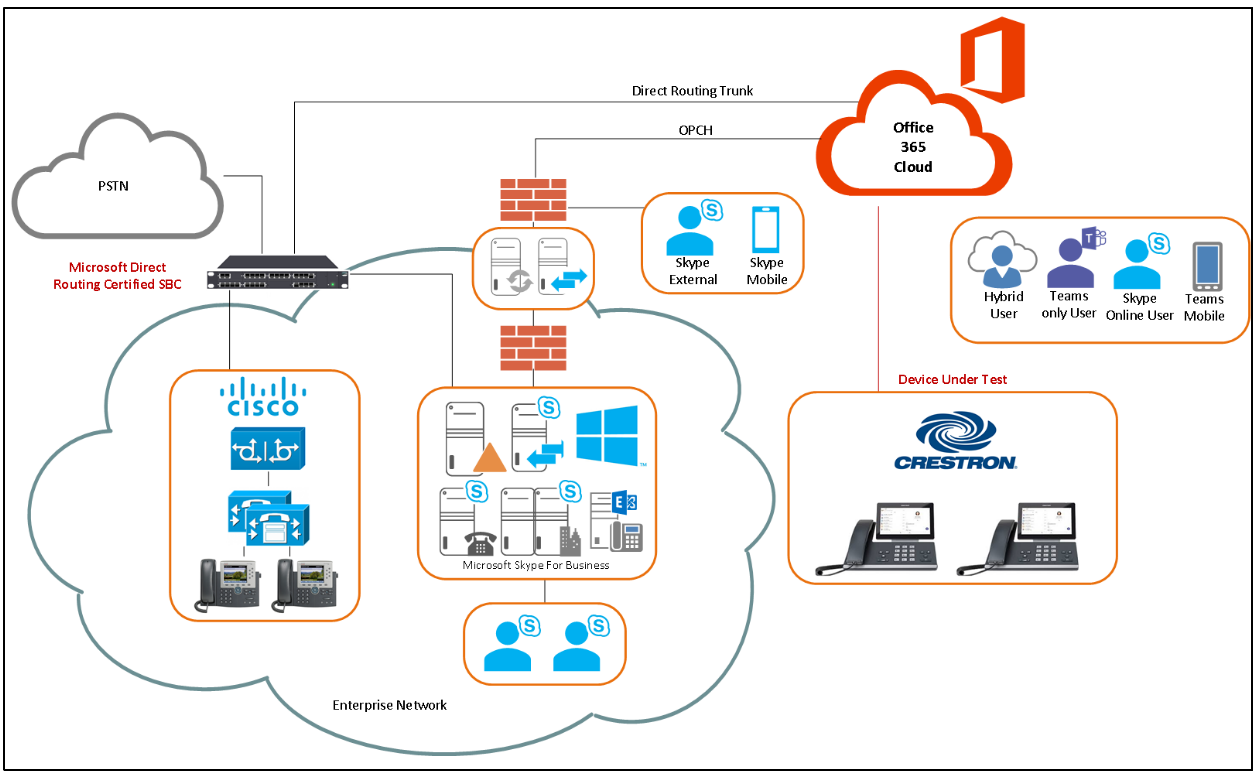


*Figure 1 Network Topology*

Numbering Plan

- Cisco UCM users are configured with 4 digit extension 65XX
- Teams users are configured with E164 numbers +197259809XX
- Skype for Business Server 2015 with E164 numbers +197259809XX

Dialing Plan

- Teams users and Cisco users call PSTN either doing 10 digits 11 digits dialing or E164 dialing
- Teams users call Cisco users by dialing 65XX
- Cisco 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
- Cisco UCM running on ESXi
- Cisco Unity Connection running on ESXi
- Cisco UBE v CISCO2921/K9
- PSTN Gateway

## 2.2 Software Requirements

- AudioCodes Mediant VE SBC v7.20A.250.003
- Skype For Business 2015 Version (6.0.9319)
- Cisco UCM v11.5.1.16900-16
- Cisco Unity Connection v11.5.1.12900-21
- Cisco UBE v 11.5.2
- 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, Cisco UBE, Cisco UCM, 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 | Cisco UBE Configuration | Section 4.5 |
| Step 4 | Cisco UCM Configuration | Section 4.6 |

## 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 |
| **Cisco UCM** | |
| IP Address | 172.16.29.81 |
| Subnet Mask | 255.255.255.0 |
| **Cisco UBE** | |
| LAN IP Address | 10.64.4.182 |
| LAN Subnet Mask | 255.255.255.0 |
| WAN IP Address | 10.70.69.70 |
| WAN Subnet Mask | 255.255.255.0 |
| **Cisco Unity** | |
| LAN IP Address | 10.80.18.5 |
| LAN 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.

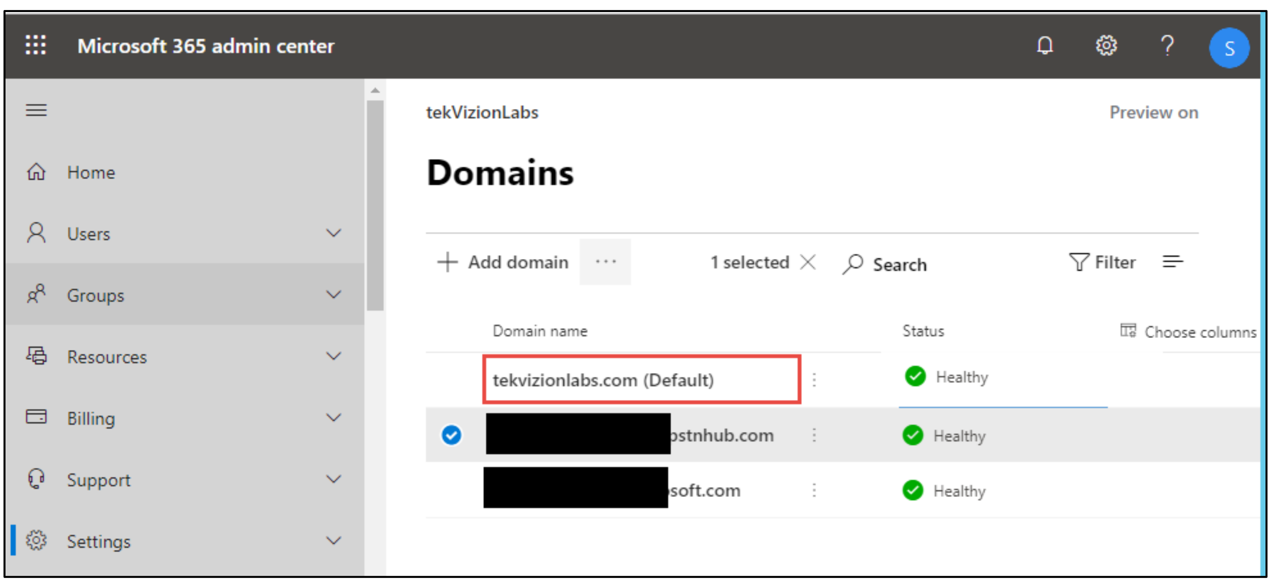


*Figure 2 – Add Domain*

### 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.

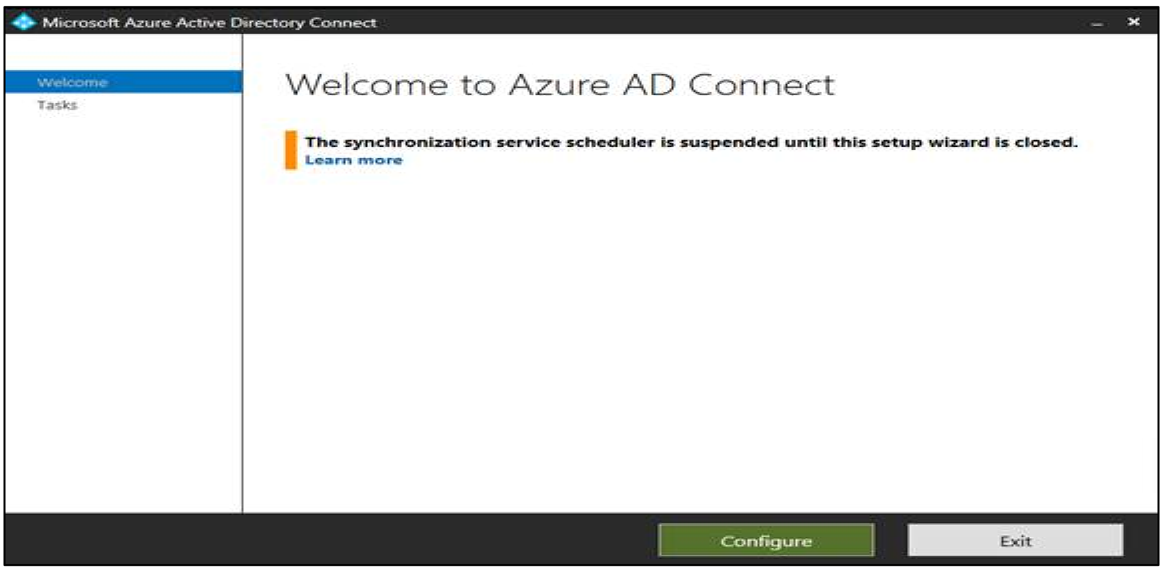


*Figure 3 – Azure AD – Sync Process*

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.

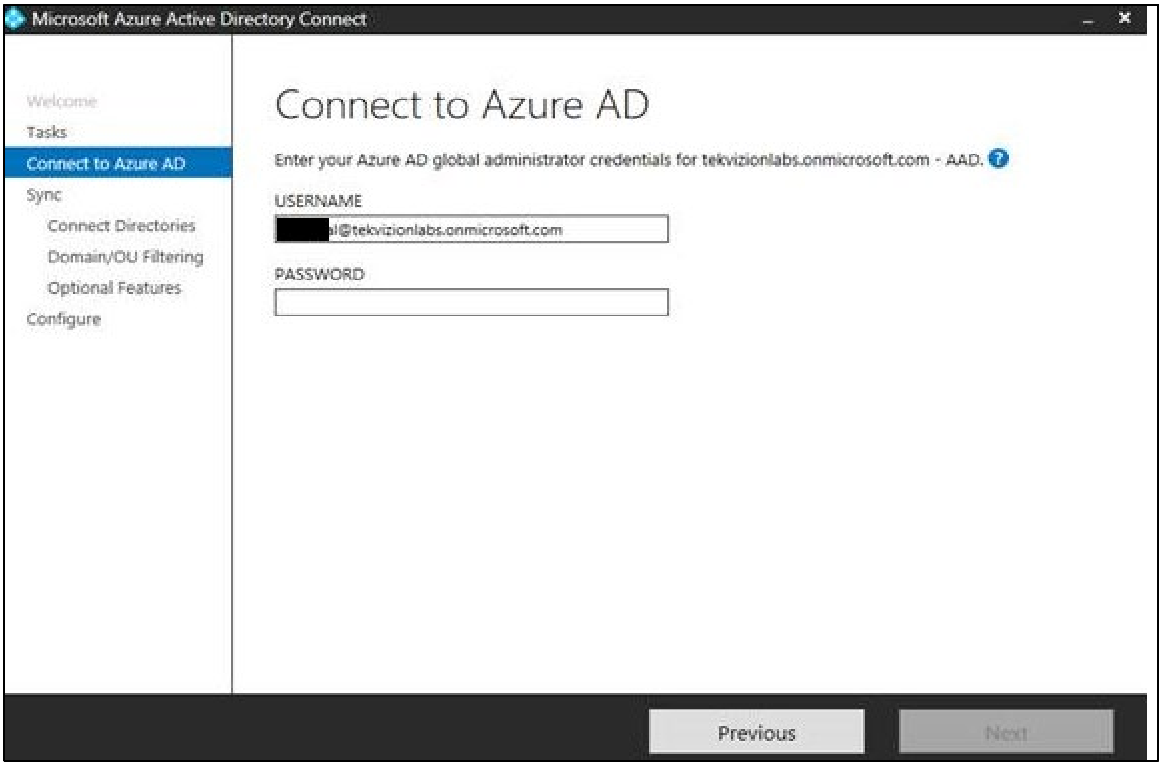


*Figure 4 – Azure AD – Sync Process*

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

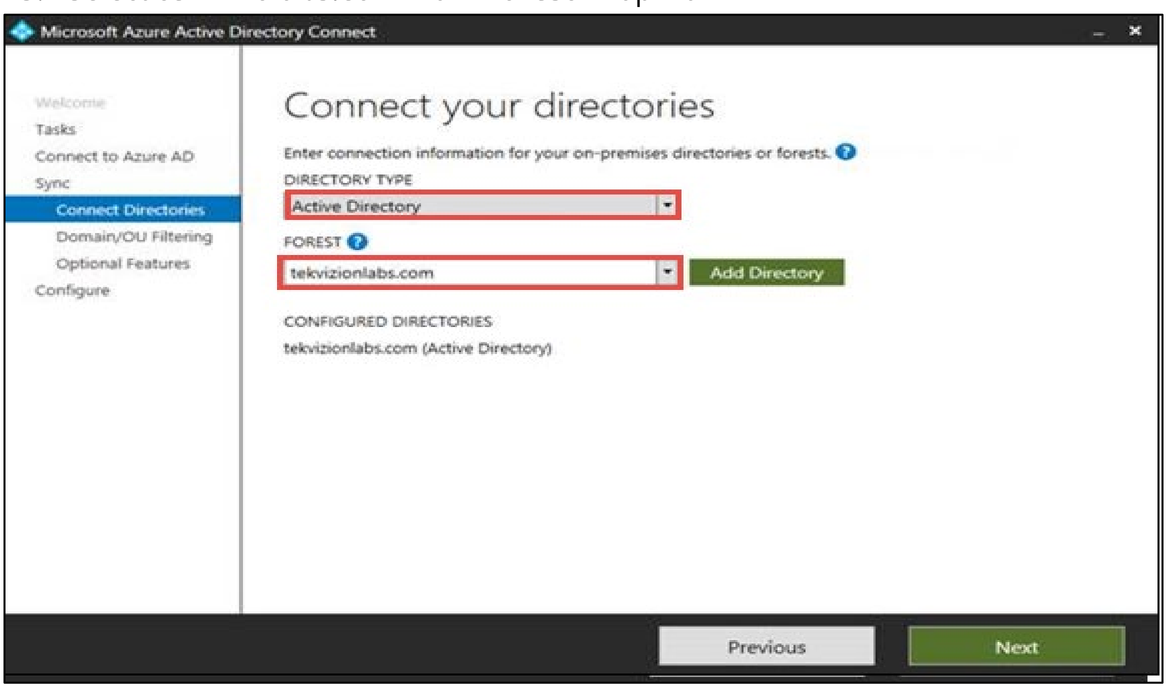


*Figure 5 – Azure AD – Sync Process*

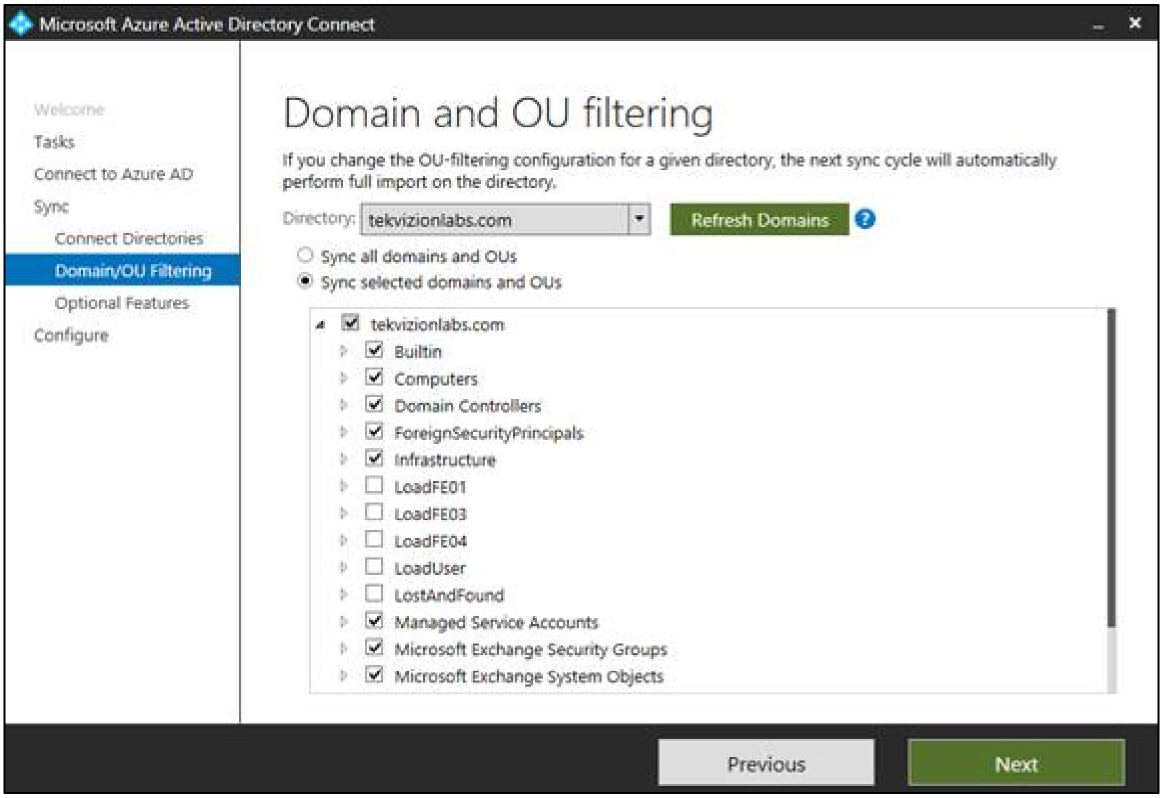


*Figure 6 – Azure AD – Sync Process*

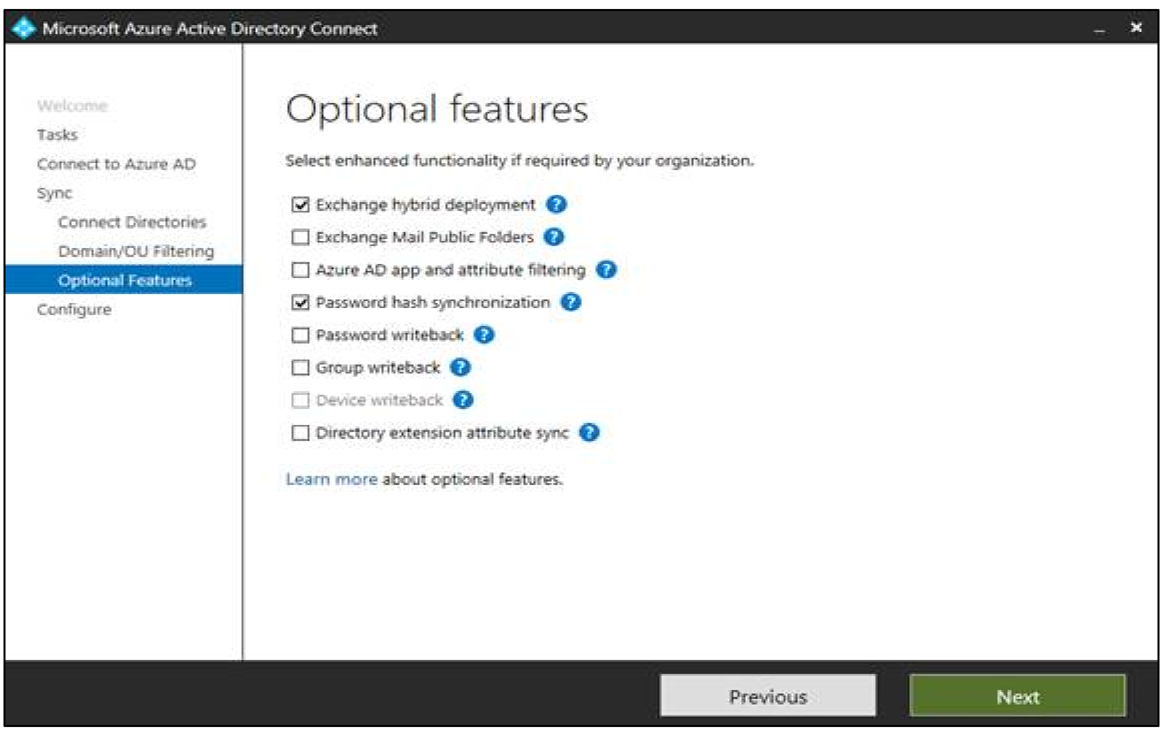


*Figure 7 – Azure AD – Sync Process*

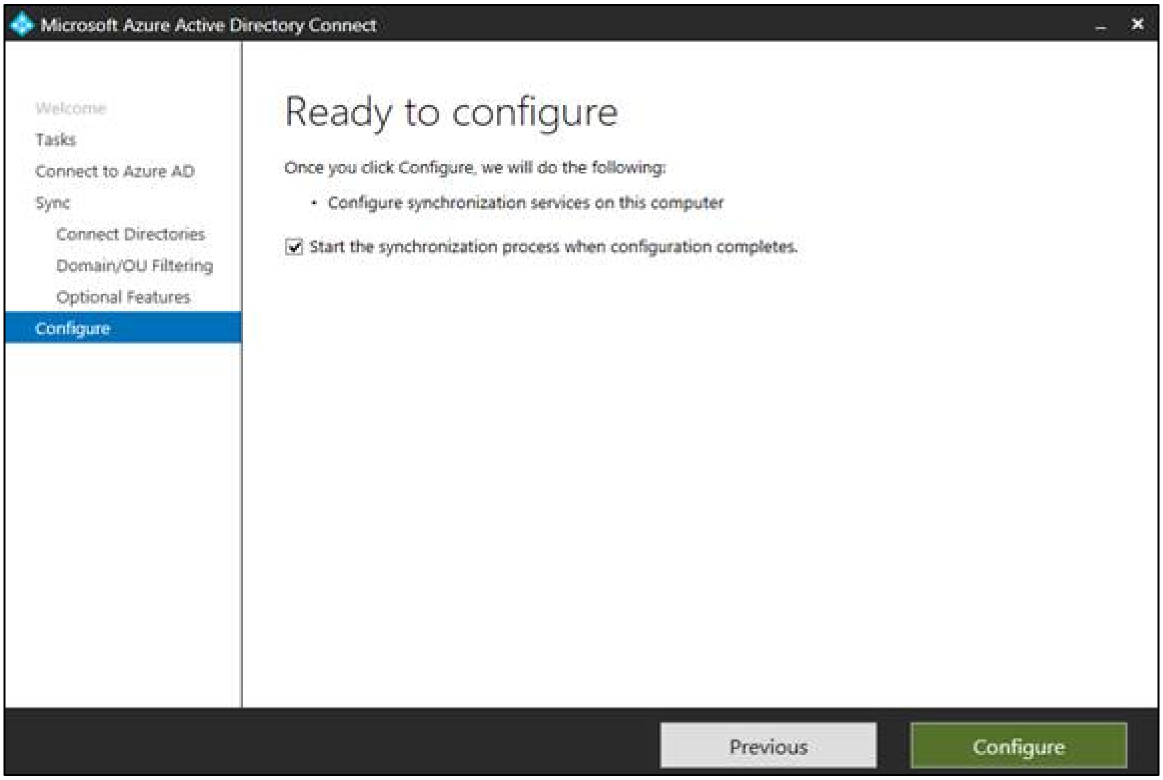


*Figure 8 – Azure AD – Sync Process*

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

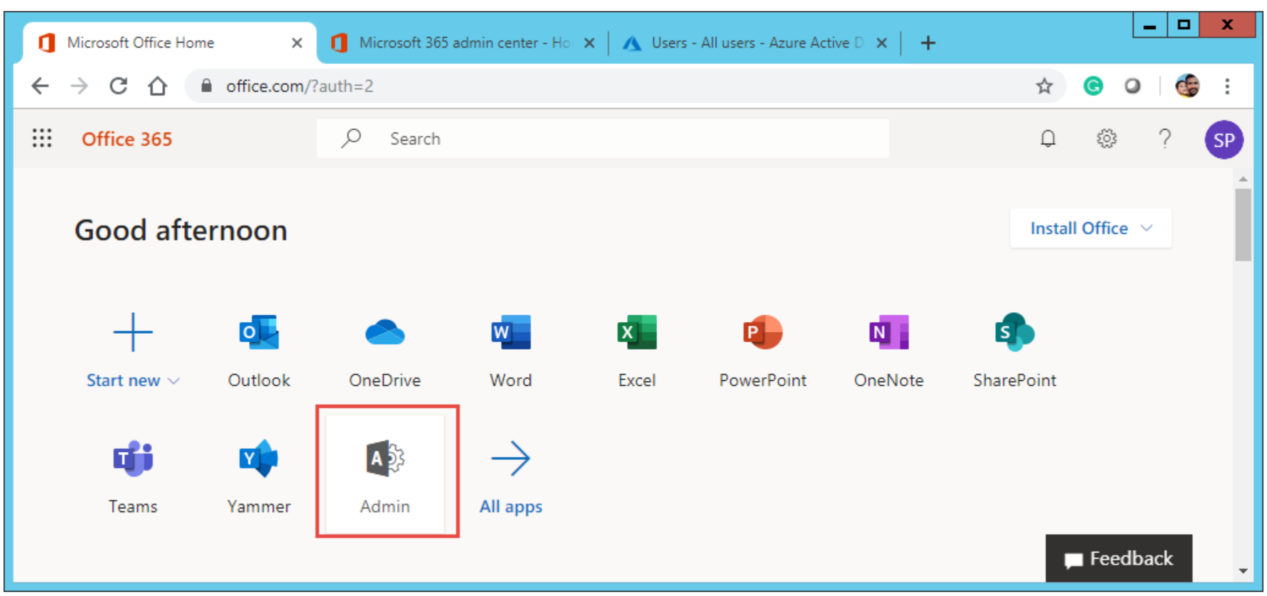


*Figure 9 – Azure AD – Sync Process*

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

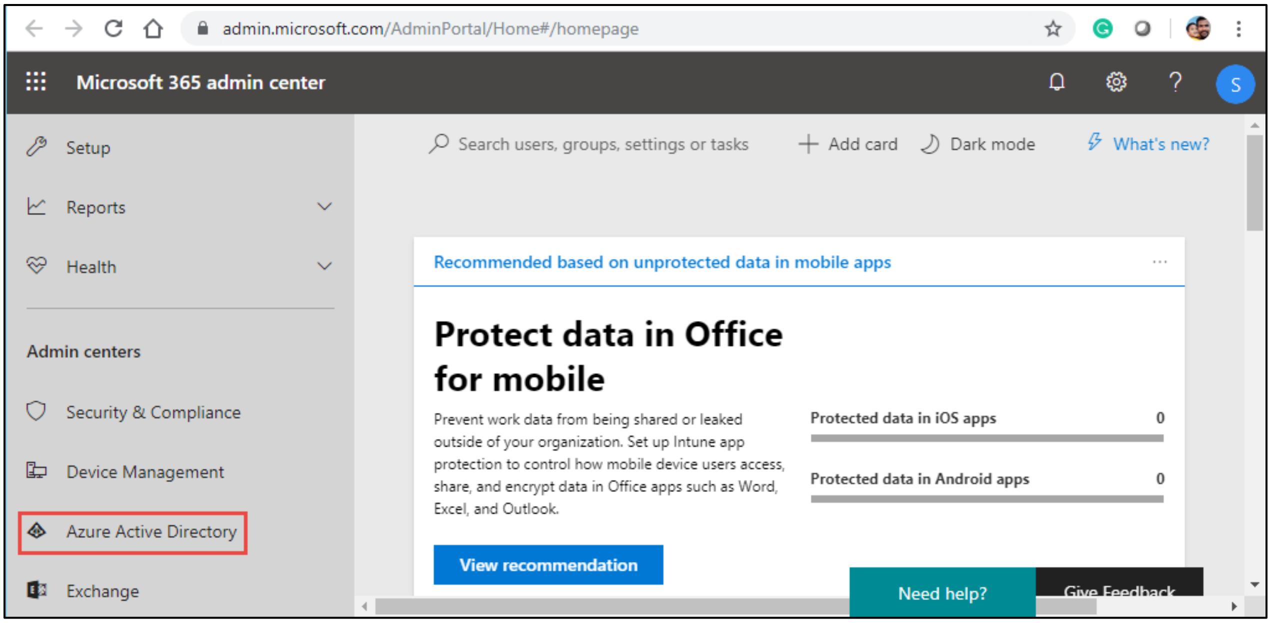


*Figure 10 – Azure AD – Sync Process*

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

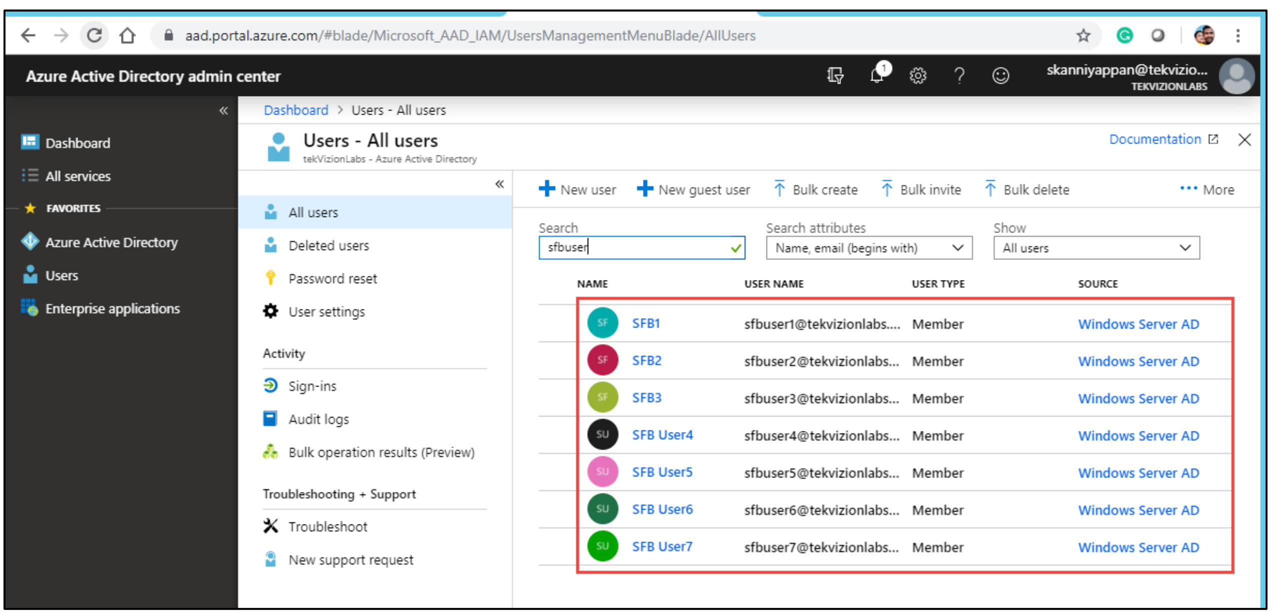


*Figure 11 – Azure AD – Sync Process*

### 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**

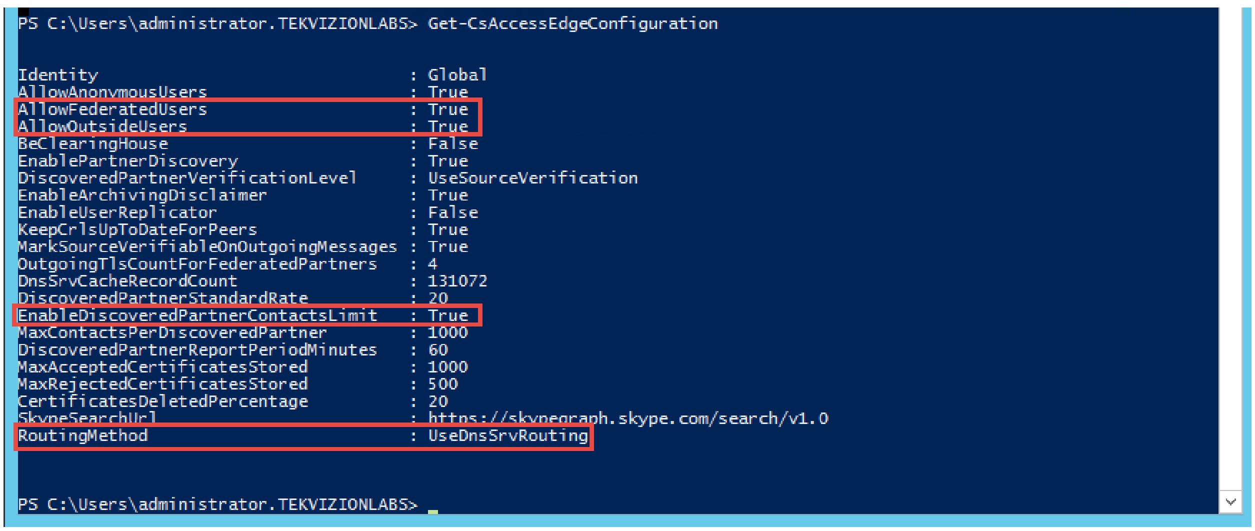


*Figure 12 – Edge Federation*

#### 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

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


Identity                  : LyncOnline
Name                      : LyncOnline
ProxyFqdn                 : sipfed.online.lync.com
VerificationLevel         : AlwaysVerifiable
Enabled                   : True
EnabledSharedAddressSpace : True
HostsOCSUsers             : True
IsLocal                   : False
AutodiscoverUrl           : https://webdir1a.online.lync.com/Autodiscover/AutodiscoverService.svc/root


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

*Figure 13 – Hosting Provider*

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

In addition to the change made in on-premises deployment, make the corresponding change in Office 365 tenant to 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

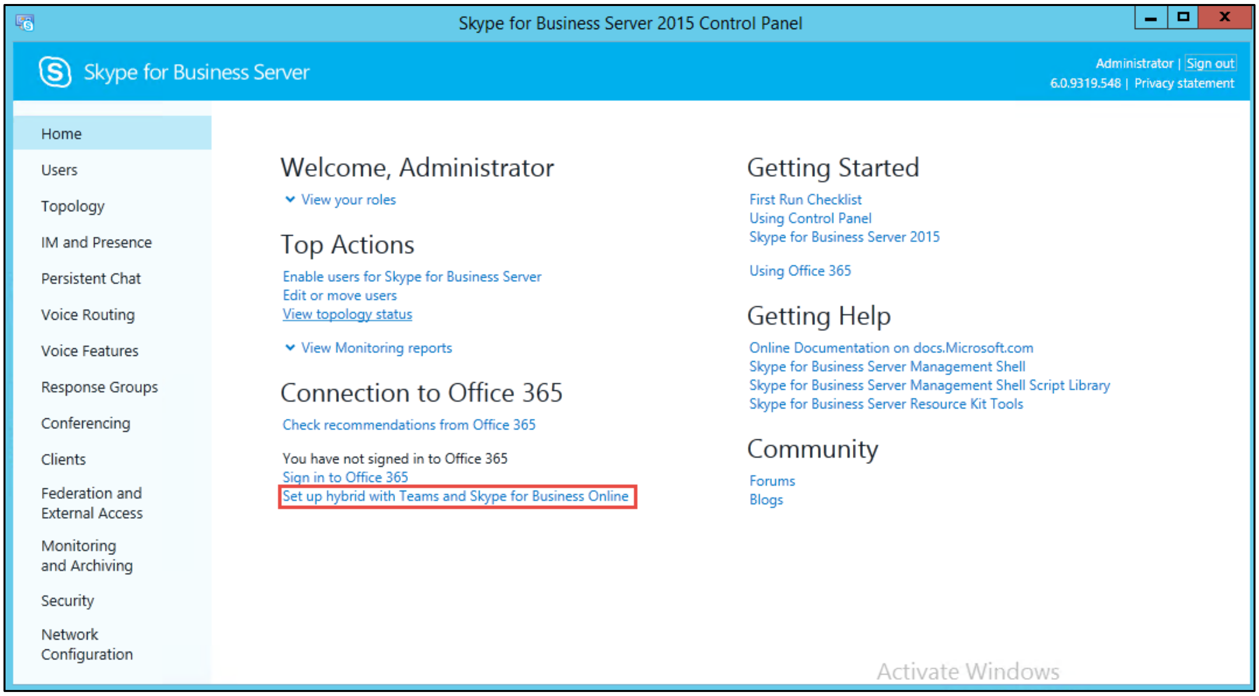


*Figure 14 – Hybrid with Teams and SFB*

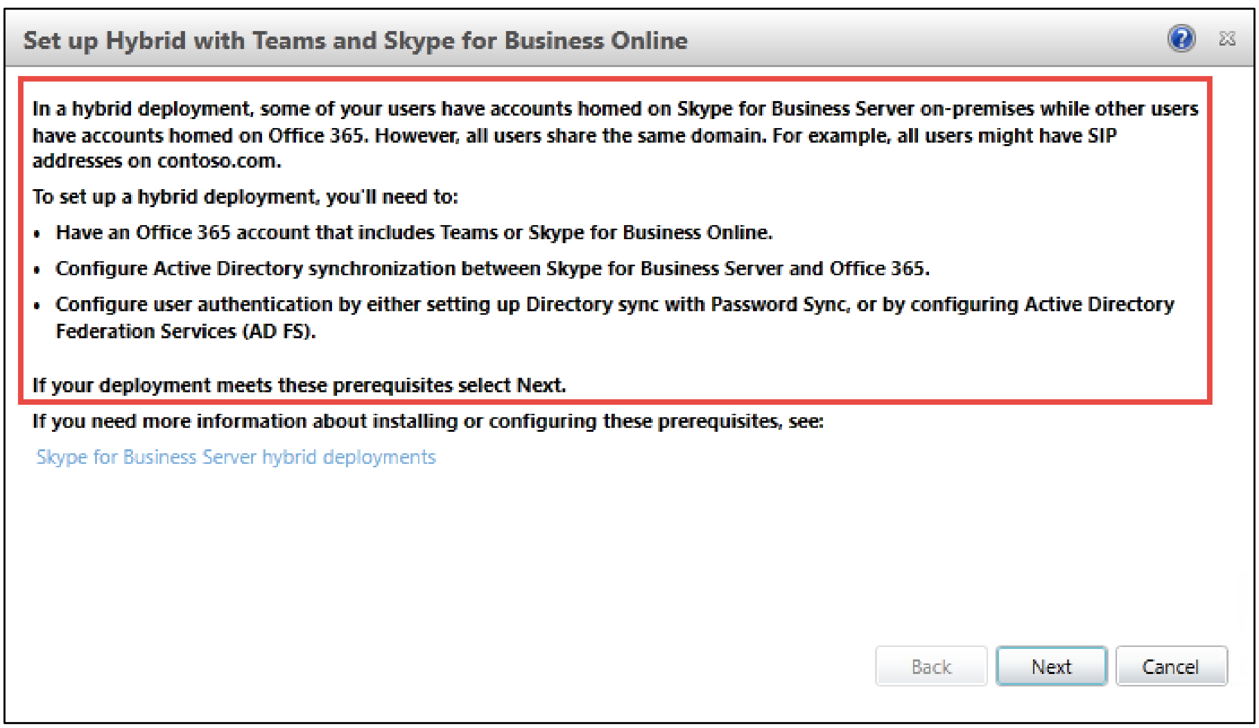


*Figure 15 – Hybrid with Teams and SFB Contd.*

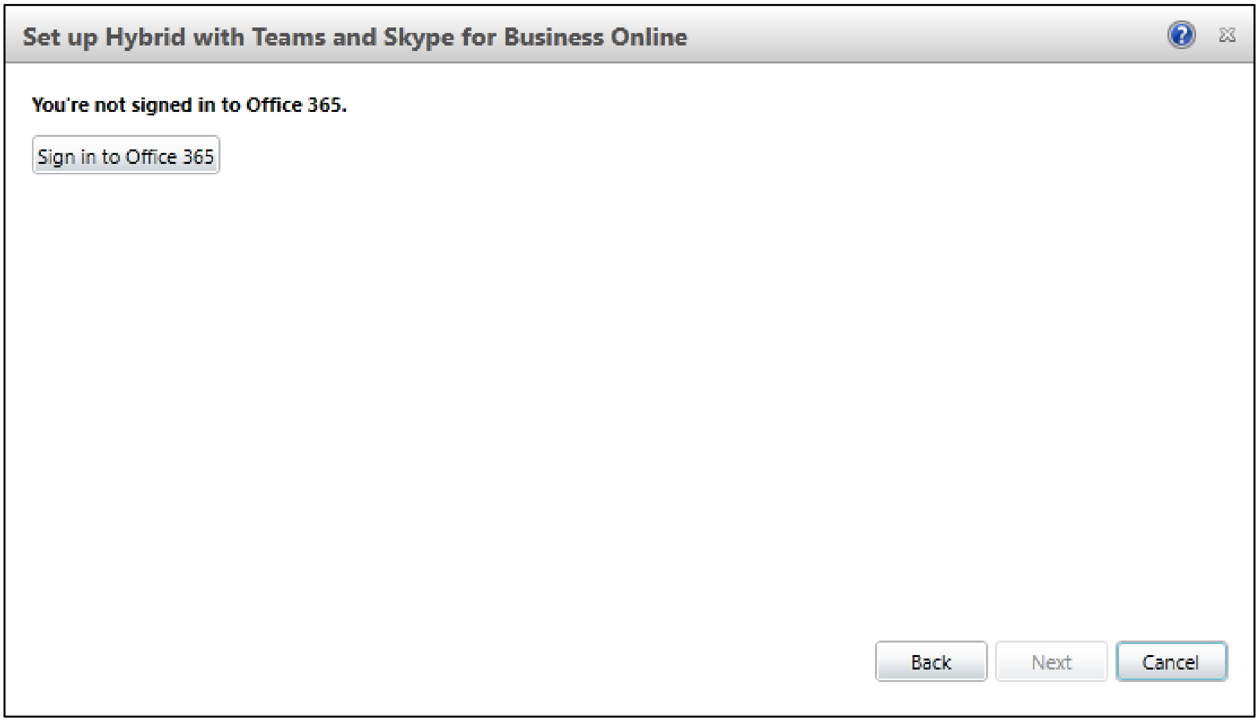


*Figure 16 – Hybrid with Teams and SFB Contd.*

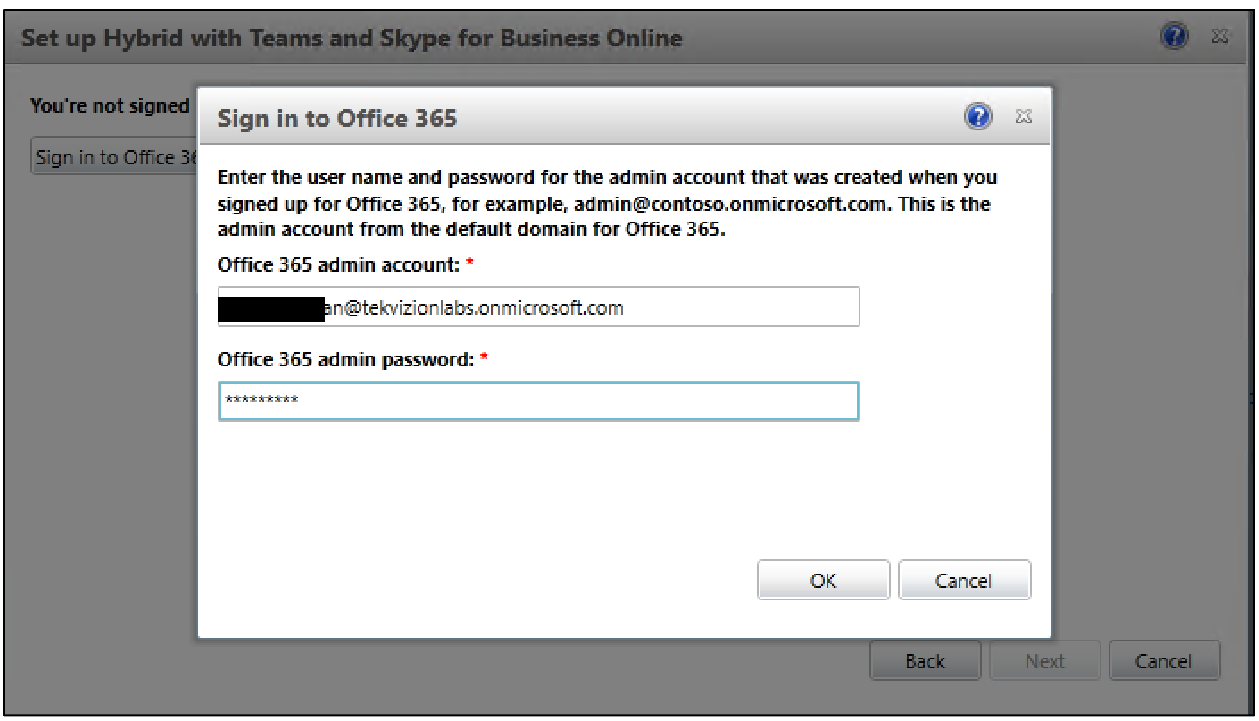


*Figure 17 – Hybrid with Teams and SFB Contd.*

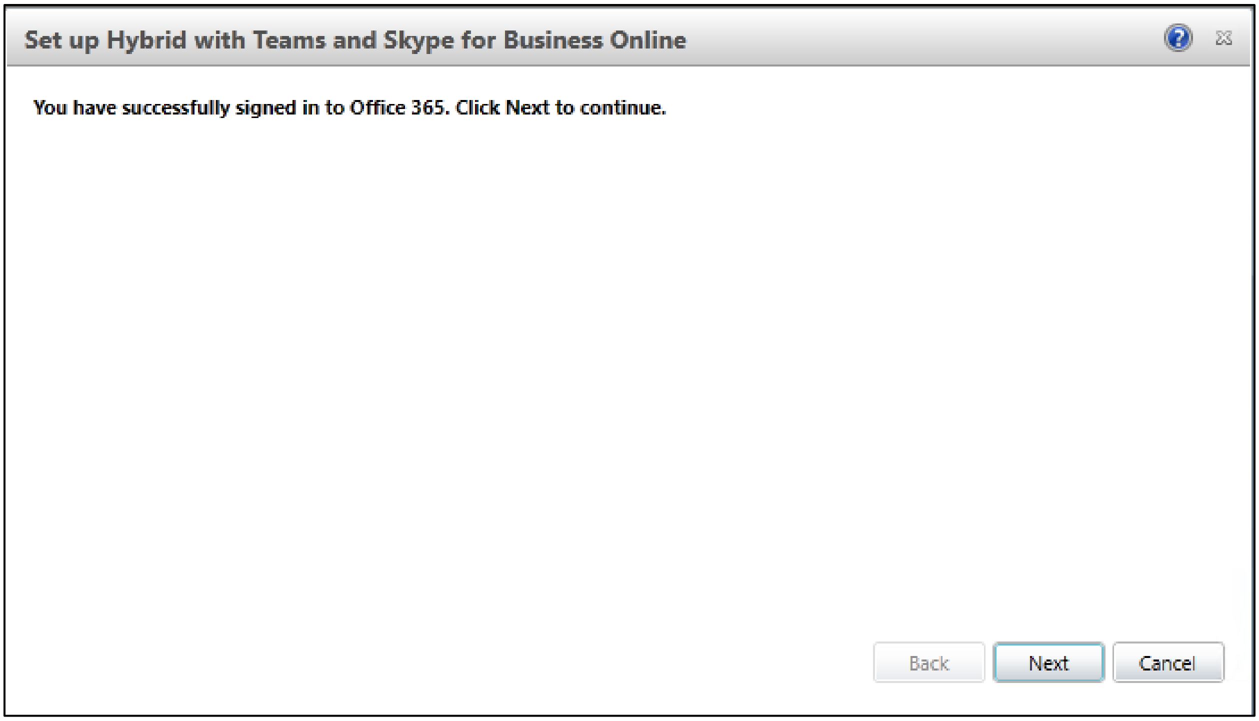


*Figure 18 – Hybrid with Teams and SFB Contd.*

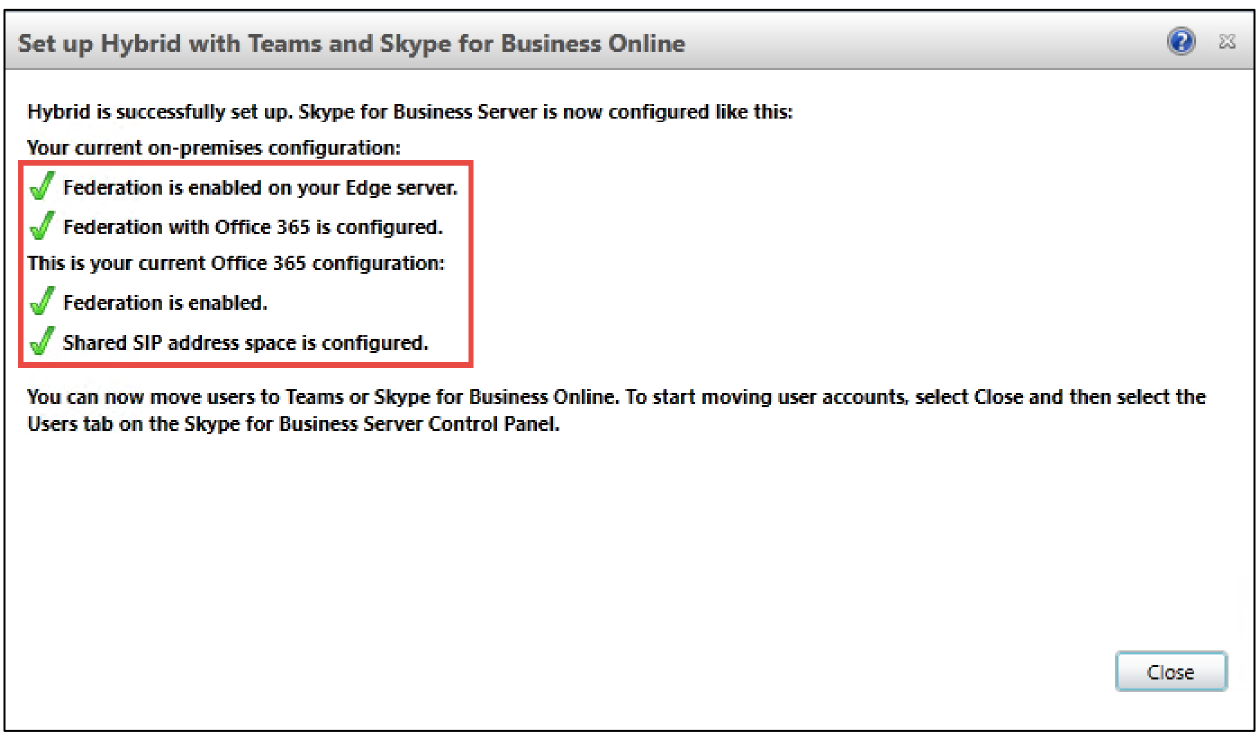


*Figure 19 – Hybrid with Teams and SFB Contd.*

#### 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.

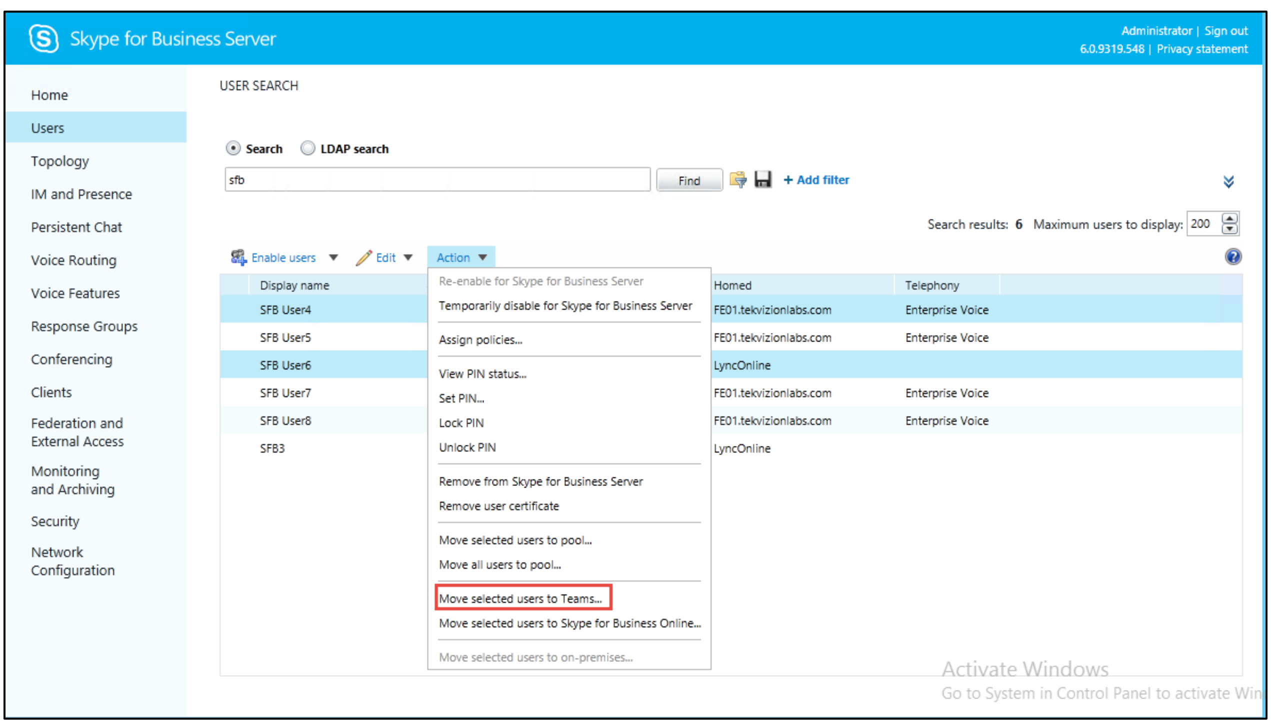


*Figure 20 – Move Users to Teams*

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

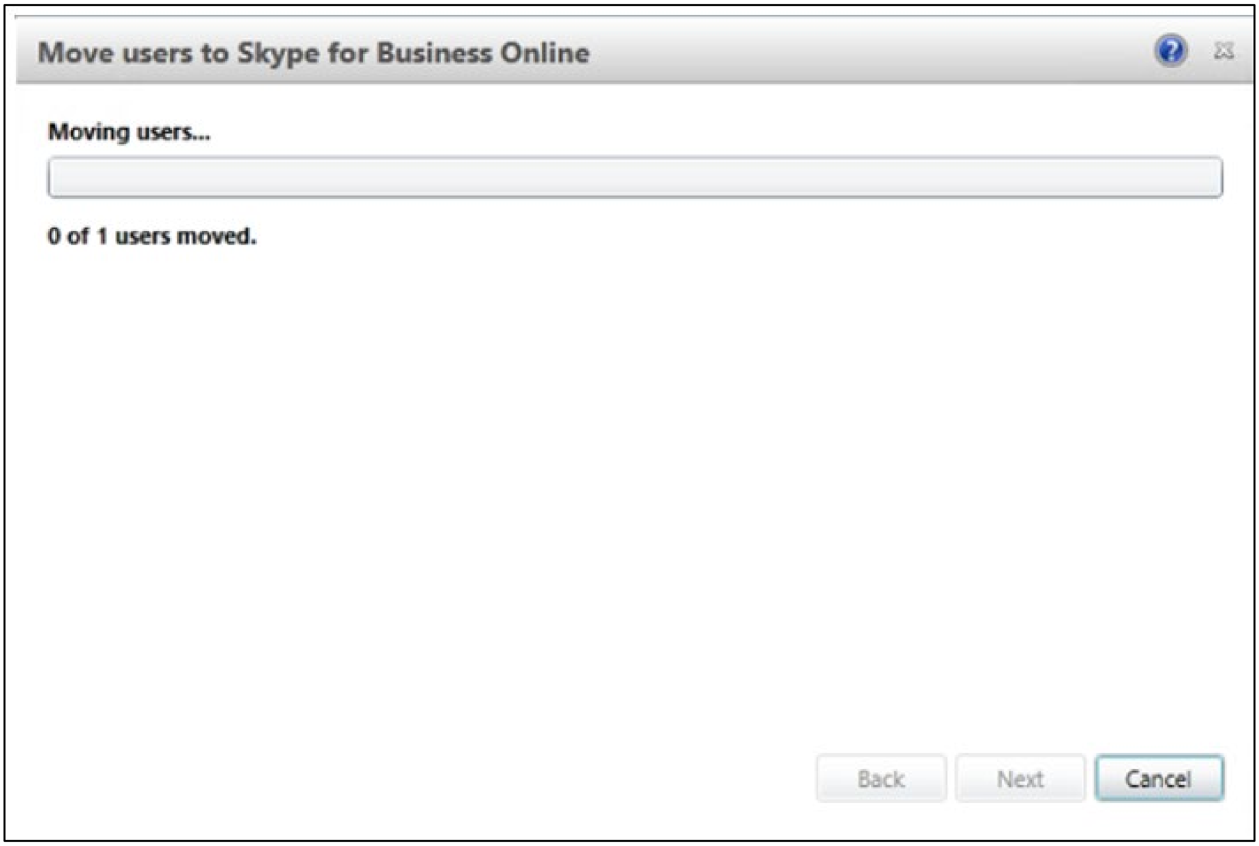


*Figure 21 – Move Users to Teams Contd.*

The migration of the selected user is complete.

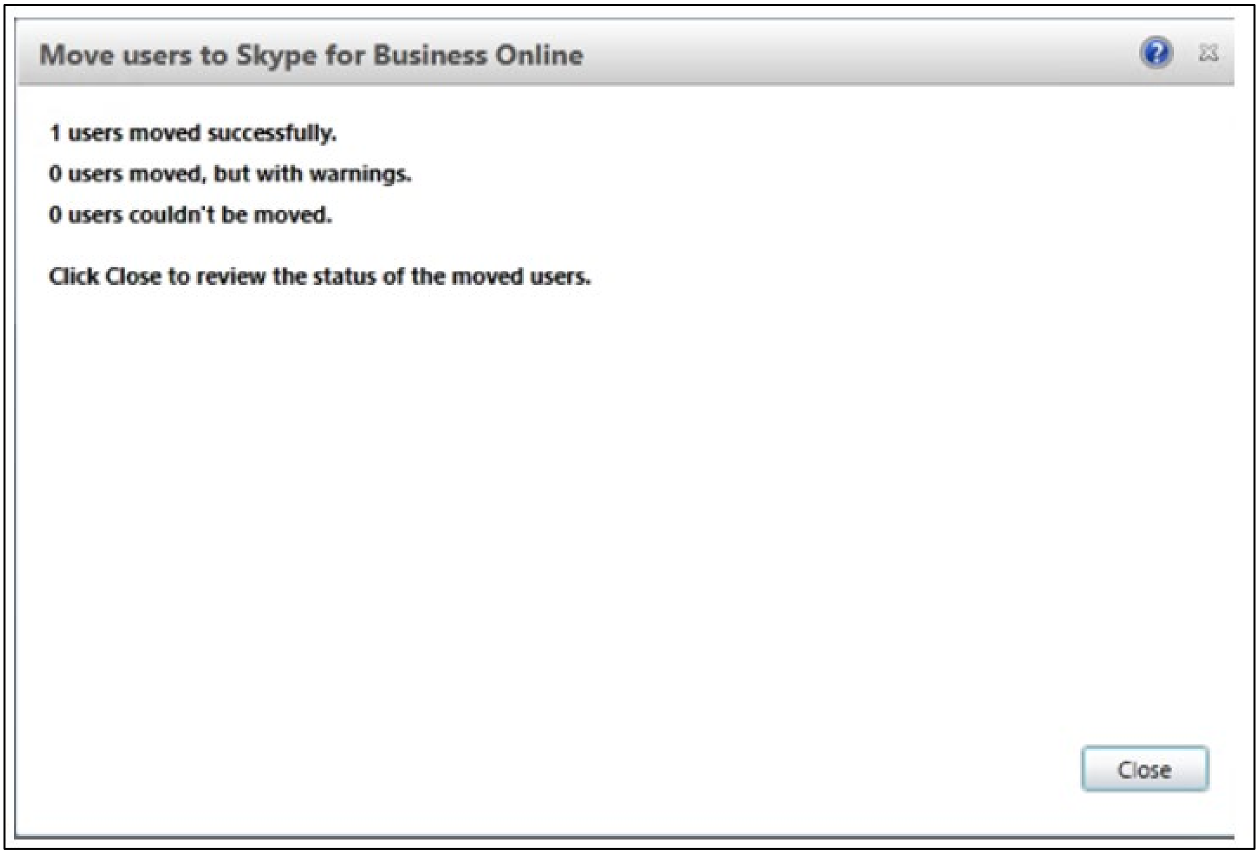


*Figure 22 – Move Users to Teams Contd.*

Administration of the user ‘sfbuser6’ from the on-site control panel

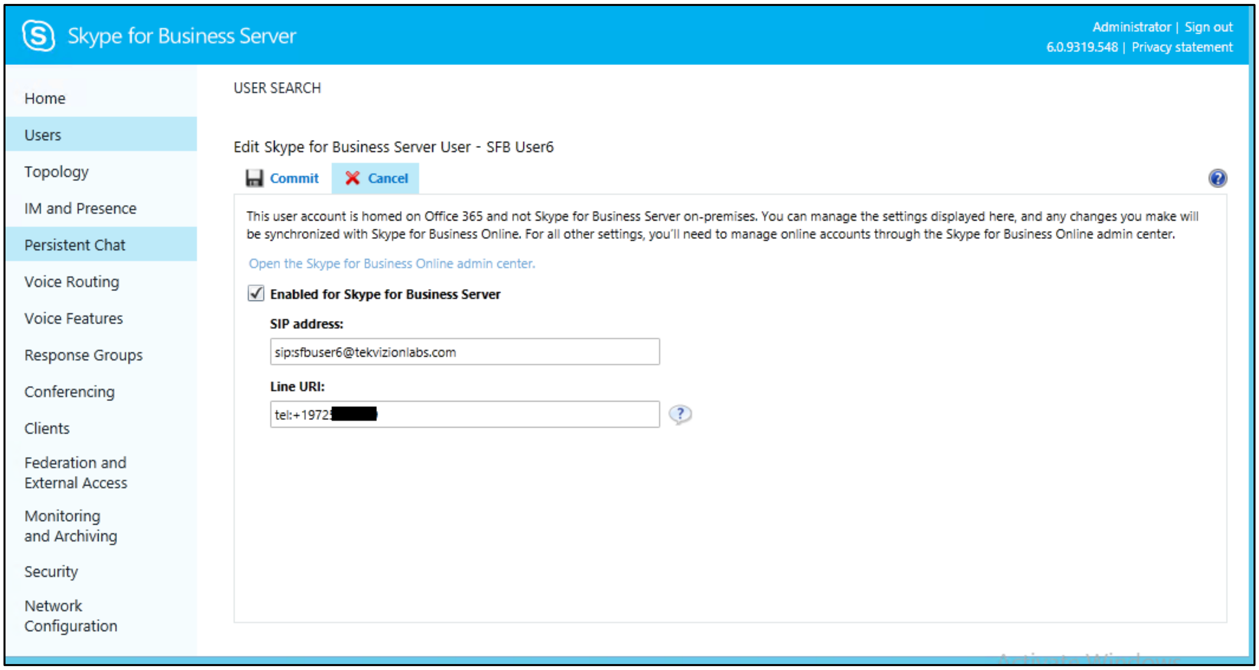


*Figure 23 – Move Users to Teams*

## 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.

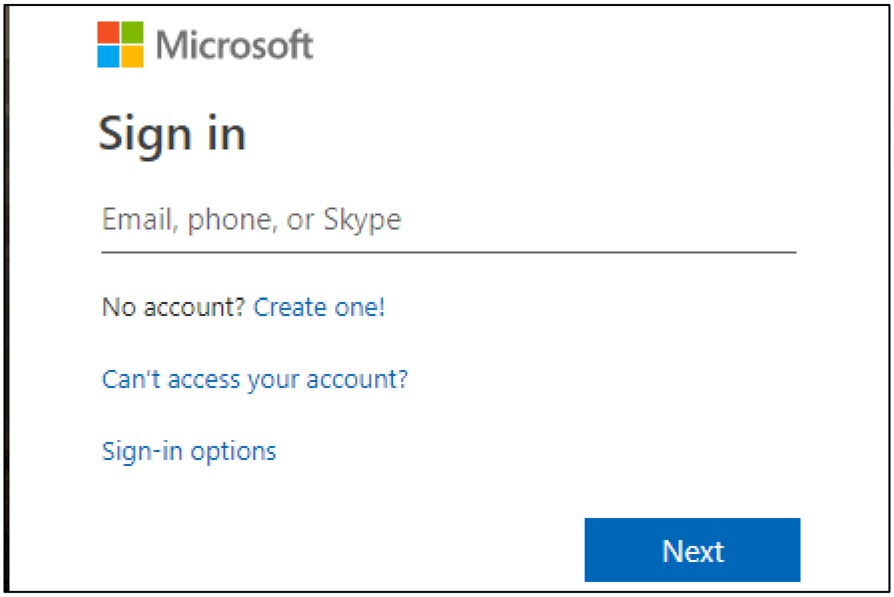


*Figure 24: Office 365 Portal Login*

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

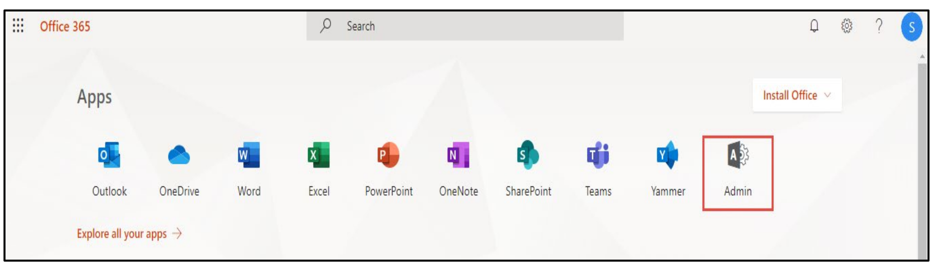


*Figure 25: Office 365 Portal Login*

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

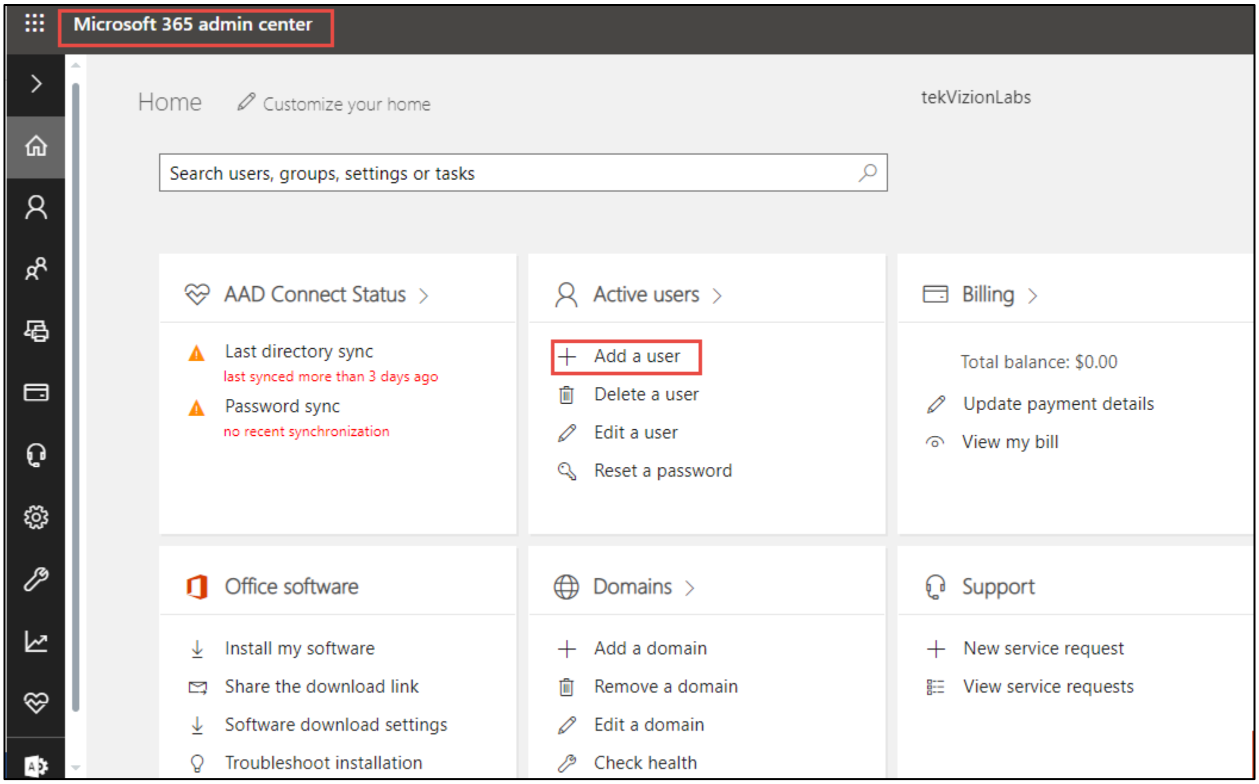


*Figure 26: Teams User Creation*

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

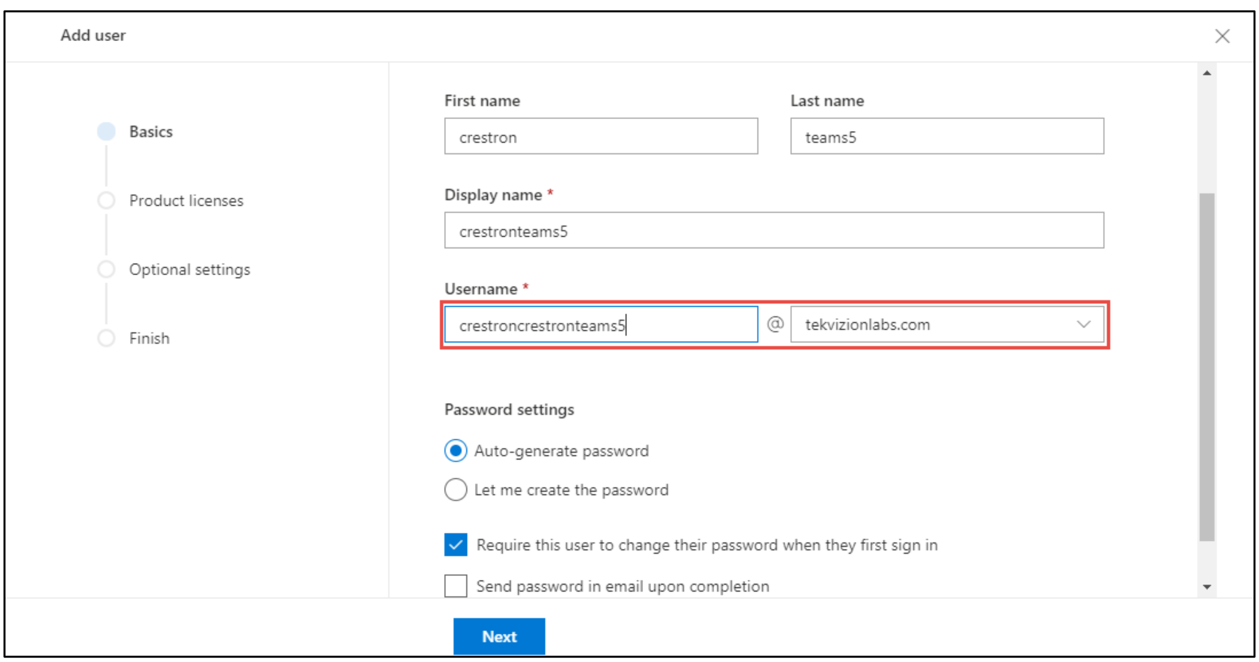


*Figure 27: Teams User Creation – Contd.*

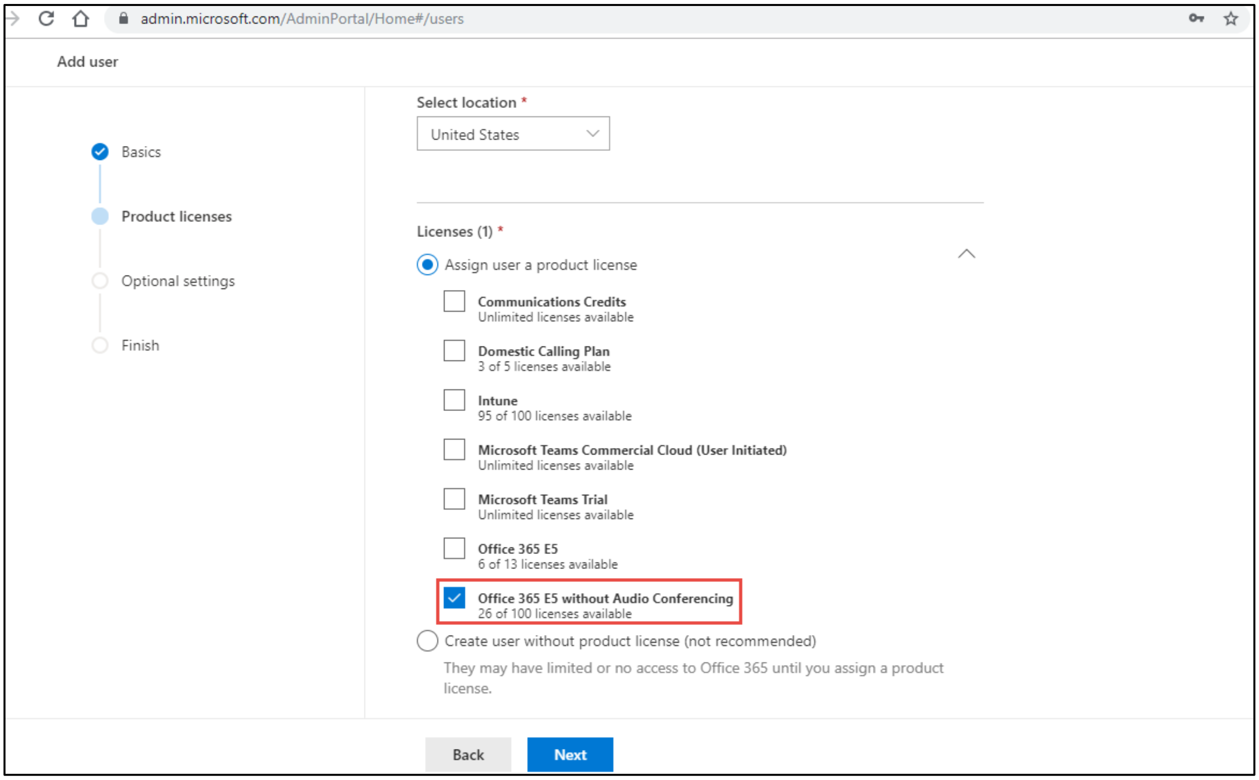


*Figure 28: Teams User Creation – Contd.*

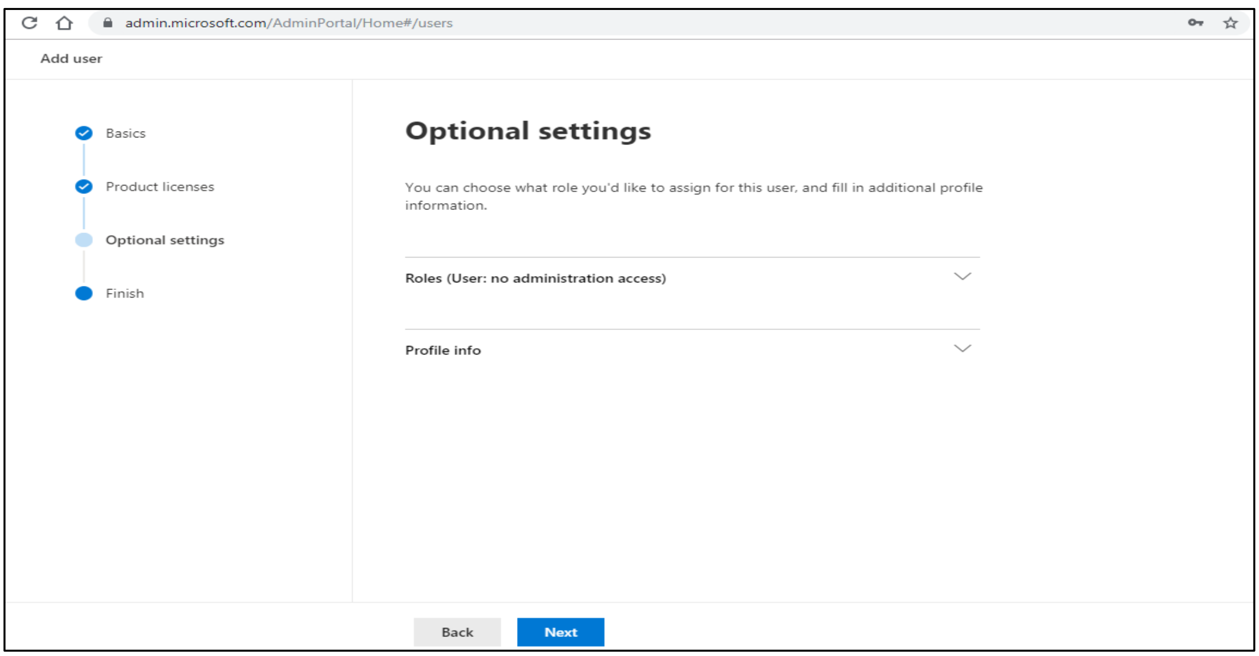


*Figure 29: Teams User Creation – Contd.*

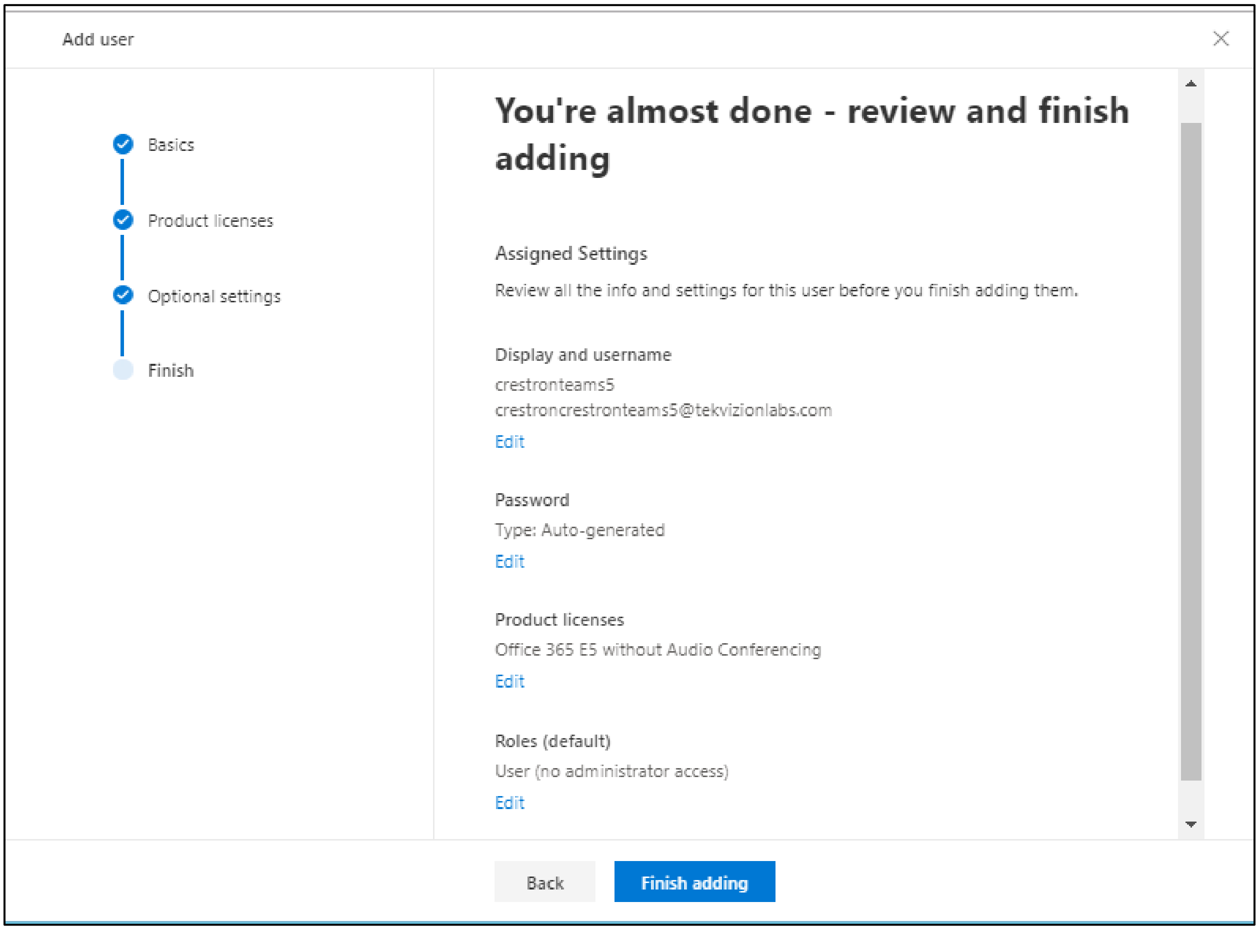


*Figure 30: Teams User Creation – Contd.*

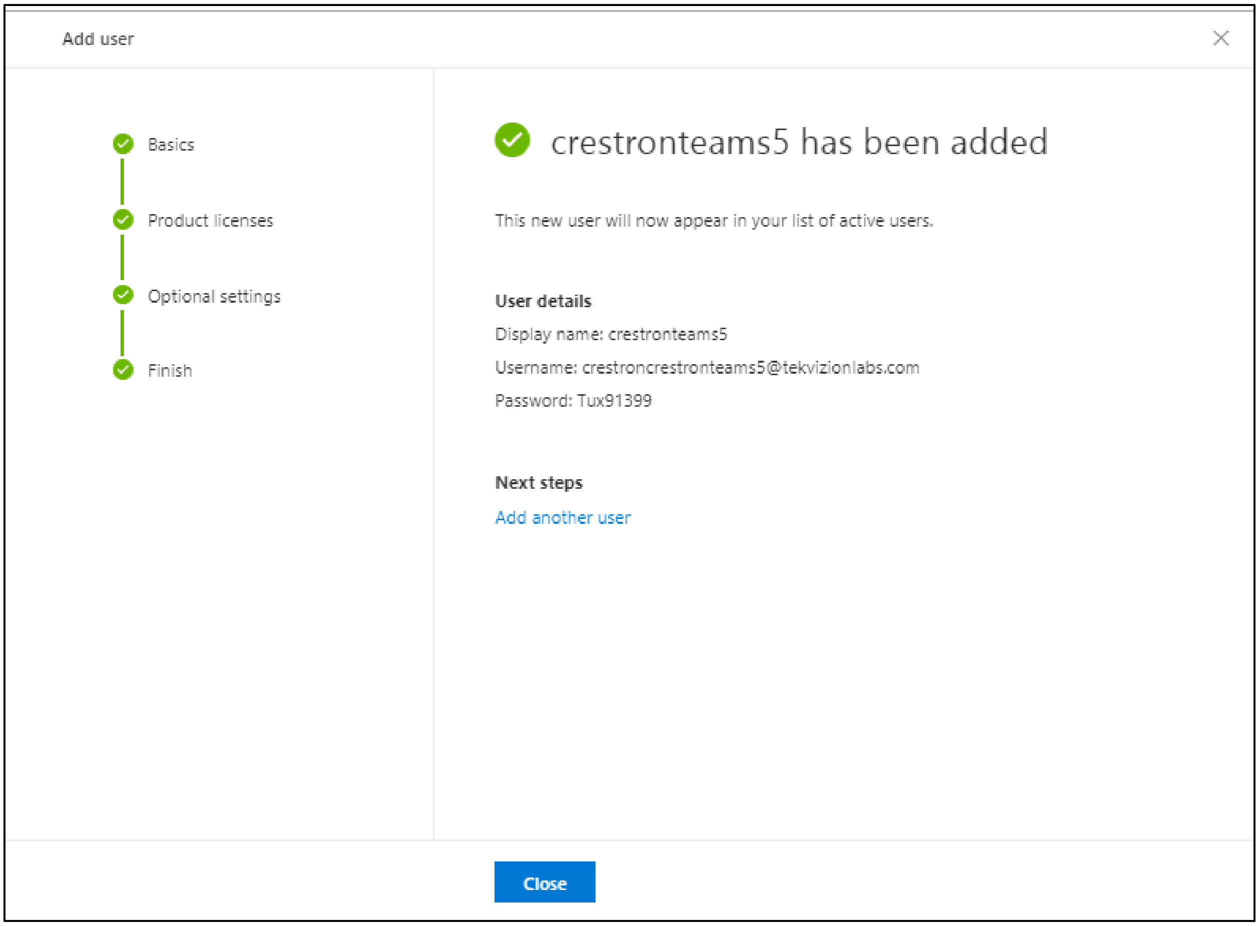

*Figure 31: Teams User Creation – Contd.*

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

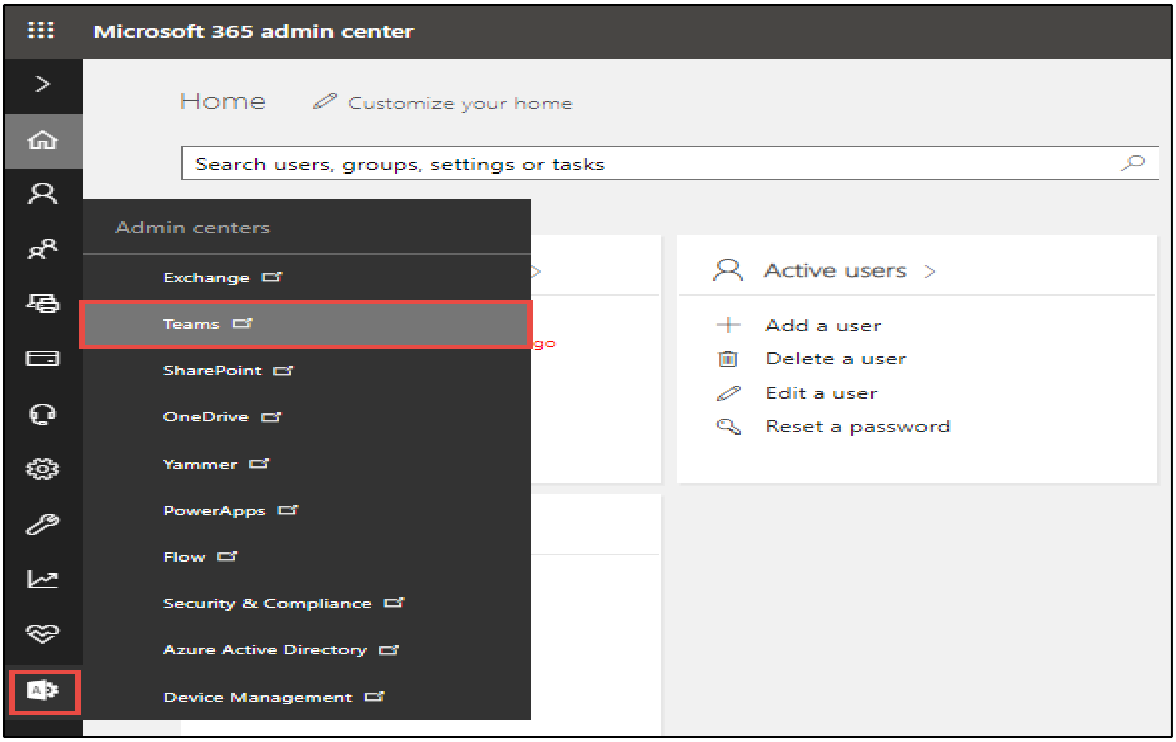


*Figure 32: Microsoft O365 admin*

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

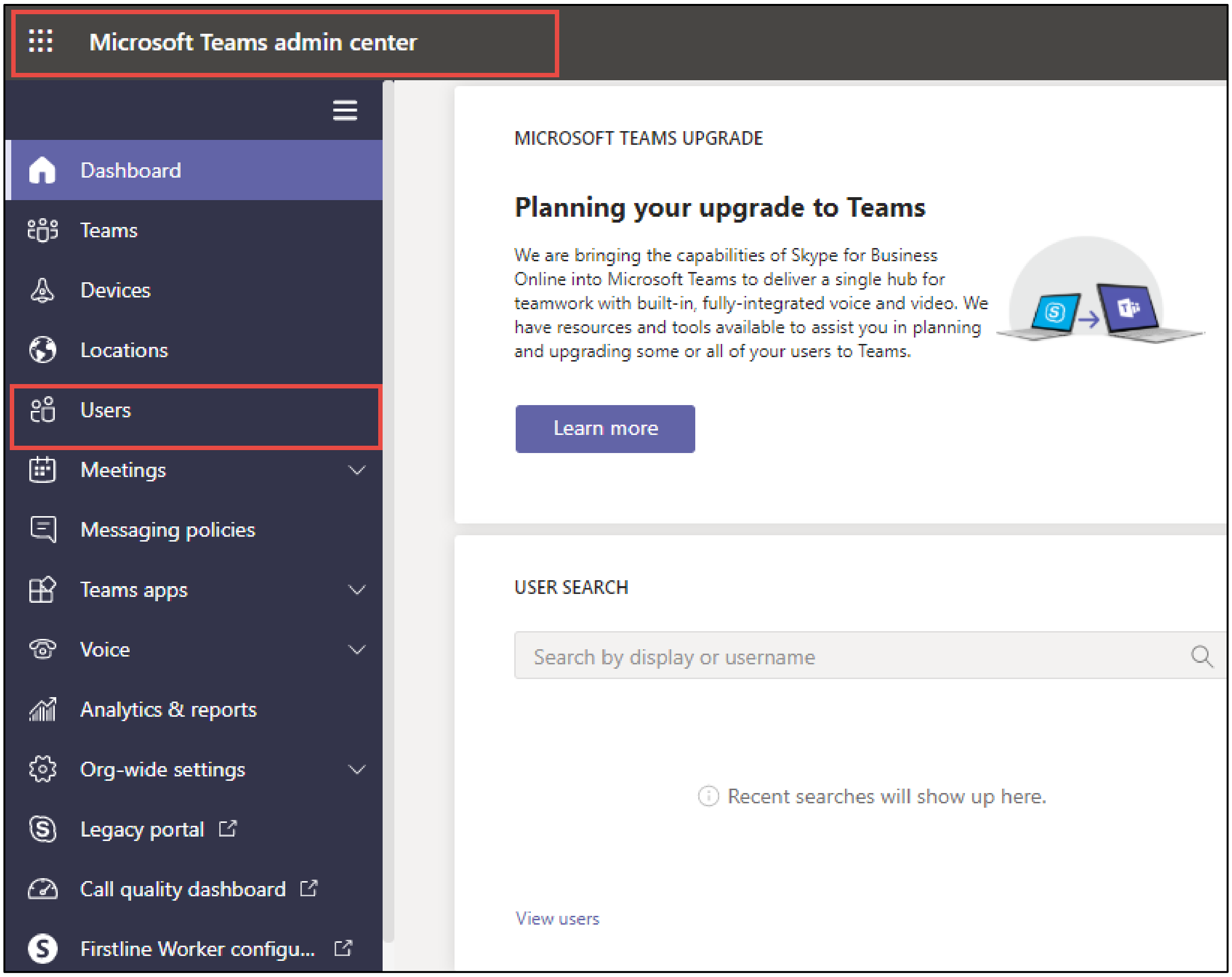


*Figure 33: Microsoft O365 admin*

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

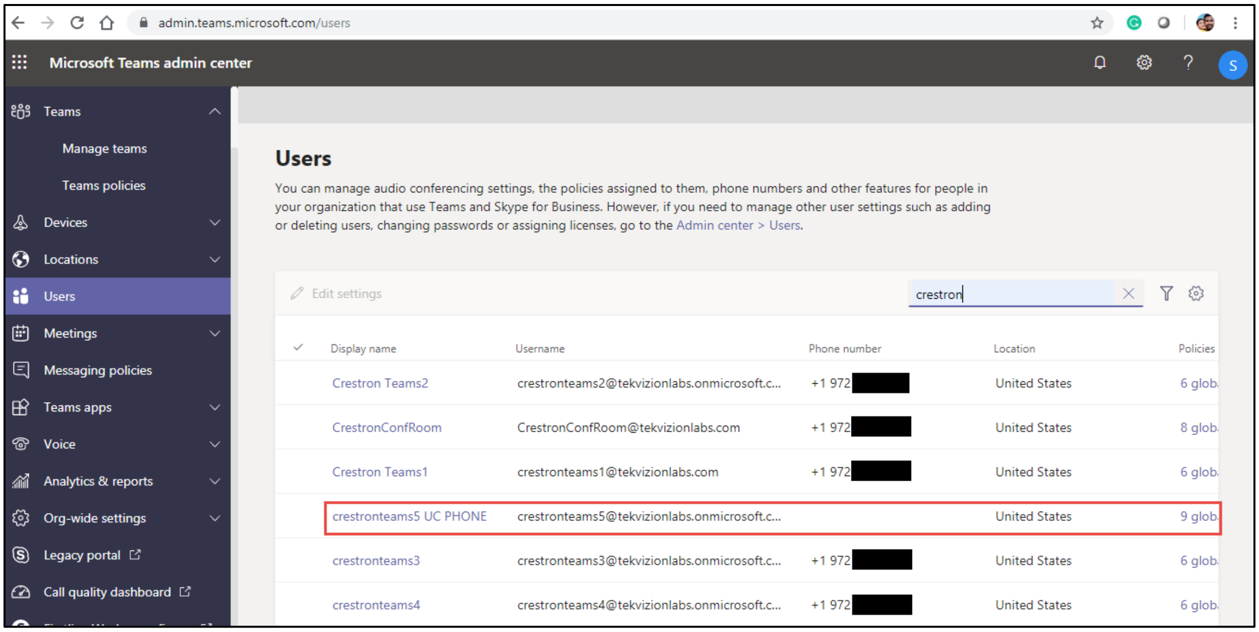


*Figure 34: Microsoft O365 admin*

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

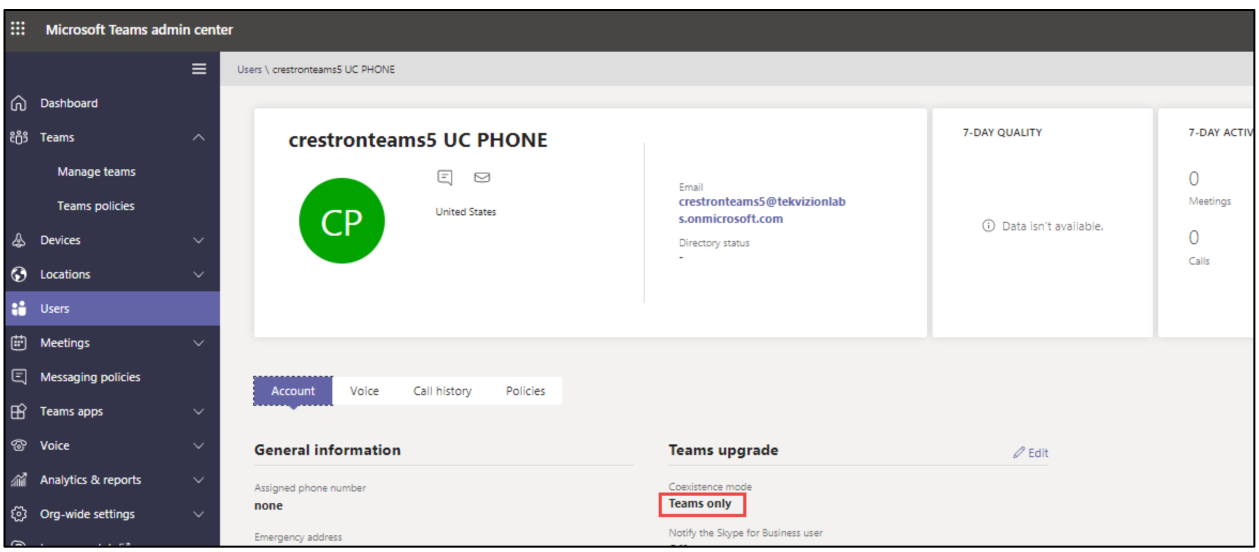


*Figure 35: Teams User*

### 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.

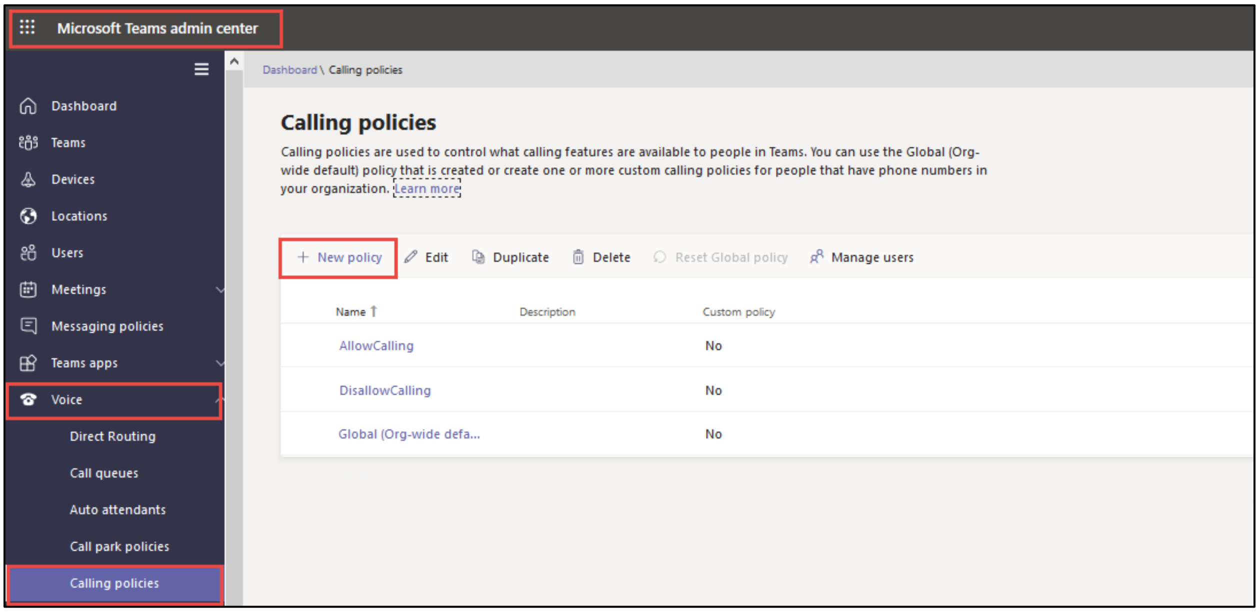


*Figure 36 – Calling Policy*

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

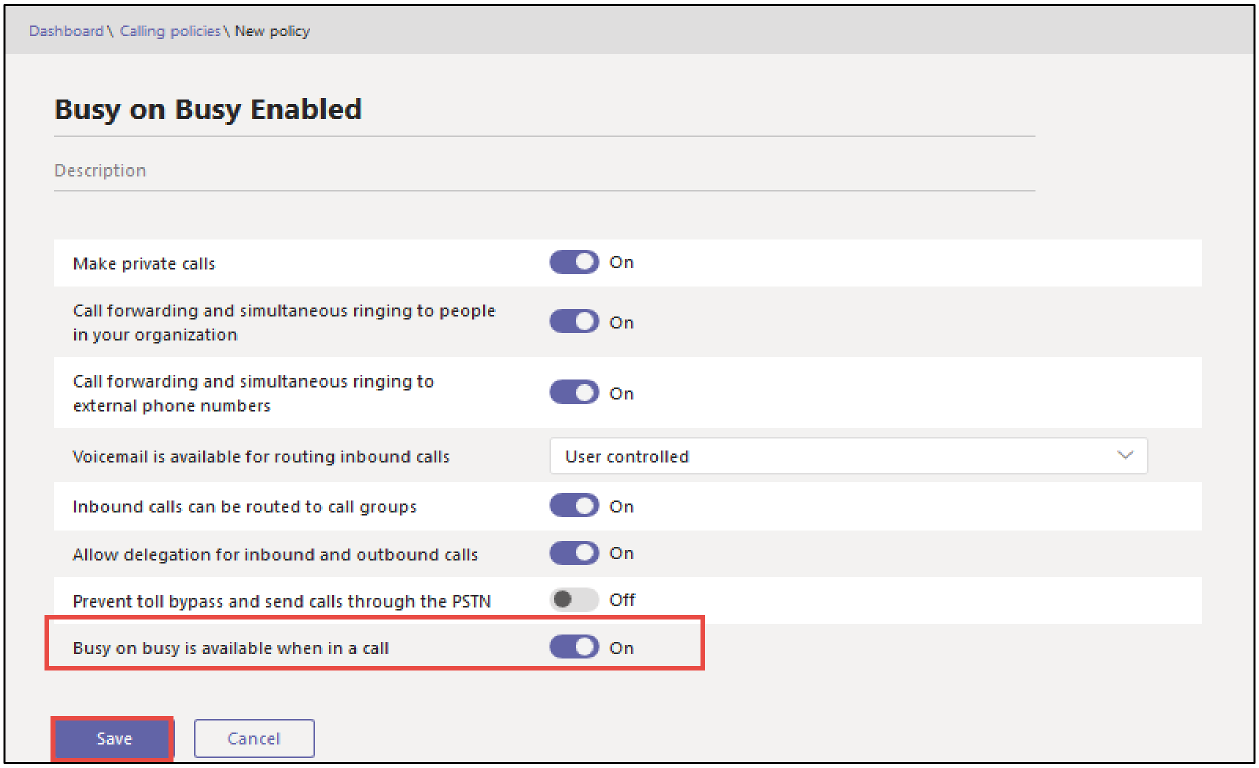


*Figure 37 – Calling Policy*

### 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. Cisco UBE 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.

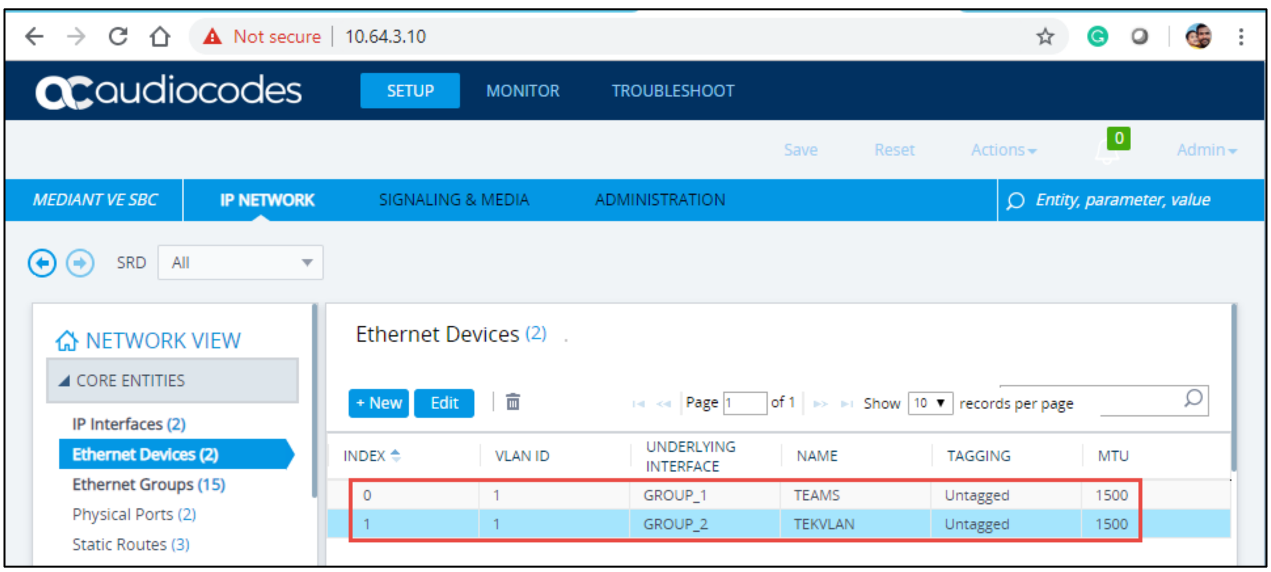


*Figure 48 – Ethernet Devices*

### 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:

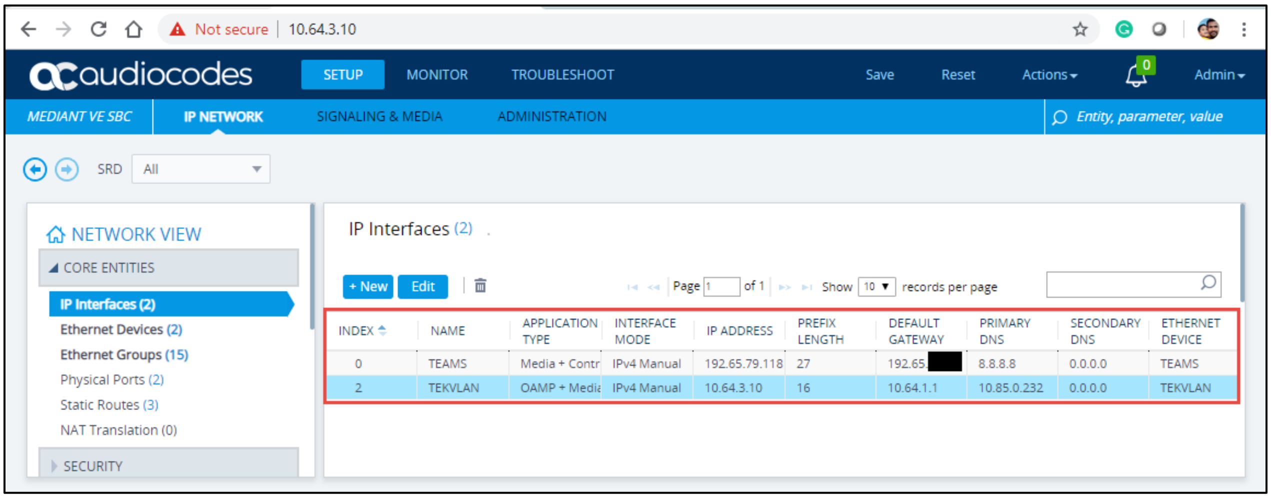


*Figure 49 – IP interface Devices*

IP interface TEAMS

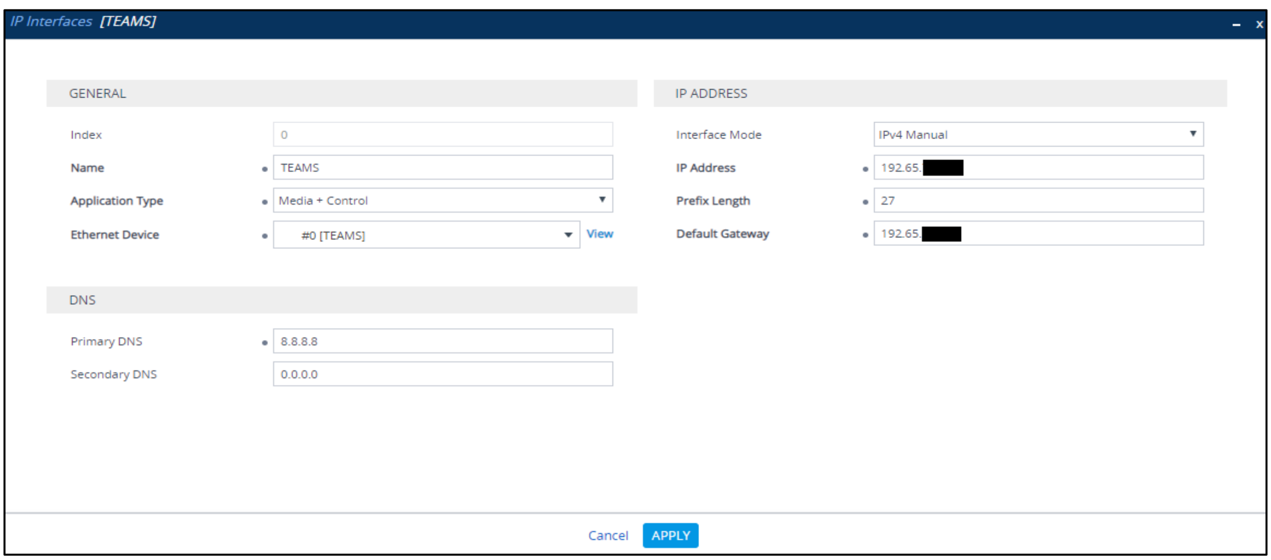


*Figure 50 – IP interface Devices*

IP Interfaces – TEKVLAN

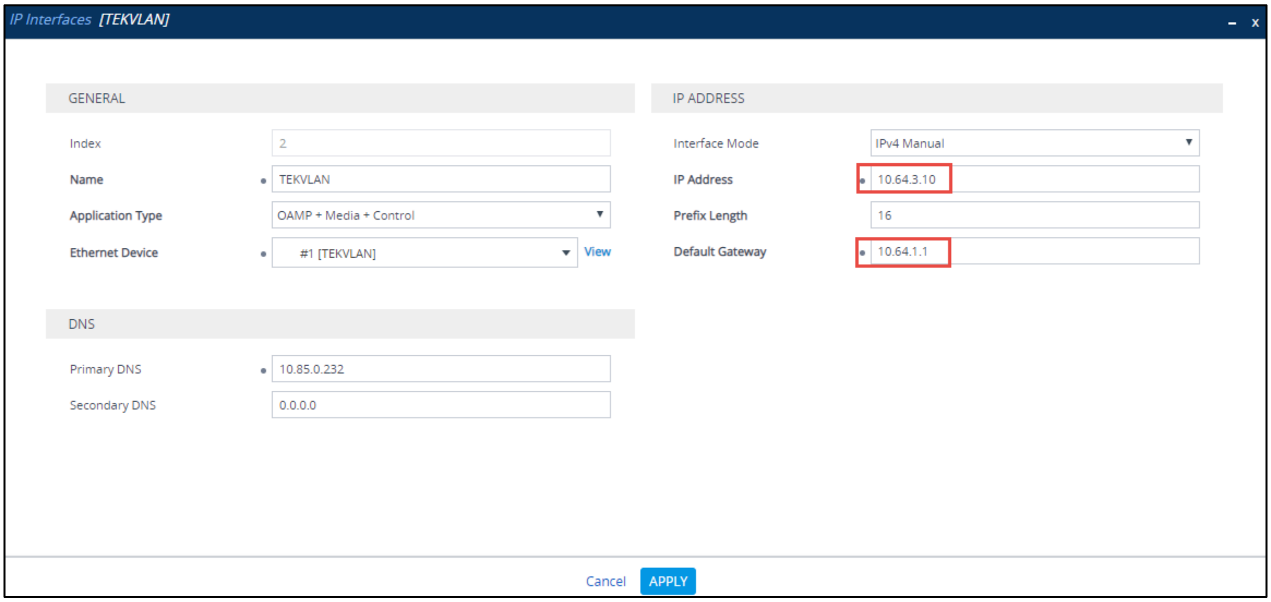


*Figure 51 – IP interface Devices*

### 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

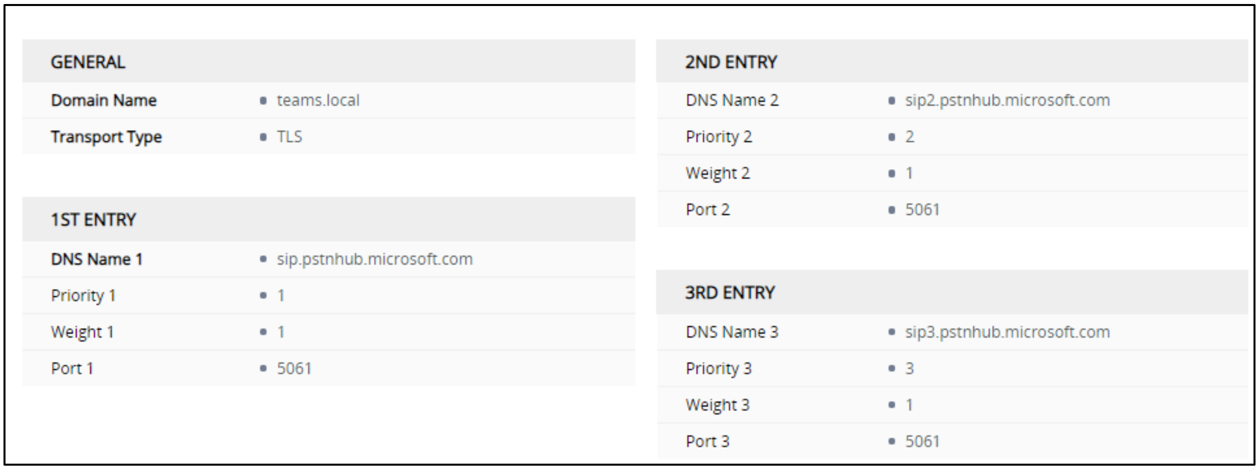


*Figure 52 – DNS SRV Records*

### 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

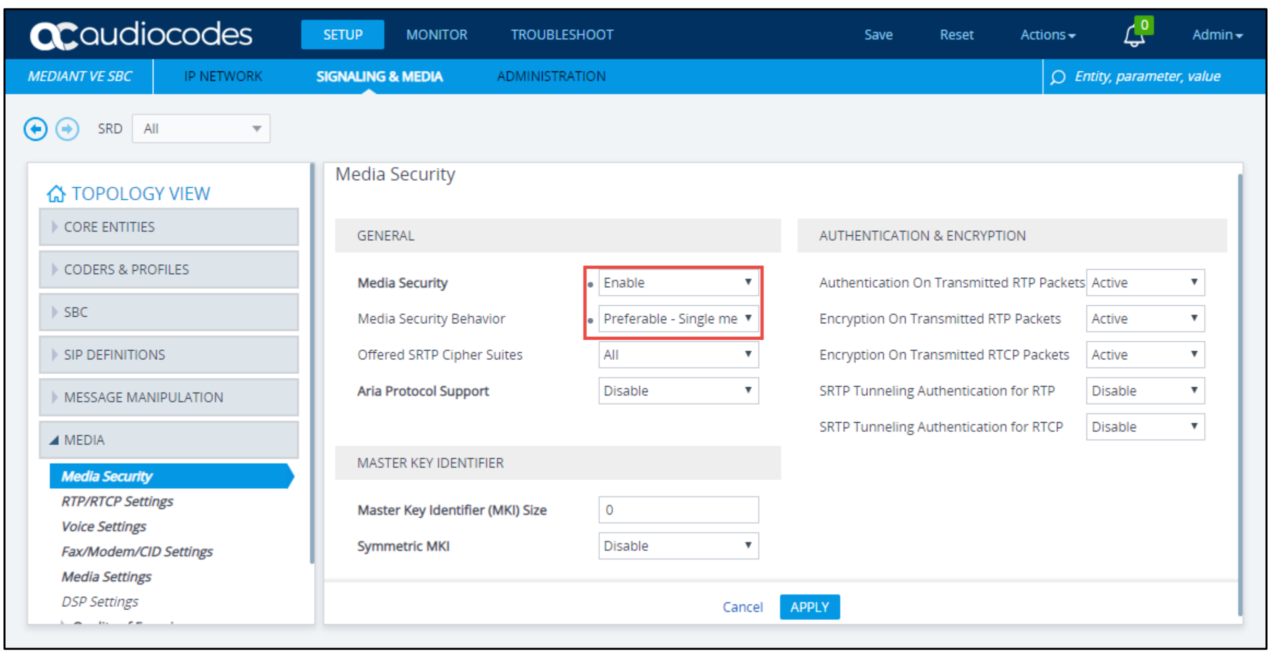


*Figure 53 – Media Security*

### 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.

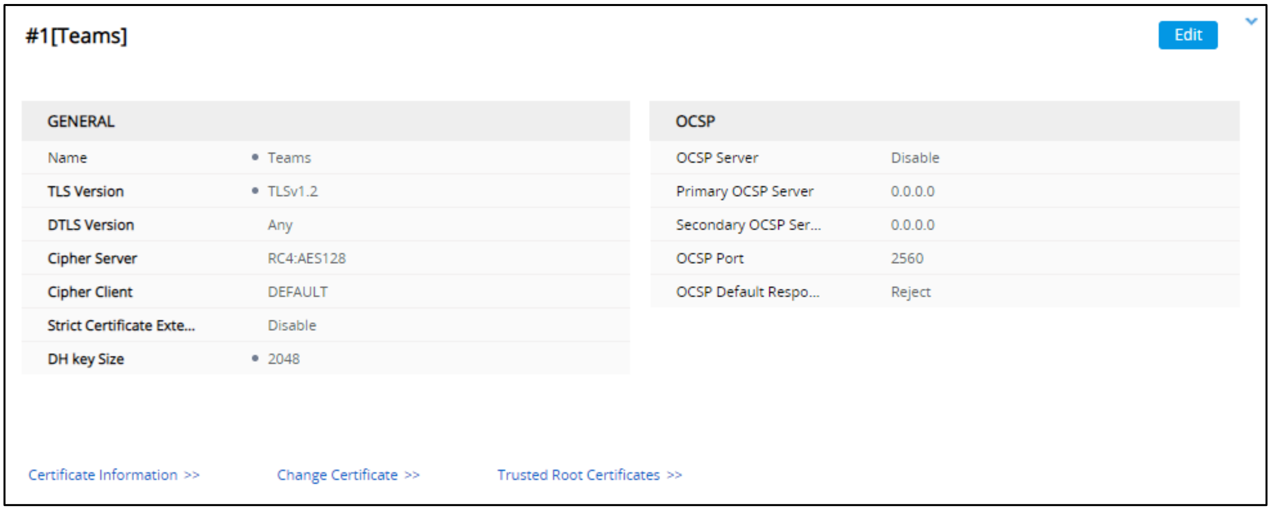


*Figure 54 – Teams TLS*

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.**

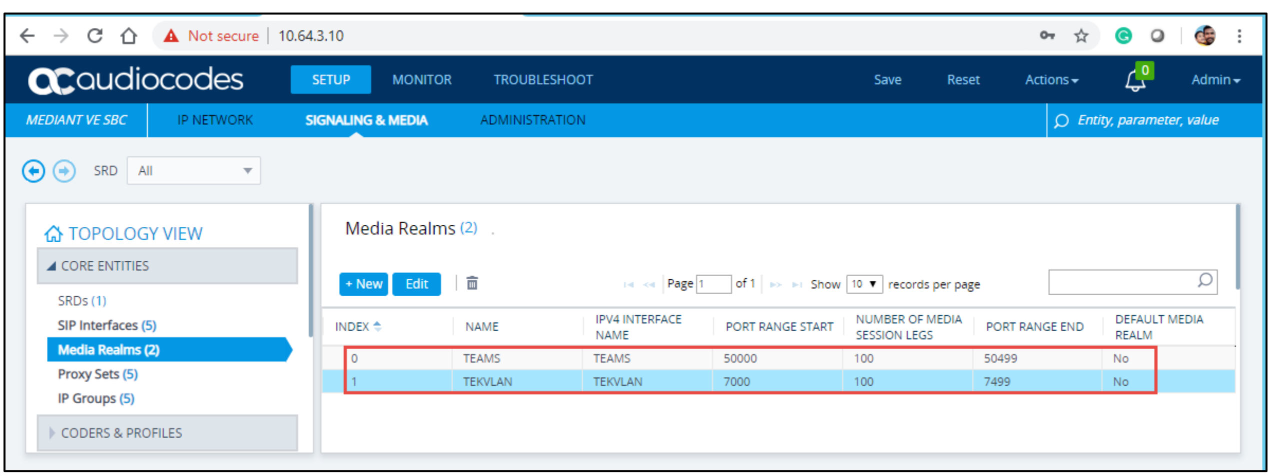


*Figure 55 – Media Realms*

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

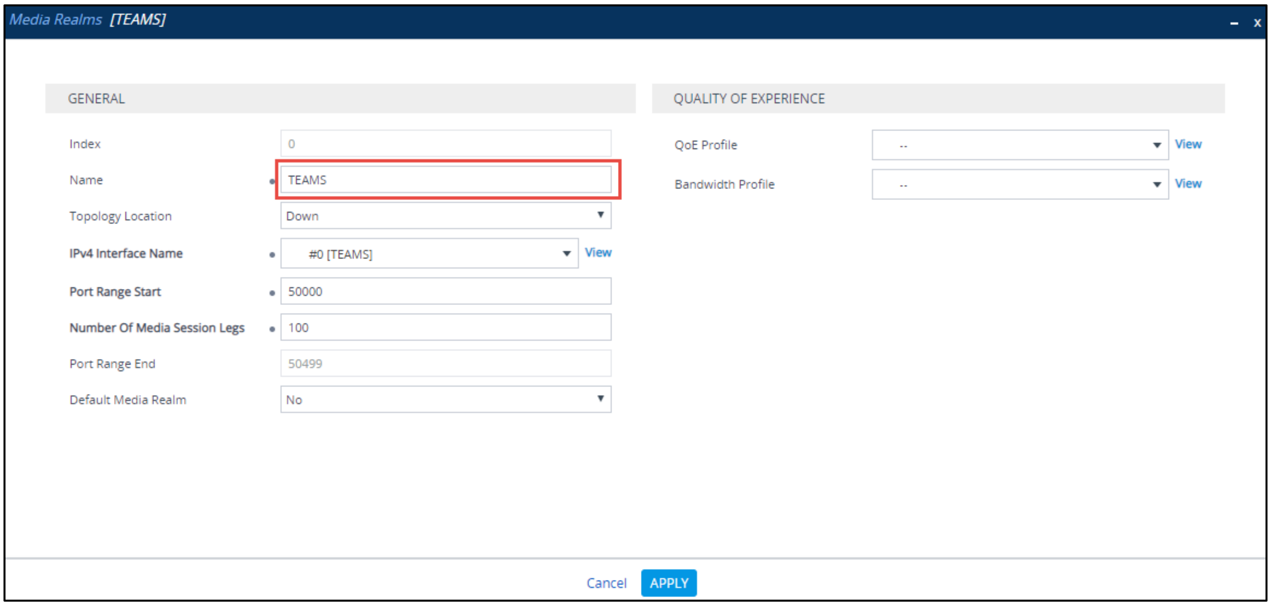


*Figure 56 – Teams*

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

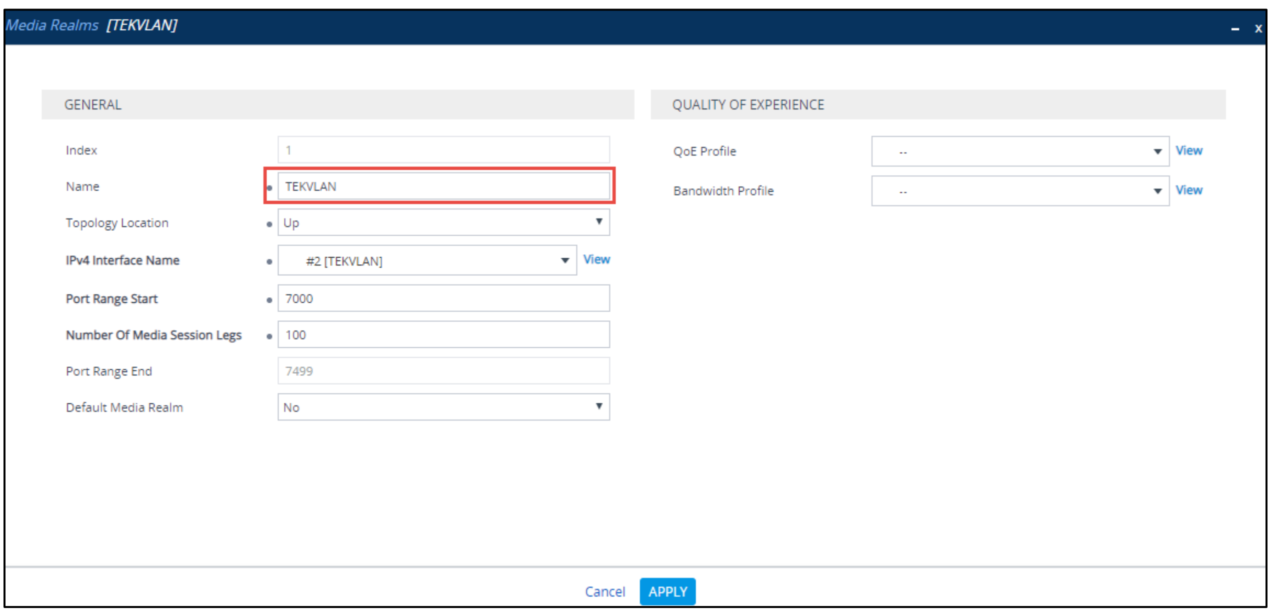


*Figure 57 – LAN LAB*

### 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.

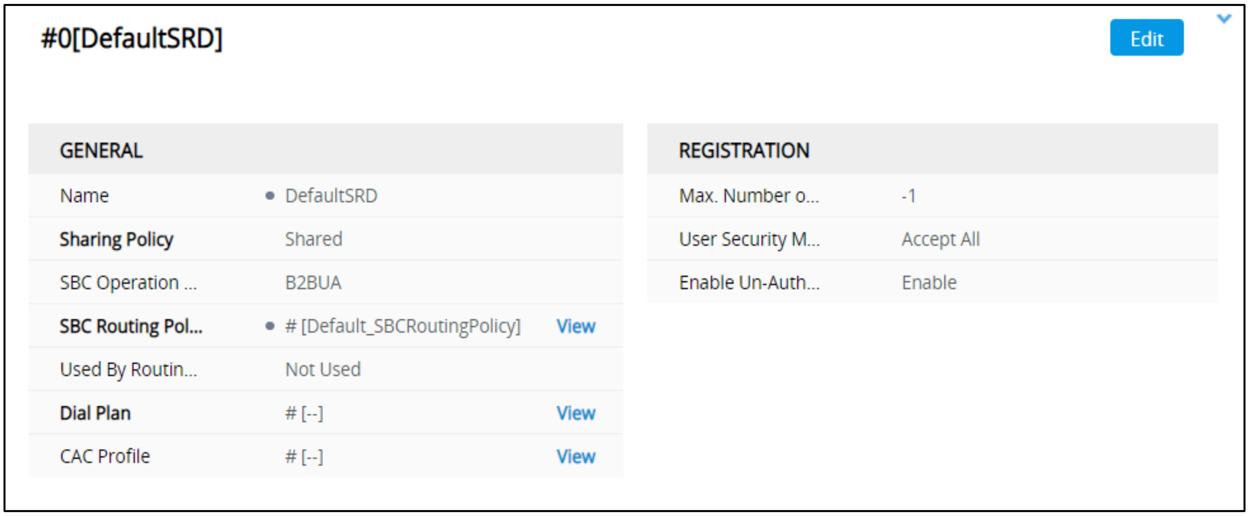


*Figure 58 – Default SRD*

### 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.

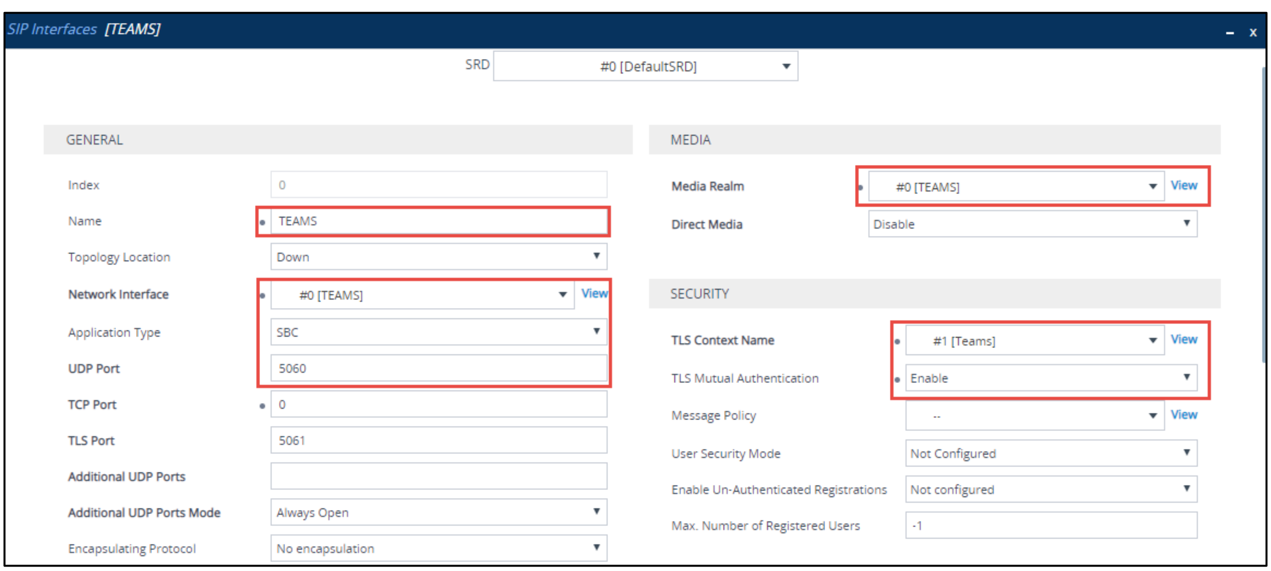

Figure 59 – Teams

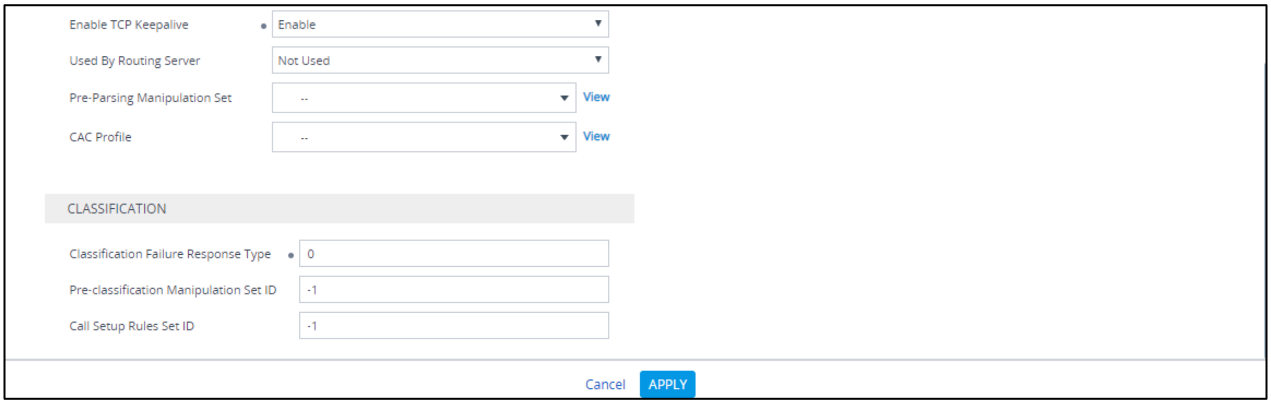


Figure 60 – Teams

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

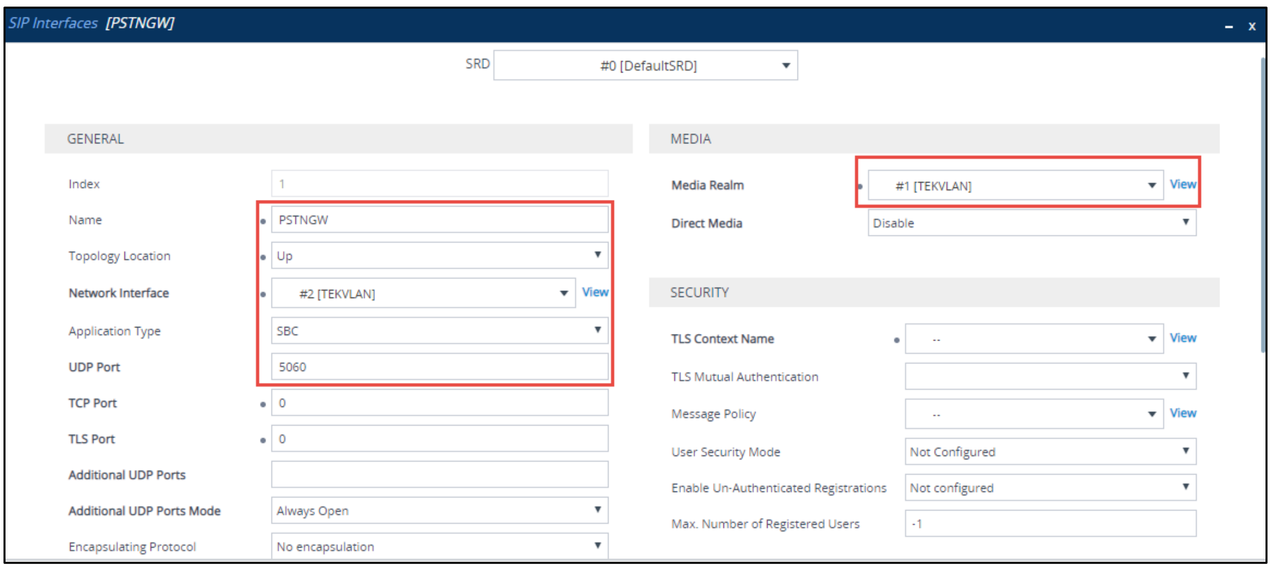


Figure 61 – PSTN

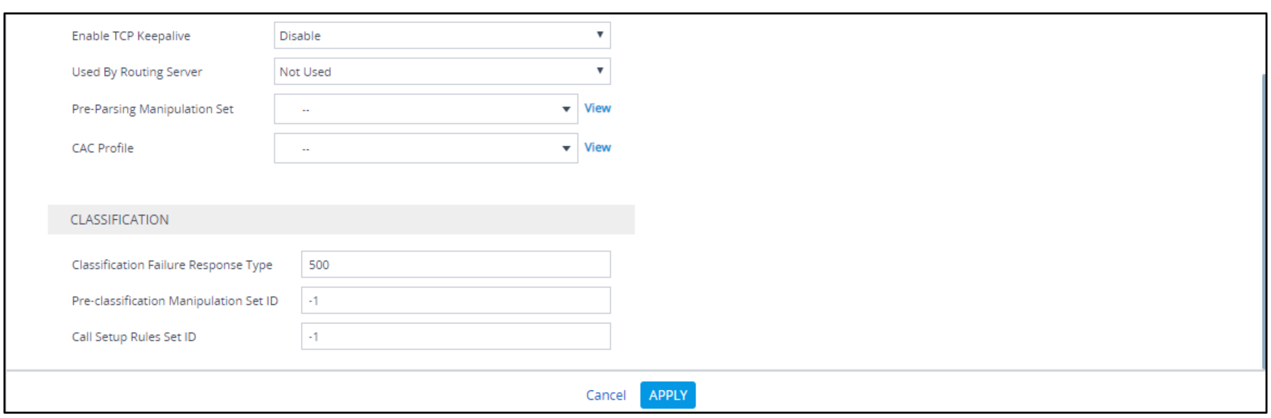


Figure 62 – PSTN

Configure a SIP interface for the LAN (towards Cisco UCM) as shown below.

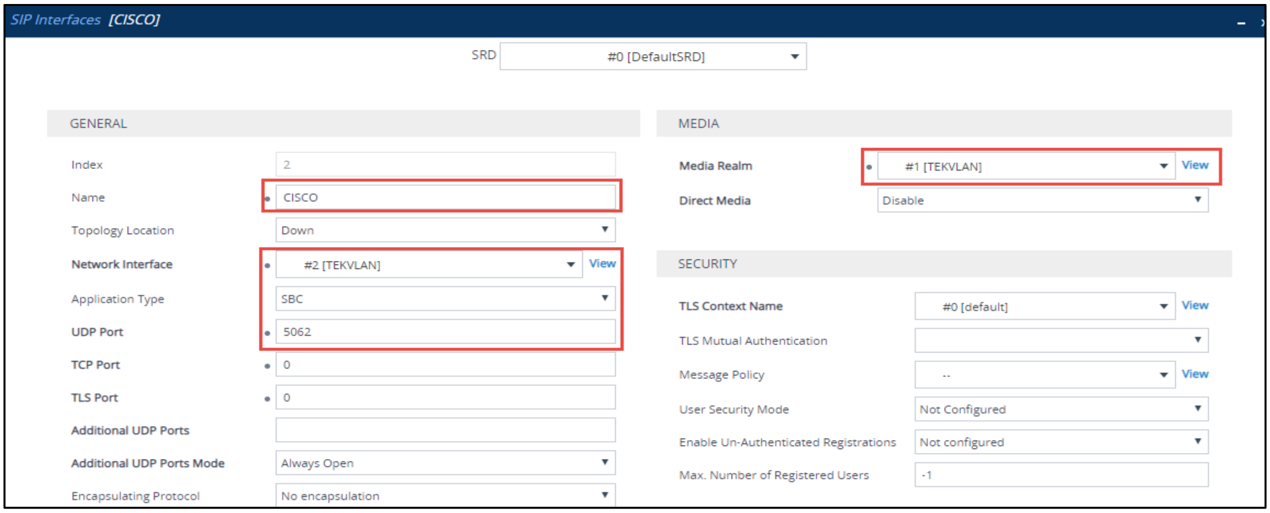


Figure 63 – Cisco

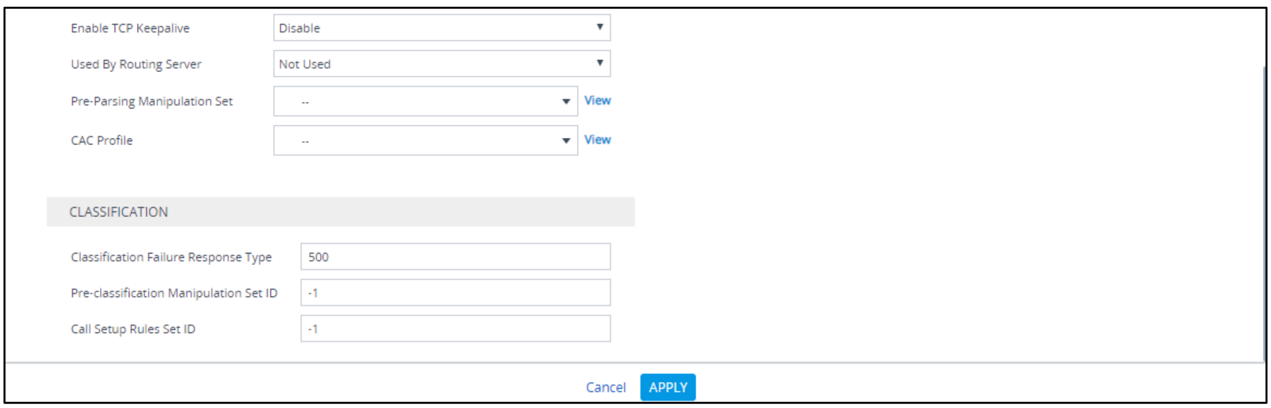


Figure 64 – Cisco

### 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 Cisco UBE. 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.

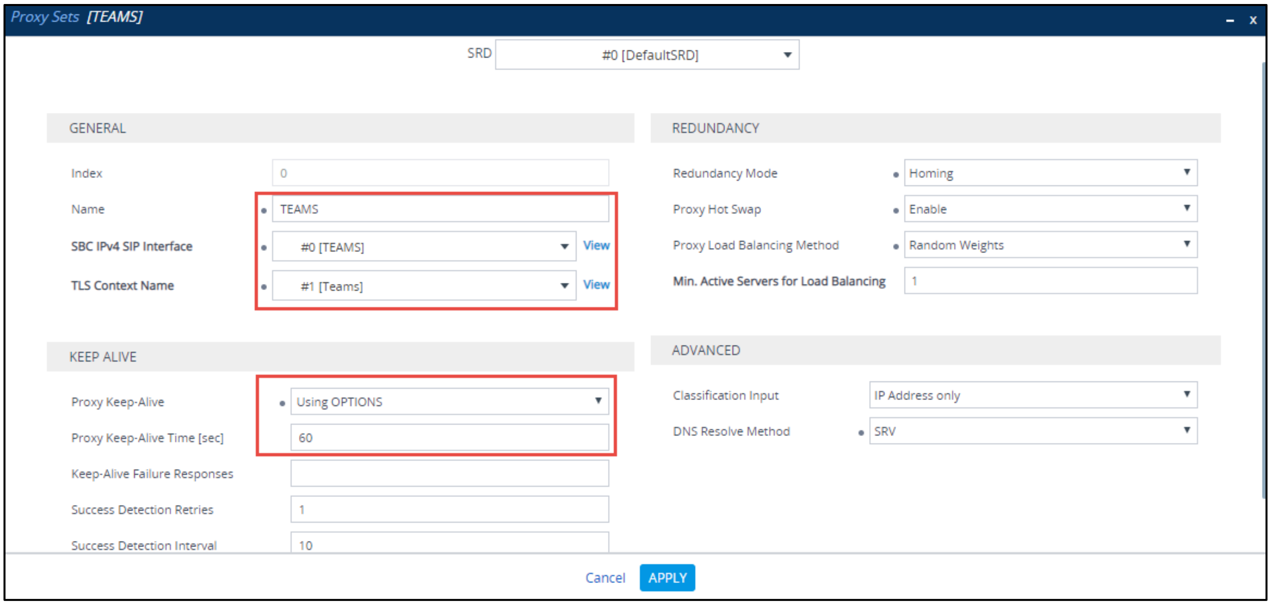


*Figure 65 – Teams*

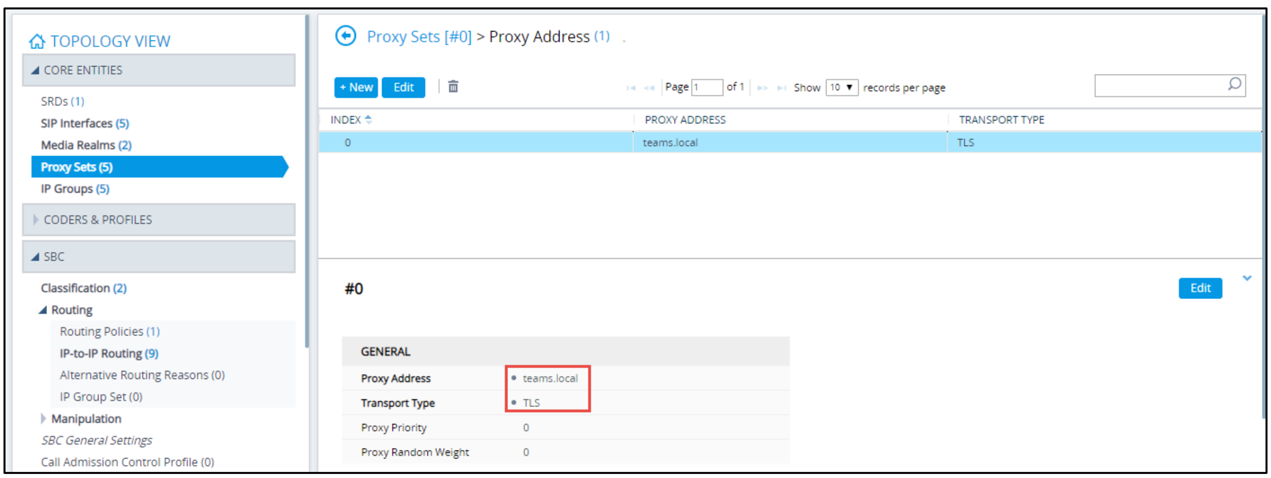


*Figure 66 – Teams*

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

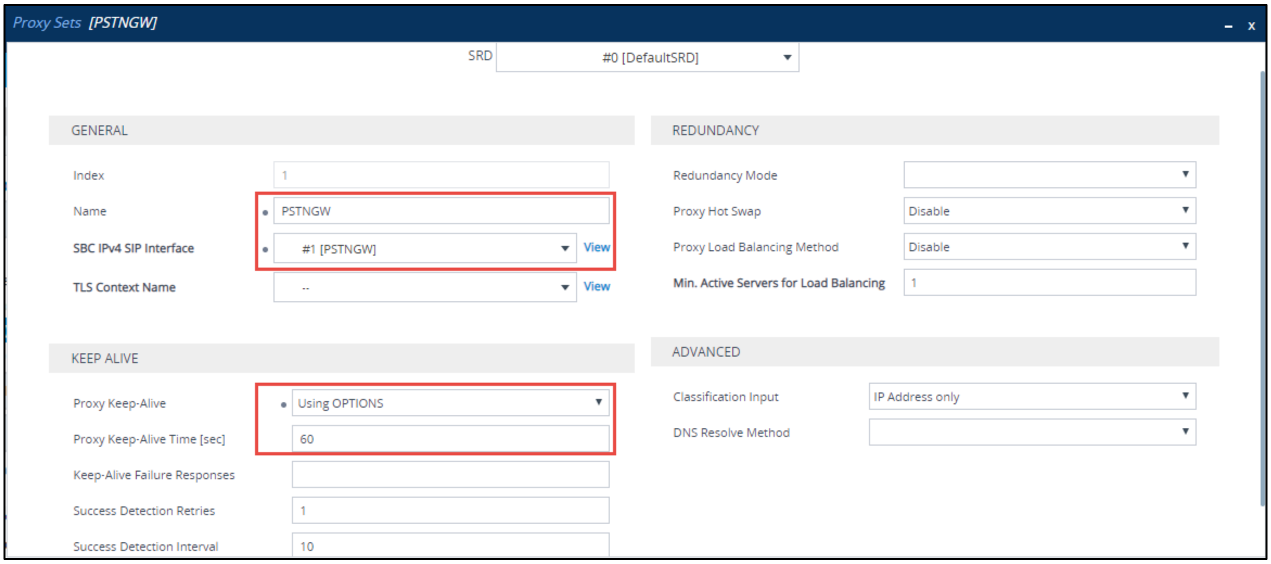

*Figure 67 – PSTN Gateway*

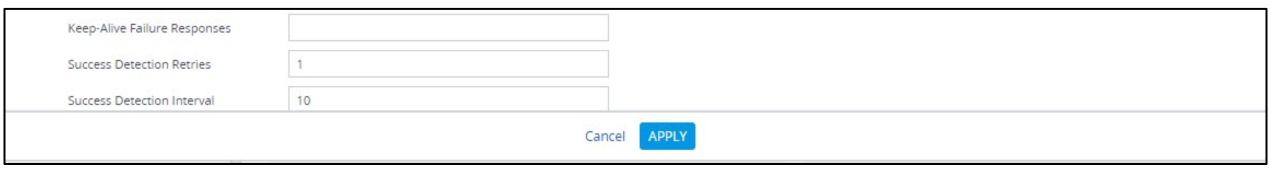

*Figure 68 – PSTN Gateway*

Configure a Proxy Set for the Cisco UCM as shown below.

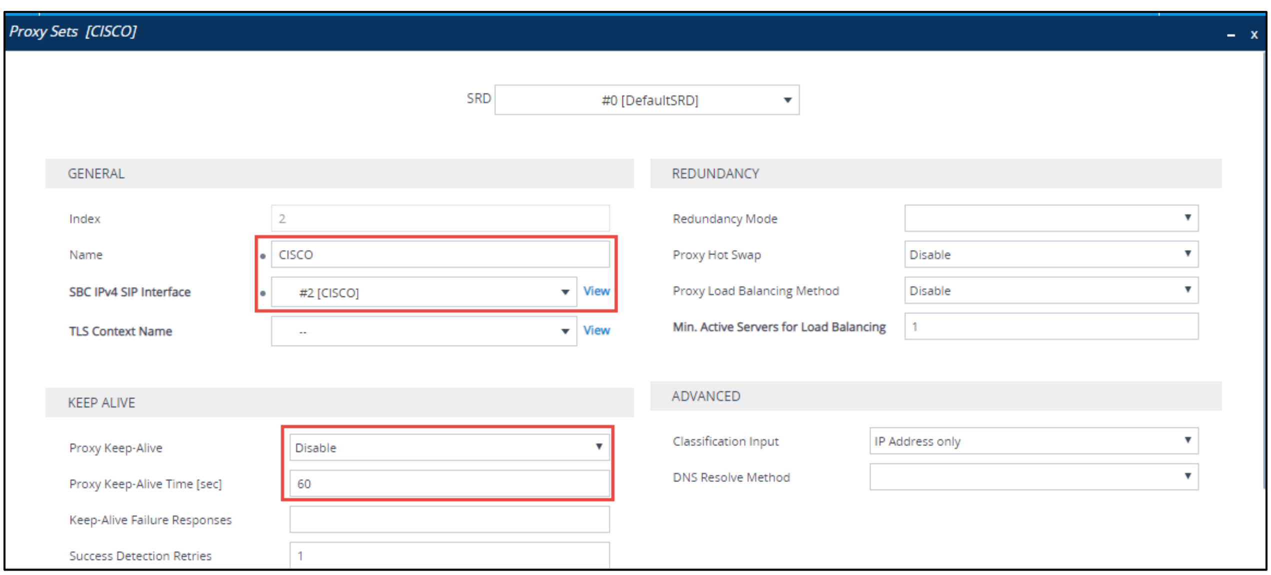

*Figure 69 – Cisco UCM*

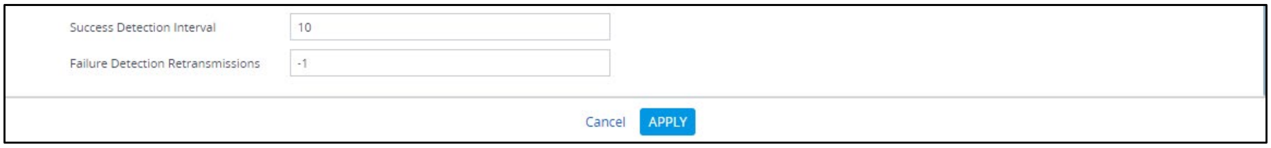

*Figure 70 – Cisco UCM*

### 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
- Cisco – 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

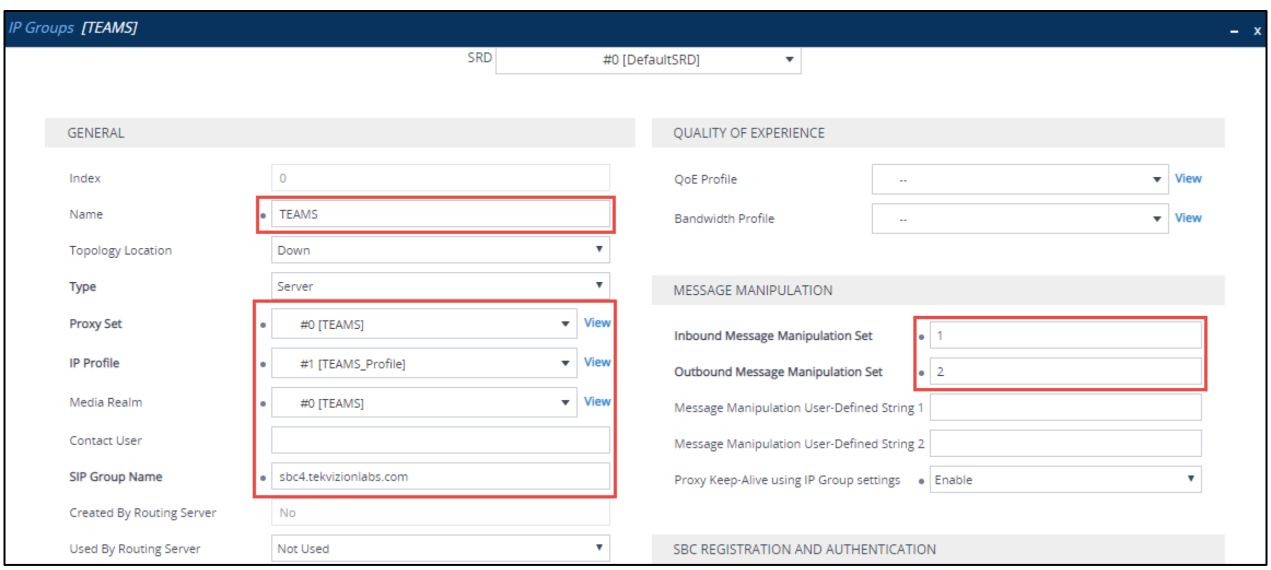

*Figure 71 – IP Group – Teams – Contd.*

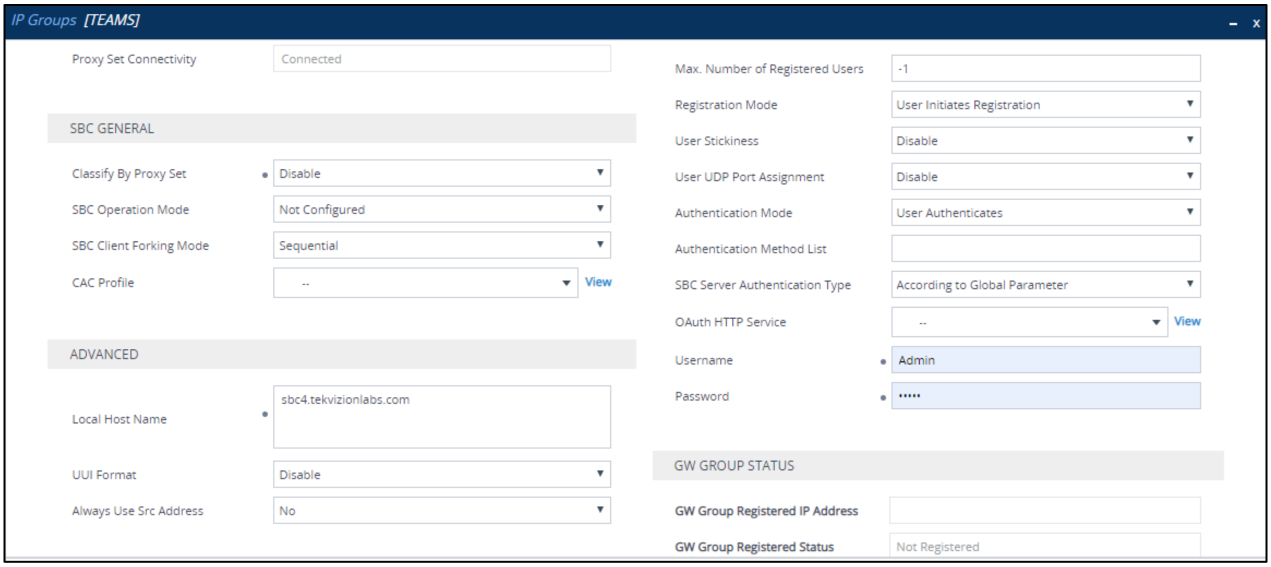


*Figure 72 – IP Group – Teams – Contd.*

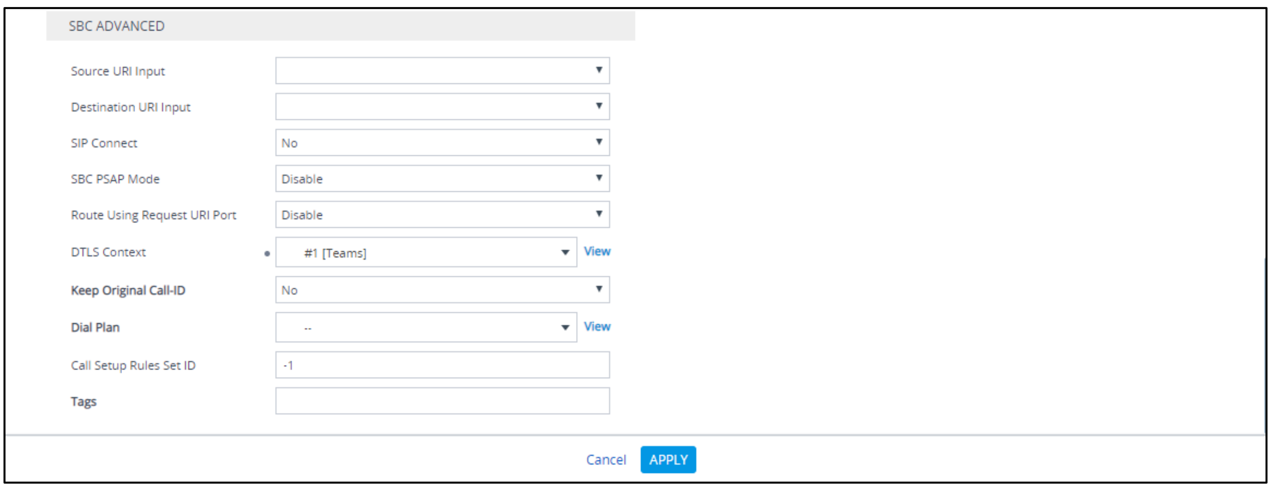


*Figure 73 – IP Group – Teams*

Configure an IP Group for PSTN Gateway as shown below

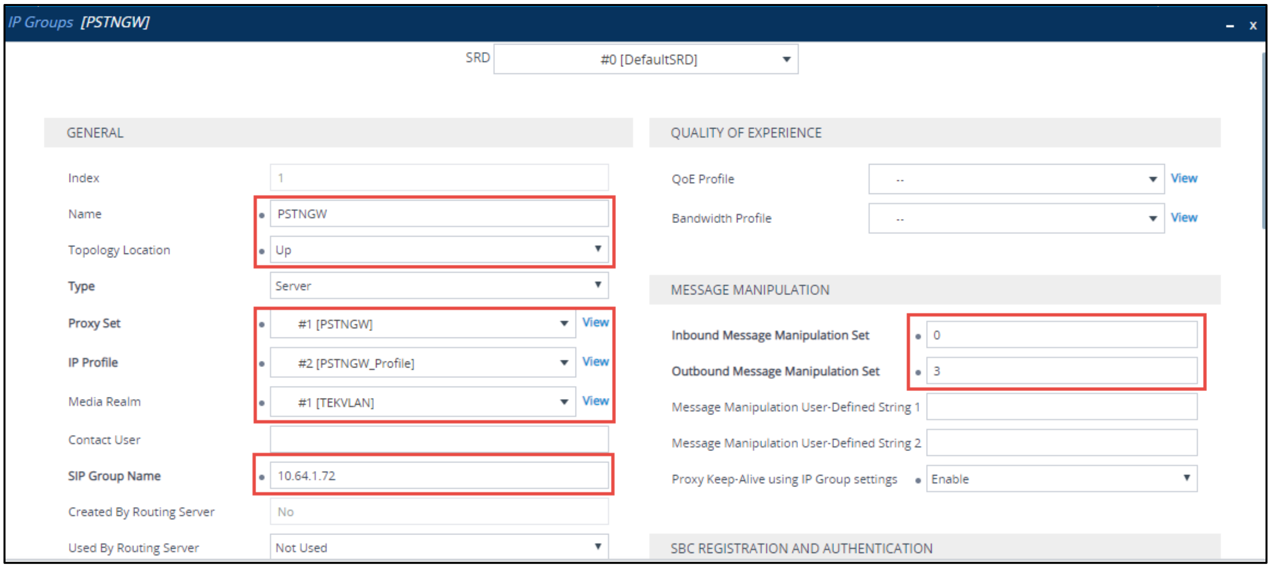

Figure 74 – IP Group – PSTN – Contd.

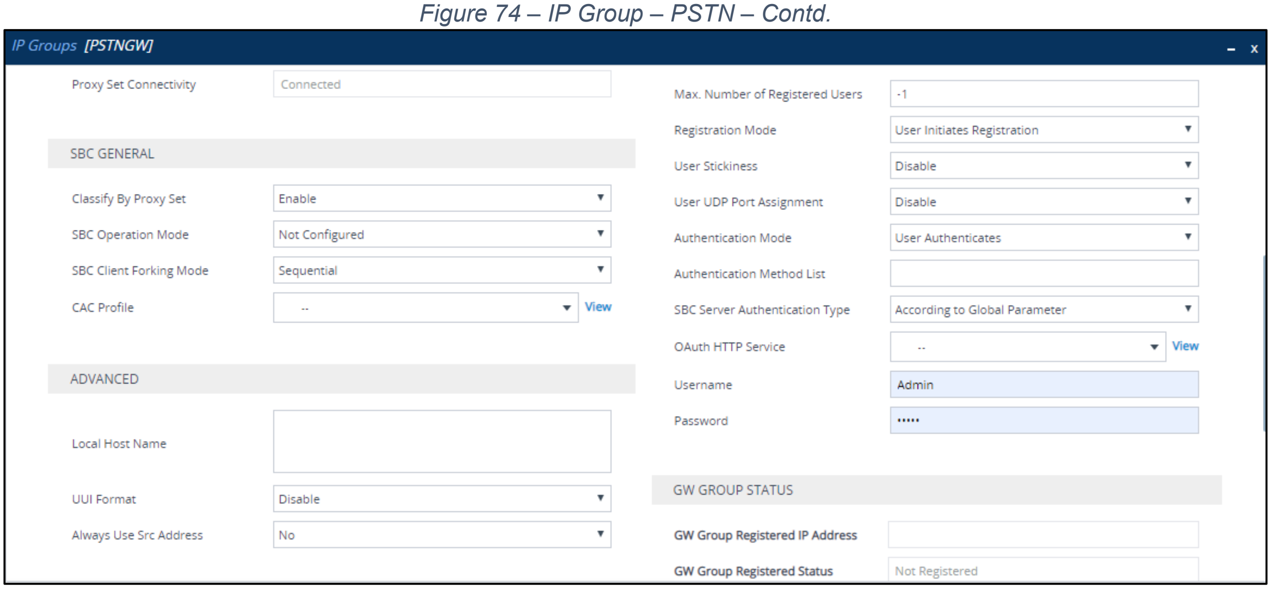

Figure 75 – IP Group – PSTN – Contd.

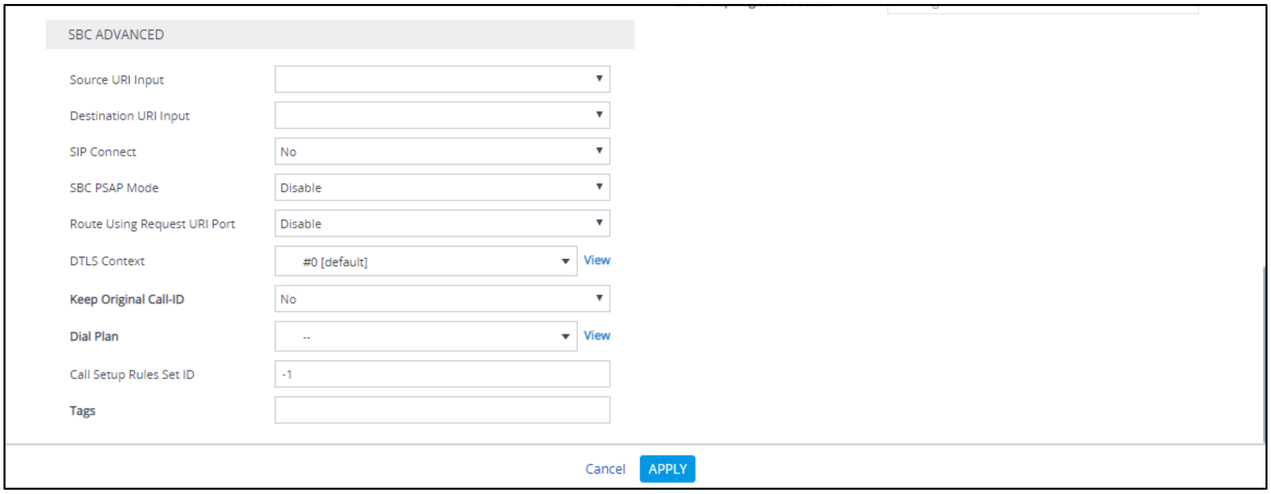


*Figure 76 – IP Group*

Configure an IP Group for Cisco UCM as shown below

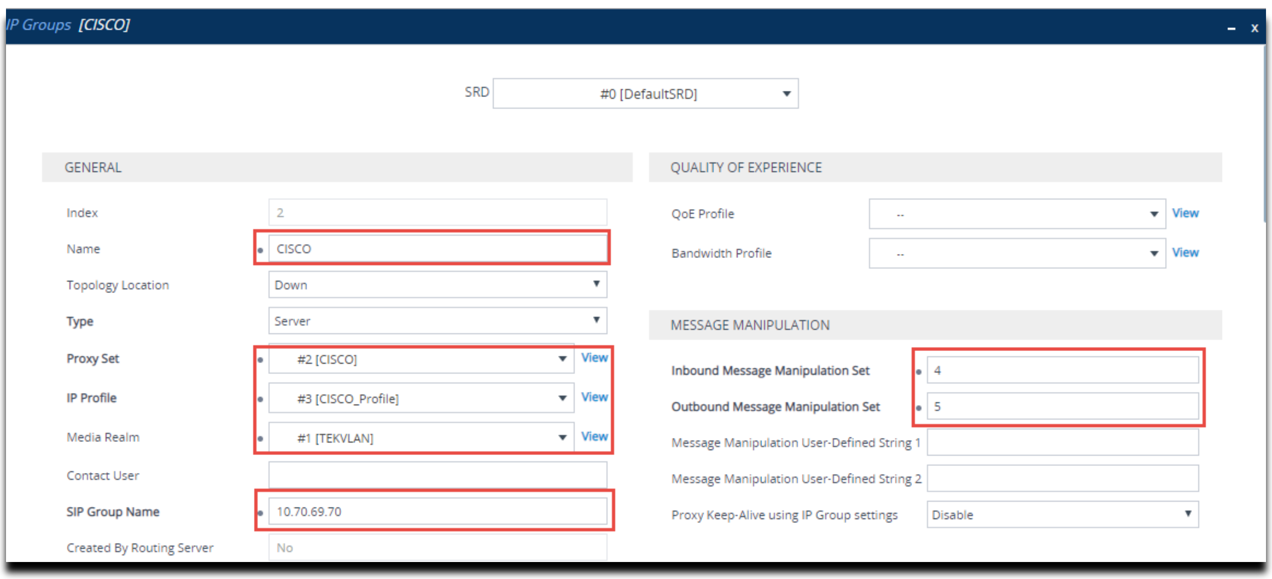


*Figure 77 – IP Group – Cisco – Contd.*

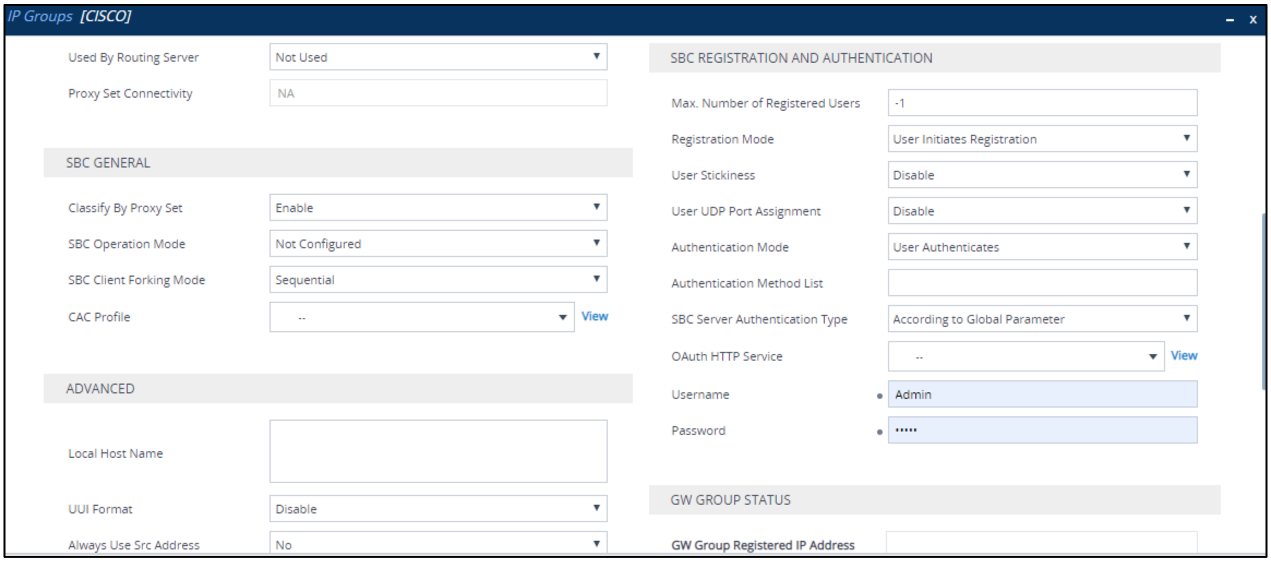


Figure 78 – IP Group – Cisco – Contd.

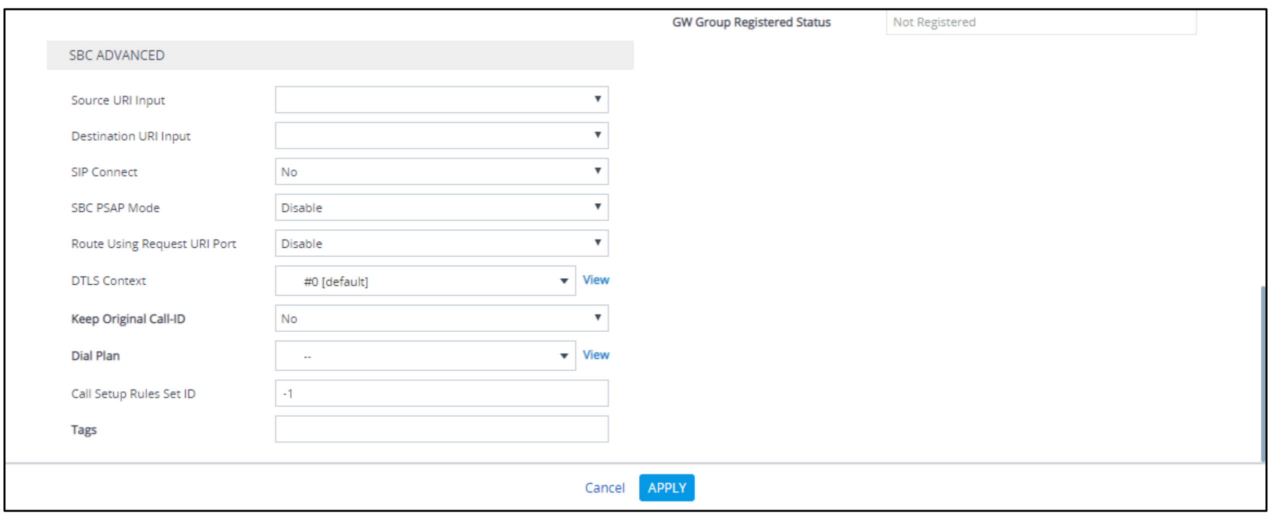


Figure 79 – IP Group

## 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
- Cisco – 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.

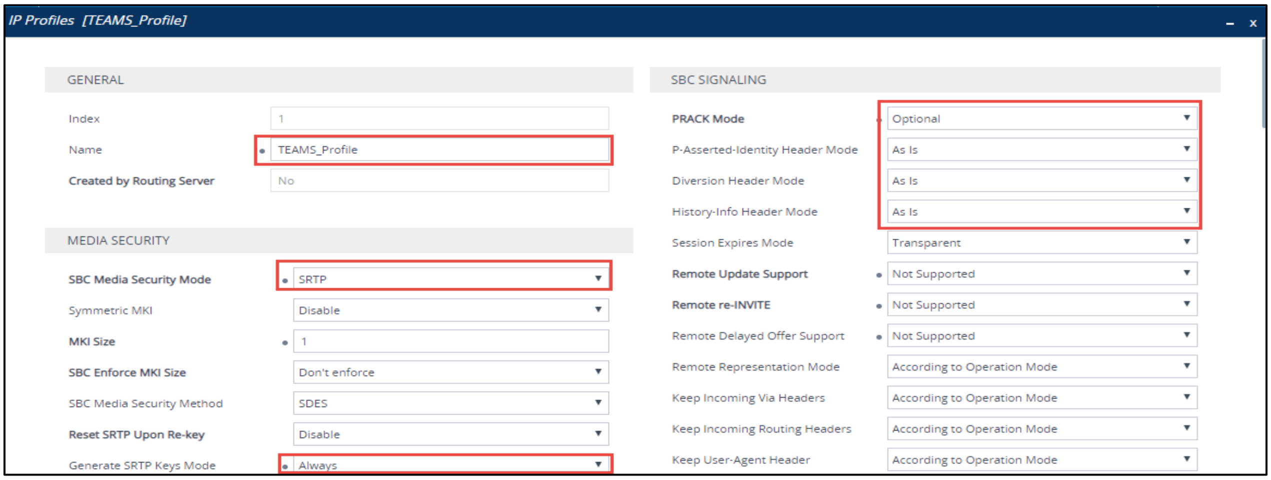


*Figure 80 – IP Profile – Teams – Contd.*

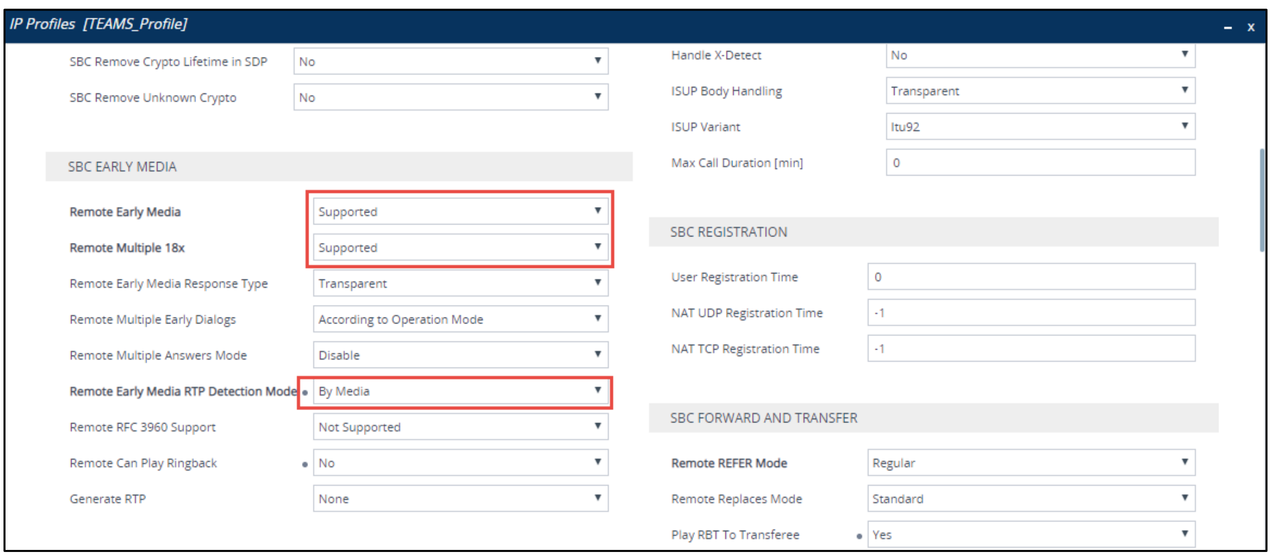


*Figure 81 – IP Profile – Teams – Contd.*

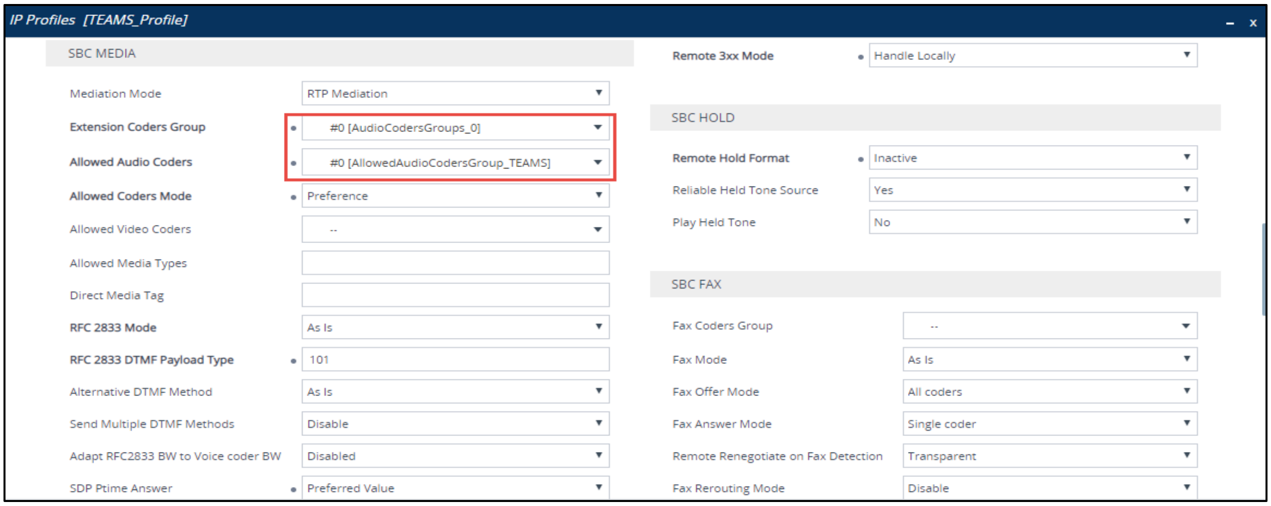

*Figure 82 – IP Profile – Teams – Contd.*

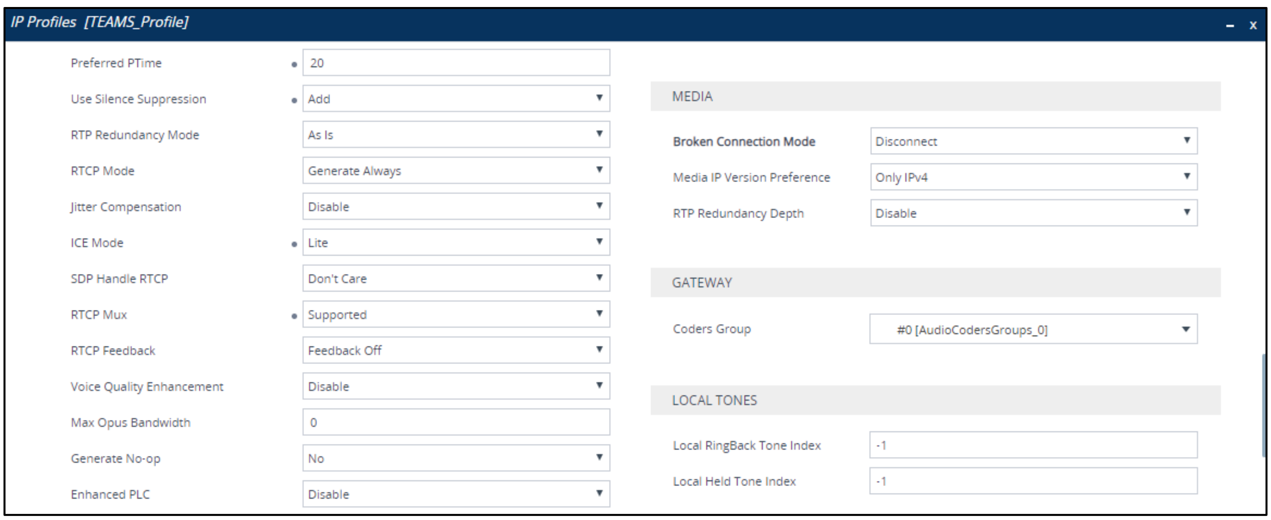

*Figure 83 – IP Profile – Teams – Contd.*

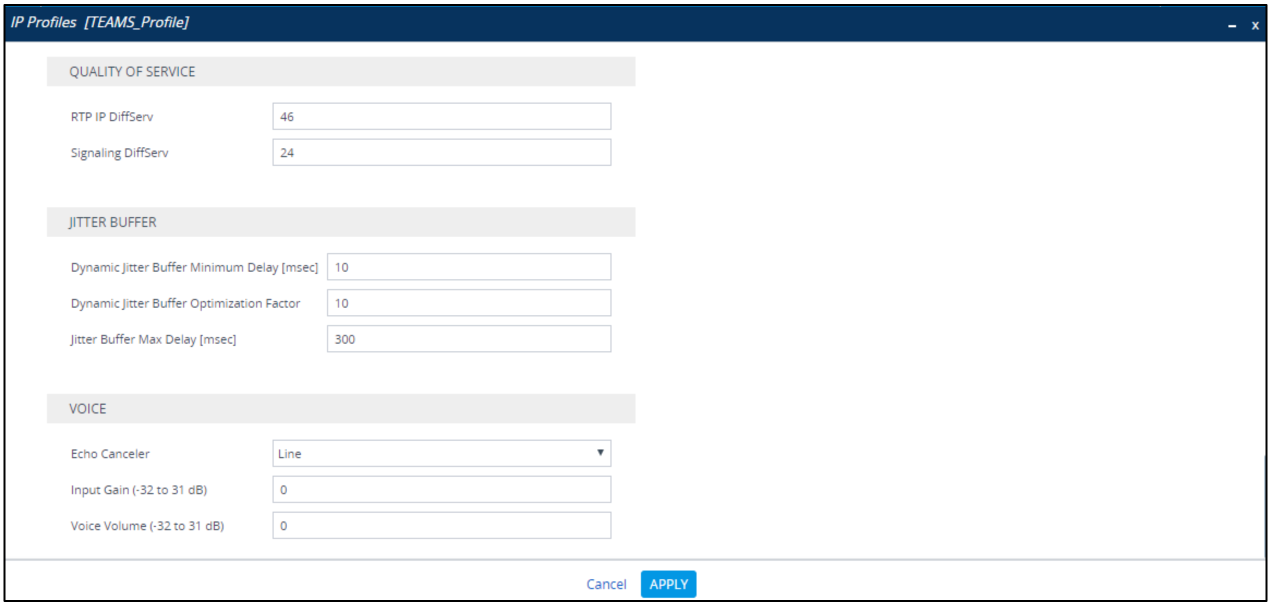


*Figure 84 – IP Profile – Teams – Contd.*

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

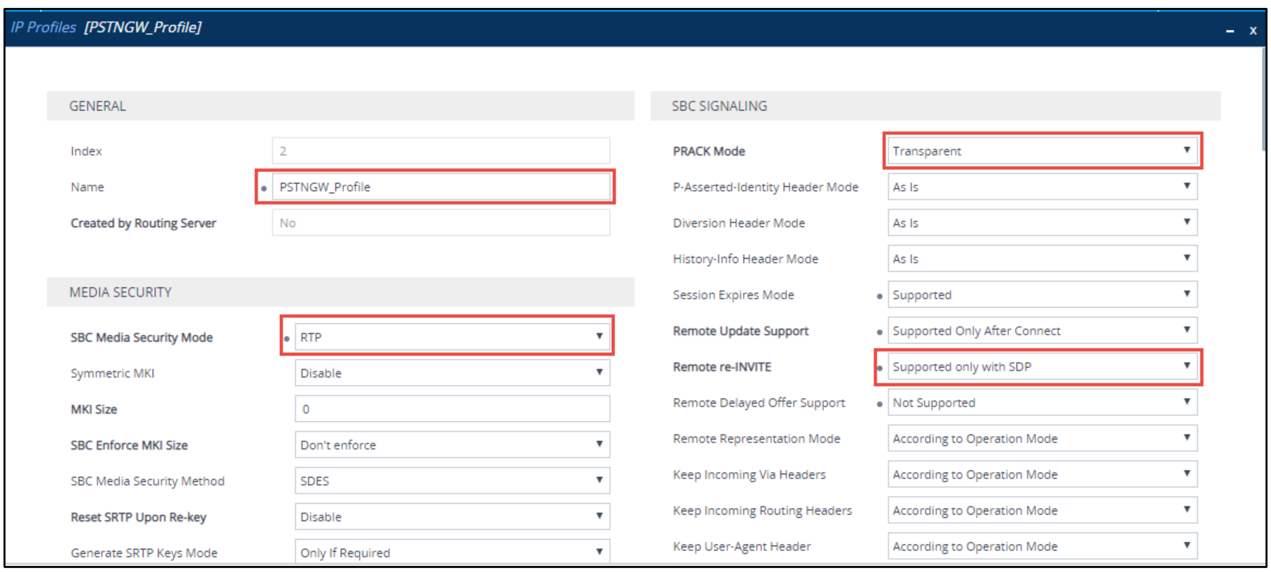


*Figure 85 – IP Profile – PSTN Gateway – Contd.*

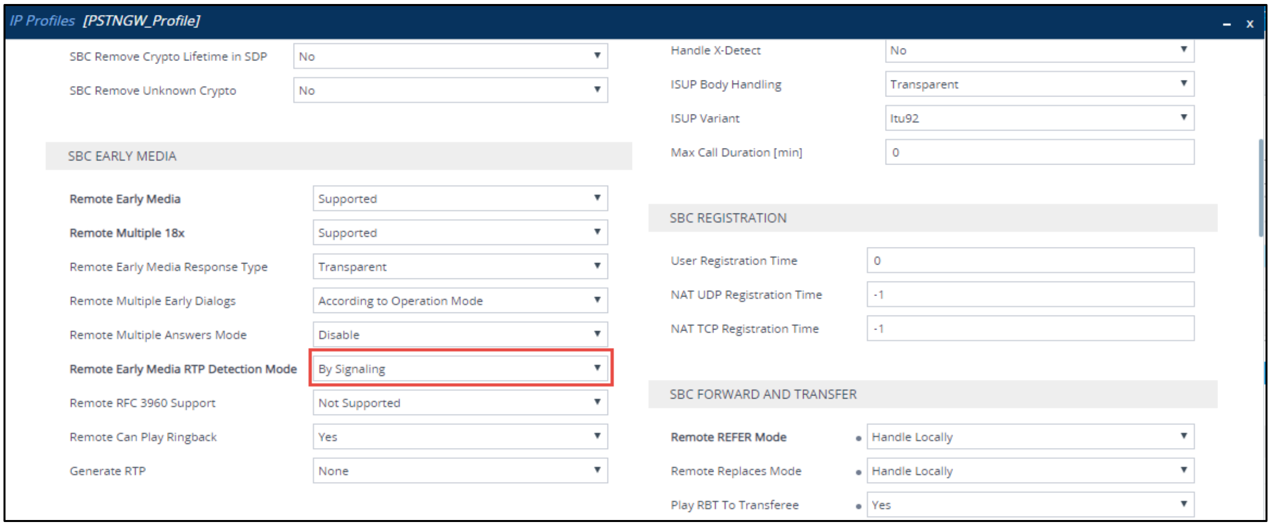


*Figure 86 – IP Profile – PSTN Gateway – Contd.*

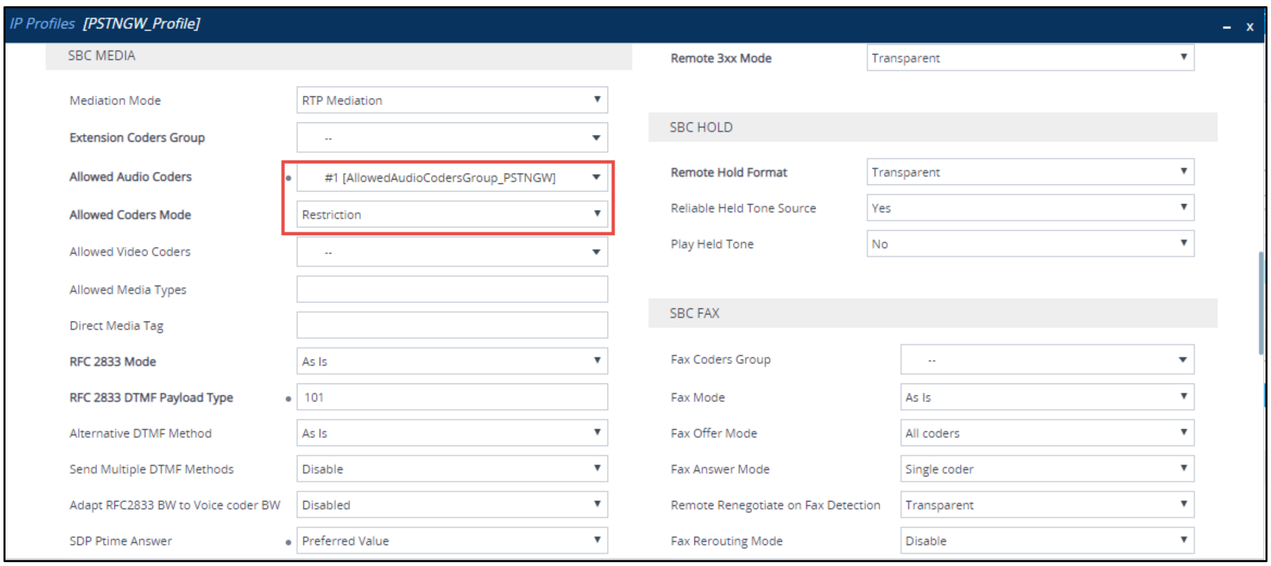


*Figure 87 – IP Profile – PSTN Gateway – Contd.*

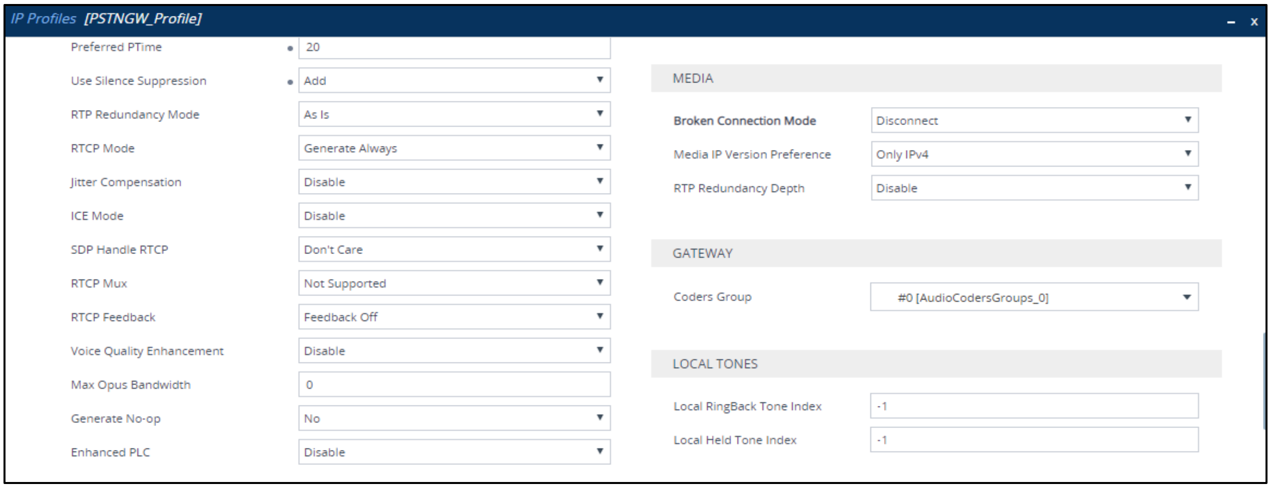


*Figure 88 – IP Profile – PSTN Gateway – Contd.*

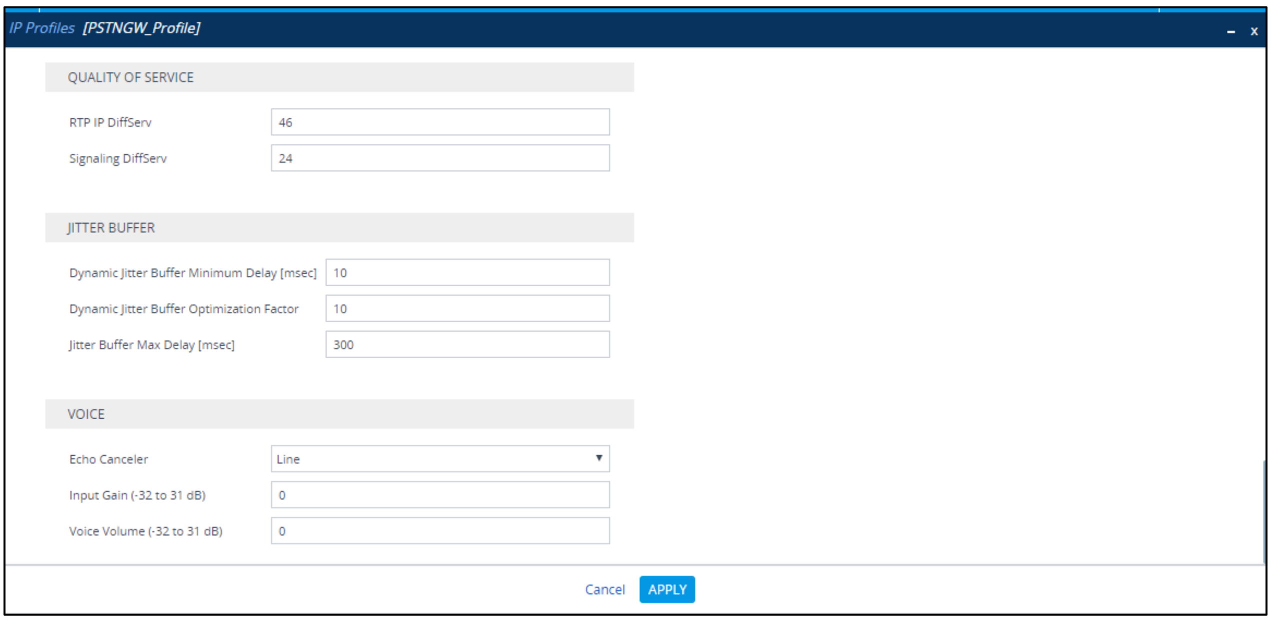


*Figure 89 – IP Profile – PSTN Gateway*

Configure the IP Profile for the Cisco UCM as shown below.

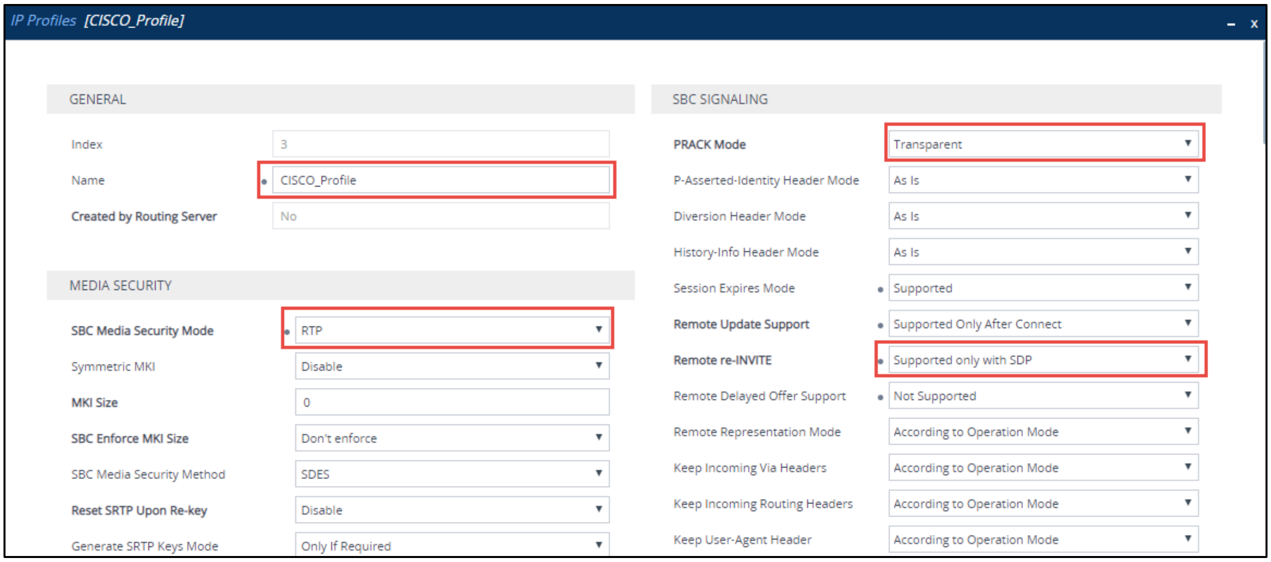

Figure 90 – IP Profile – Cisco – Contd.

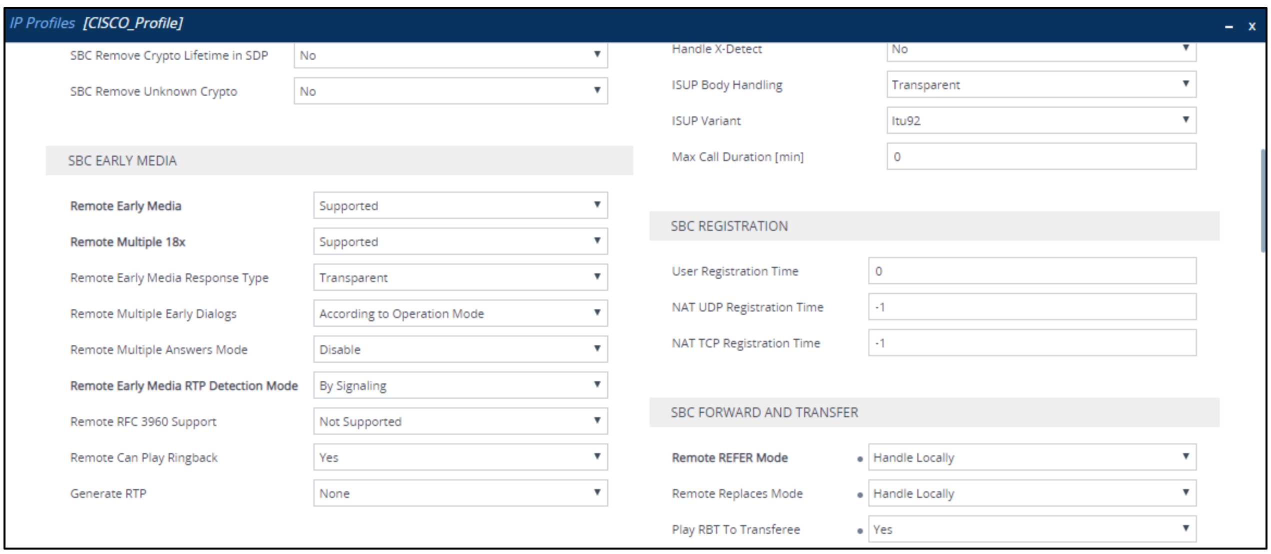

Figure 91 – IP Profile – Cisco – Contd.

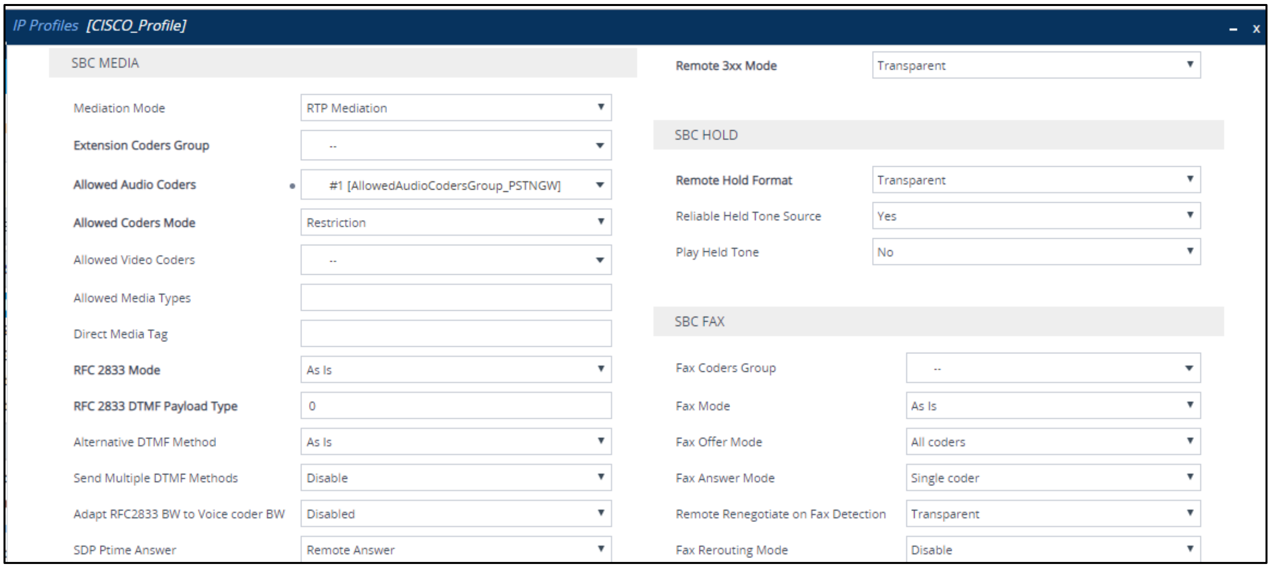


*Figure 92 – IP Profile – Cisco – Contd.*

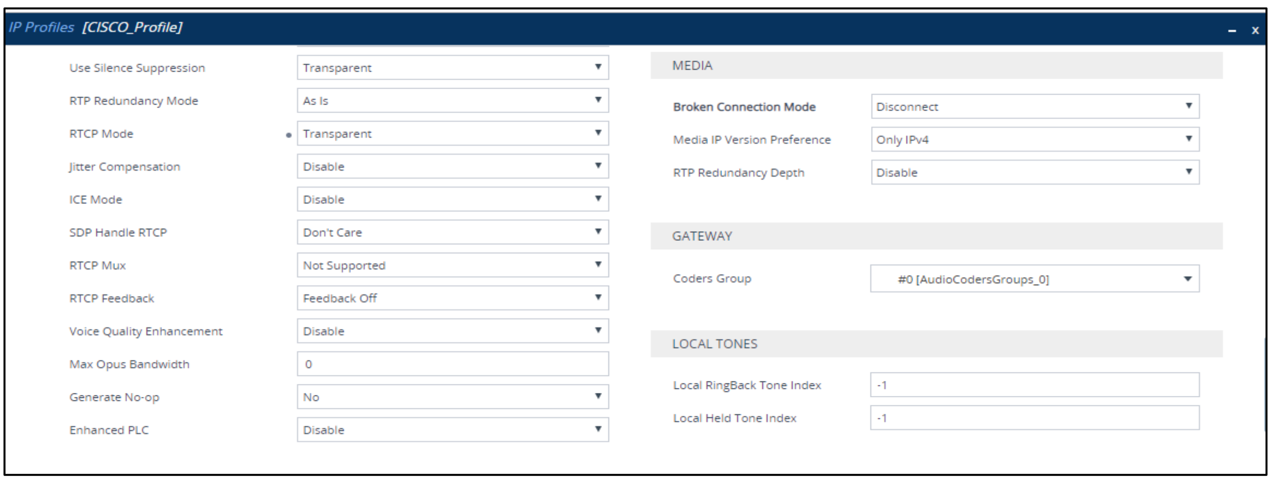


*Figure 93 – IP Profile – Cisco – Contd.*

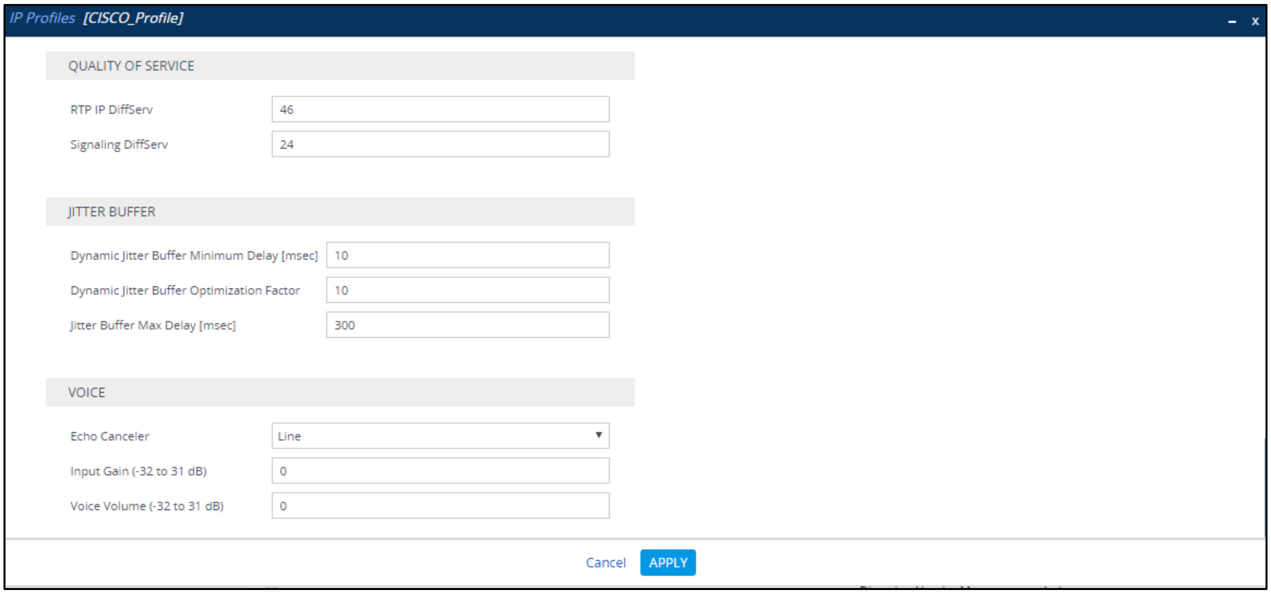


*Figure 94 – IP Profile – Cisco*

### 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

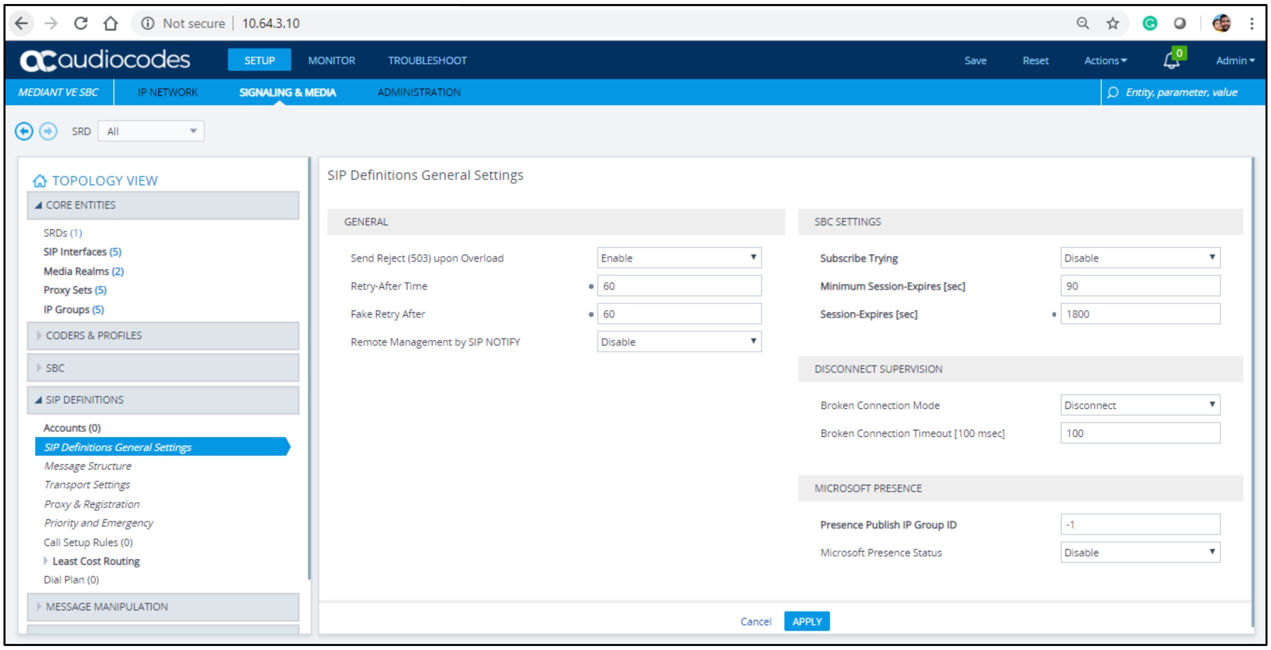


*Figure 95 – SIP Definition*

## 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.

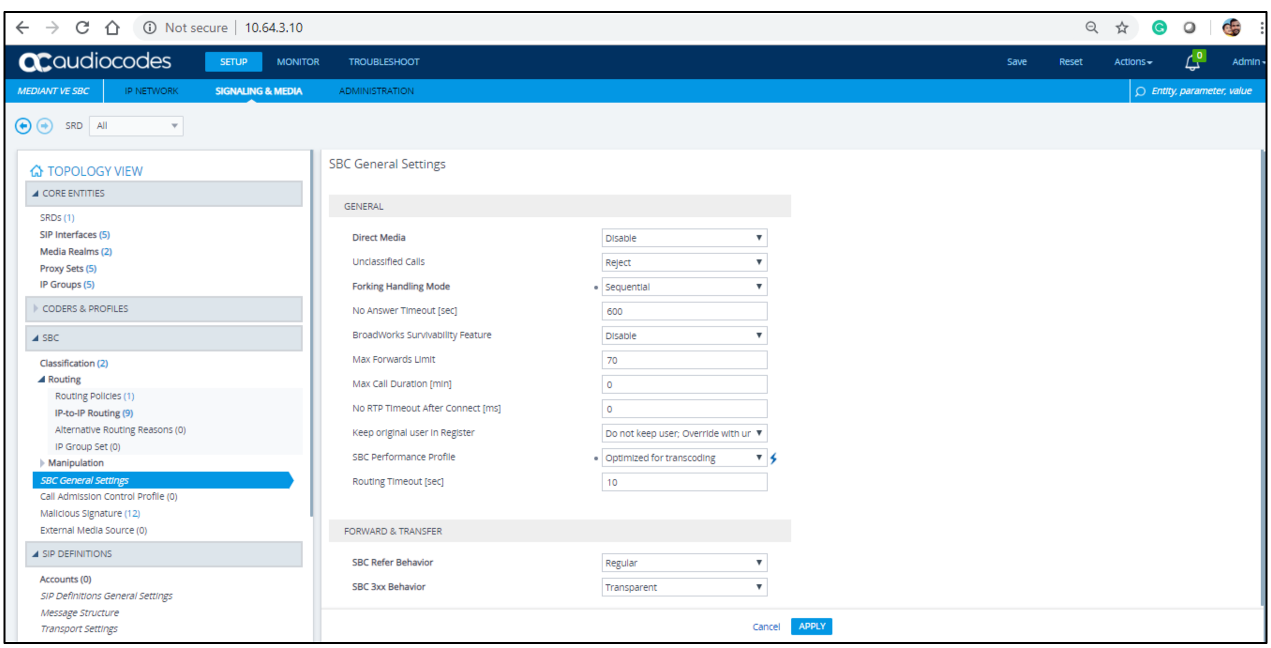


*Figure 96 – SBC General Setting – Contd.*

### 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 Cisco
- Calls from Cisco 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

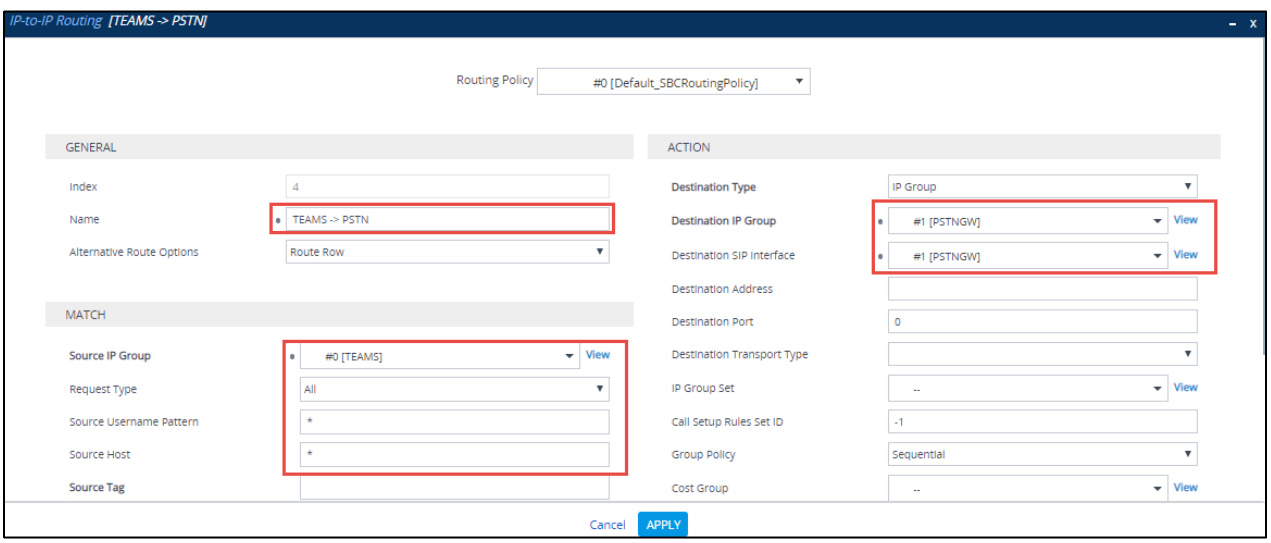

*Figure 97 – Teams to PSTN – Contd.*

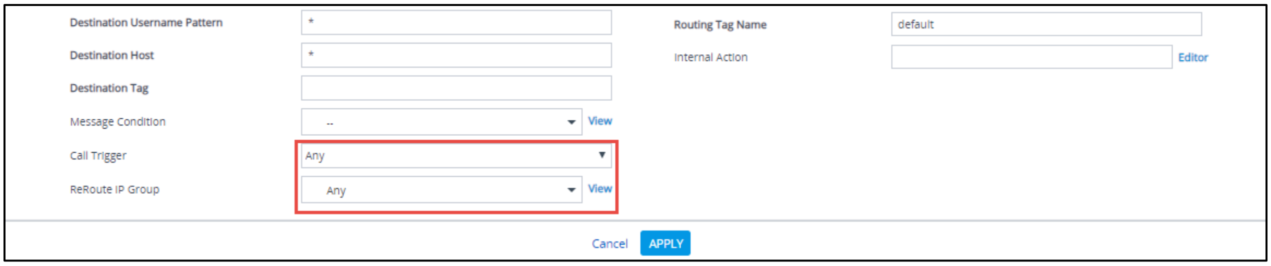


*Figure 98 – Teams to PSTN*

## Calls from PSTN Gateway to Teams

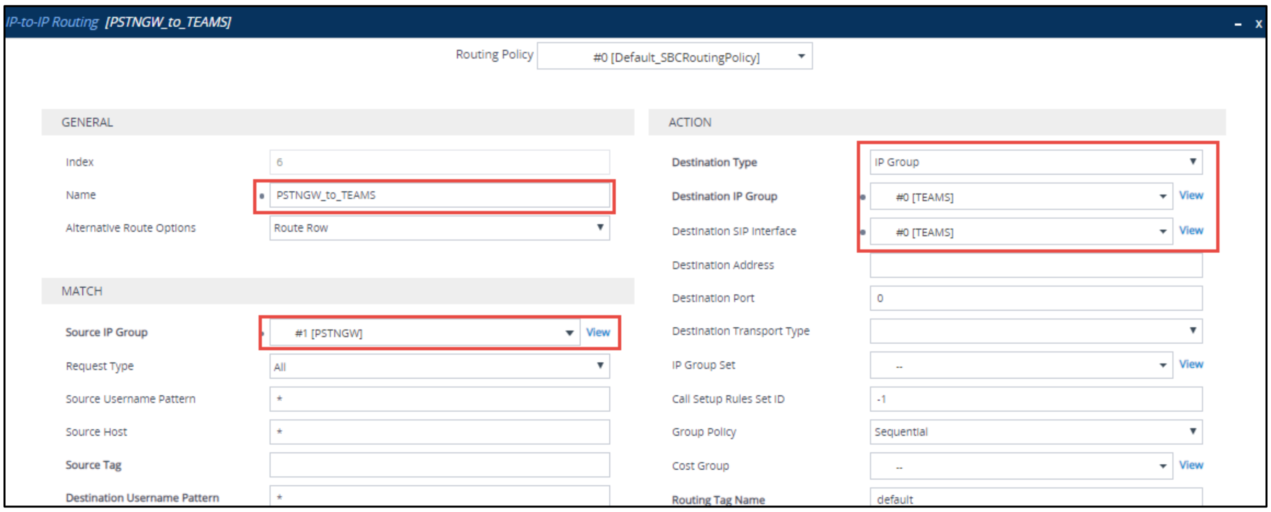


*Figure 99 – PSTN to Teams – Contd.*

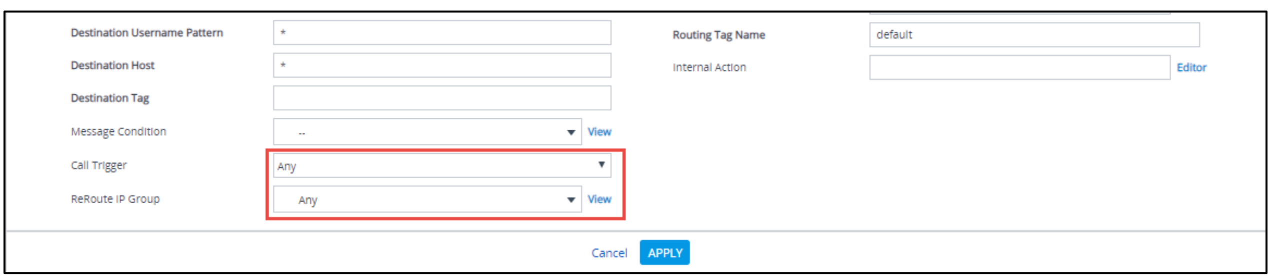


*Figure 100 – PSTN to Teams*

## Calls from Teams to Cisco

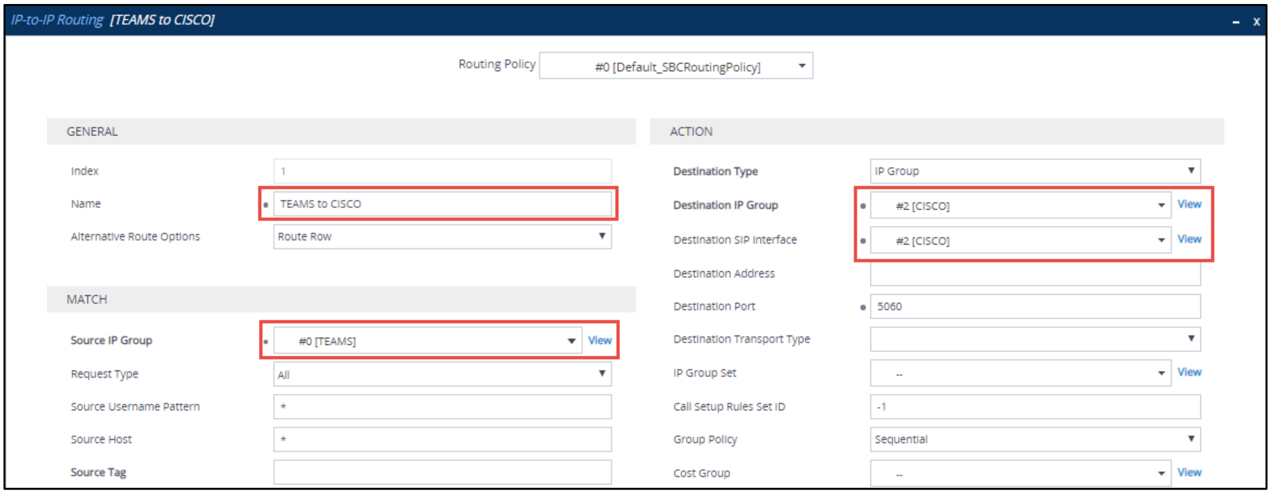


Figure 101 –Teams to Cisco – Contd.

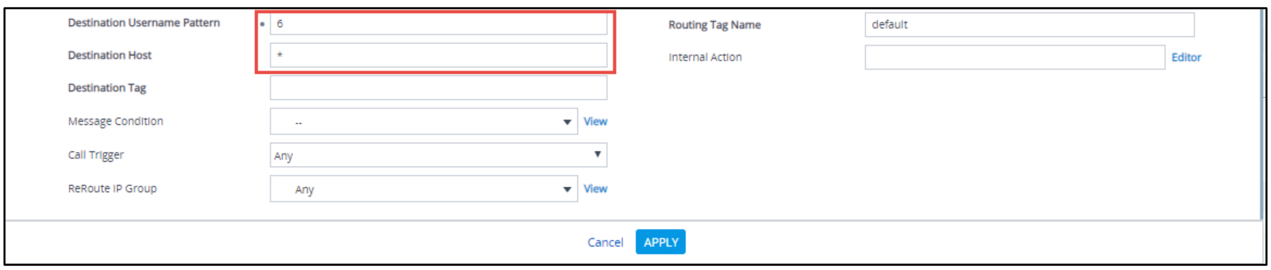


Figure 102 –Teams to Cisco

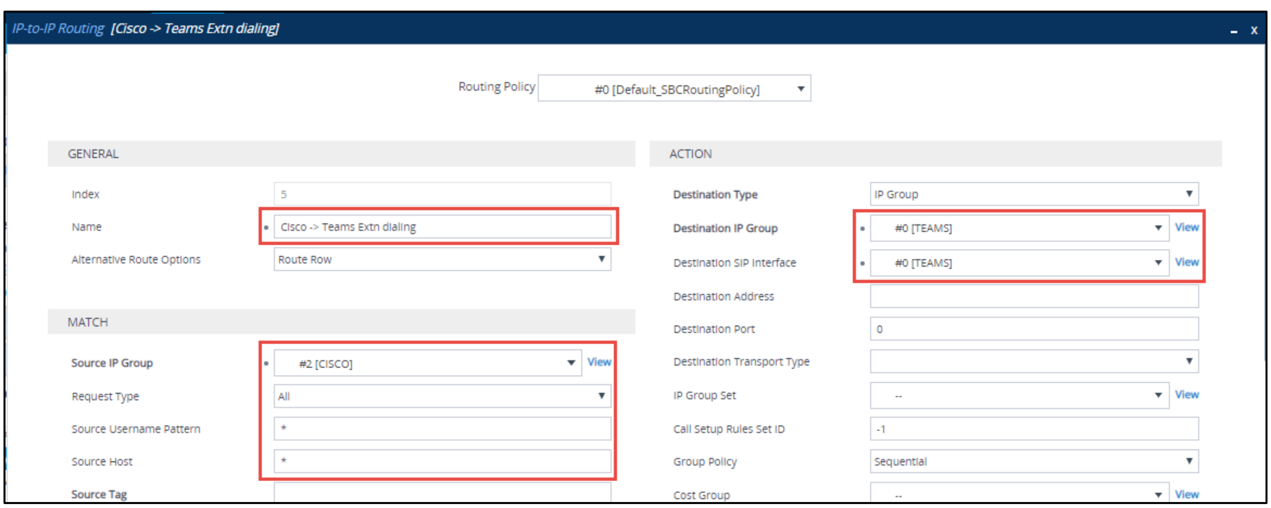


Figure 103 –Cisco to Teams – Contd.

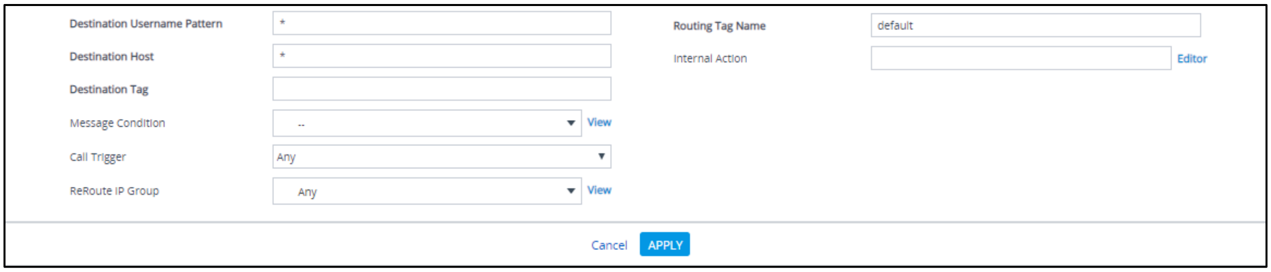


*Figure 104 –Cisco to Teams*

### 4.5.16 IP Group

IP Group – Teams

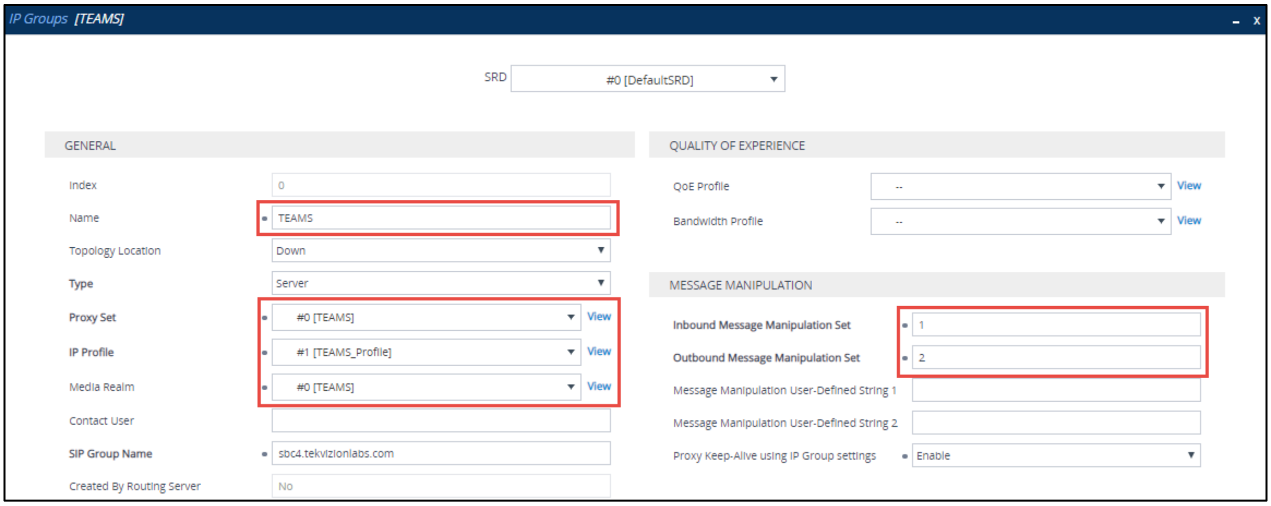


*Figure 105 – IP Groups Teams – Contd.*

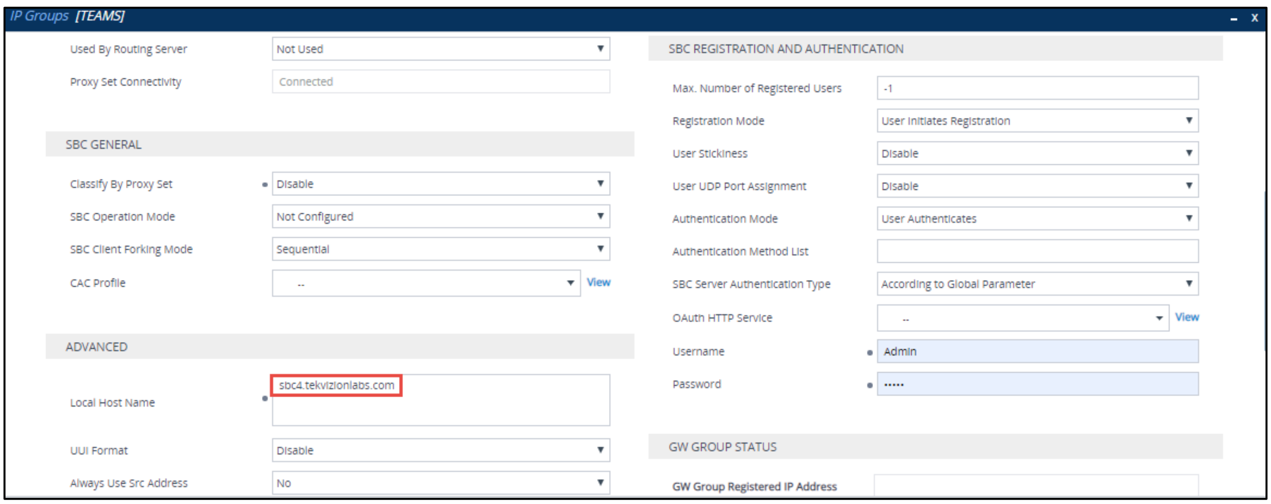


*Figure 106 – IP Groups Teams – Contd.*

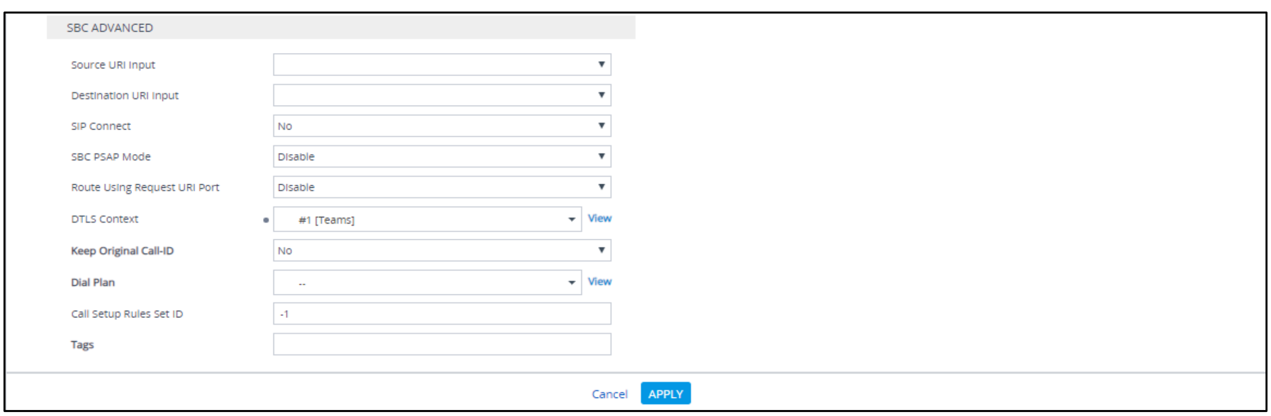


*Figure 107 – IP Groups Teams*

## IP Group – PSTN Gateway

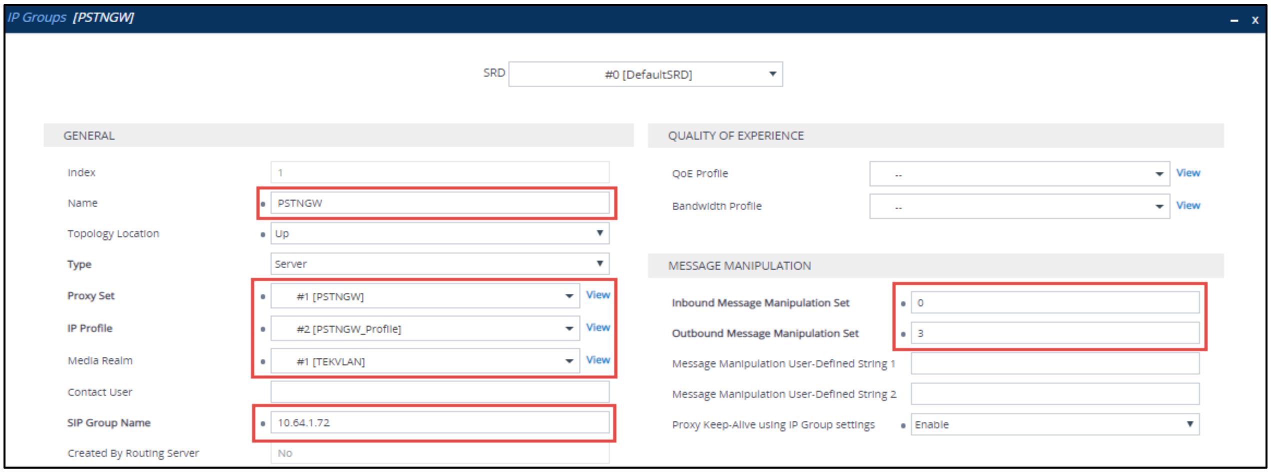

Figure 108 – IP Groups PSTN – Contd.

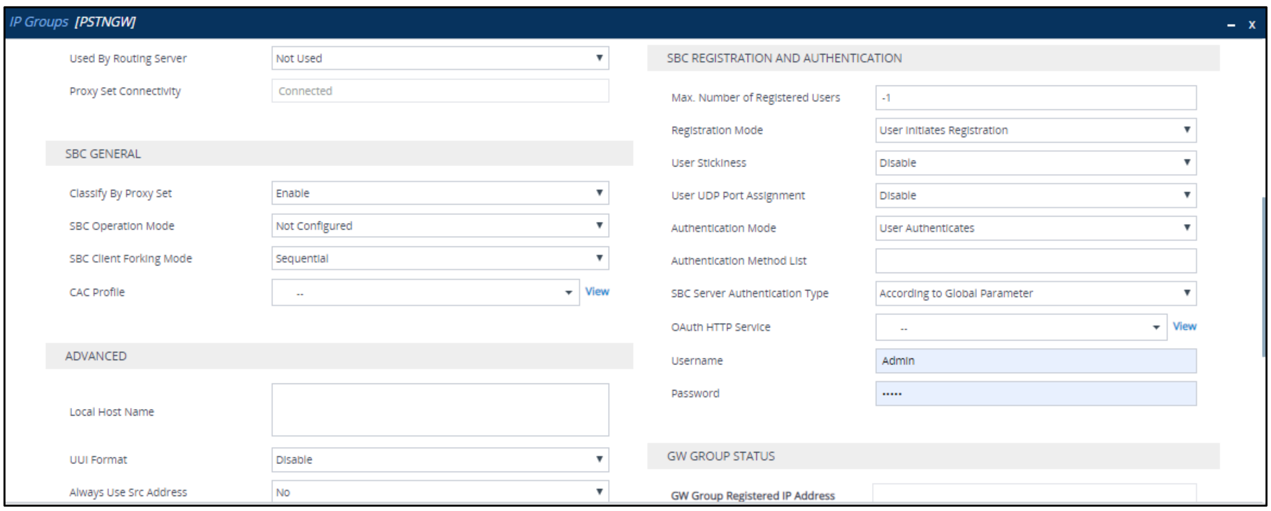

Figure 109 – IP Groups PSTN – Contd.

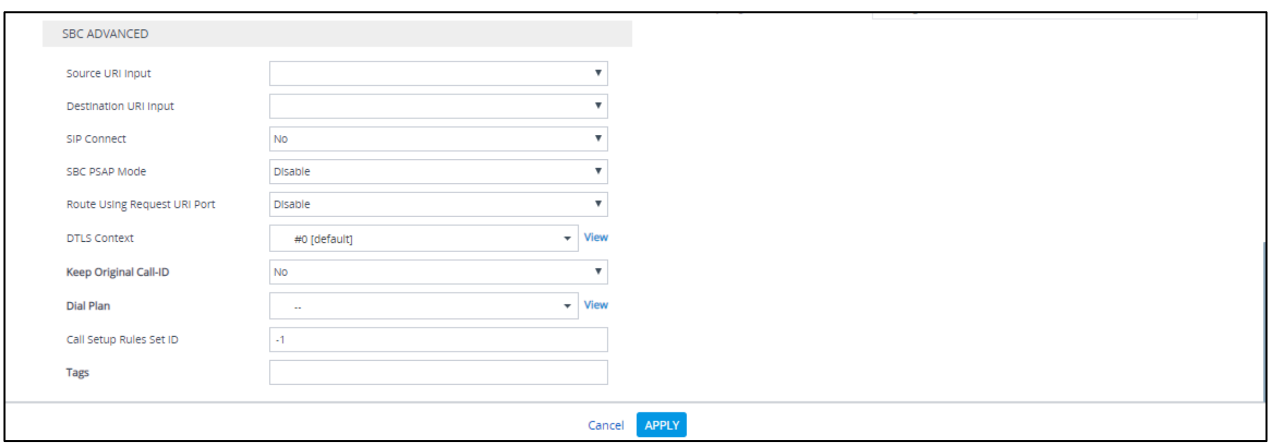

Figure 110 – IP Groups PSTN

## IP Group – Cisco

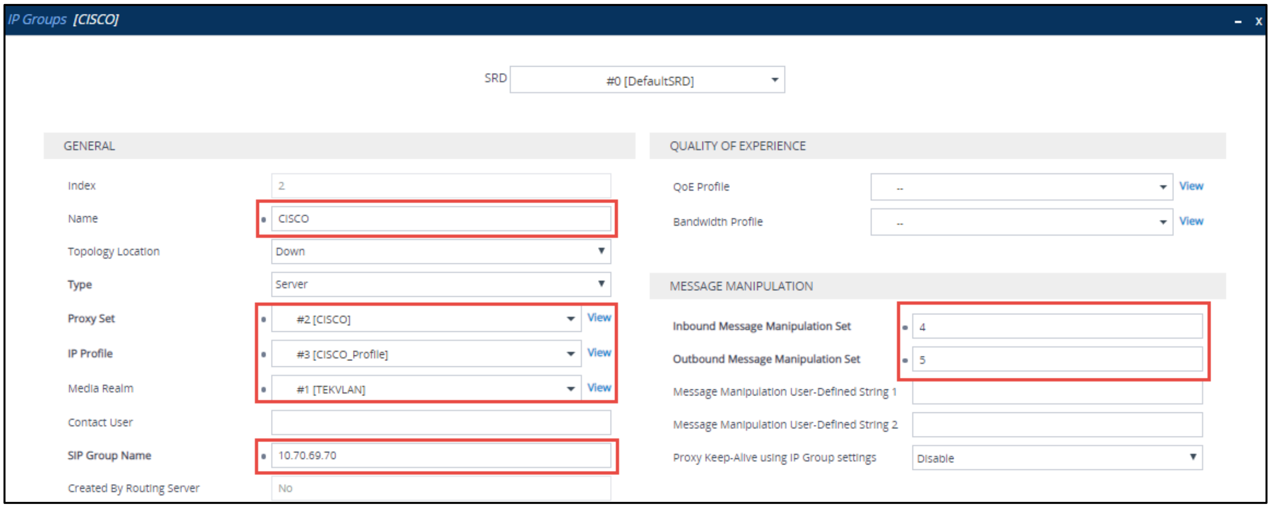


*Figure 111 – IP Groups Cisco – Contd.*

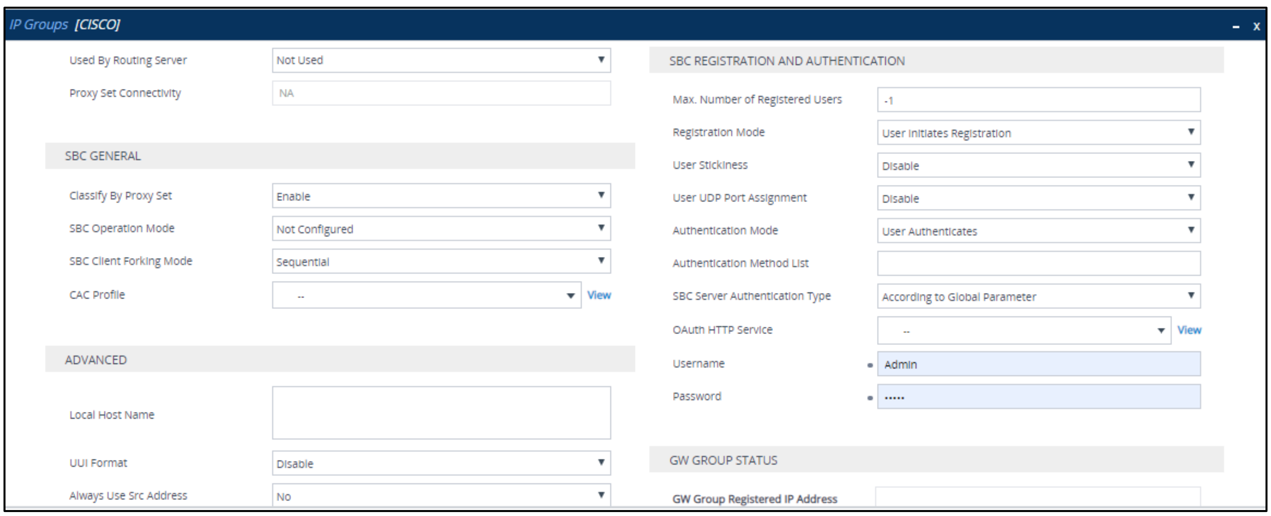


*Figure 112 – IP Groups Cisco – Contd.*

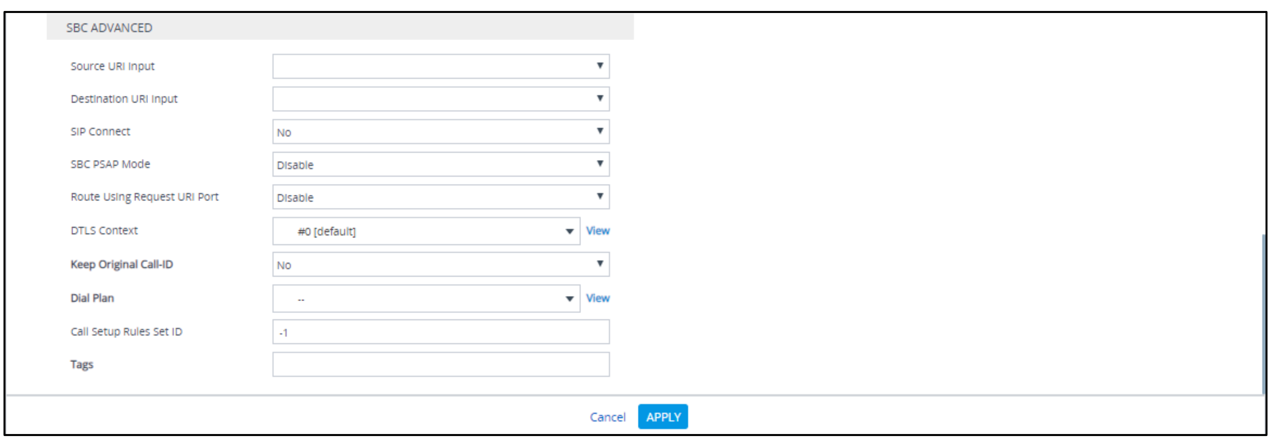


*Figure 113 – IP Groups Cisco*

### 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:

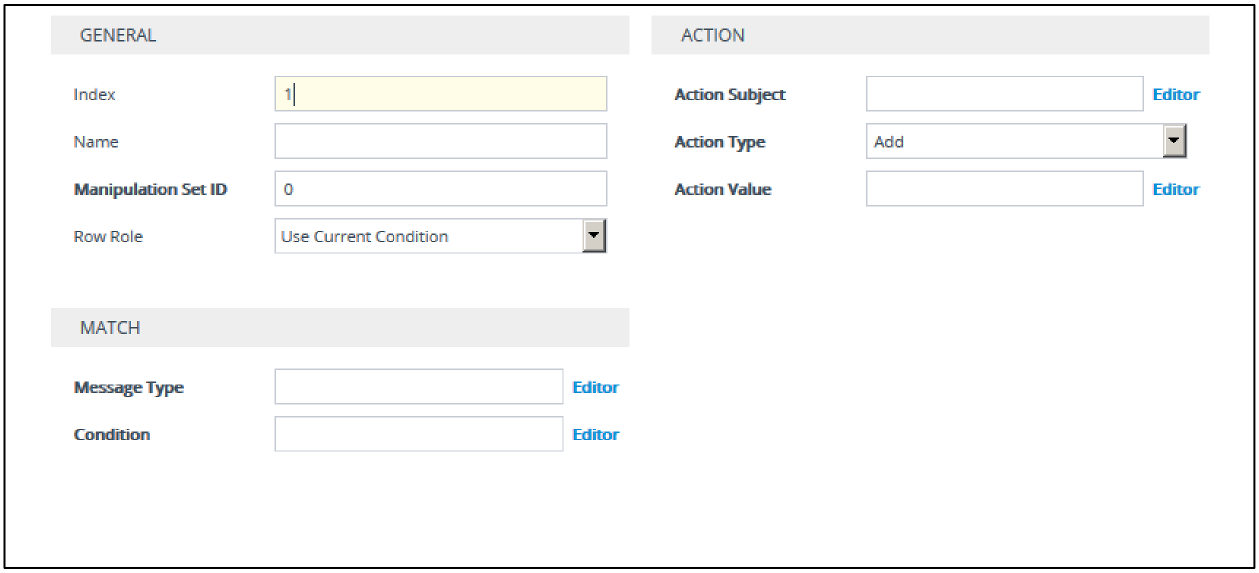

*Figure 114 – SIP Message Manipulation*

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 = 4: Manipulation from Cisco
Manipulation set ID = 5: Manipulation to Cisco

Manipulation from Teams

- To Remove “Privacy” header: To Remove Privacy Header from Teams

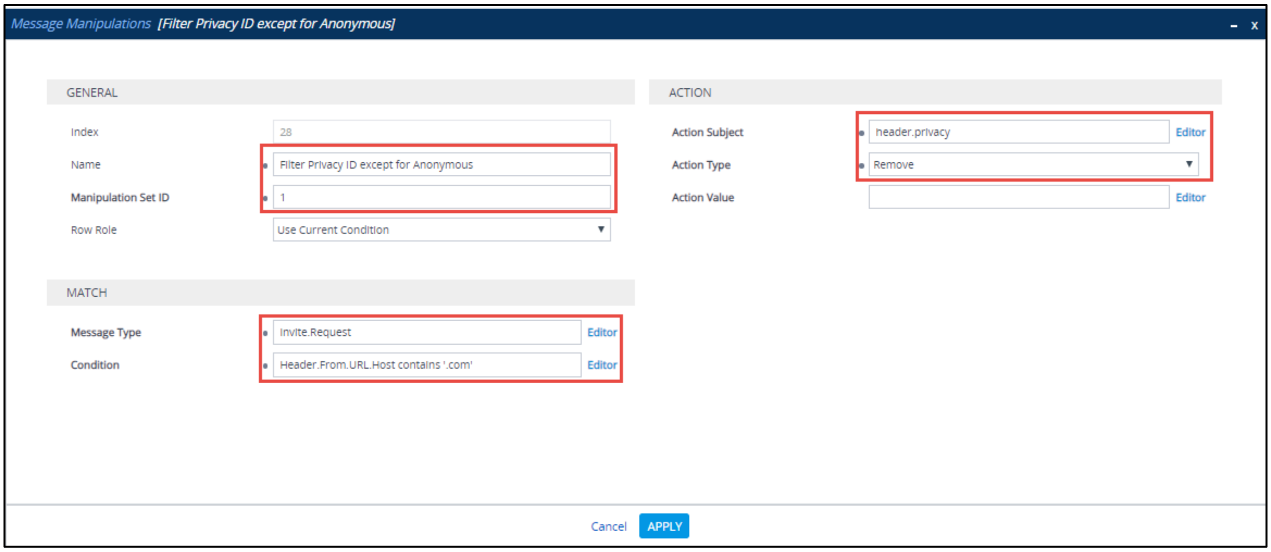


*Figure 115 – SIP Message Manipulation - Privacy*

Manipulation to Teams

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

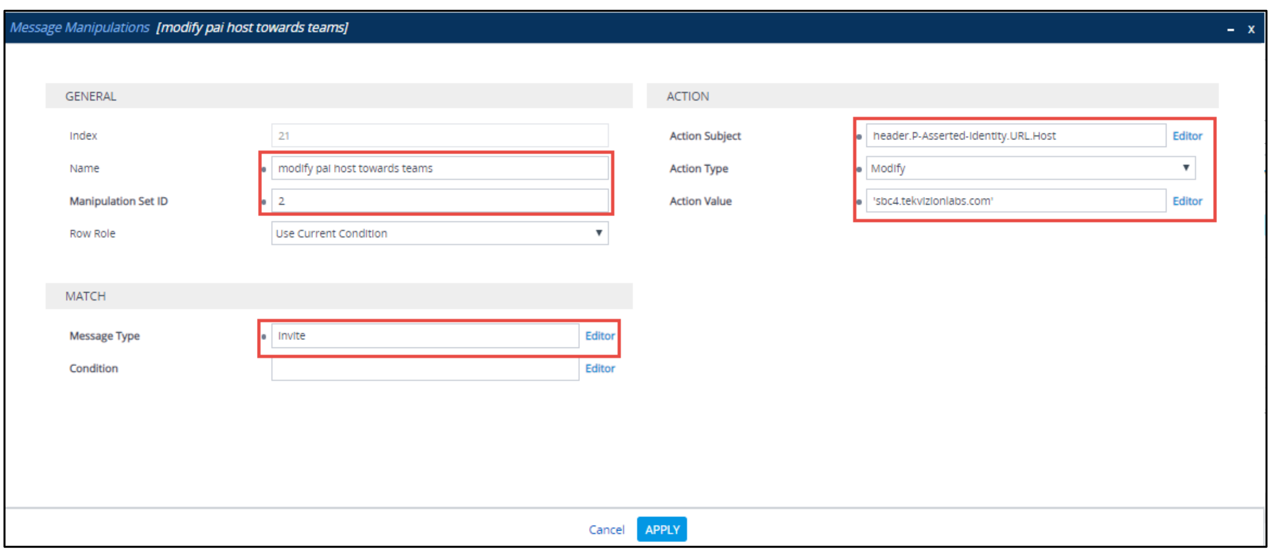


*Figure 116 – SIP Message Manipulation - PAI*

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

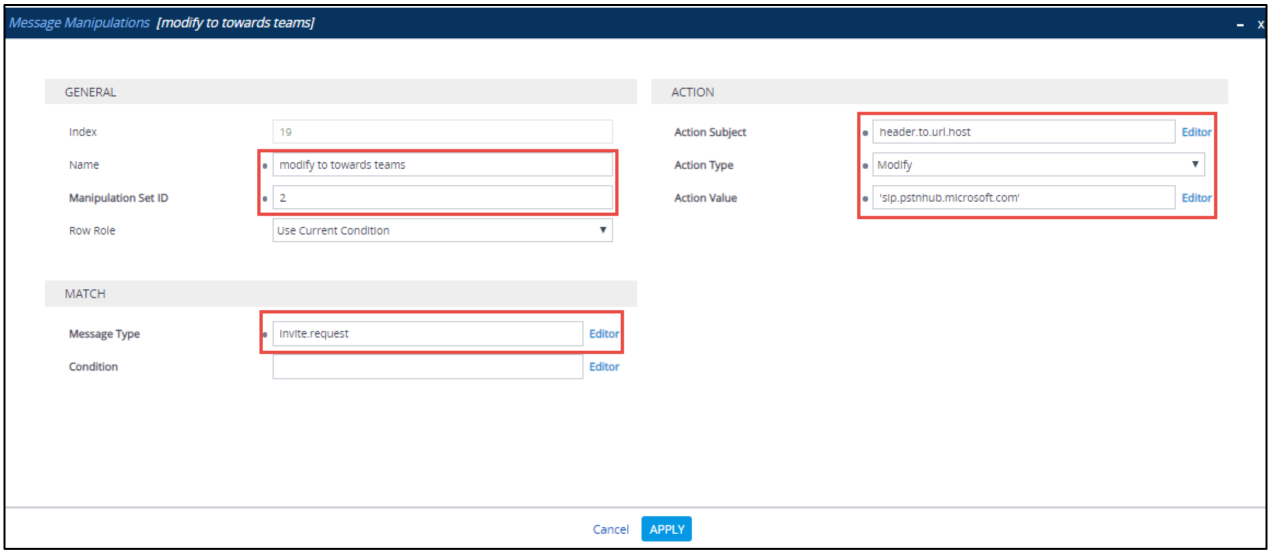


*Figure 117 – SIP Message Manipulation - To*

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

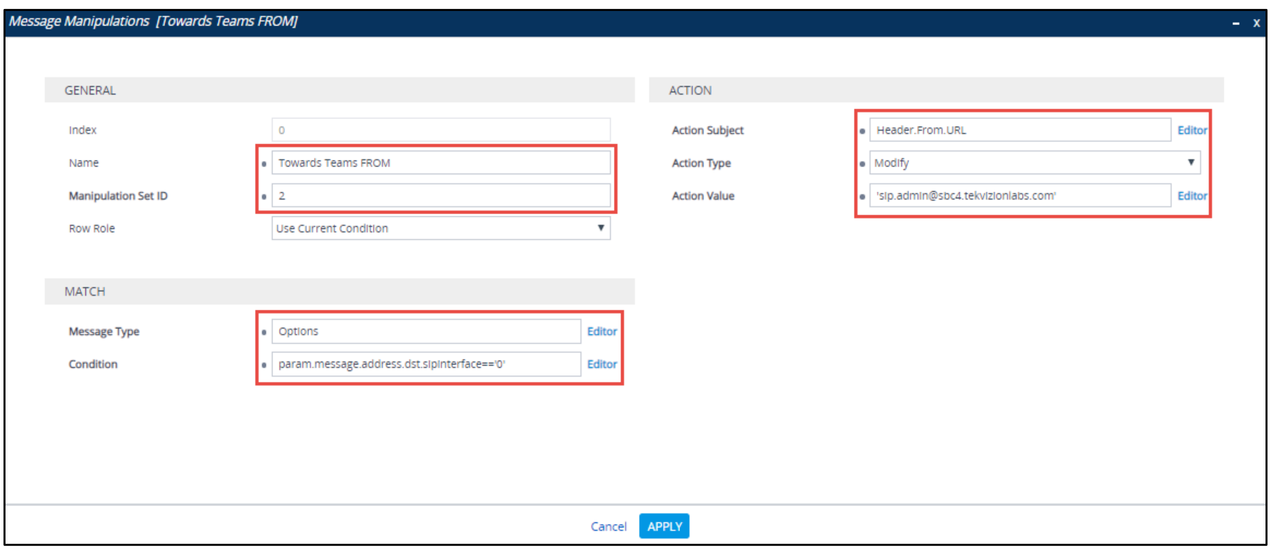


*Figure 118 – SIP Message Manipulation - From*

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

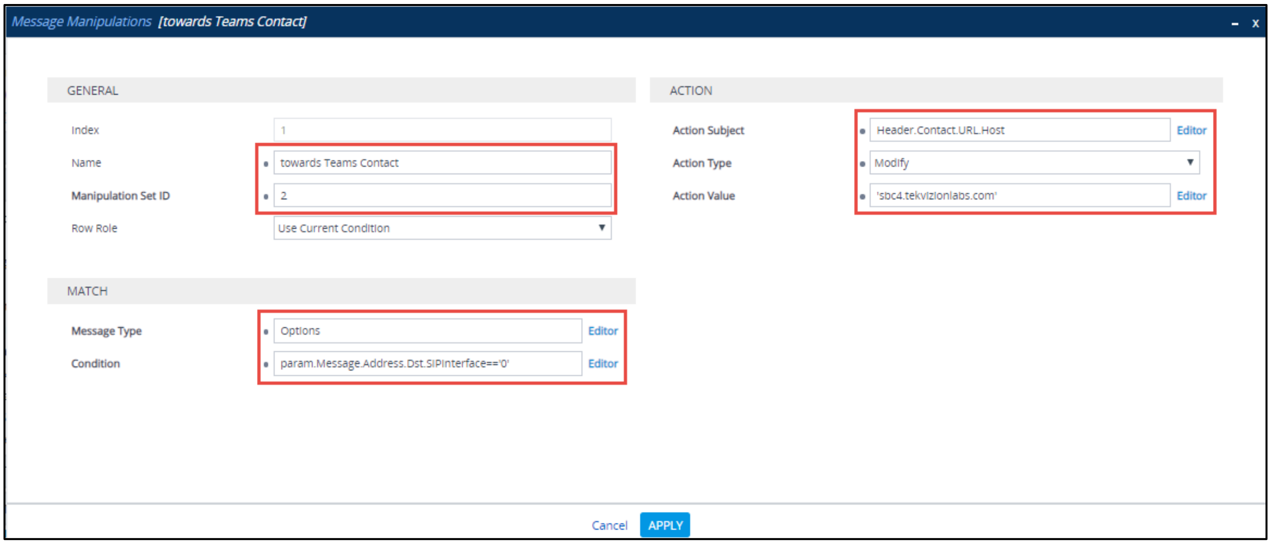


*Figure 119 – SIP Message Manipulation - Contact*

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

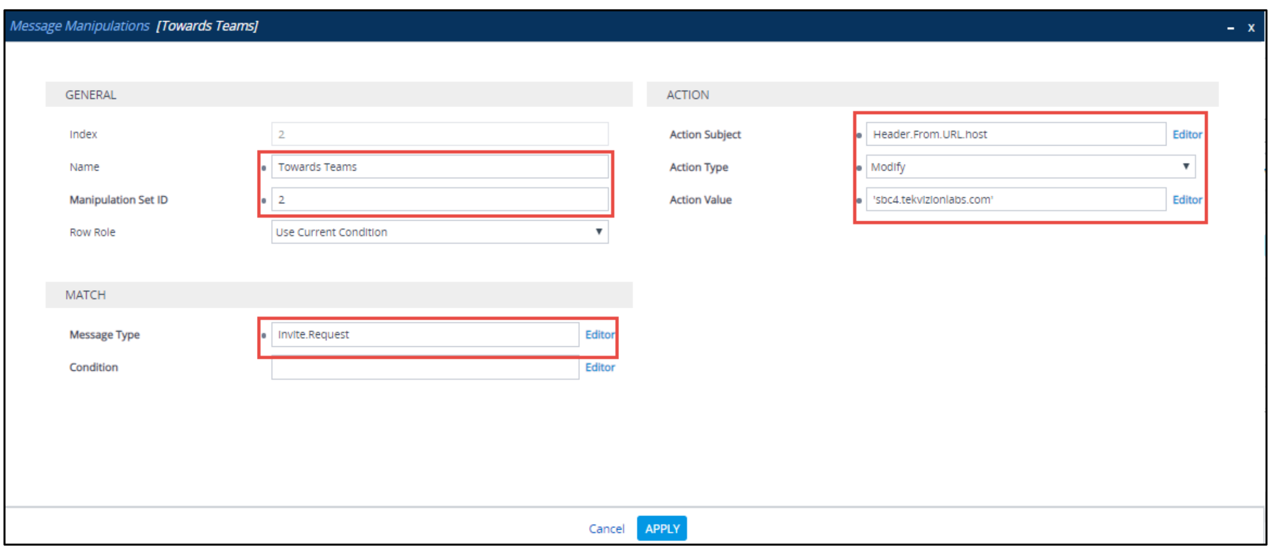


*Figure 120 – SIP Message Manipulation - From*

Manipulation to PSTN

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

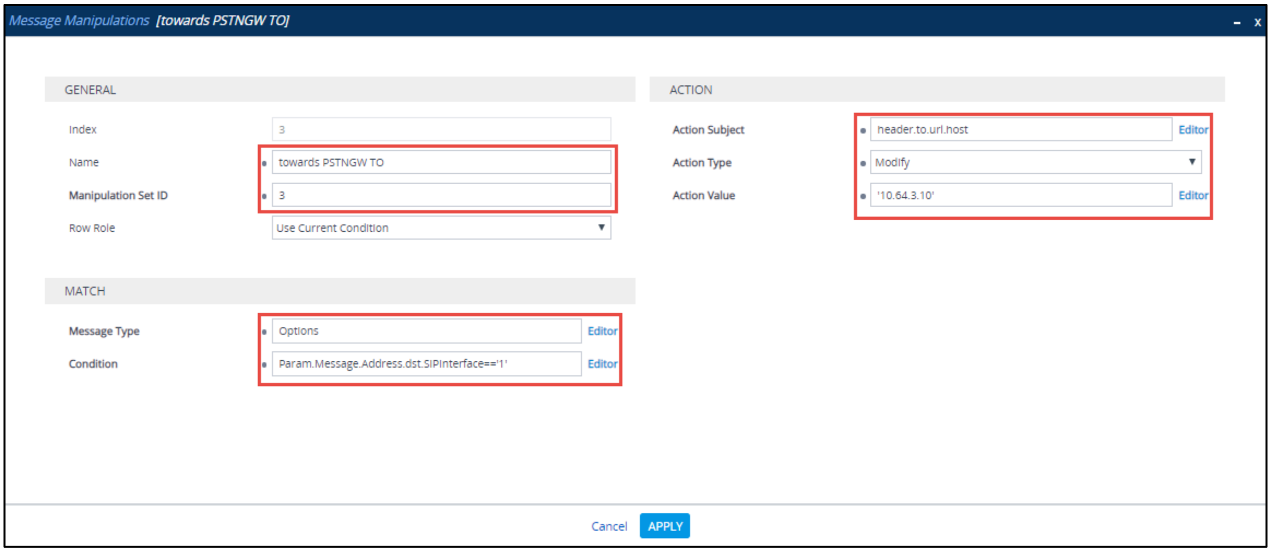


*Figure 121 – SIP Message Manipulation – To*

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

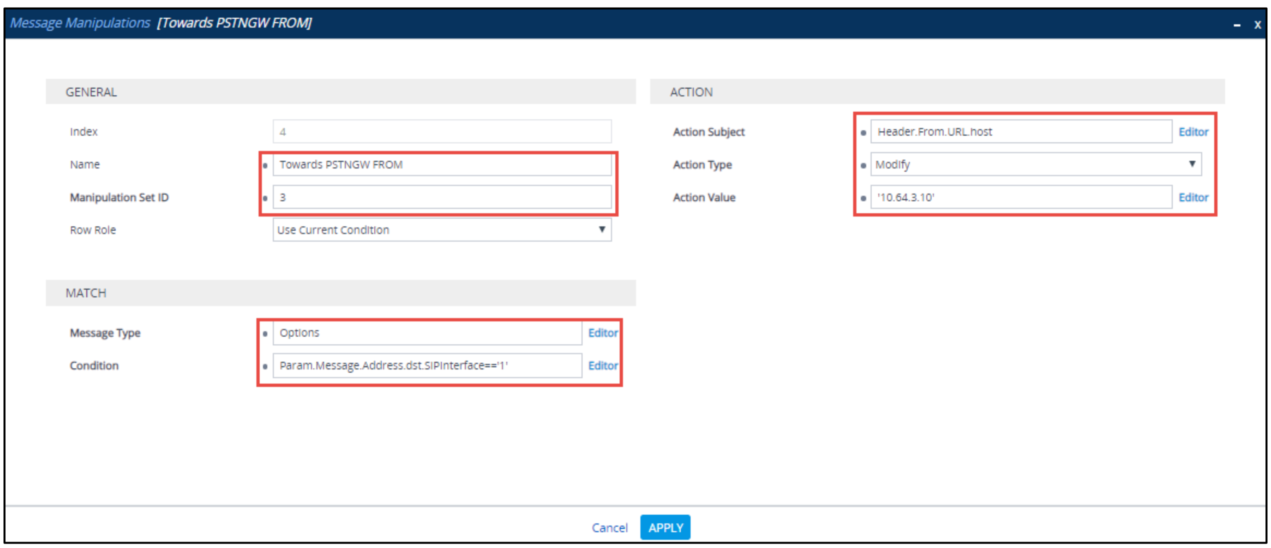


*Figure 122 – SIP Message Manipulation – From*

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

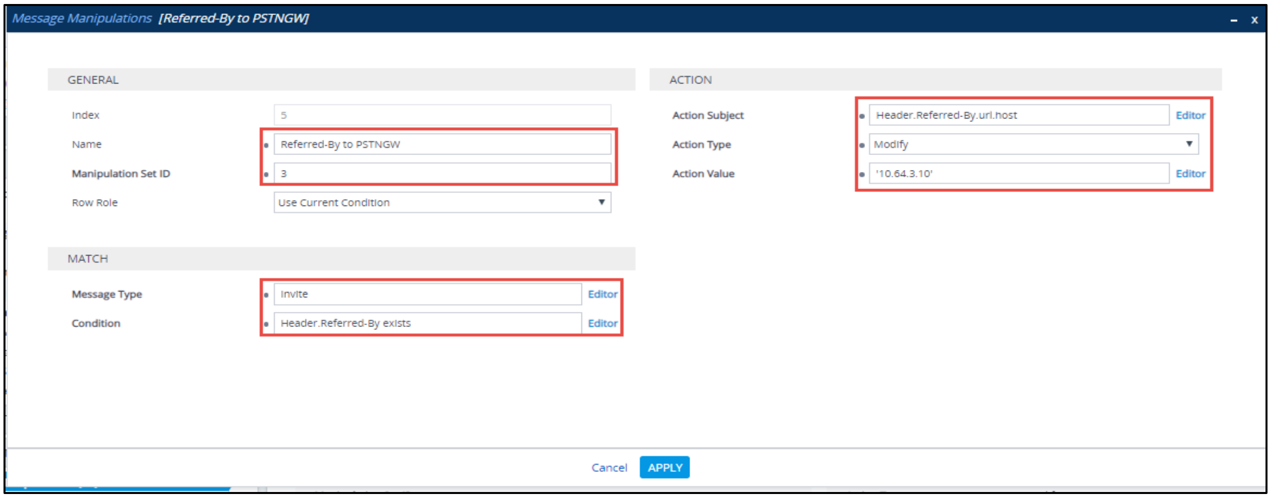


*Figure 123 – SIP Message Manipulation – Referred - By*

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

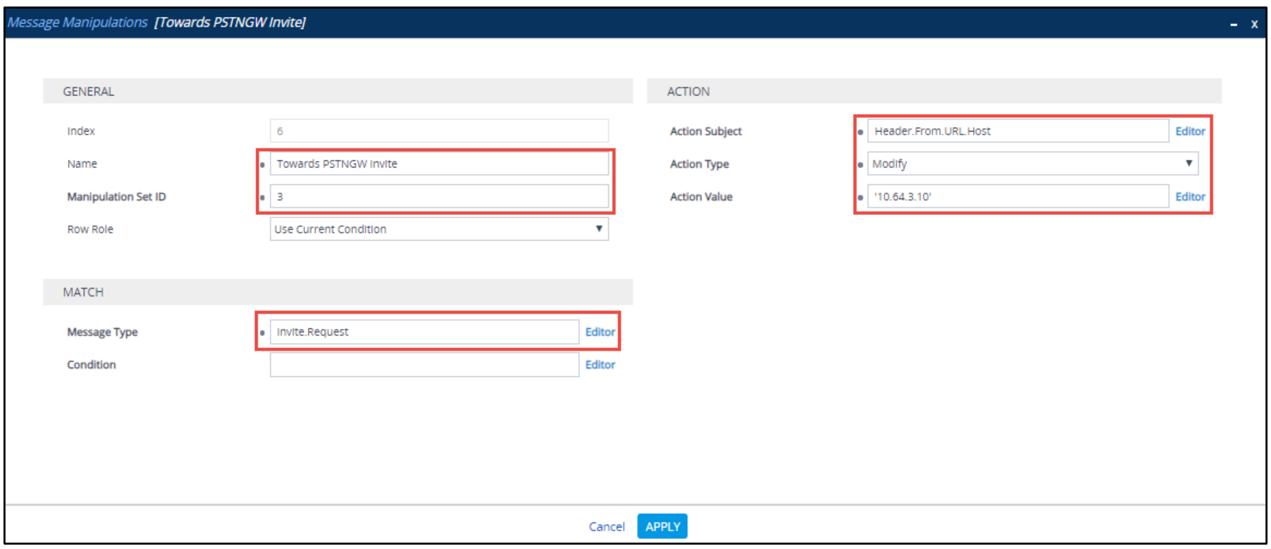


*Figure 124 – SIP Message Manipulation – From*

Manipulation to Cisco

- To Remove "Privacy" header: To Filter Privacy ID except for Anonymous in Host

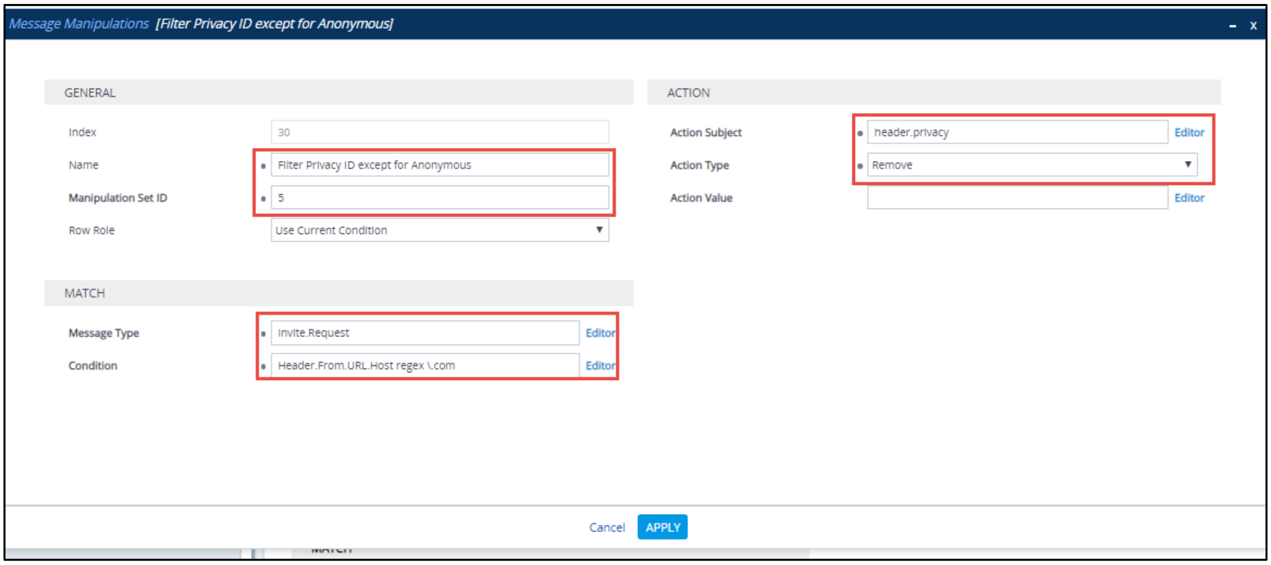


Figure 125 – SIP Message Manipulation – Privacy

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

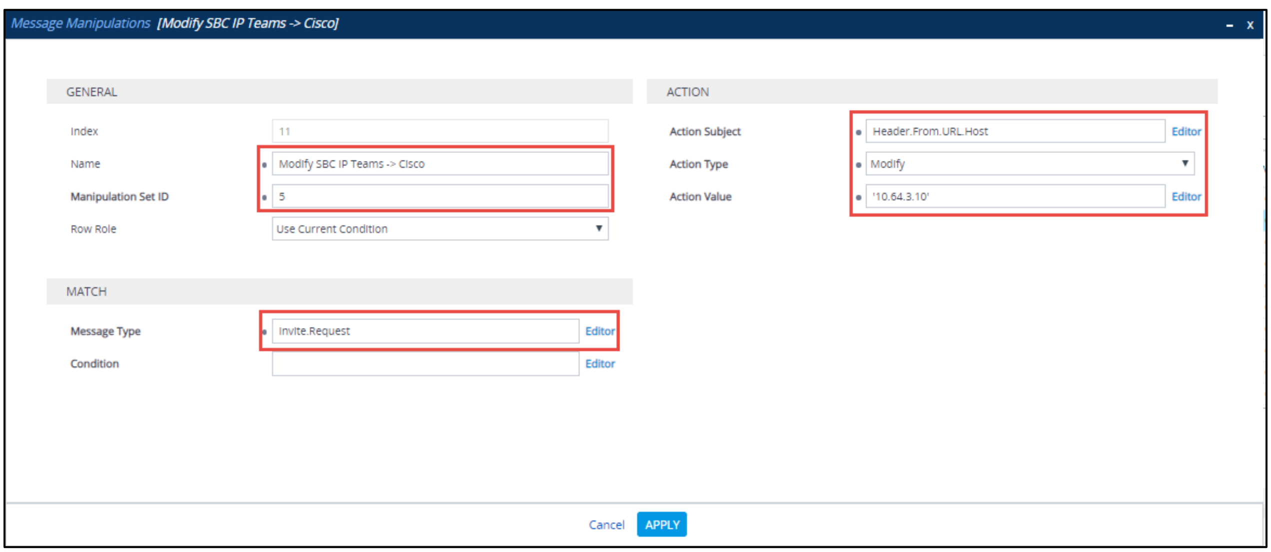


Figure 126 – SIP Message Manipulation – From

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

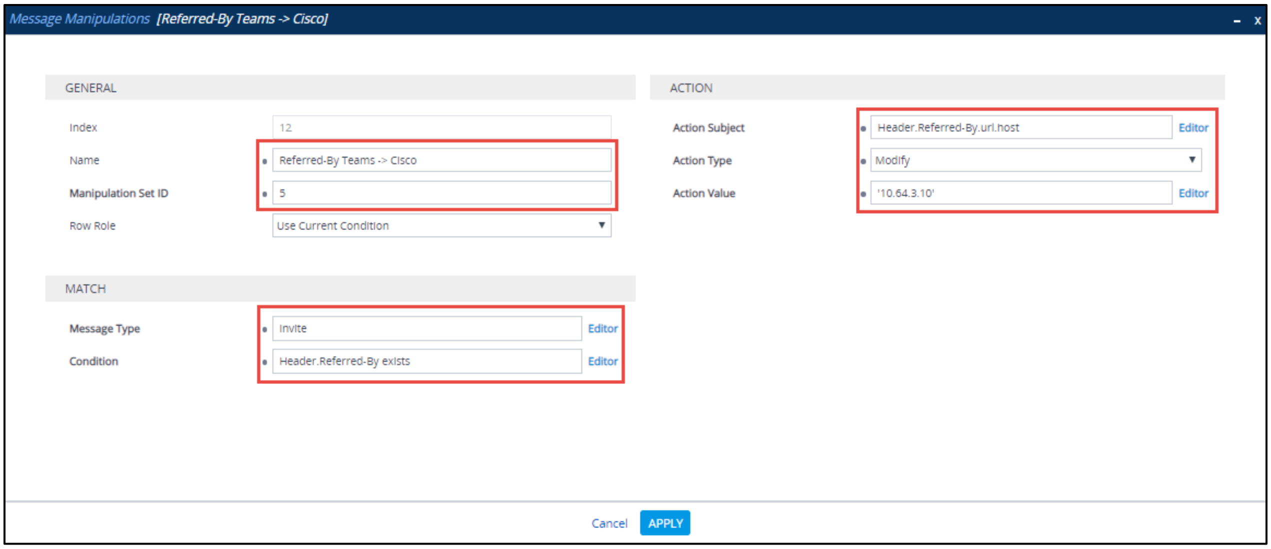


*Figure 127 – SIP Message Manipulation – Referred By*

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

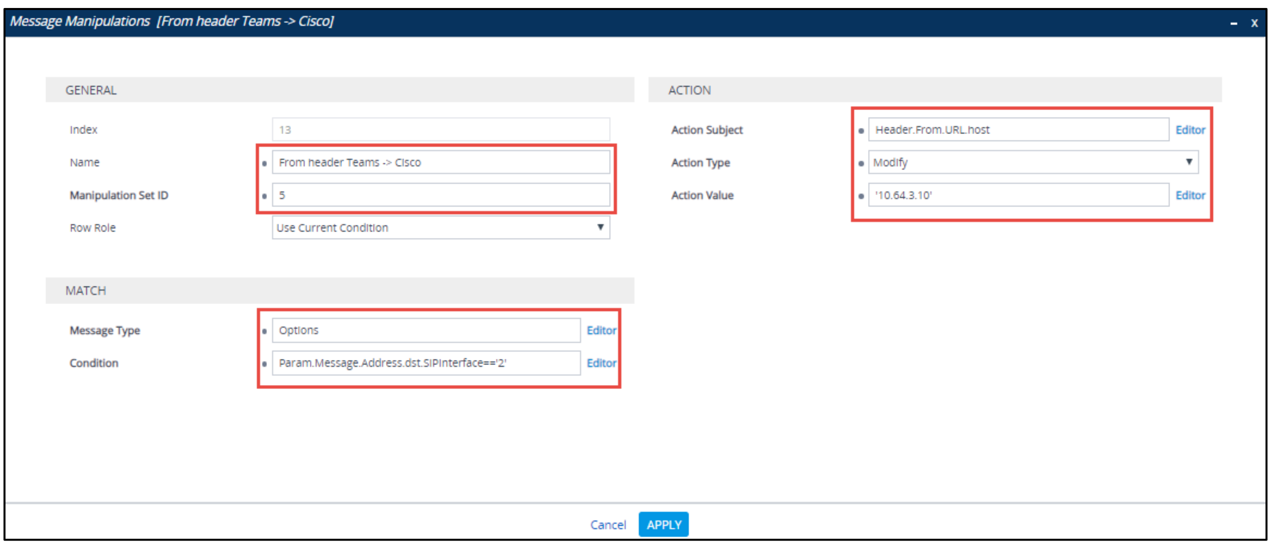


*Figure 128 – SIP Message Manipulation – From*

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

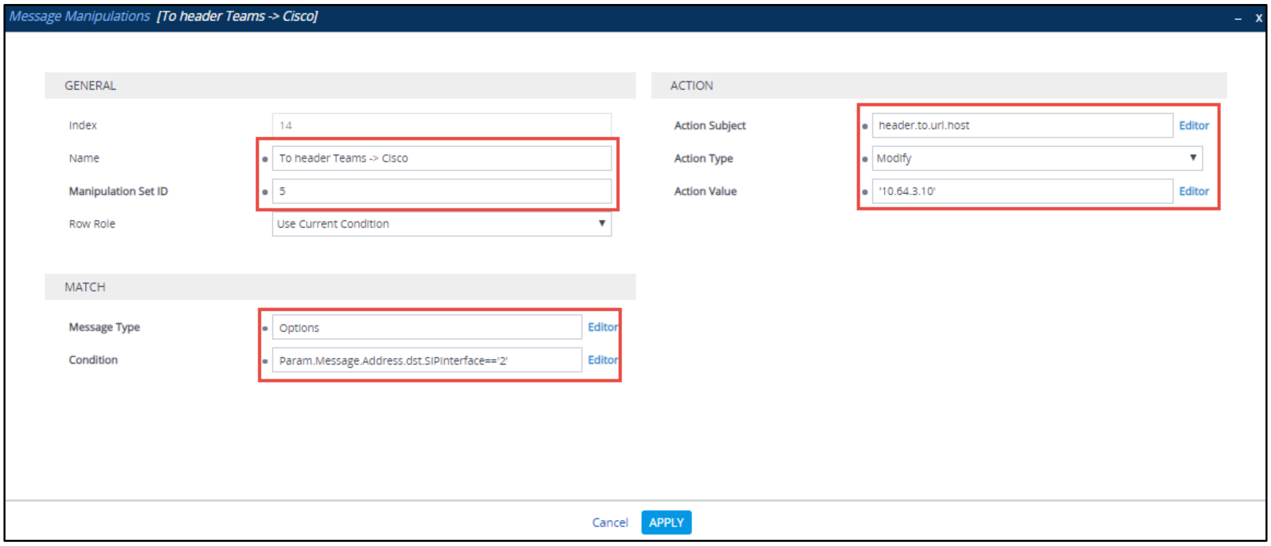


*Figure 129 – SIP Message Manipulation – to*

## 4.6 Cisco UBE Configuration

```
Crestron_Teams#sh run
Building configuration...
Current configuration : 6699 bytes
!

!
version 15.7
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname Crestron_Teams
!
boot-start-marker
boot system tftp c2900-universalk9-mz.SPA.157-3.M1.bin 255.255.255.255
boot-end-marker

!
enable secret 4 sKPgCY/XPea3wk8xoeSWo7UGFaNVwzXDEyXWhuDjeLk
enable password 7 071B244778580354471C
!
!
voice service voip
 no ip address trusted authenticate
 address-hiding1
 mode border-element license capacity 20[2]
 allow-connections sip to sip[3]
 fax protocol pass-through g711ulaw
 sip
  bind control source-interface GigabitEthernet0/1
  bind media source-interface GigabitEthernet0/1
  session refresh
  asserted-id pa[4]i
  early-offer forced
```

---

[1] Hide signaling and media peer addresses from endpoints other than gateway.
[2] If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the Cisco
[3] This command enables Cisco UBE basic IP-to-IP voice communication feature.
[4] This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).

```
  midcall-signaling passthru[5]
  privacy-policy passthru
  g729 annexb-all
!
voice class codec 3
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
 codec preference 3 g729r8
!
!
!
!
username cisco privilege 15 password 7 083549453F481F464205
!
redundancy inter-device
 scheme standby SB
!
!
redundancy
!
!
!
!
!
track 1 interface GigabitEthernet0/0 line-protocol
!
track 2 interface GigabitEthernet0/1 line-protocol
```

[5] This command must be enabled at a global level to maintain integrity of SIP signaling between AudioCodes network and Cisco Unified Communications Manager (Cisco UCM) across Cisco UBE.

```
!
!
!
!
!
!
!
!
!
!
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0[6]
 description LAN to CUCM
 ip address 10.64.4.182 255.255.0.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1[7]
 description WAN to AudioCodes
 ip address 10.70.69.70 255.255.255.0
 duplex auto
```

[6] LAN interface pointing to CUCM
[7] WAN interface pointing to AudioCodes

```
 speed auto
!
interface GigabitEthernet0/2
 no ip address
 shutdown
 duplex auto
 speed auto
!
no ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 10.79.69.1
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.71.9.0 255.255.255.0 10.64.1.1
ip route 10.80.18.0 255.255.255.0 10.64.1.1
ip route 172.16.24.0 255.255.248.0 10.64.1.1
!
!
!
!
control-plane
!
 !
 !
 !
```

```
 !
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 10 voip[8]
 description Ingress CUCM to Audio Codes LAN Interface
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 8....
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 no vad
!
```

[8] Inbound Dial-peer for Cisco UCM facing network

```
dial-peer voice 11 voip[9]
 description Egress CUCM to Audio Codes LAN Interface
 huntstop
 destination-pattern 8....
 session protocol sipv2
 session target ipv4:10.64.3.10:5062
 session transport udp
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 12 voip[10]
 description Ingress  Audio Codes LAN Interface to CUCM
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 6...
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
 no vad
```

[9] Outbound Dial-peer towards AudioCodes

[10] Inbound Dial peer from AudioCodes

```
!
dial-peer voice 13 voip[11]
 description CUBE to CUCM_LAN interface
 huntstop
 destination-pattern 6...
 session protocol sipv2
 session target ipv4:172.16.29.81
 session transport udp
 voice-class codec 3
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0
 voice-class sip bind media source-interface GigabitEthernet0/0
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 14 voip
 description Ingress  CGW Interface to CUBE
 huntstop
 session protocol sipv2
 session transport udp
 incoming called-number 6...
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 dtmf-relay rtp-nte
```

[11] Outbound Dial peer towards Cisco UCM

```
 no vad
!
dial-peer voice 16 voip
 description Egress CUCM to Audio Codes LAN Interface
 huntstop
 destination-pattern 97259800..
 session protocol sipv2
 session target ipv4:10.64.3.10:5062
 session transport udp
 voice-class codec 3
 voice-class sip bind control source-interface GigabitEthernet0/1
 voice-class sip bind media source-interface GigabitEthernet0/1
 no vad
!
!
gatekeeper
 shutdown
!
!
 vstack
!
line con 0
 password 7 111D1C0E2143115D5424
 login
line aux 0
line 2
 no activation-character
```

```
 no exec
 transport preferred none
 transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
 stopbits 1
line vty 0 4
 exec-timeout 0 0
 privilege level 15
 password 7 071B244778580354471C
 login local
 transport input telnet
!
no scheduler allocate
!
end

Crestron_Teams#
```

## 4.7 Cisco UCM Configuration

The configuration screen shots shows general over view of lab configuration for this interoperability testing.

### 4.7.1 Version

Cisco UCM version

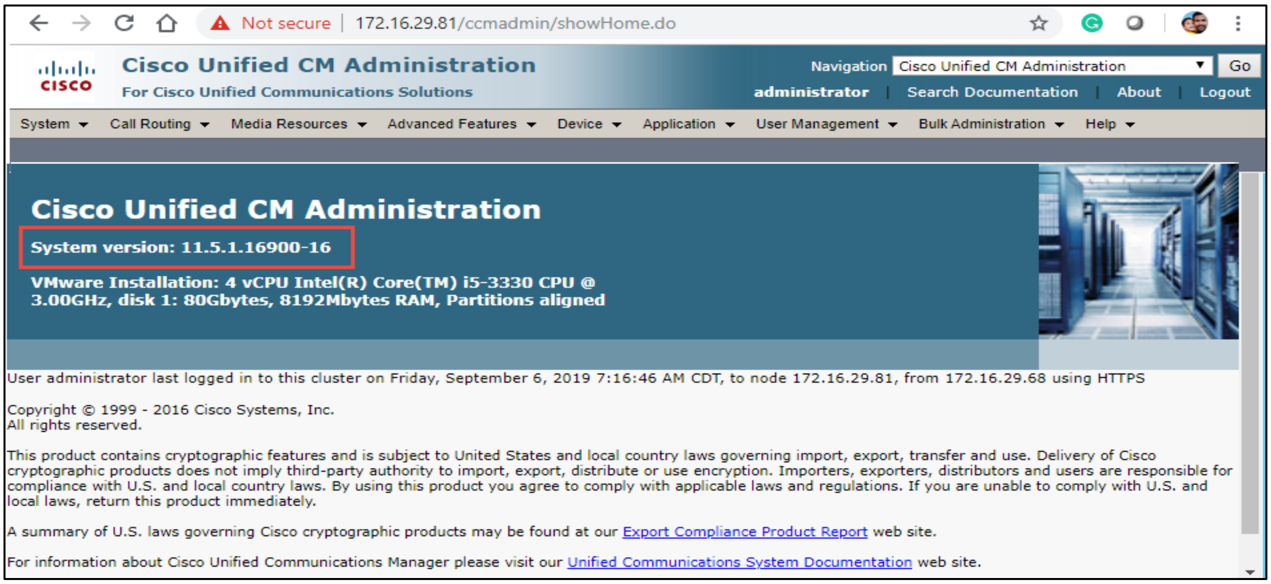


*Figure 130 – Cisco UCM Version*

### 4.7.2 Cisco UCM Audio Codec Preference List

To Configure Audio Codec Preference list**, navigate to** System **→ Region Information →Audio codec preference list**

Cisco UCM 9.0 introduced a new feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)

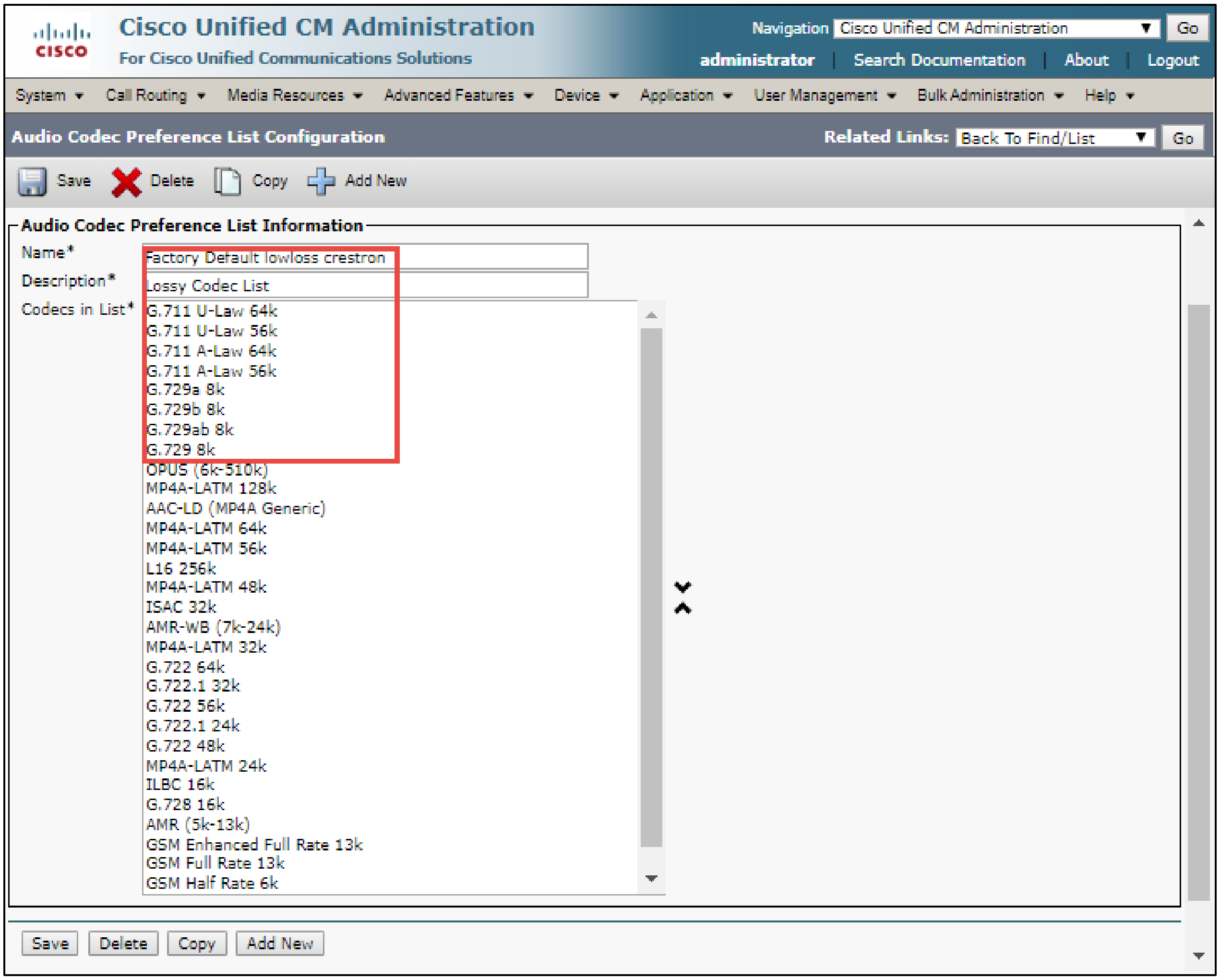

*Figure 131 – Audio Codec Preference List*

### 4.7.3 Cisco UCM Region Configuration

To configure Region Configuration, navigate to **System → Region Information → Region**

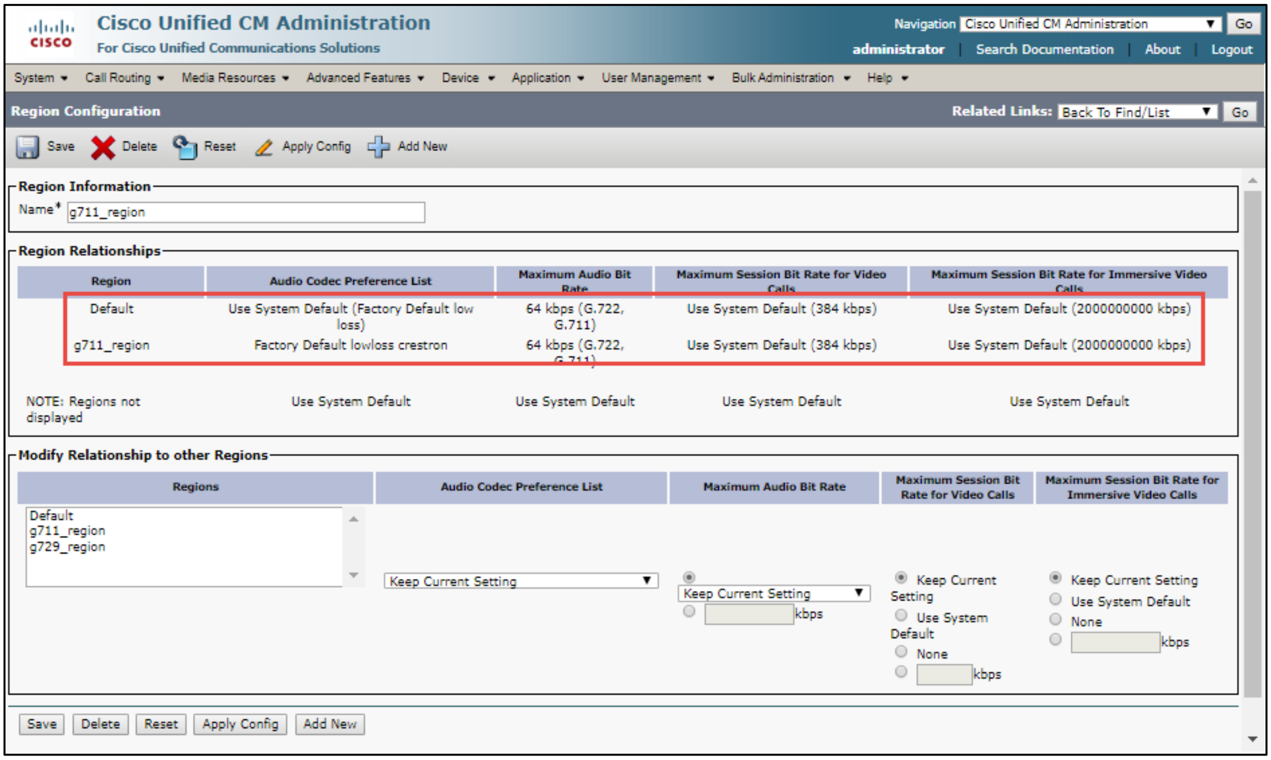


*Figure 132 – Cisco UCM Region*

### 4.7.4 Cisco UCM Device Pool

To configure Device Pool, navigate to **System → Device Pool**

"G711_Pool" Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices

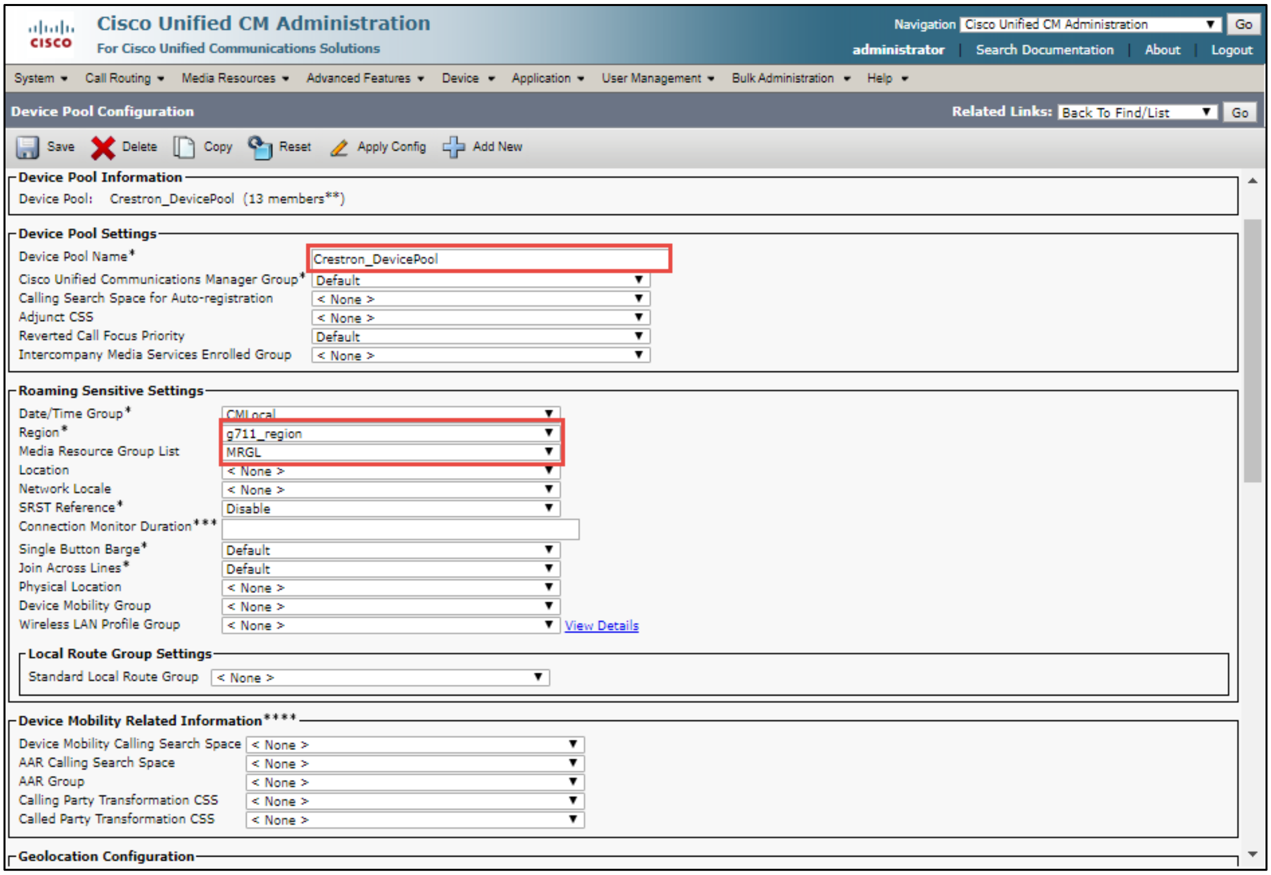


*Figure 133 – Cisco UCM Device Pool – Contd.*

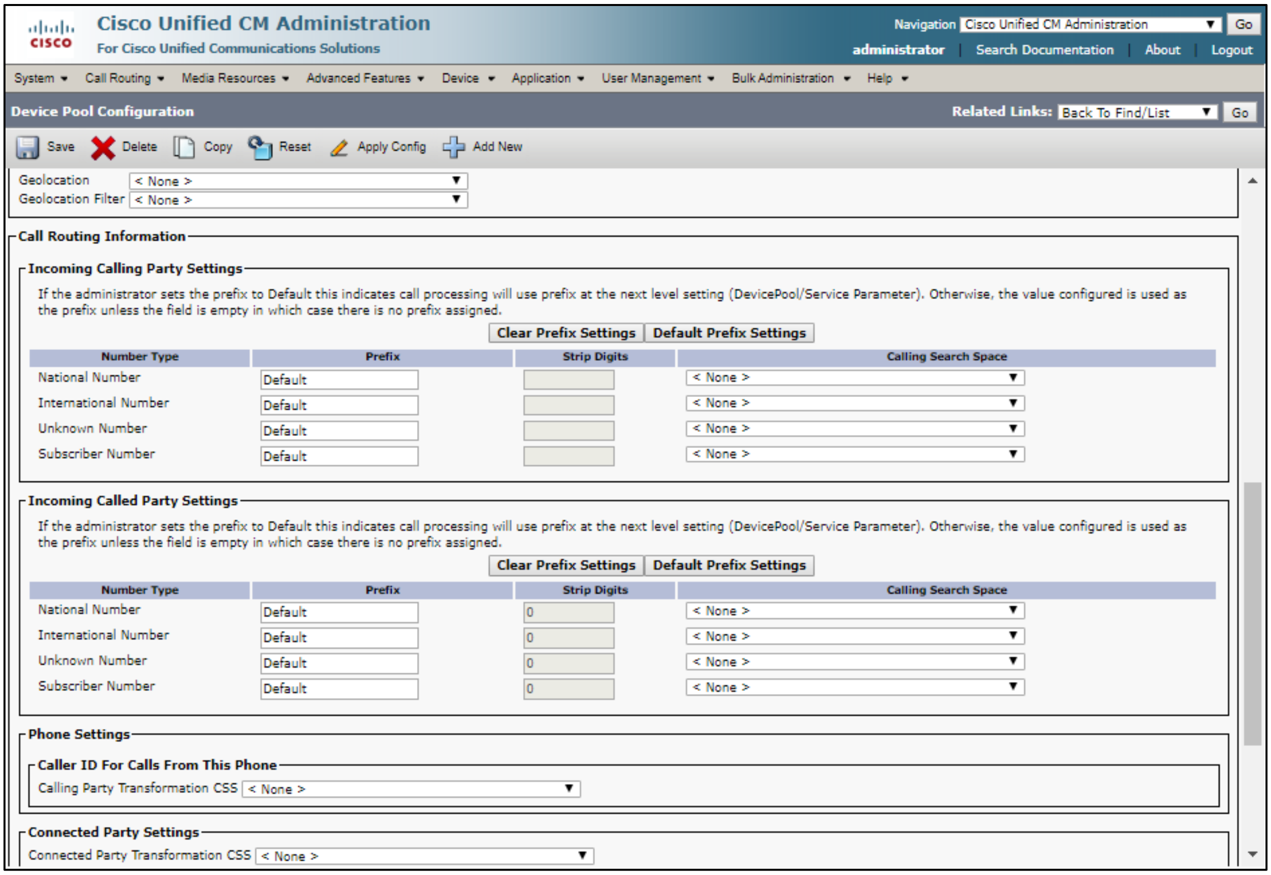

*Figure 134 – Cisco UCM Device Pool*

### 4.7.5 Cisco UCM Annunciator Configuration

To configure Annunciator, navigate to **Media Resource → Annunciator**

Set Name* = ANN_2.
Set Description = ANN_tekcucm5-cucmpub. This is used for this example
Set Device Pool* = Crestron_DevicePool

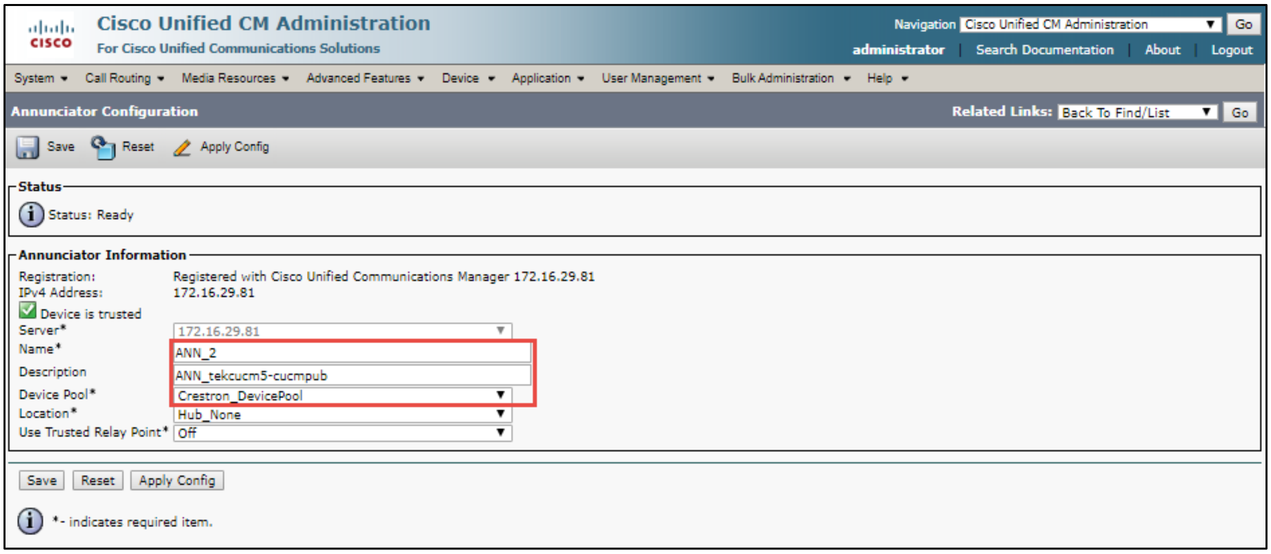


*Figure 135 – Cisco UCM Annunciator*

### 4.7.6 Cisco UCM Conference Bridge

To configure Conference Bridge, navigate to **Media Resource → Annunciator**

Set Name* = CFB_2
Set Description = ANN_tekcucm5-cucmpub. This is used for this example
Set Device Pool* = Crestron_DevicePool

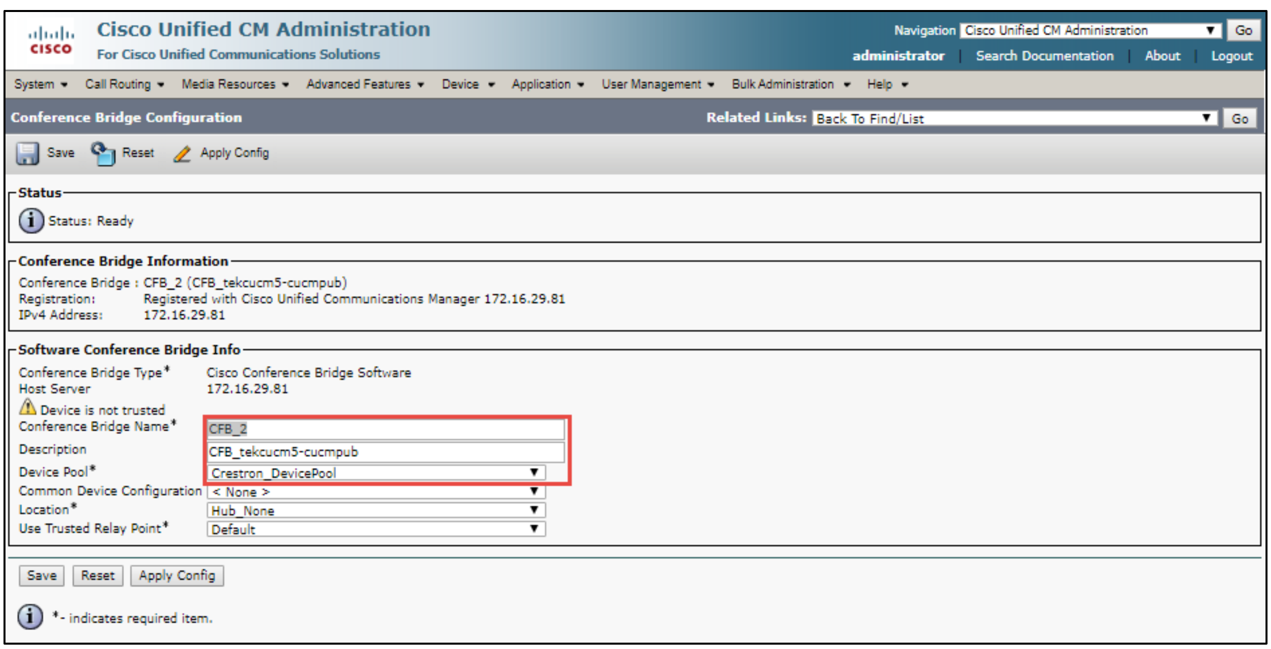


*Figure 136 – Cisco UCM Conference Bridge*

### 4.7.7 Cisco UCM MTP

To configure MTP, navigate to **Media Resource →MTP**

Set Name* = MTP_2
Set Description = MTP_tekcucm5-cucmpub. This is used for this example
Set Device Pool* = Crestron_DevicePool

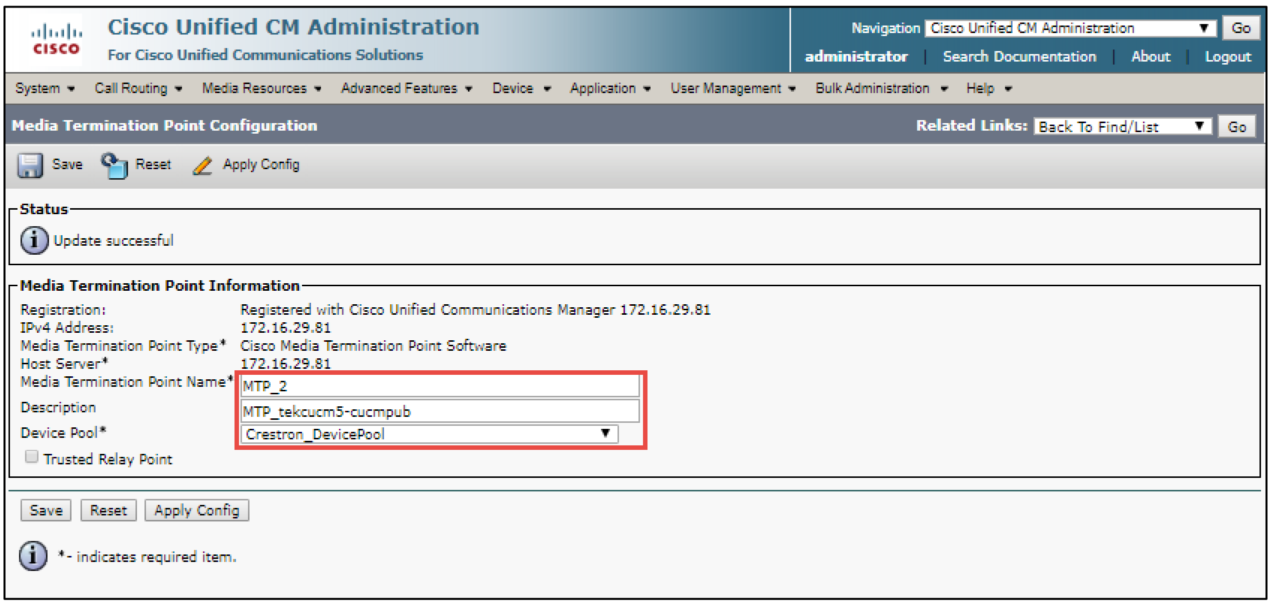

*Figure 137 – Cisco UCM MTP*

### 4.7.8 Cisco Media Resource Group

To configure IP-to-IP routing rules, navigate to **Media Resource → MRG**

Set Name* = MRG
Selected Media Resources Group as shown on below Screen used in this example

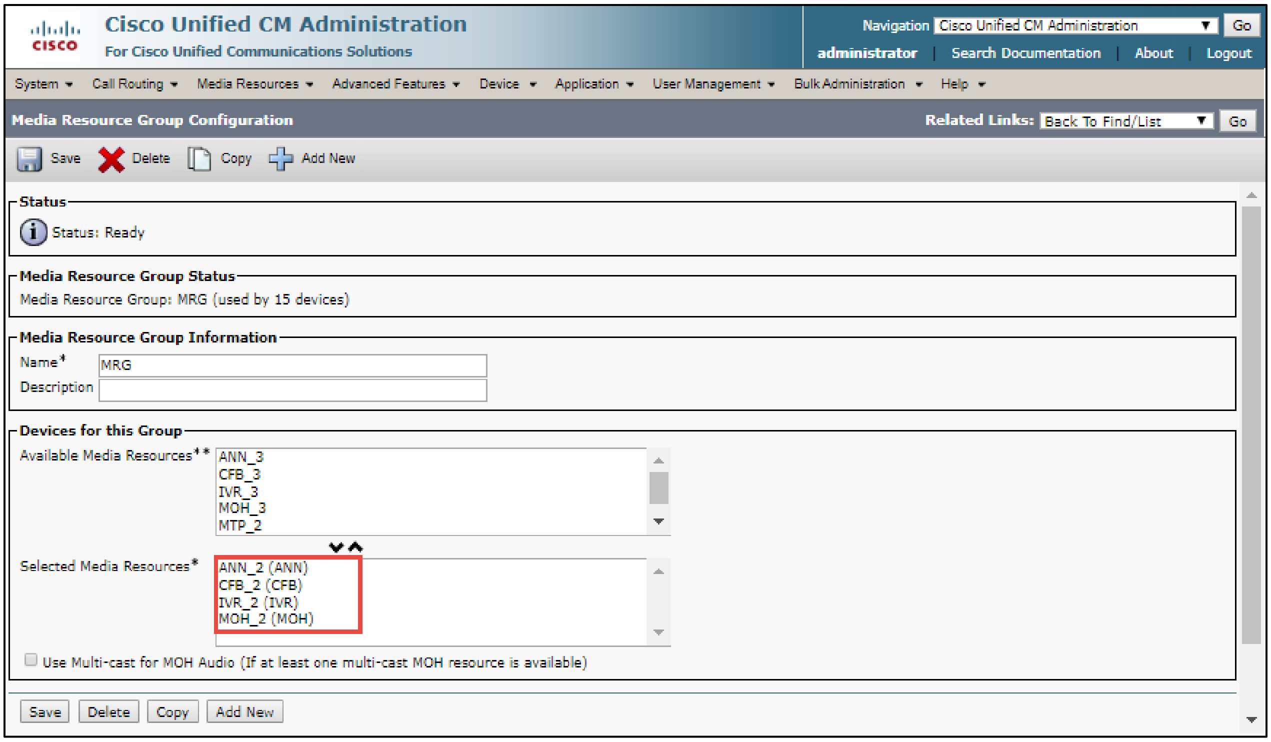


*Figure 138 – Cisco UCM MTP*

## 4.7.9 Cisco Media Resource Group List

To configure Media Resource Group List, navigate to **Media Resource → MRGL**

Set Name* = MRGL
Selected Media Resources Groups = MRG

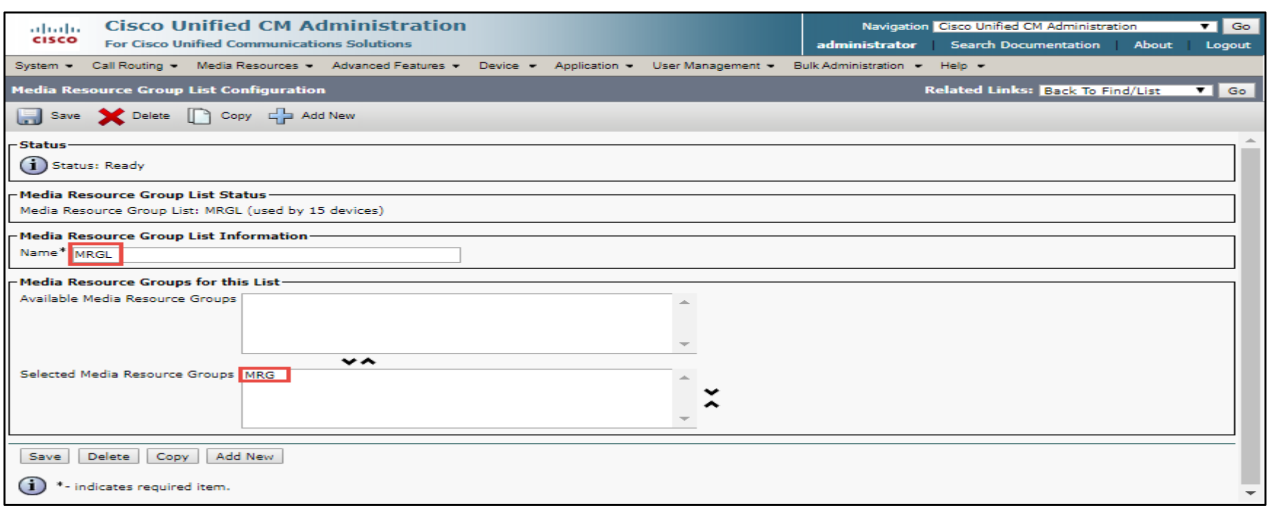


*Figure 139 – Cisco UCM MRGL*

### 4.7.10 Cisco UCM SIP Trunk towards Cisco UBE

To configure SIP Trunk**,** navigate to **Device -> Trunk**

Set **Device Name*** = Cube_Crestron_Teams. This is used for this example
Set **Description** = Cube_Crestron_Teams. This is used for this example
Set **Device Pool*** = Crestron_Devicepool. This is used for this example
Set **Media Resource Group List** = MRGL

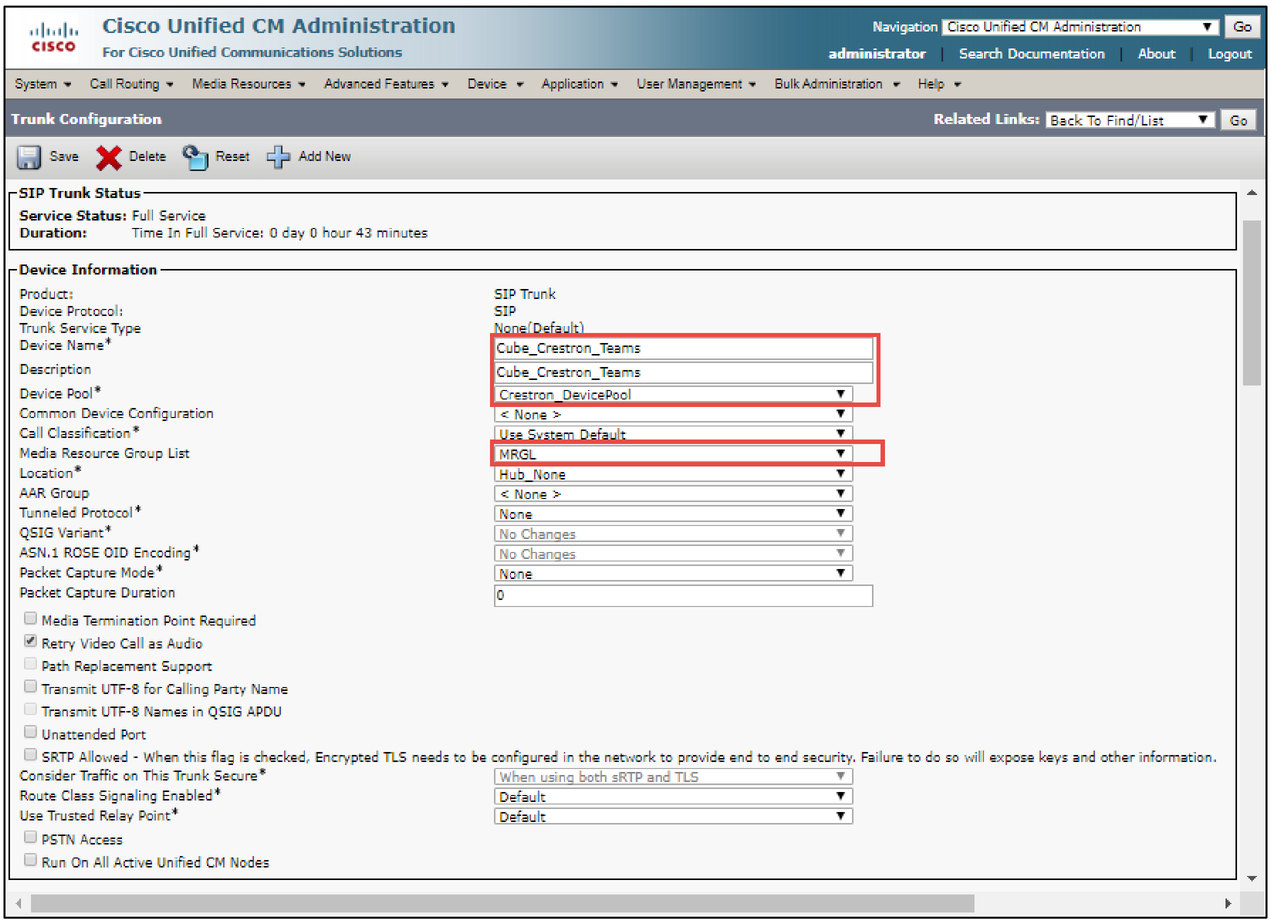


*Figure 140 – SIP Trunk – Cisco UBE – Contd.*

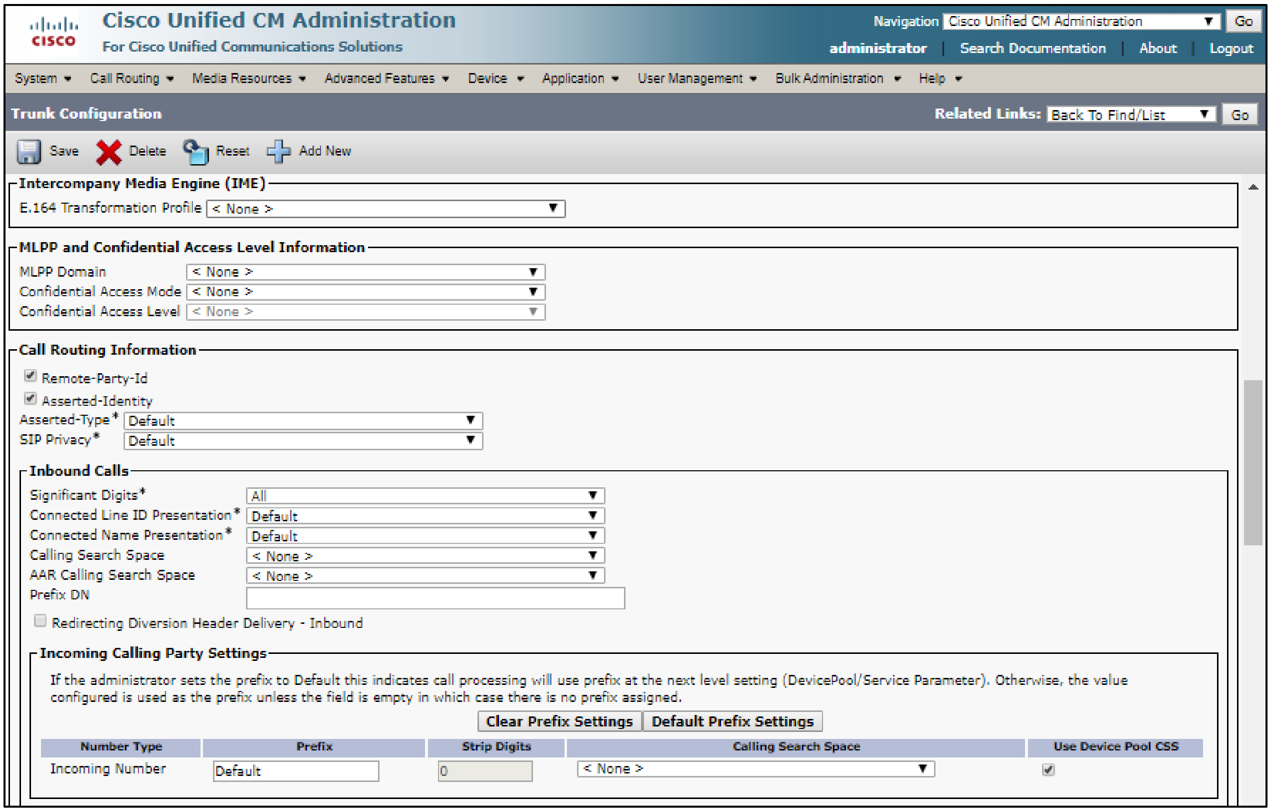

*Figure 141 – SIP Trunk – Cisco UBE – Contd.*

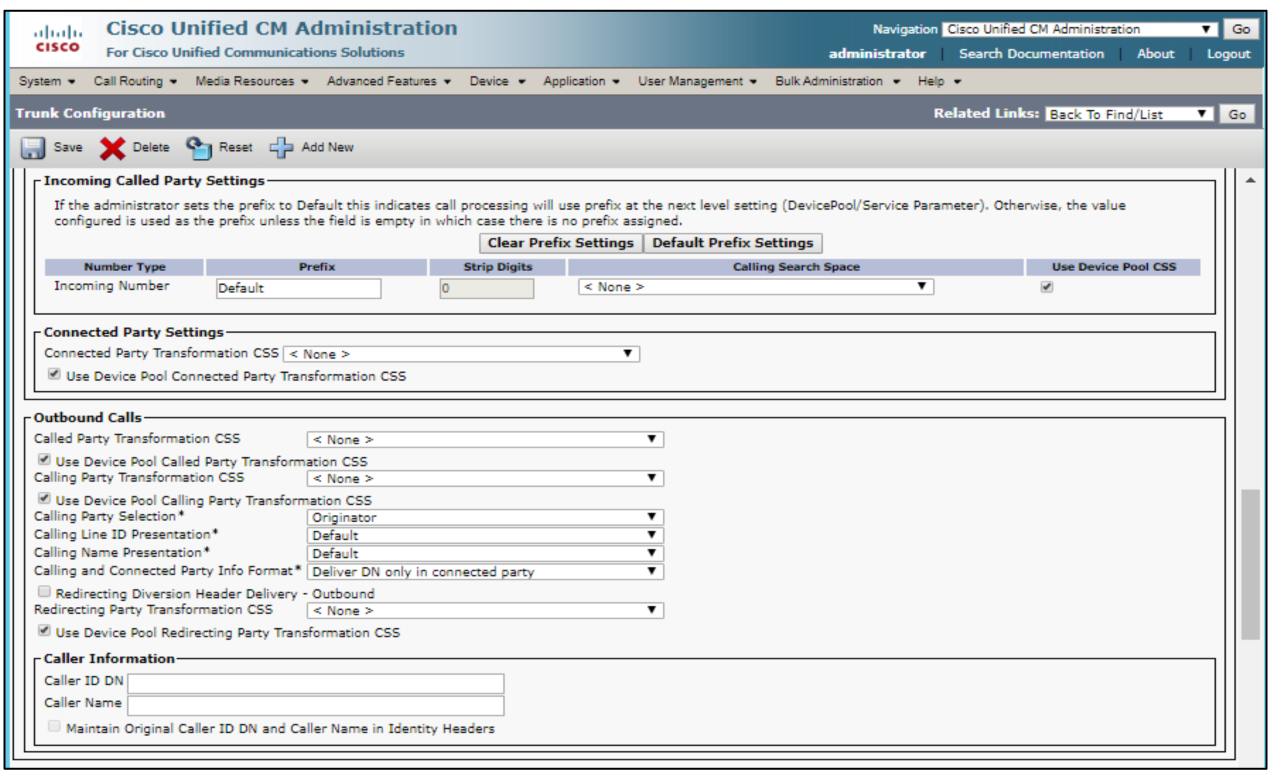

*Figure 142 – SIP Trunk – Cisco UBE – Contd.*

Set **Destination Address** = Set IP address of Cisco UBE.

Set **SIP Trunk Security Profile*** = Non Secure Sip Trunk Profile.
Set **SIP Profile*** = Crestron Profile. This is used in this example.
Set **DTMF Signaling Method*** = No Preference. This is used in this example.

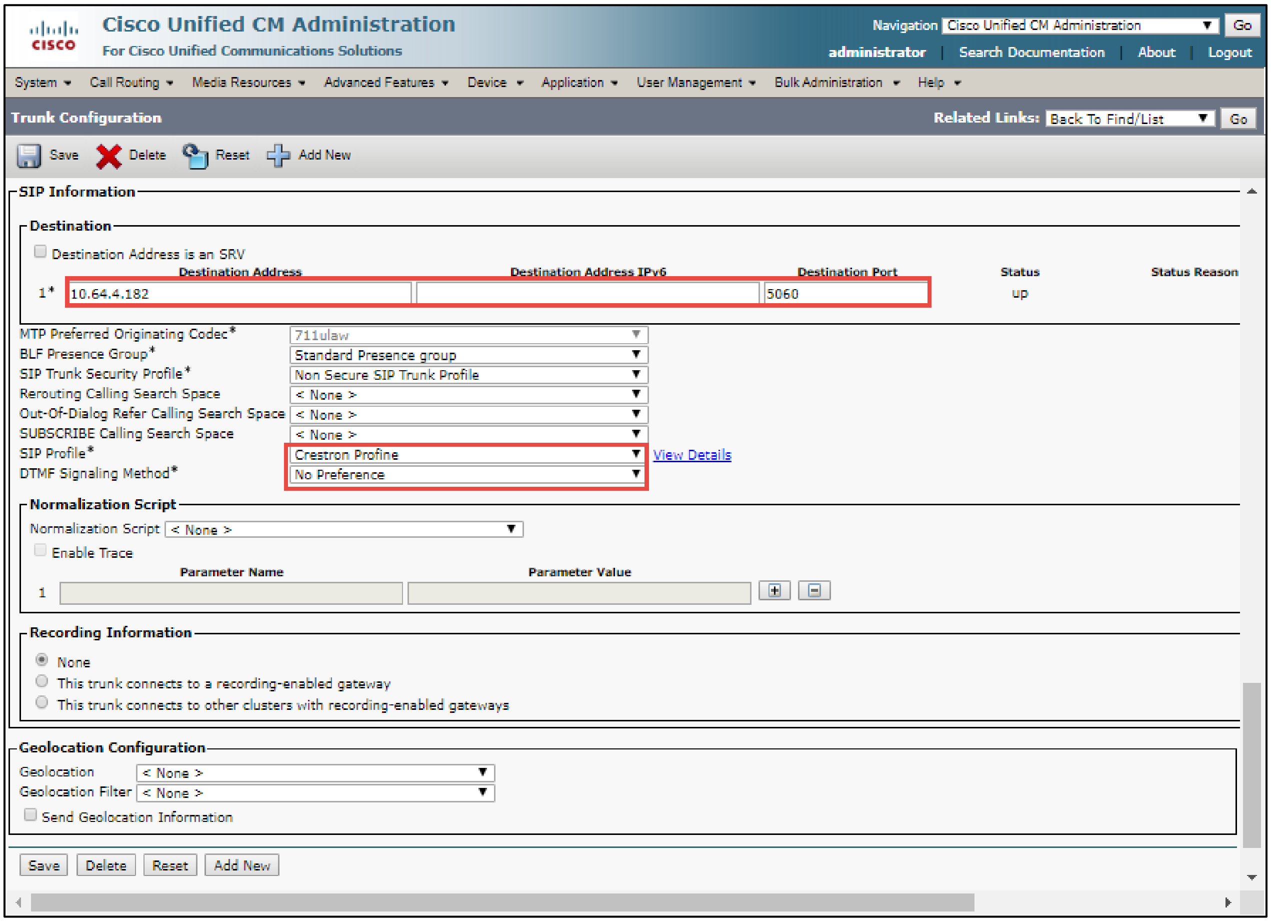

*Figure 143 – SIP Trunk – Cisco UBE*

### 4.7.11 Cisco UCM SIP Trunk towards Cisco Unity

Set **Device Name*** = Unity. This is used for this example
Set **Description** = Unity Connection. This is used for this example
Set **Media Resource Group List** = MRGL

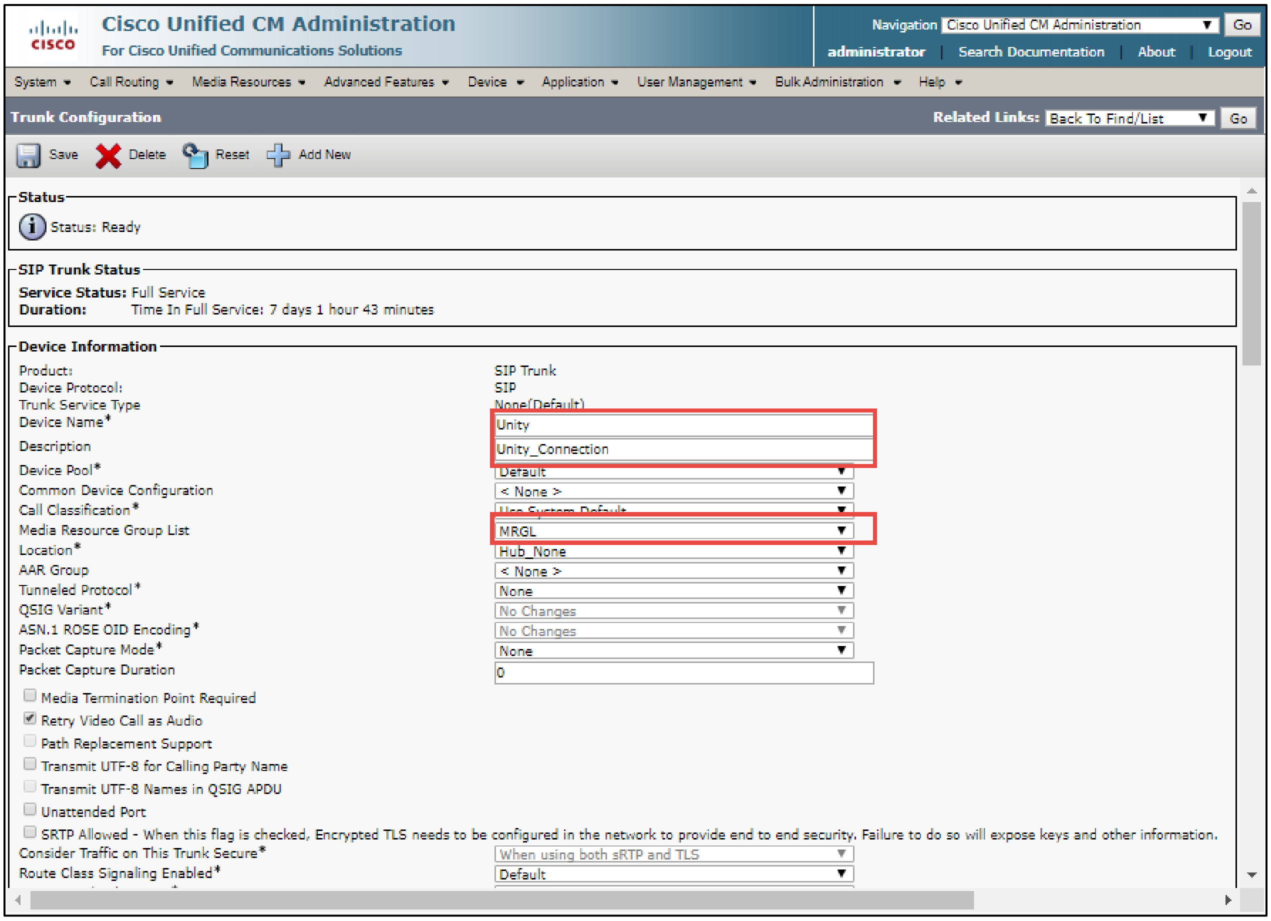


*Figure 144 – SIP Trunk – Cisco Unity – Contd.*

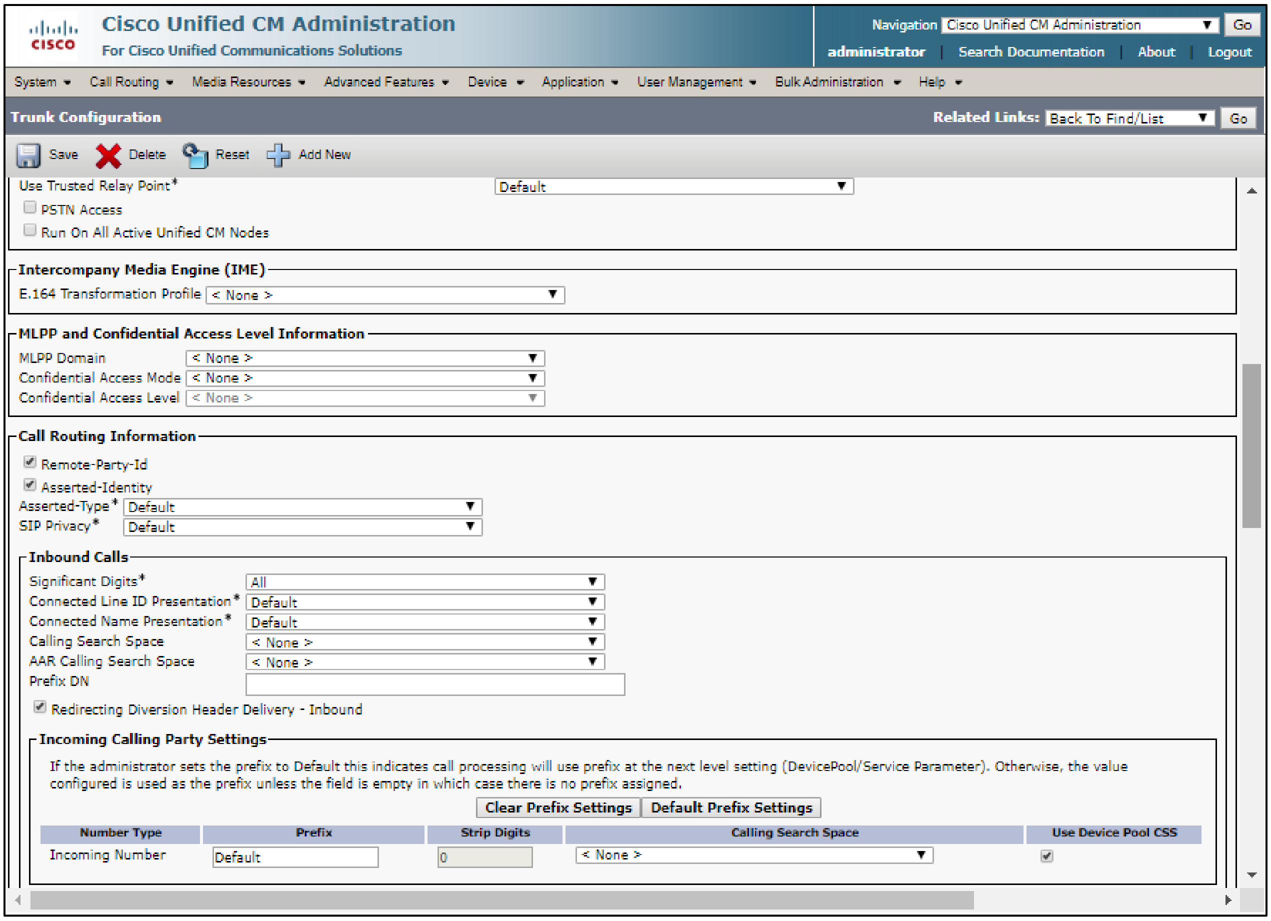


*Figure 145 – SIP Trunk – Cisco Unity – Contd.*

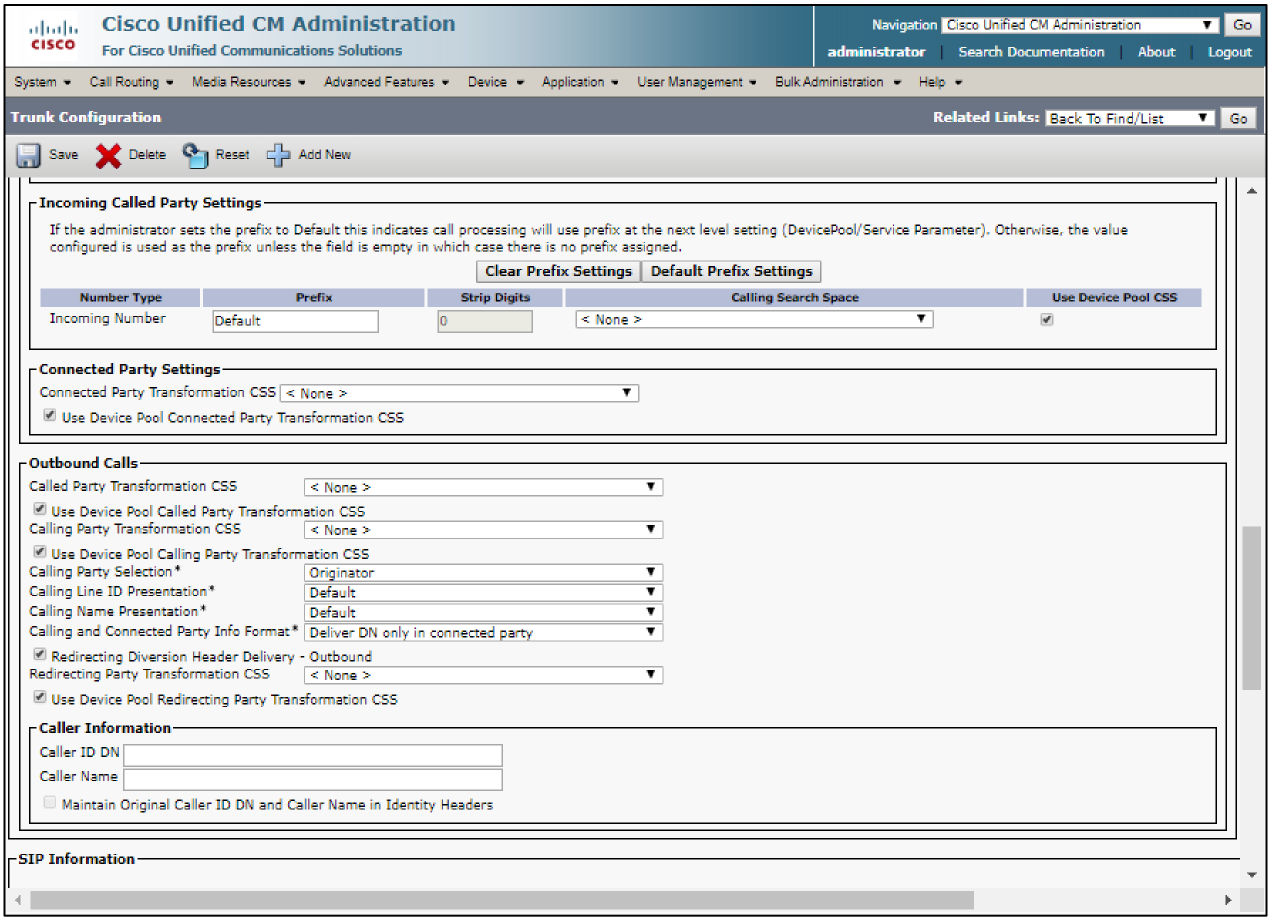


*Figure 146 – SIP Trunk – Cisco Unity – Contd.*

Set **Destination Address** = Set IP address of Cisco UBE.
Set **SIP Trunk Security Profile*** = Non Secure Sip Trunk Profile.
Set **SIP Profile*** = Standard SIP Profile - OPTIONS. This is used in this example.
Set **DTMF Signaling Method*** = No Preference. This is used in this example.

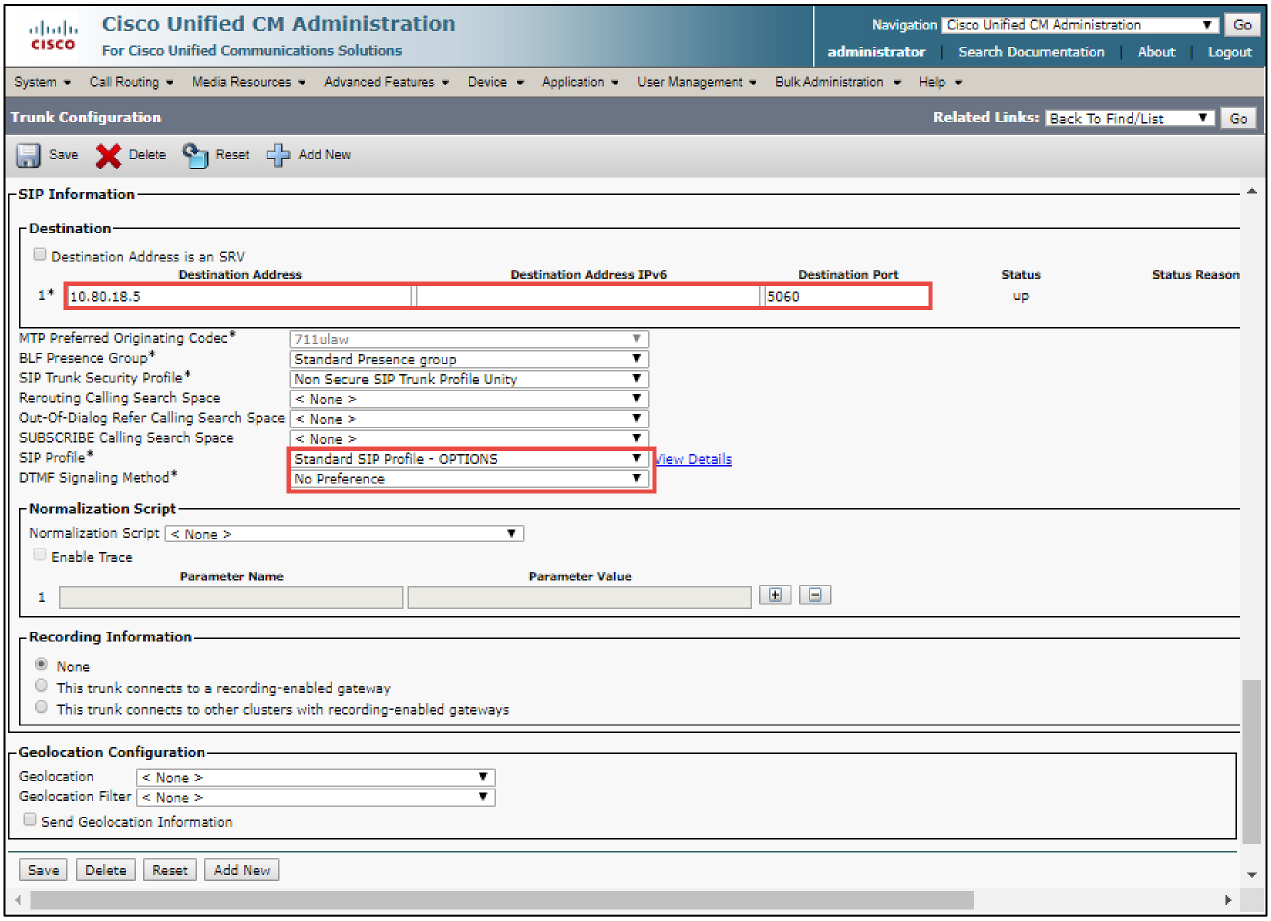


*Figure 147 – SIP Trunk – Cisco Unity*

## 4.8 Cisco Unity Connection (CUC)

### 4.8.1 Telephony Integration – Phone System

To configure CUC, **navigate to Telephony Integrations → Phone system**
Add New
Set Phone System Name* = Cisco_Crestron. This Name used for this test

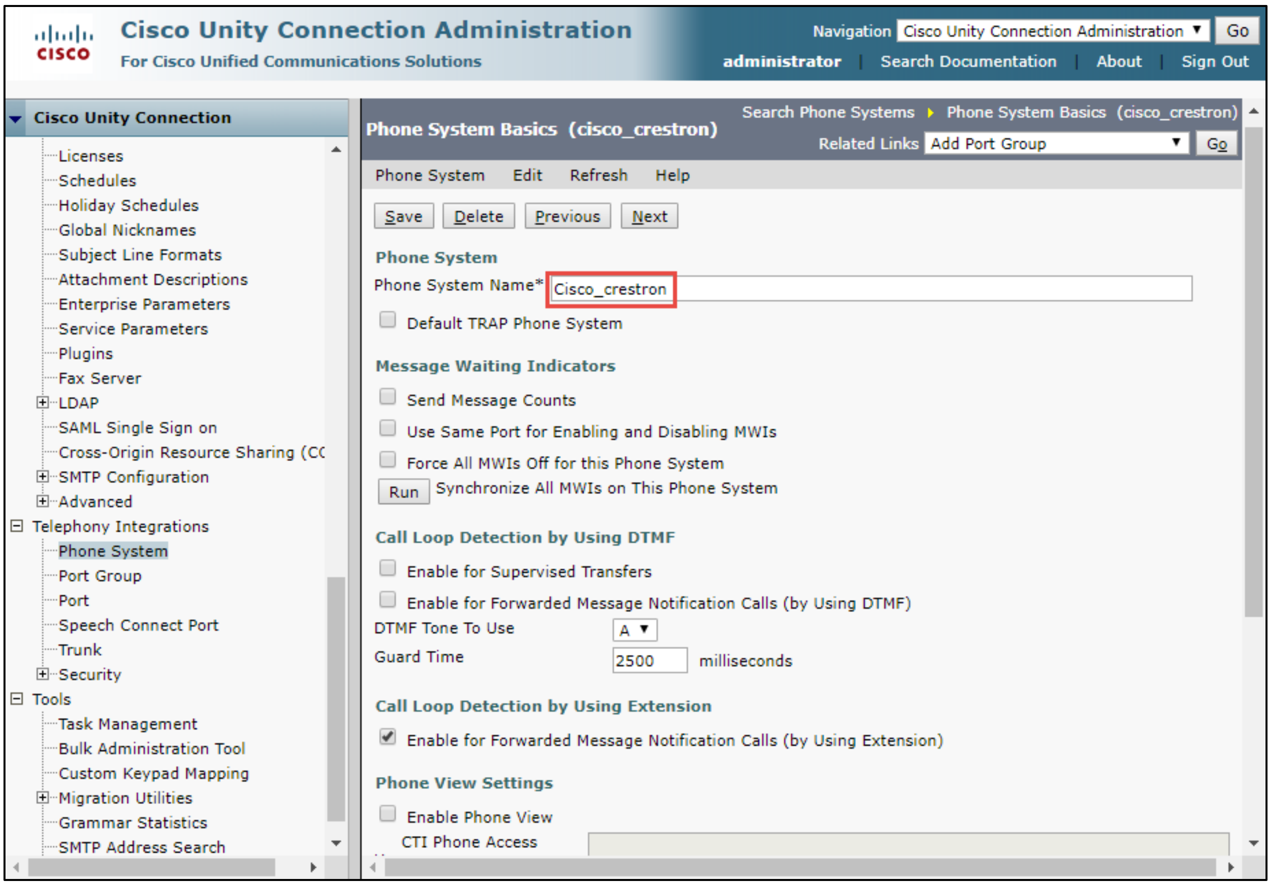


*Figure 148 – SIP Trunk – Phone System – Contd.*

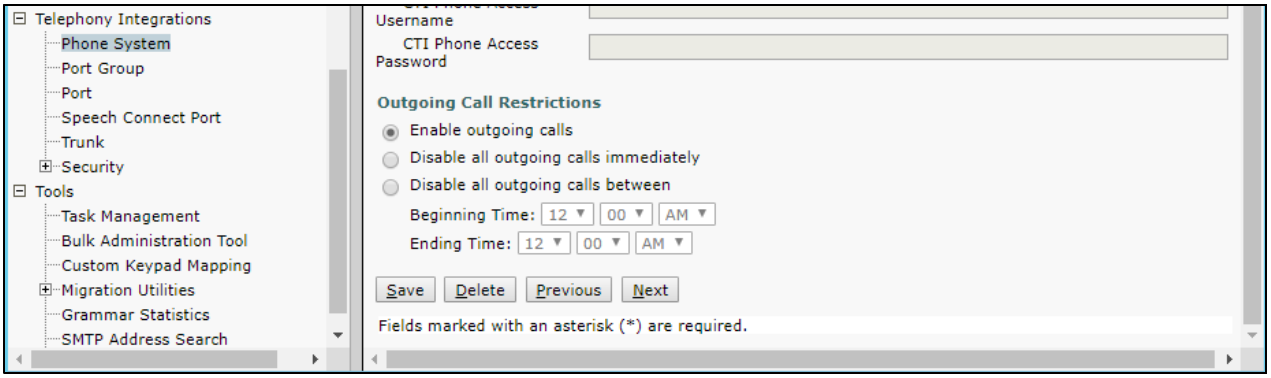


*Figure 149 – SIP Trunk – Phone System – Contd.*

### 4.8.2 Phone Group

To configure Port Group, navigate to **Telephony Integrations** -> **Port Group**

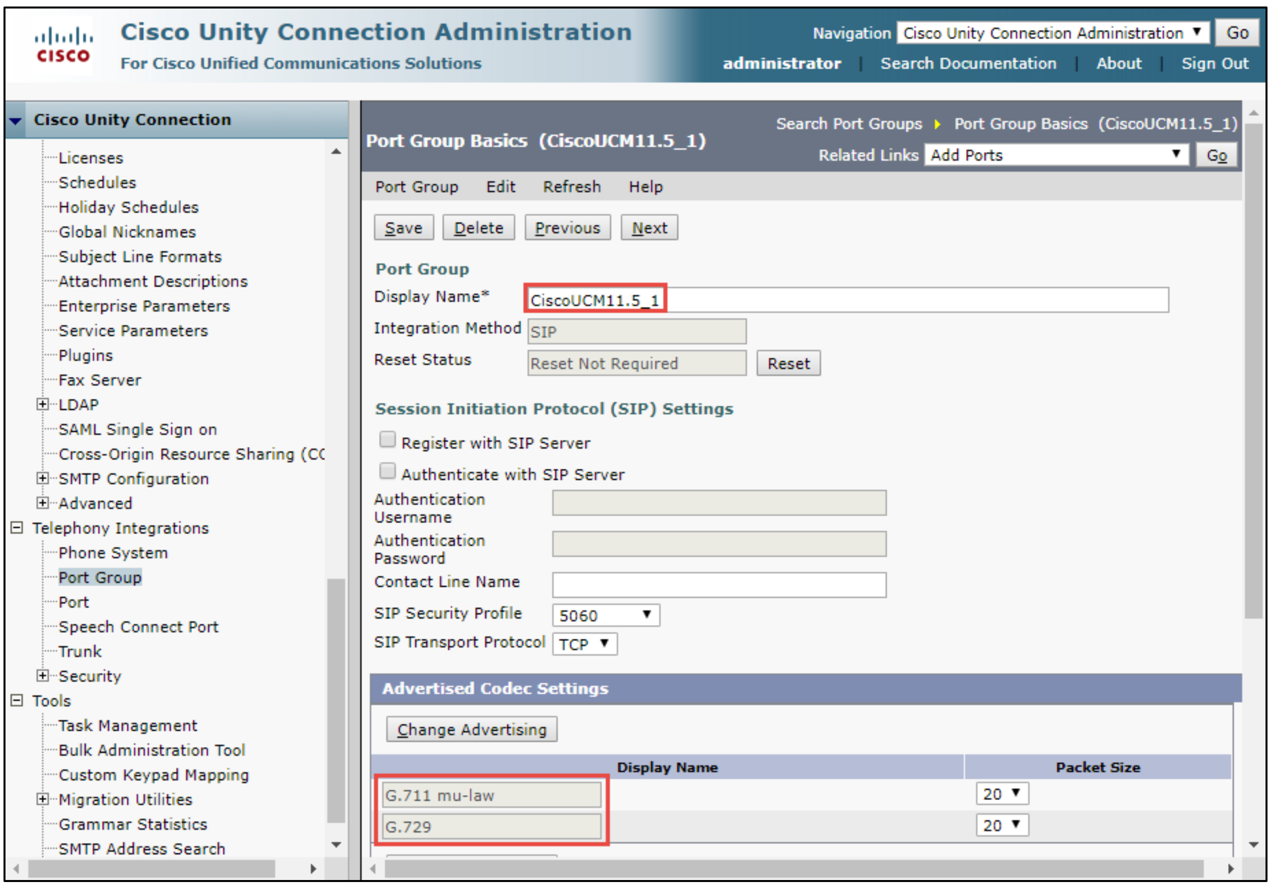


*Figure 150 –Phone Group – Contd.*

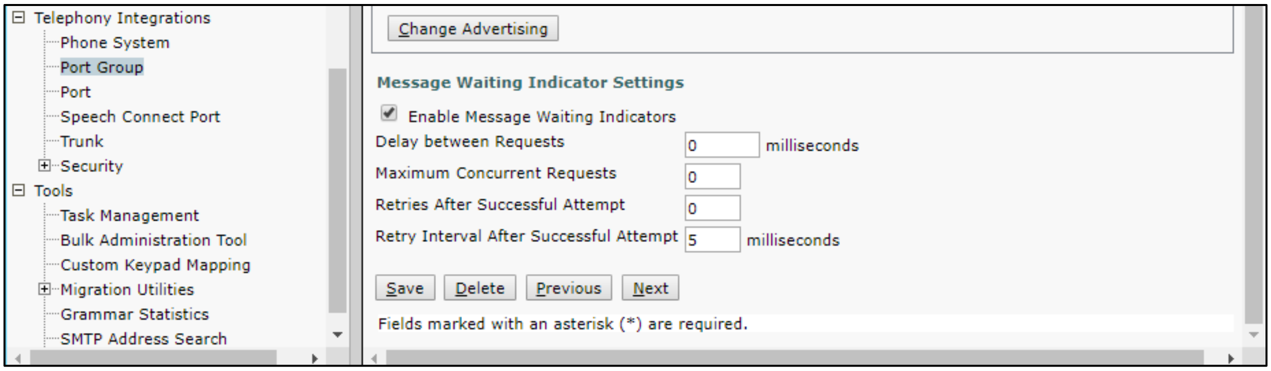


*Figure 151 –Phone Group*

### 4.8.3 Port

To configure Port, navigate to **Telephony Integrations → Port**

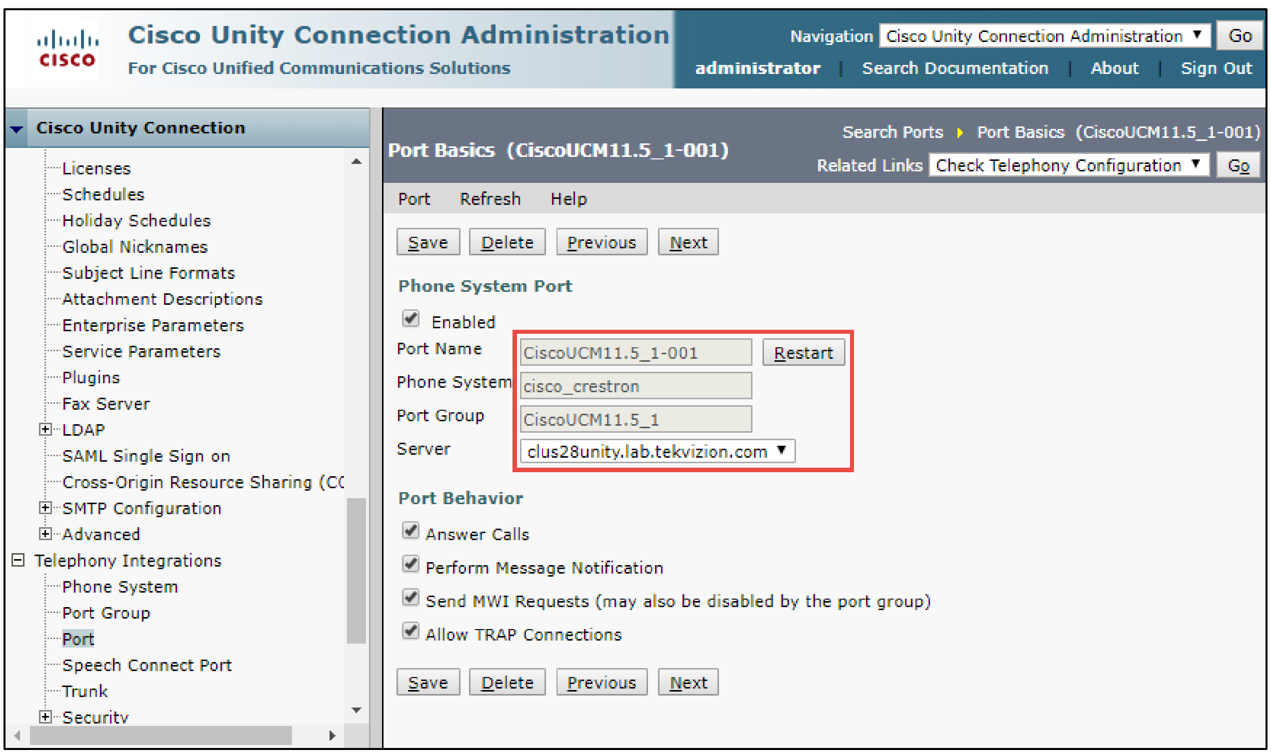


*Figure 152 – Port*

### 4.8.4 User

**To configure User, navigate to** Cisco Unity Connection → Users → Users

Set **Alias***= **6500** - This is used for the test

Set **First Name** = **CUCM** - This is used to identify the User

Set **Extension*** = **6500** - This is user's extension number

All other values are default.

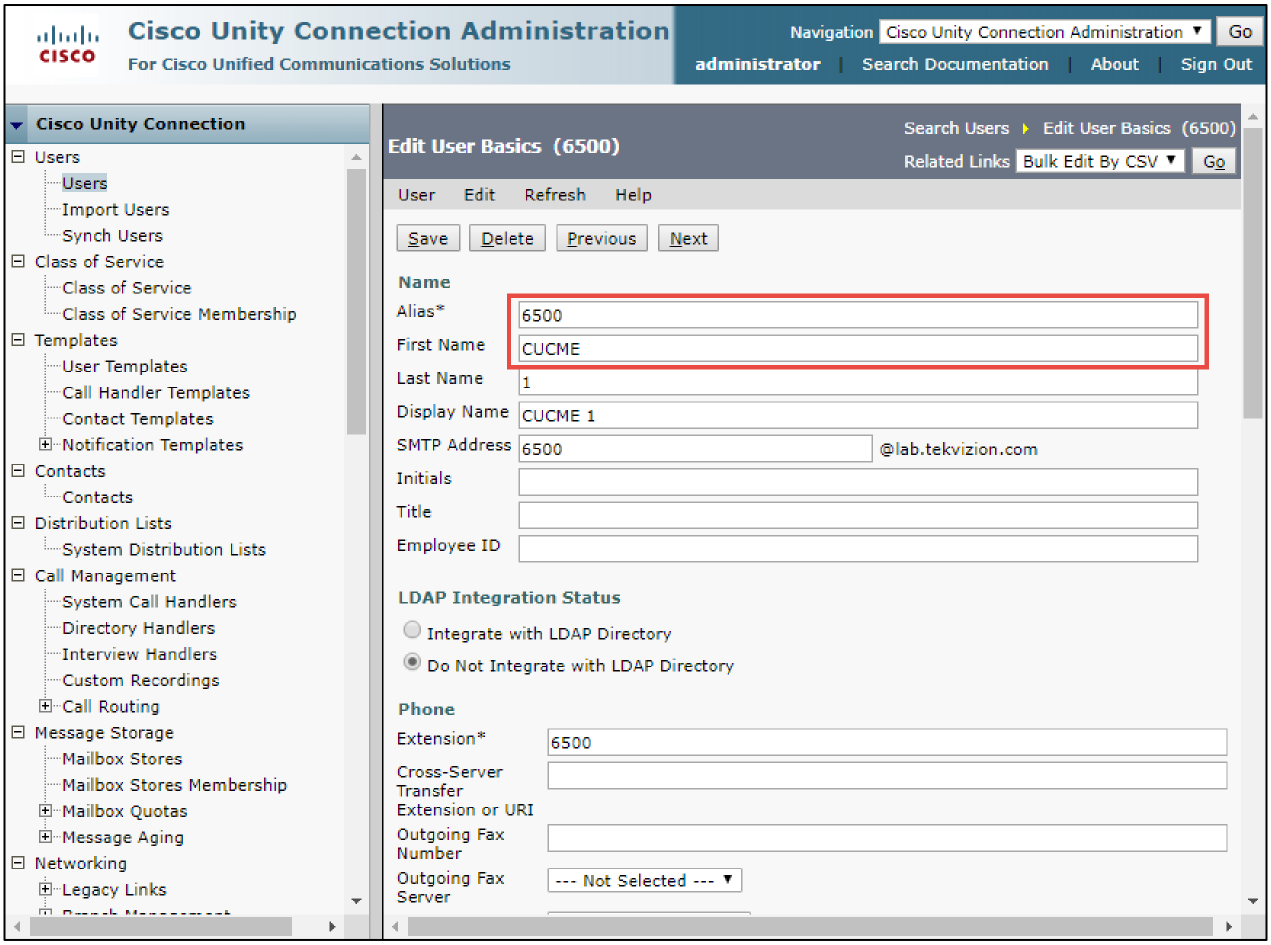


*Figure 153 – User*

Set **Users*** = **cisco_crestron** - Phone system used in this example

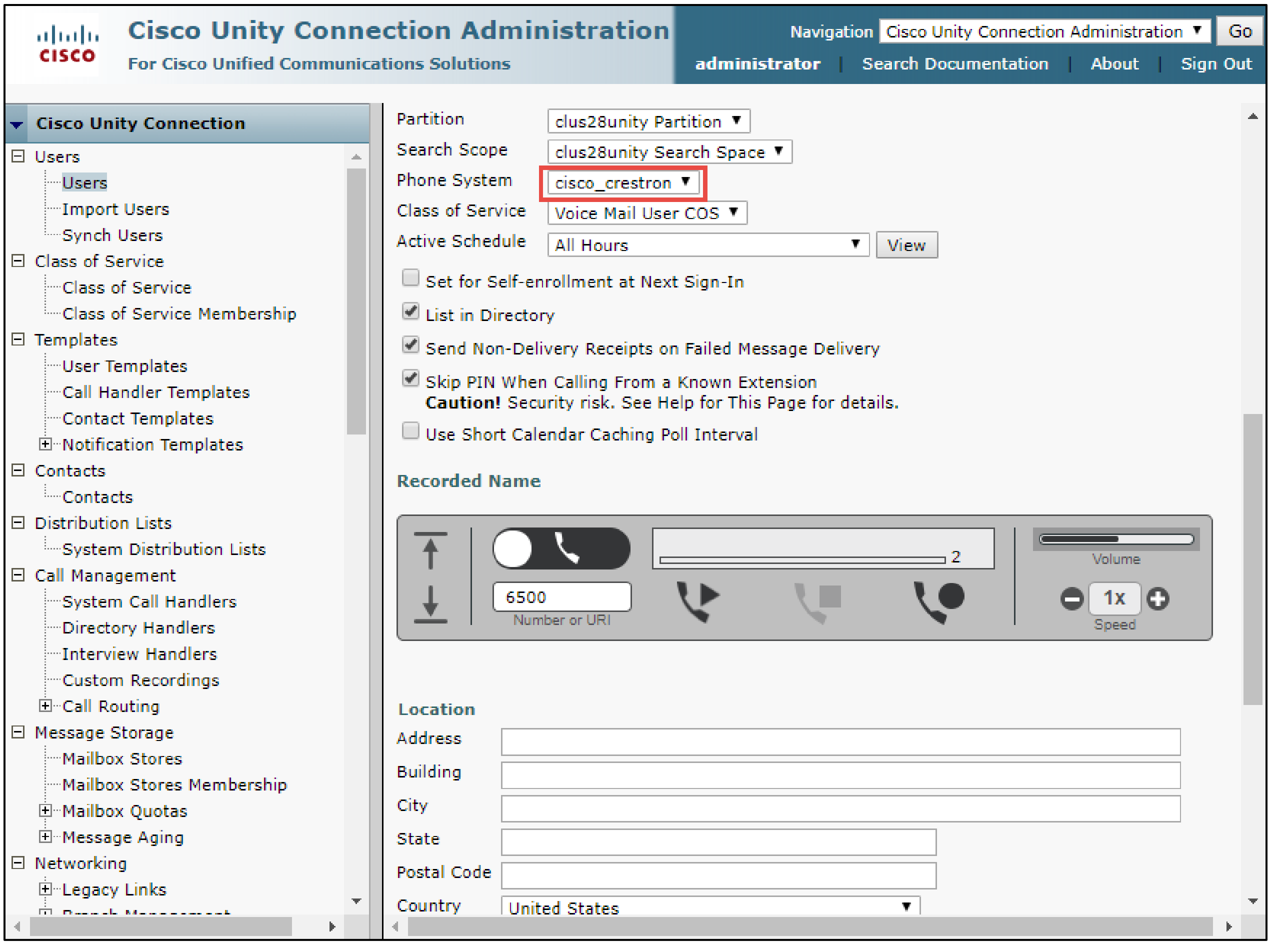


Figure 154 – User

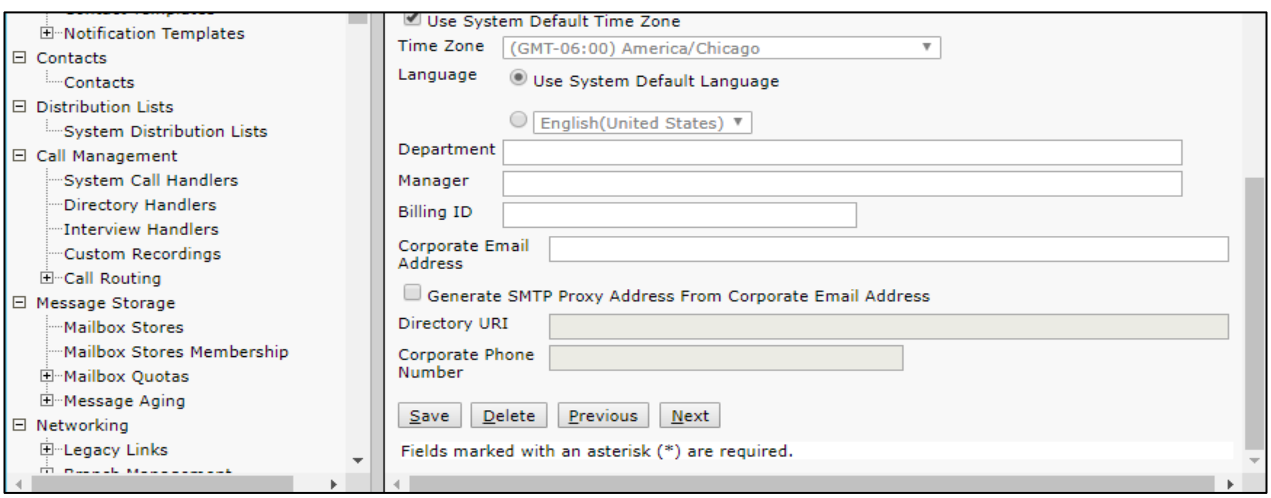


Figure 155 – User

# 5 Acronyms

| Acronym | Definition |
|---|---|
| Cisco UCM | Cisco Unified Communications Manager |
| 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 |
| **Cisco UBE** | **Cisco Unified Border Element** |

# 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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. PBX A user hangs up the call<br>9. Verify call is cleared successfully | 1. Call is connected with bi-directional audio, voice is clear, no echo<br>2. Call is disconnected | PASSED | |
| 2 | Teams user Calls PBX B user | 1. Make a voice call from Teams user to PBX B user<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. PBX B user hangs up the call<br>9. Verify call is cleared successfully | 1. Call is connected with bi-directional audio, voice is clear, no echo<br>2. Call is disconnected | PASSED | |
| 3 | Teams user Calls PSTN user | 1. Make a voice call from Teams user to PSTN user<br>2. Teams user hears Ring back Tone | 1. Call is connected with bi-directional audio, voice is clear, | PASSED | |

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| | | 3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. PSTN user hangs up the call<br>9. Verify call is cleared successfully | no echo<br>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<br>2. PBX A user starts ringing<br>3. Teams user hears Ring back Tone<br>4. Teams user hangs up the call while PBX A user is ringing<br>5. PBX A user stops ringing<br>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<br>2. PBX B user starts ringing<br>3. Teams user hears Ring back Tone<br>4. Teams user hangs up the call while PBX B user is ringing<br>5. PBX B user stops ringing<br>6. Verify call is cleared successfully | 1. Call is disconnected before answer | PASSED | |

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| 6 | Teams user Calls PSTN user and hangs up before answer | 1. Make a voice call from Teams user to PSTN user<br>2. PSTN user starts ringing<br>3. Teams user hears Ring back Tone<br>4. Teams user hangs up the call while PSTN user is ringing<br>5. PSTN user stops ringing<br>6. Verify call is cleared successfully | 1. Call is disconnected before answer | PASSED | |
| 7 | PBX A user Calls Teams user | 1. Make a voice call from PBX A user to Teams user<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. PBX A user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. Teams user hangs up the call<br>9. Verify call is cleared successfully | 1. Call is connected with bi-directional audio, voice is clear, no echo<br>2. Call is disconnected | PASSED | |
| 8 | PBX B user Calls Teams user | 1. Make a voice call from PBX B user to Teams user<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. PBX B user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. Teams user hangs up the call<br>9. Verify call is cleared successfully | 1. Call is connected with bi-directional audio, voice is clear, no echo<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. PSTN user hangs up the call<br>6. Verify call is cleared successfully<br>7. Repeat steps 1 to 4<br>8. Teams user hangs up the call<br>9. Verify call is cleared successfully | 1. Call is connected with bi-directional audio, voice is clear, no echo<br>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<br>2. Teams user starts ringing<br>3. PBX A user hears Ring back Tone<br>4. PBX A user hangs up the call while Teams user is ringing<br>5. Teams user stops ringing<br>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<br>2. Teams user starts ringing<br>3. PBX B user hears Ring back Tone<br>4. PBX B user hangs up the call while Teams user is ringing<br>5. Teams user stops ringing<br>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<br>2. Teams user starts ringing<br>3. PSTN user hears Ring back Tone<br>4. PSTN user hangs up the call while Teams user is ringing<br>5. Teams user stops ringing<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is on hold<br>7. Teams user resumes the call<br>8. Verify two way audio is re-established between the two end points<br>9. Teams user hangs up the call<br>10. Verify call is cleared successfully | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>4. Two way audio | PASSED | |

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| | | on hold<br>7. Teams user resumes the call<br>8. Verify two way audio is re-established between the two end points<br>9. Teams user hangs up the call<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is on hold<br>7. Teams user resumes the call<br>8. Verify two way audio is re-established between the two end points<br>9. Teams user hangs up the call<br>10. Verify call is cleared successfully | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is on hold<br>7. Teams user resumes the call | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>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<br>9. PBX A user hangs up the call<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is on hold<br>7. Teams user resumes the call<br>8. Verify two way audio is re-established between the two end points<br>9. PBX B user hangs up the call<br>10. Verify call is cleared successfully | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user initiates call hold<br>6. Verify no audio is present while call is on hold<br>7. Teams user resumes the call<br>8. Verify two way audio is re-established between the two end points | 1. Call is placed on hold successfully<br>2. No audio present during hold<br>3. Call is resumed successfully<br>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<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user 1 answers the call<br>4. Verify two way audio<br>5. Teams user places a consultation call to PBX A user 2<br>6. Verify PBX A user 1 is placed on hold<br>7. PBX A user 2 answers the call<br>8. Verify two way audio<br>9. Teams user completes the transfer<br>10. Verify two way audio between PBX A user 1 and PBX A user 2<br>11. PBX A user 1 hangs up the call<br>12. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user places a consultation call to PBX B user<br>6. Verify PBX A user is placed on hold<br>7. PBX B user answers the call<br>8. Verify two way audio<br>9. Teams user completes the transfer | 1. Call is transferred successfully<br>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<br>11. PBX A user hangs up the call<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user places a consultation call to PSTN user<br>6. Verify PBX A user is placed on hold<br>7. PSTN user answers the call<br>8. Verify two way audio<br>9. Teams user completes the transfer<br>10. Verify two way audio between PBX A user and PSTN user<br>11. PBX A user hangs up the call<br>12. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user 1 answers the call<br>4. Verify two way audio<br>5. Teams user places a consultation call to PBX B user 2<br>6. Verify PBX B user 1 is placed on hold<br>7. PBX B user 2 answers the call | 1. Call is transferred successfully<br>2. Two way audio present after call is transferred | PASSED | |

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

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| | | 8. Verify two way audio<br>9. Teams user completes the transfer<br>10. Verify two way audio between PSTN user and PBX A user<br>11. PSTN user hangs up the call<br>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<br>2. PSTN user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user places a consultation call to PSTN user 2<br>6. Verify PSTN user 1 is placed on hold<br>7. PSTN user 2 answers the call<br>8. Verify two way audio<br>9. Teams user completes the transfer<br>10. Verify two way audio between PSTN user 1 and PSTN user 2<br>11. PSTN user 1 hangs up the call<br>12. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user 1 answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user 2 | 1. Call is transferred successfully<br>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<br>7. PBX A user 1 hears Ring back Tone<br>8. PBX A user 2 answers the call<br>9. Verify two way audio between PBX A user 1 and PBX A user 2<br>10. PBX A user 1 hangs up the call<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user<br>6.PBX B user starts ringing<br>7. PBX A user hears Ring back Tone<br>8. PBX B user answers the call<br>9. Verify two way audio between PBX A user and PBX B user<br>10. PBX A user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user<br>6.PSTN user starts ringing<br>7. PBX A user hears Ring back Tone<br>8. PSTN user answers the call<br>9. Verify two way audio between PBX A user and PSTN user<br>10. PBX A user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user 1 answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user 2<br>6.PBX B user 2 starts ringing<br>7. PBX B user 1 hears Ring back Tone<br>8. PBX B user 2 answers the call<br>9. Verify two way audio between PBX B user 1 and PBX B user 2<br>10. PBX B user 1 hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user<br>6.PBX A user starts ringing<br>7. PBX B user hears Ring back Tone<br>8. PBX A user answers the call<br>9. Verify two way audio between PBX B user and PBX A user<br>10. PBX B user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user<br>6.PSTN user starts ringing<br>7. PBX B user hears Ring back Tone<br>8. PSTN user answers the call<br>9. Verify two way audio between PBX B user and PSTN user<br>10. PBX B user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user<br>6.PBX B user starts ringing<br>7. PSTN user hears Ring back Tone<br>8. PBX B user answers the call<br>9. Verify two way audio between PSTN user and PBX B user<br>10. PSTN user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user<br>6.PBX A user starts ringing<br>7. PSTN user hears Ring back Tone<br>8. PBX A user answers the call<br>9. Verify two way audio between PSTN user and PBX A user<br>10. PSTN user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user 1 answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user 2<br>6.PSTN user 2 starts ringing<br>7. PSTN user 1 hears Ring back Tone<br>8. PSTN user 2 answers the call<br>9. Verify two way audio between PSTN user 1 and PSTN user 2<br>10. PSTN user 1 hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX A user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user 2<br>6.PBX A user 2 starts ringing<br>7. PBX A user 1 hears Ring back Tone<br>8. PBX A user 2 answers the call<br>9. Verify two way audio between PBX A user 1 and PBX A user 2<br>10. PBX A user 1 hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user<br>6.PBX B user starts ringing<br>7. PBX A user hears Ring back Tone<br>8. PBX B user answers the call<br>9. Verify two way audio between PBX A user and PBX B user<br>10. PBX A user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user<br>6.PSTN user starts ringing<br>7. PBX A user hears Ring back Tone<br>8. PSTN user answers the call<br>9. Verify two way audio between PBX A user and PSTN user<br>10. PBX A user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX B user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user 2<br>6.PBX B user 2 starts ringing<br>7. PBX B user 1 hears Ring back Tone<br>8. PBX B user 2 answers the call<br>9. Verify two way audio between PBX B user 1 and PBX B user 2<br>10. PBX B user 1 hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user<br>6.PBX A user starts ringing<br>7. PBX B user hears Ring back Tone<br>8. PBX A user answers the call<br>9. Verify two way audio between PBX B user and PBX A user<br>10. PBX B user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user<br>6.PSTN user starts ringing<br>7. PBX B user hears Ring back Tone<br>8. PSTN user answers the call<br>9. Verify two way audio between PBX B user and PSTN user<br>10. PBX B user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX B user<br>6.PBX B user starts ringing<br>7. PSTN user hears Ring back Tone<br>8. PBX B user answers the call<br>9. Verify two way audio between PSTN user and PBX B user<br>10. PSTN user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PBX A user<br>6.PBX A user starts ringing<br>7. PSTN user hears Ring back Tone<br>8. PBX A user answers the call<br>9. Verify two way audio between PSTN user and PBX A user<br>10. PSTN user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PSTN user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user transfers the call to PSTN user 2<br>6.PSTN user 2 starts ringing<br>7. PSTN user 1 hears Ring back Tone<br>8. PSTN user 2 answers the call<br>9. Verify two way audio between PSTN user 1 and PSTN user 2<br>10. PSTN user 1 hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user 1 answers the call<br>4. Verify two way audio<br>5. Teams user 1 transfers the call to Teams user 2<br>6.Teams user 2 starts ringing<br>7. Teams user 2 answers the call<br>8. Verify two way audio between PSTN user and Teams user 2<br>10. PSTN user hangs up the call<br>11. Verify call is cleared successfully | 1. Call is transferred successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user 1 answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user 2 to the ongoing call<br>6.PBX A user 2 starts ringing<br>7. PBX A user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX B user to the ongoing call<br>6.PBX B user starts ringing<br>7. PBX B user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user to the ongoing call<br>6.PSTN user starts ringing<br>7. PSTN user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user 1 answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX B user 2 to the ongoing call<br>6.PBX B user 2 starts ringing<br>7. PBX B user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user to the ongoing call<br>6.PBX A user starts ringing<br>7. PBX A user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user to the ongoing call<br>6. PSTN user starts ringing<br>7. PSTN user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX B user to the ongoing call<br>6. PBX B user starts ringing<br>7. PBX B user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user to the ongoing call<br>6. PBX A user starts ringing<br>7. PBX A user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. Teams user hears Ring back Tone<br>3. PSTN user 1 answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user 2 to the ongoing call<br>6. PSTN user 2 starts ringing<br>7. PSTN user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX A user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user 2 to the ongoing call<br>6. PBX A user 2 starts ringing<br>7. PBX A user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBXB user to the ongoing call<br>6. PBX B user starts ringing<br>7. PBX B user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user to the ongoing call<br>6. PSTN user starts ringing<br>7. PSTN user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX B user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX B user 2 to the ongoing call<br>6. PBX B user 2 starts ringing<br>7. PBX B user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user to the ongoing call<br>6.  PBX A user starts ringing<br>7.  PBX A user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user to the ongoing call<br>6. PSTN user starts ringing<br>7. PSTN user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX B user to the ongoing call<br>6. PBX B user starts ringing<br>7. PBX B user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PBX A user to the ongoing call<br>6. PBX A user starts ringing<br>7. PBX A user answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PSTN user 1 hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Teams user adds PSTN user 2 to the ongoing call<br>6. PSTN user 2 starts ringing<br>7. PSTN user 2 answers the call<br>9. Verify all three users are able to hear each other<br>10. Teams user hangs up the call<br>11. Verify call is cleared successfully | 1. Third user is added to the call successfully<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user 1 answers the call<br>4. Verify two way audio<br>5. Teams user 1 adds Teams user 2 to the ongoing call<br>6. Verify Teams user 2 is added successfully to the call<br>7. Teams user 1 adds PBX A user to the ongoing call<br>9. Verify PBX A user is added successfully to the call<br>10. Teams user 1 adds PBX B user to the ongoing call<br>11. Verify PBX B user is added successfully to the call<br>12. Verify all four users are able to hear each other<br>13. All the users hang up and call is cleared successfully for all the users | 1. Third user is added to the call successfully<br>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<br>2. Make a voice call from PBX A user 1 to Teams user<br>3. PBX A user 2 starts ringing<br>4. PBX A user 2 answers the call<br>5. Verify two way audio<br>6. PBX A user 1 hangs up the call<br>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<br>2. Make a voice call from PBX A user to Teams user<br>3. PBX B user starts ringing<br>4. PBX B user answers the call<br>5. Verify two way audio<br>6. PBX A user hangs up the call<br>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<br>2. Make a voice call from PBX A user to Teams user<br>3. PSTN user starts ringing<br>4. PSTN user answers the call<br>5. Verify two way audio<br>6. PBX A user hangs up the call<br>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<br>2. Make a voice call from PBX B user 1 to Teams user<br>3. PBX B user 2 starts ringing<br>4. PBX B user 2 answers the call<br>5. Verify two way audio<br>6. PBX B user 1 hangs up the call<br>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<br>2. Make a voice call from PBX B user to Teams user<br>3. PBX A user starts ringing<br>4. PBX A user answers the call<br>5. Verify two way audio<br>6. PBX B user hangs up the call<br>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<br>2. Make a voice call from PBX B user to Teams user<br>3. PSTN user starts ringing<br>4. PSTN user answers the call<br>5. Verify two way audio<br>6. PBX B user hangs up the call<br>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<br>2. Make a voice call from PSTN user to Teams user<br>3. PBX B user starts ringing<br>4. PBX B user answers the call<br>5. Verify two way audio<br>6. PSTN user hangs up the call<br>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<br>2. Make a voice call from PSTN user to Teams user<br>3. PBX A user starts ringing<br>4. PBX A user answers the call<br>5. Verify two way audio<br>6. PSTN user hangs up the call<br>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<br>2. Make a voice call from PSTN user 1 to Teams user<br>3. PSTN user 2 starts ringing<br>4. PSTN user 2 answers the call<br>5. Verify two way audio<br>6. PSTN user 1 hangs up the call<br>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<br>2. Make a voice call from PBX A user 1 to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX A user 2 starts ringing<br>4. PBX A user 2 answers the call<br>5. Verify two way audio<br>6. PBX A user 1 hangs up the call<br>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<br>2. Make a voice call from PBX A user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX B user starts ringing<br>4. PBX B user answers the call<br>5. Verify two way audio<br>6. PBX A user hangs up the call<br>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<br>2. Make a voice call from PBX A user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PSTN user starts ringing<br>4. PSTN user answers the call<br>5. Verify two way audio<br>6. PBX A user hangs up the call<br>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<br>2. Make a voice call from PBX B user 1 to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX B user 2 starts ringing<br>4. PBX B user 2 answers the call<br>5. Verify two way audio<br>6. PBX B user hangs up the call<br>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<br>2. Make a voice call from PBX B user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX A user starts ringing<br>4. PBX A user answers the call<br>5. Verify two way audio<br>6. PBX B user hangs up the call<br>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<br>2. Make a voice call from PBX B user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PSTN user starts ringing<br>4. PSTN user answers the call<br>5. Verify two way audio<br>6. PBX B user hangs up the call<br>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<br>2. Make a voice call from PSTN user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX B user starts ringing<br>4. PBX B user answers the call<br>5. Verify two way audio<br>6. PSTN user hangs up the call<br>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<br>2. Make a voice call from PSTN user to Teams user<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PBX A user starts ringing<br>4. PBX A user answers the call<br>5. Verify two way audio<br>6. PSTN user hangs up the call<br>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<br>3. Teams user starts ringing<br>4. Teams user does not answer the call<br>5. Call gets forwarded after the no answer timeout value is reached<br>6. PSTN user 2 starts ringing<br>4. PSTN user 2 answers the call<br>5. Verify two way audio<br>6. PSTN user 1 hangs up the call<br>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<br>2. Make a voice call from PSTN user to Teams user<br>3. Teams user, PBX A user and PBX B user starts ringing<br>4. PBX A user answers the call<br>5. Verify two way audio<br>6. PSTN user hangs up<br>7. Verify call is cleared successfully<br>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<br>2.Teams user hears Ring back Tone<br>3. PBX A user starts ringing<br>4. Verify caller ID displayed on PBX A user is Unavailable/Private/Anonymous<br>5. PBX A user answers the call<br>6. Verify two way audio<br>7. Teams user hangs up the call<br>8. Verify call is cleared successfully | 1. Teams user is able to dial an outbound call with restricted caller ID<br>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<br>2.Teams user hears Ring back Tone<br>3. PBX B user starts ringing<br>4. Verify caller ID displayed on PBX B user is Unavailable/Private/Anonymous<br>5. PBX B user answers the call<br>6. Verify two way audio<br>7. Teams user hangs up the call<br>8. Verify call is cleared successfully | 1. Teams user is able to dial an outbound call with restricted caller ID<br>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<br>2.Teams user hears Ring back Tone<br>3. PSTN user starts ringing<br>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<br>2. Call is successful with two way audio | PASSED | |

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| | | 5. PSTN user answers the call<br>6. Verify two way audio<br>7. Teams user hangs up the call<br>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<br>2. PBX A user hears Ring back Tone<br>3. Teams user starts ringing<br>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous<br>5. Teams user answers the call<br>6. Verify two way audio<br>7. PBX A user hangs up the call<br>8. Verify call is cleared successfully | 1. Teams user is able to receive an inbound call with restricted caller ID<br>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<br>2. PBX B user hears Ring back Tone<br>3. Teams user starts ringing<br>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous<br>5. Teams user answers the call<br>6. Verify two way audio<br>7. PBX B user hangs up the call<br>8. Verify call is cleared successfully | 1. Teams user is able to receive an inbound call with restricted caller ID<br>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<br>2. PSTN user hears Ring back Tone<br>3. Teams user starts ringing<br>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous<br>5. Teams user answers the call<br>6. Verify two way audio<br>7. PSTN user hangs up the call<br>8. Verify call is cleared successfully | 1. Teams user is able to receive an inbound call with restricted caller ID<br>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<br>2. Teams user does not answer the call<br>3. Allow the call to get forwarded to voicemail<br>4. PBX A user successfully leaves voicemail<br>5. Teams user receives voicemail notification<br>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<br>2. Teams user does not answer the call<br>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<br>4. PBX B user successfully leaves voicemail<br>5. Teams user receives voicemail notification<br>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<br>2. Teams user does not answer the call<br>3. Allow the call to get forwarded to voicemail<br>4. PSTN user successfully leaves voicemail<br>5. Teams user receives voicemail notification<br>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<br>2. PBX A user does not answer the call<br>3. Allow the call to get forwarded to voicemail<br>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<br>2. PBX B user does not answer the call<br>3. Allow the call to get forwarded to voicemail<br>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<br>2. PBX A returns 486 Busy<br>3. Verify Teams user gets appropriate notification or announcement and the call is cleared<br>4. Repeat steps 1 to 3 where PBX A returns 480, 404, 503 SIP responses<br>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<br>2. PBX A user starts ringing<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>6. Verify call is cleared successfully | 1. Teams user is able to call using SIP URI<br>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<br>2. PBX B user starts ringing | 1. Teams user is able to call using SIP URI<br>2. Call is connected | PASSED | |

| External ID | Title | Procedure | Expected Results | Status | Comments |
|---|---|---|---|---|---|
| | user using SIP URI | 3. PBX B user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>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<br>2. PBX A user starts ringing<br>3. PBX A user answers the call<br>4. Verify two way audio<br>5. PBX A user hangs up the call<br>6. Verify call is cleared successfully | 1. Teams user is able to call using SIP URI<br>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<br>2. PBX B user starts ringing<br>3. PBX B user answers the call<br>4. Verify two way audio<br>5. PBX B user hangs up the call<br>6. Verify call is cleared successfully | 1. Teams user is able to call using SIP URI<br>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<br>2. Teams user hears Ring back Tone<br>3. Skype for Business user answers the call<br>4. Verify two way audio<br>5. Teams user hangs up the call<br>6. Verify call is cleared successfully<br>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<br>2. Skype for Business user hears Ring back Tone<br>3. Teams user answers the call<br>4. Verify two way audio<br>5. Skype for Business user hangs up the call<br>6. Verify call is cleared successfully<br>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<br>2. Make a voice call from Teams user to Skype for Business user<br>3. Teams user hears Ring back Tone<br>4. Skype for Business user answers the call<br>5. Verify two way audio<br>6. Teams user hangs up the call<br>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<br>2. Make a voice call from Skype for Business user to Teams user<br>3. Skype for Business user hears Ring back Tone<br>4. Teams user answers the call<br>5. Verify two way audio<br>6. Skype for Business user hangs up the call<br>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)<br>2. Verify call is successful<br>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<br>2. Teams user 1 joins the meeting using the Join button<br>3. Teams user 2 joins the meeting using the dial-in conferencing number<br>4. Verify Teams users are able to join the meeting successfully<br>5. Verify all three users are able to hear | | PASSED | |

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