Crestron UC-PHONE and UC-PHONE-PLUS

Connecting Microsoft Teams Direct Routing using AudioCodes Mediant Virtual Edition (VE) and Cisco UCM 11.5

September 2019

Document History

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1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring **Crestron UC-PHONE and UC-PHONE-PLUS with Microsoft Teams Direct Routing using AudioCodes Mediant VE SBC and Cisco UCM 11.5 as Customer PBX.**

1.1 Crestron UC-PHONE and UC-PHONE-PLUS

The Crestron UC-PHONE and UC-PHONE-PLUS phones are designed for use with the Microsoft Teams intelligent communications platform. They enable superior voice calling and full-duplex hands-free conferencing in a stylish desktop package. A consistent user experience at every desk, workstation, and meeting space is provided via the familiar and intuitive Microsoft Teams touch screen UI, affording simple operation with comprehensive call and contact management features, built-in calendaring, and one-touch meeting joins.

The Crestron UC-PHONE and UC-PHONE-PLUS desk phones install easily and connect securely, with IoT cloud based provisioning and management via the Crestron XiO Cloud™ service. They work natively with any Microsoft Teams account for a streamlined deployment on any enterprise or SMB network.

1.2 tekVizion Labs

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

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration

- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

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

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

For more information on tekVizion and its practice areas, please visit tekVizion Labs website at www.tekVizion.com

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representation of Crestron UC-PHONE and UC-PHONE-PLUS connected O365 Cloud with Microsoft Teams Direct Routing to Cisco UCM 11.5 environment using AudioCodes Mediant VE SBC and PSTN Gateway for PSTN connectivity. Media bypass enables Configured teams side used in this topology.

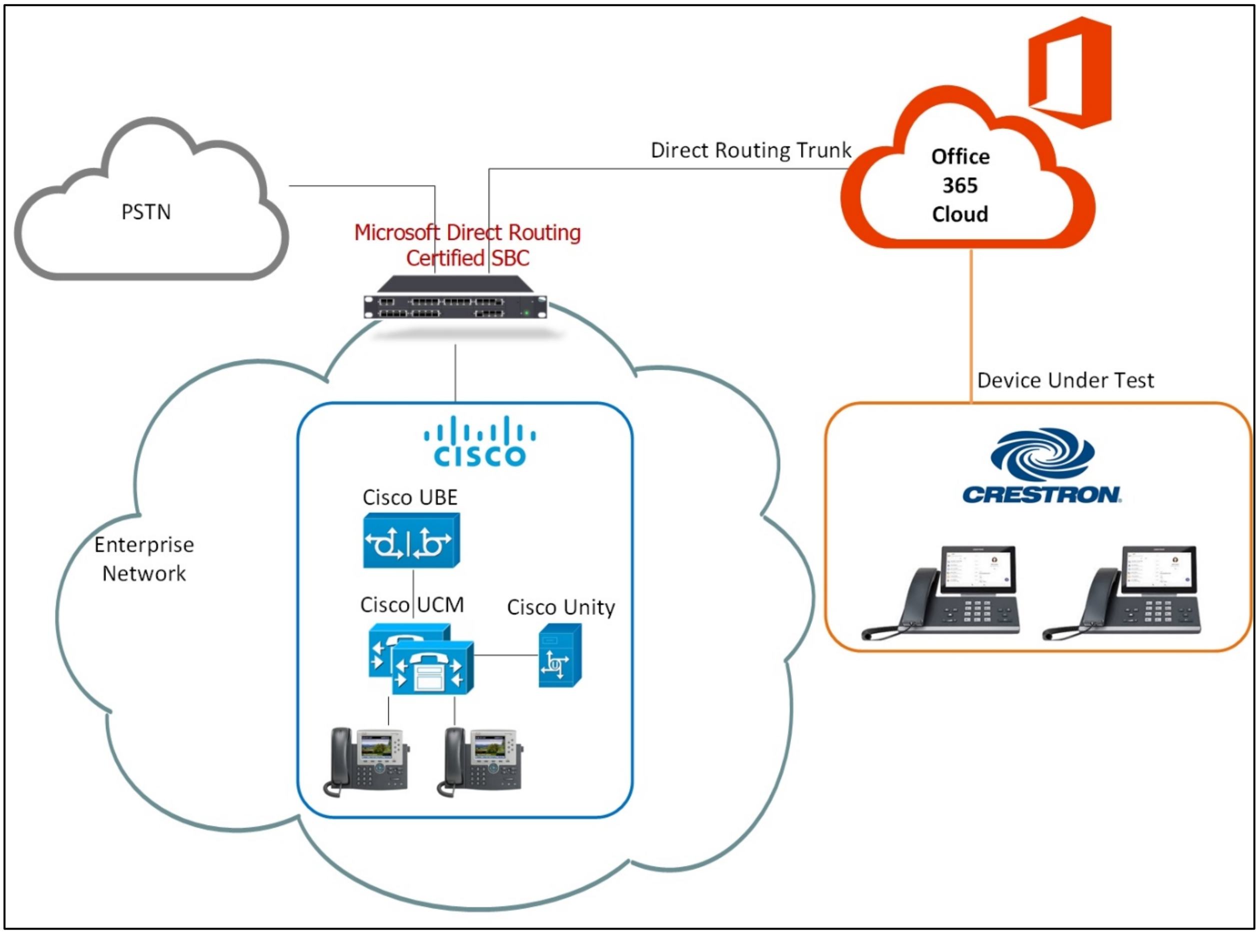

Figure 1 Network Topology

Numbering Plan

- Cisco UCM users are configured with 4 digit extension 65XX
- Teams users are configured with E164 numbers +197259809XX

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.

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
- 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
- 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)
- Call Failover

3.2 Caveats and Limitations

- Direct Routing supports call escalation to an adhoc conference without Audioconferencing license. However the UC-PHONE-PLUS and UC-PHONE desk phones could not add a user into conference without Audio Conferencing license.
- The UC-PHONE-PLUS desk phone is unable to resume a held call using soft-key, 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 and AudioCodes** for SIP Trunking with **Microsoft Teams Direct Routing.**

Table 1 – PBX Configuration Steps

Steps	Description	Reference
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.

Table 2 – IP Addresses

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

4.3 Microsoft Teams Configuration

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

4.3.1 Teams User Configuration

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

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

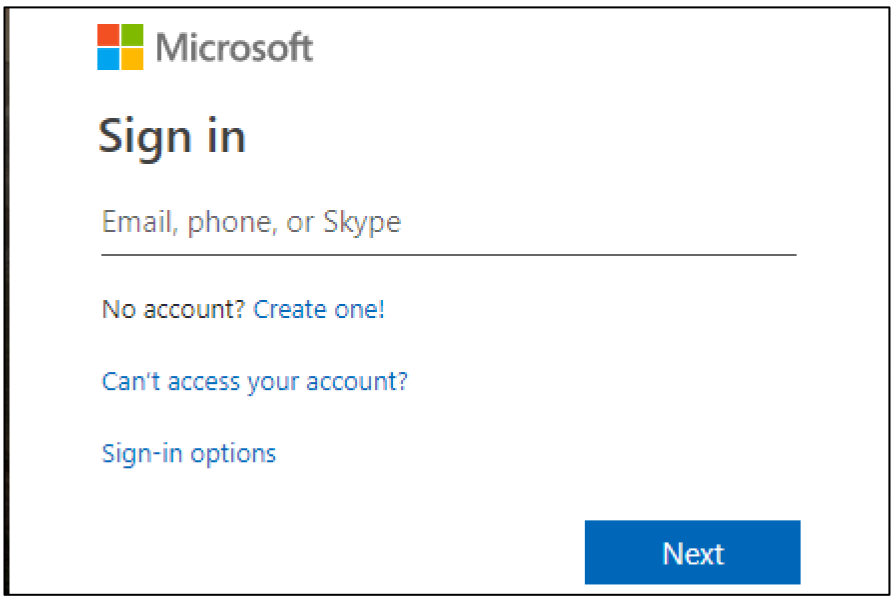

Figure 2: Office 365 Portal Login

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

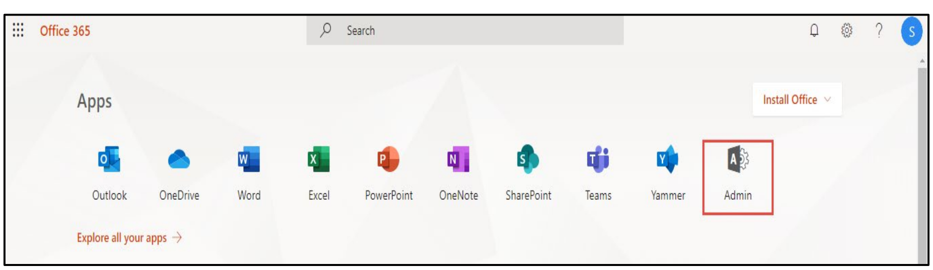

Figure 3: Office 365 Portal Login

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

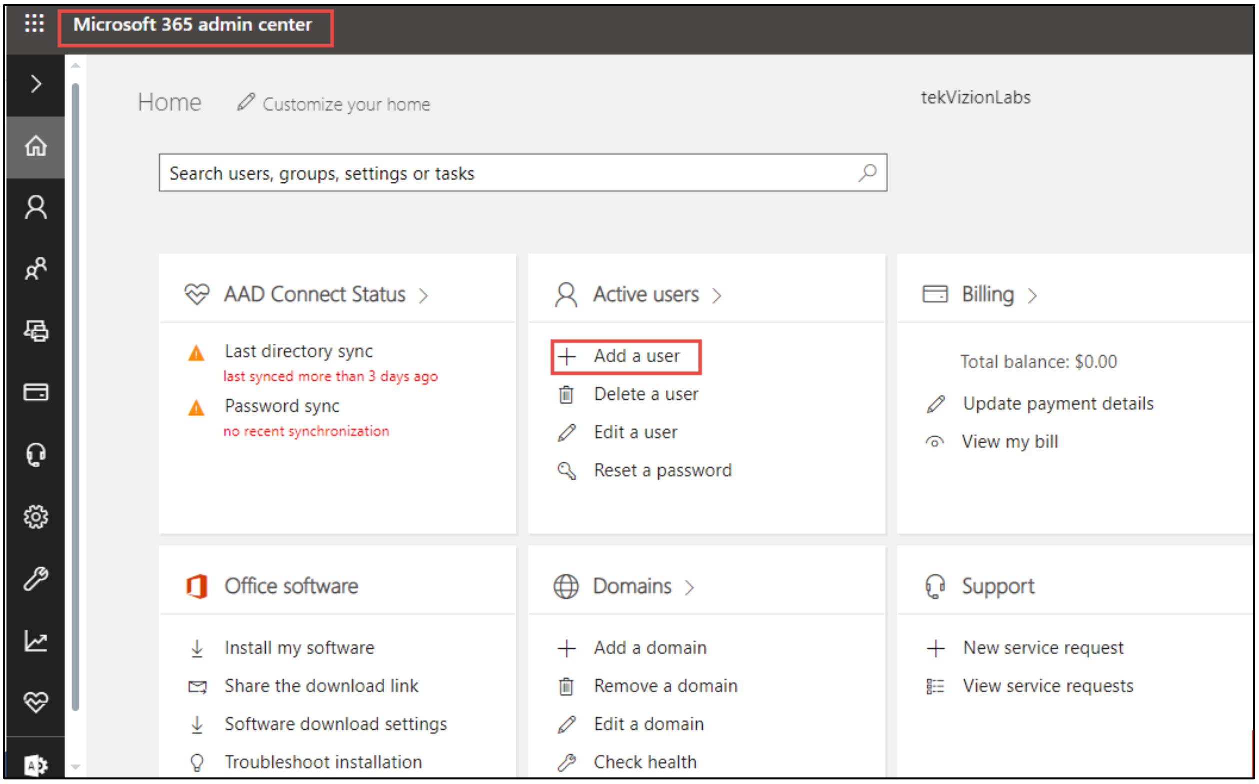

Figure 4: Teams User Creation

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

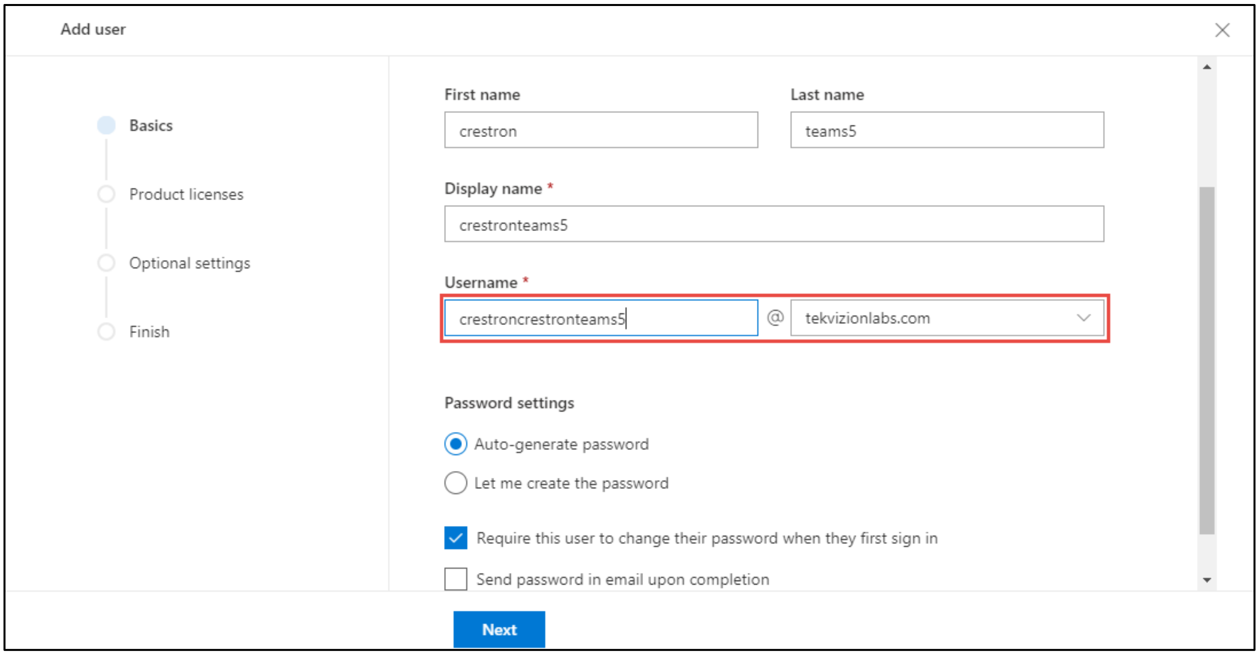

Figure 5: Teams User Creation – Contd.

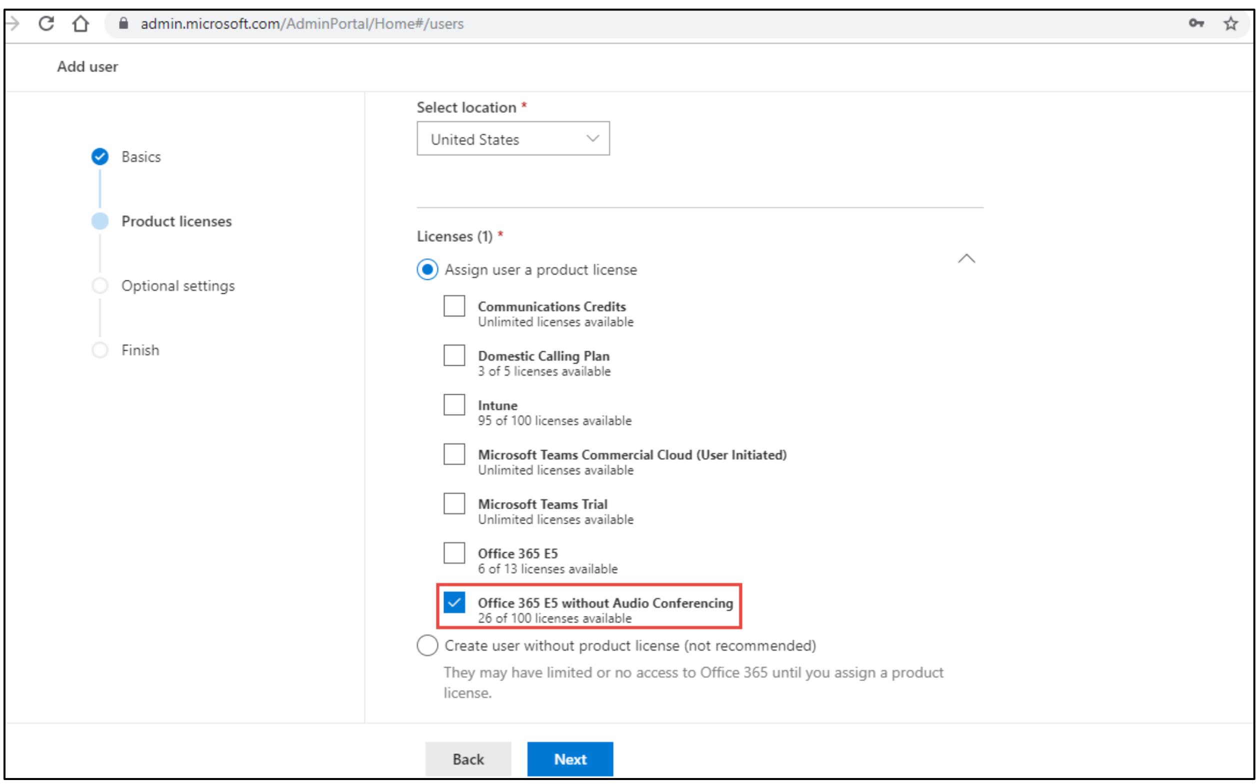

Figure 6: Teams User Creation – Contd.

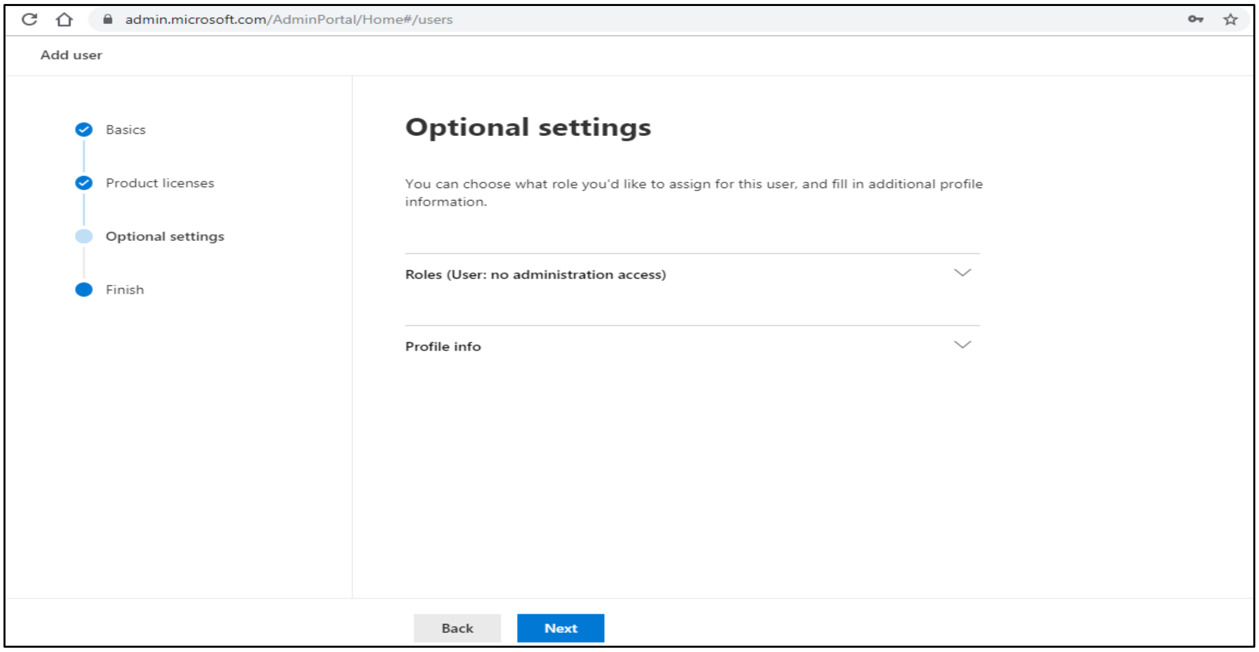

Figure 7: Teams User Creation – Contd.

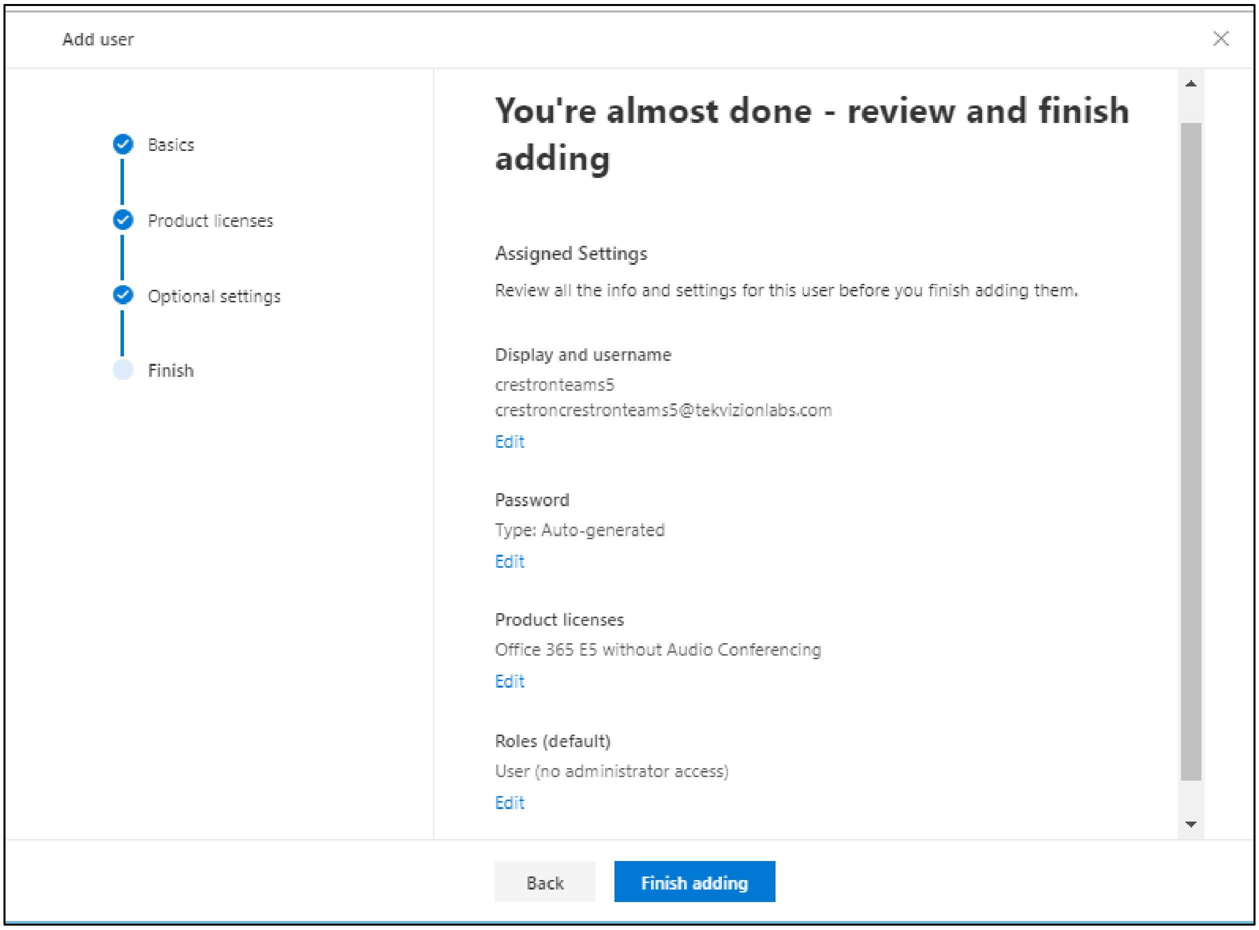

Figure 8: Teams User Creation – Contd.

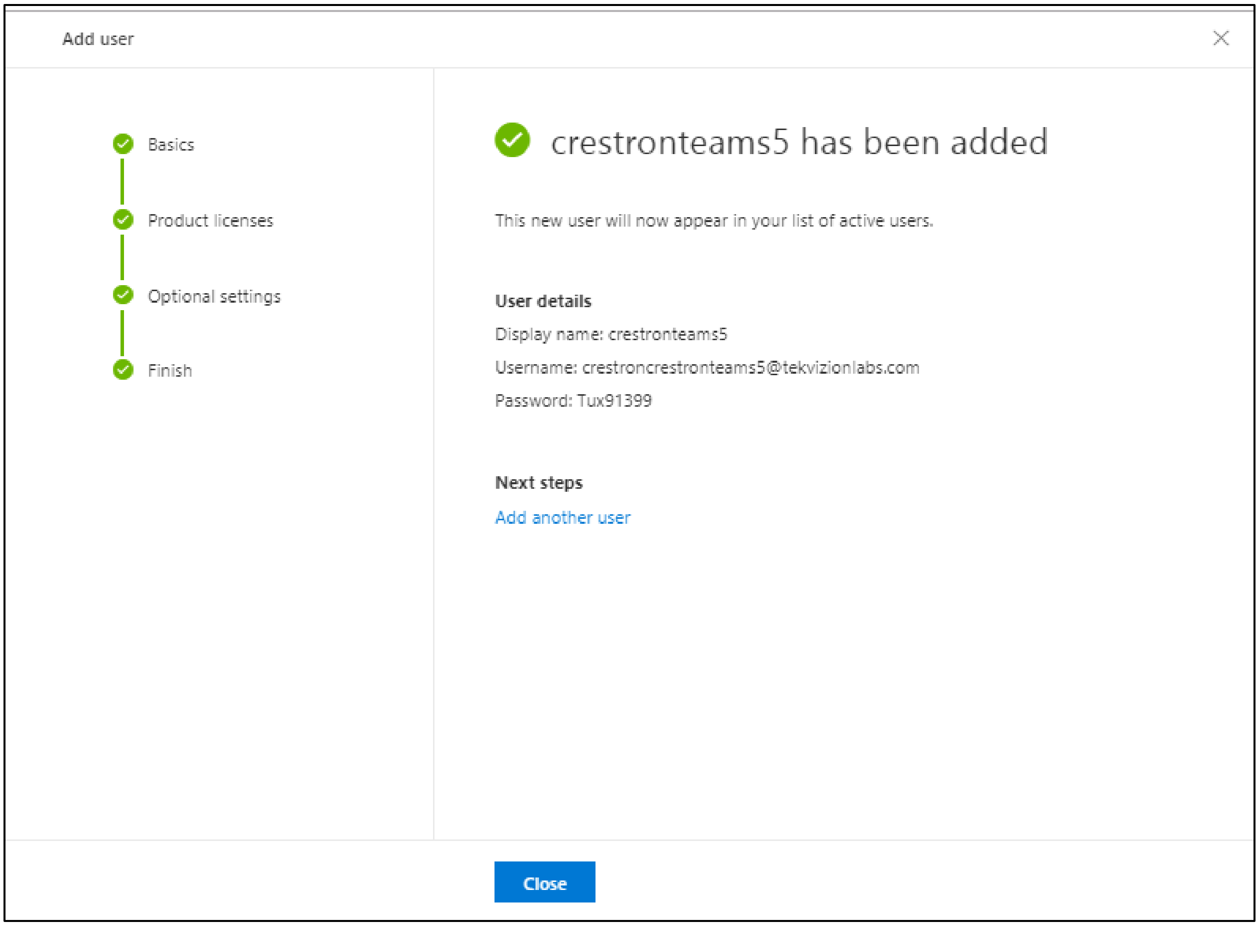

Figure 9: Teams User Creation – Contd.

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

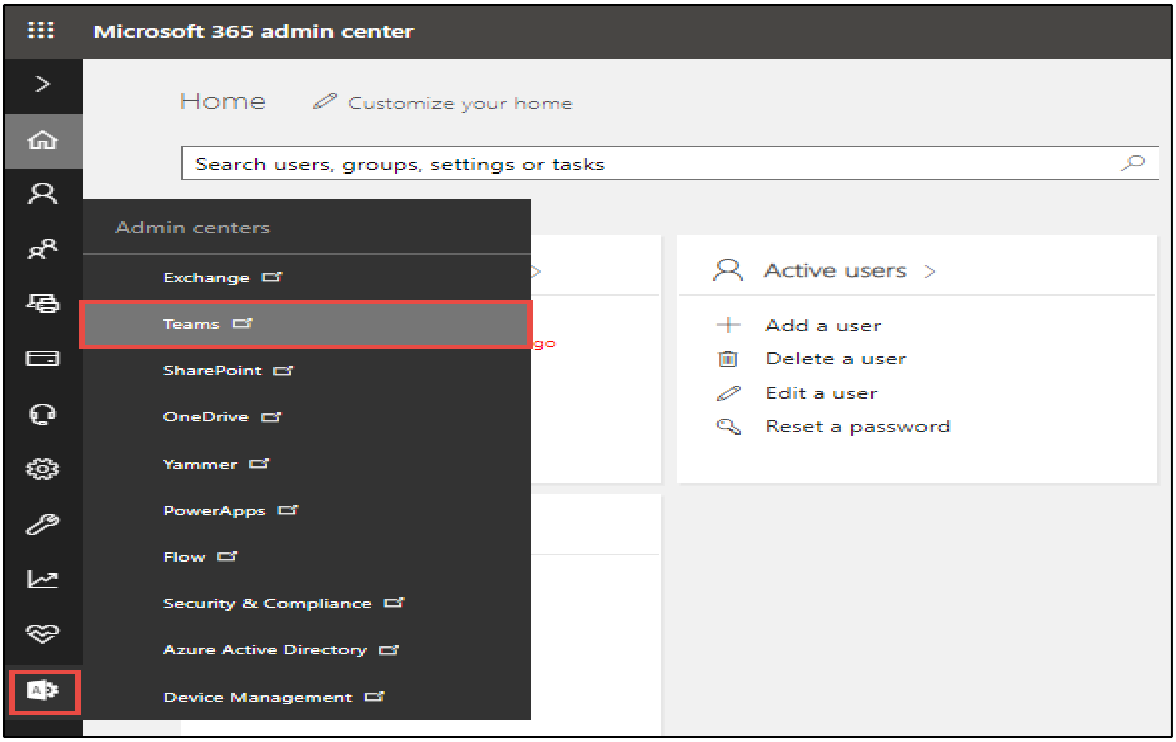

Figure 10: Microsoft O365 admin

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

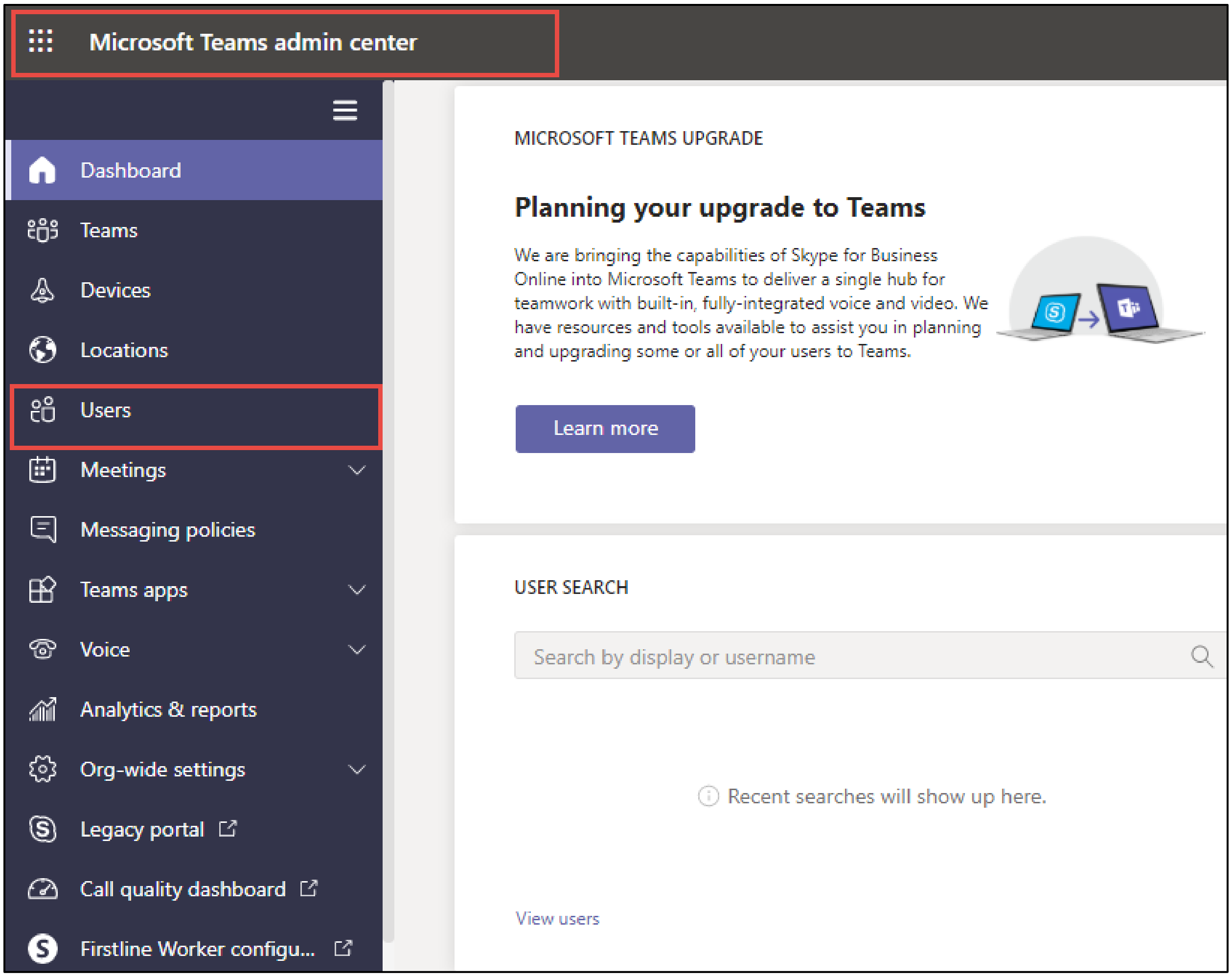

Figure 11: Microsoft O365 admin

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

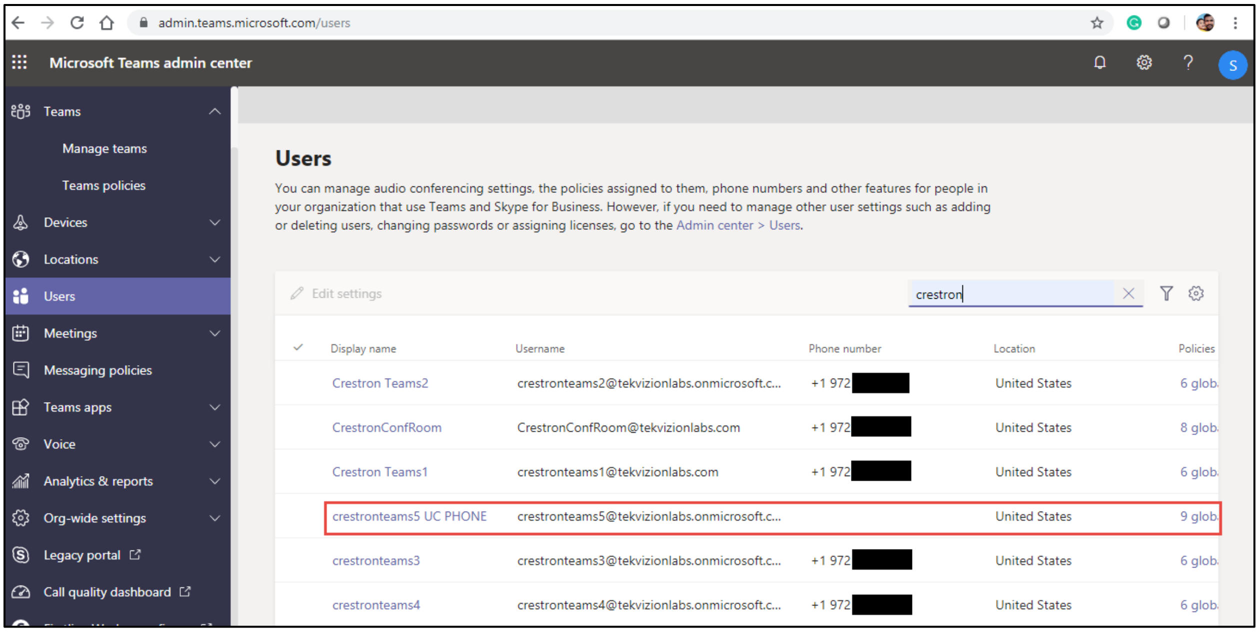

Figure 12: Microsoft O365 admin

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

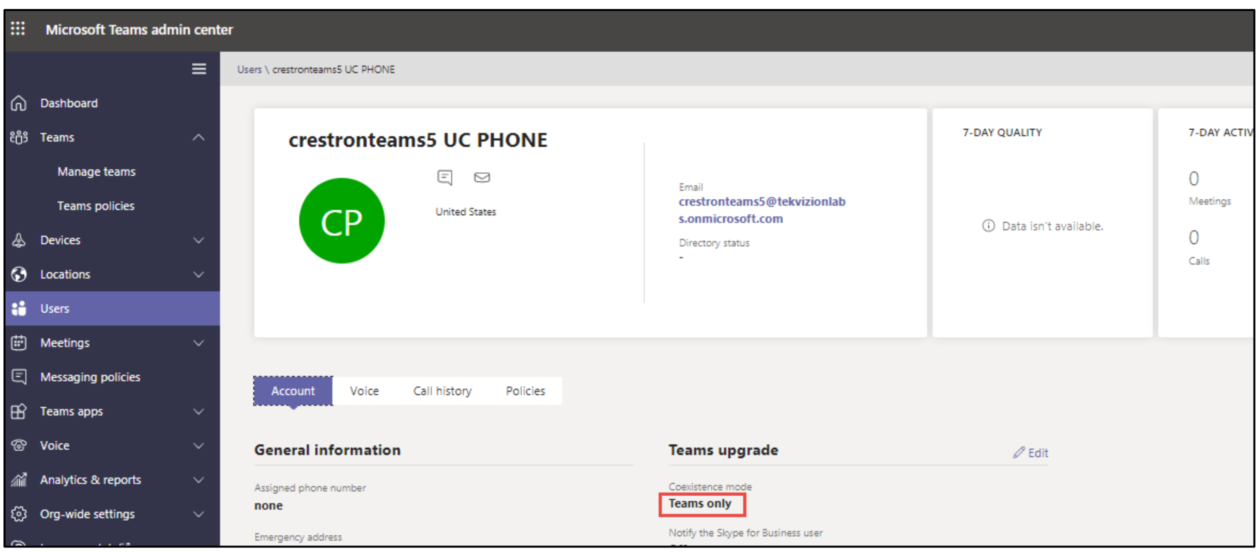

Figure 13: Teams User

4.3.2 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.

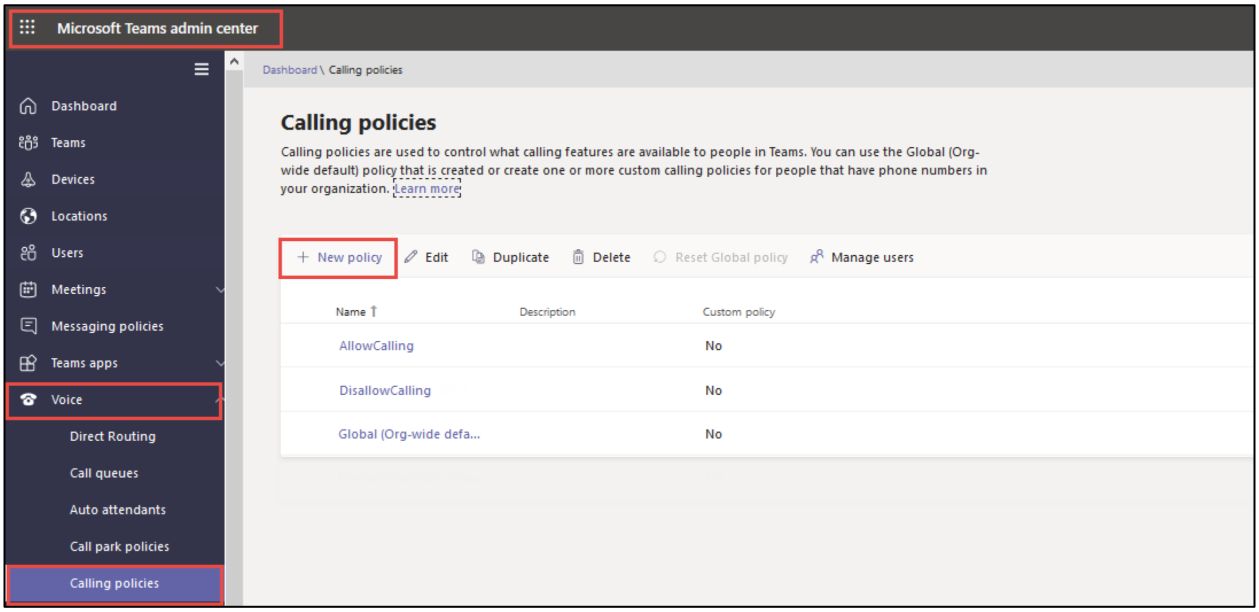

Figure 14 – Calling Policy

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

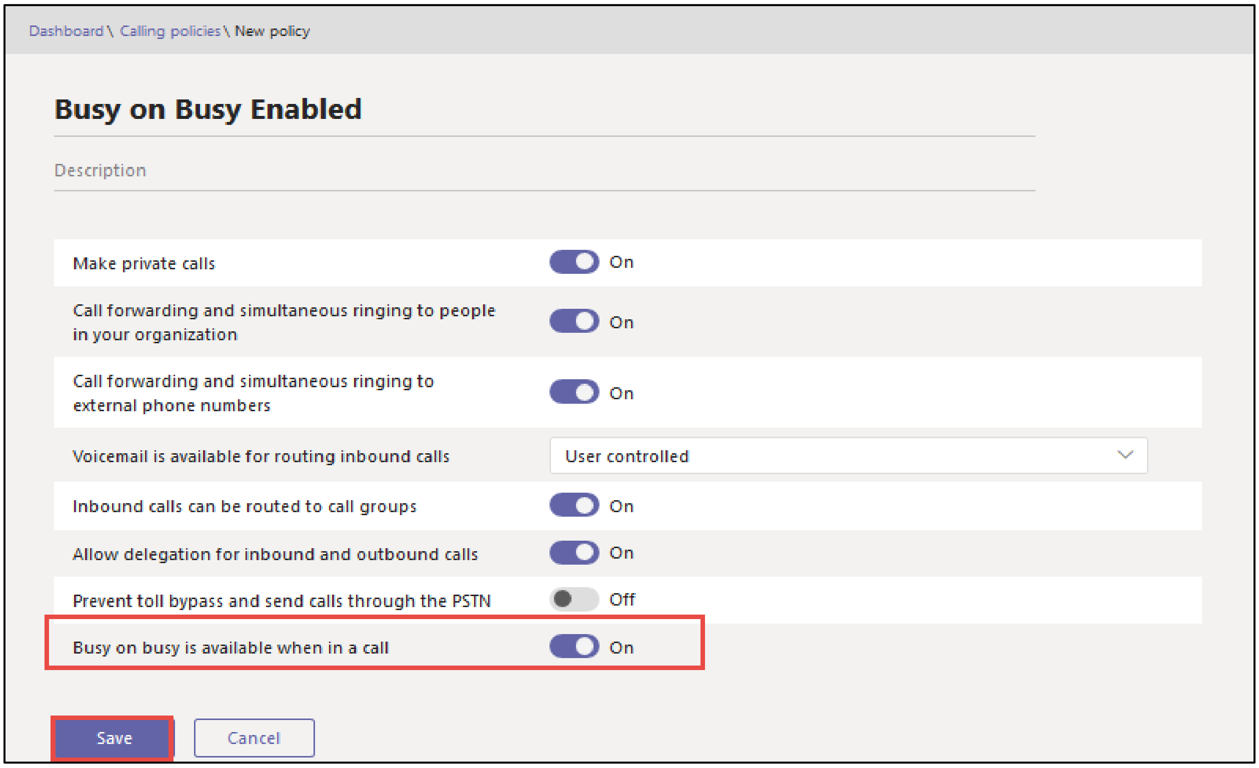

Figure 15 – Calling Policy

4.3.3 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

4.3.4 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 16 – Online PSTN Gateway

4.3.5 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 17 - Microsoft Teams - Online PSTN usage reference

4.3.6 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 18 - Microsoft Teams - Online PSTN Voice Route reference

4.3.7 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 19 - Microsoft Teams - Online Voice Route Policy

4.3.8 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 20 - Microsoft Teams *- Online Voice Route Policy to User*

4.3.9 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 21 - Microsoft Teams – Configure Tenant Dial Plan

4.3.10 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 22 - Microsoft Teams – Normalization Rule

4.3.11 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 23 - Microsoft Teams – Normalization Rule to tenant dial plan

4.3.12 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 24 - Microsoft Teams – tenant dial plan to user

4.3.13 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 25 – 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.

4.4 AudioCodes VE SBC Configuration

4.4.1 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.

4.4.2 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.

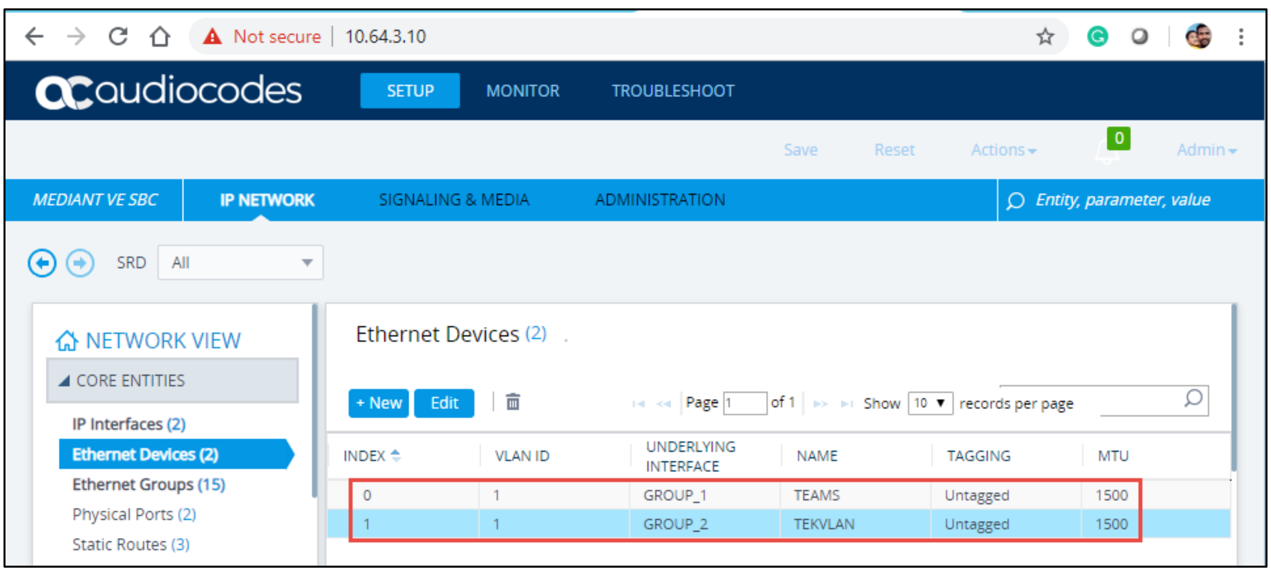

Figure 26 – Ethernet Devices

4.4.3 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:

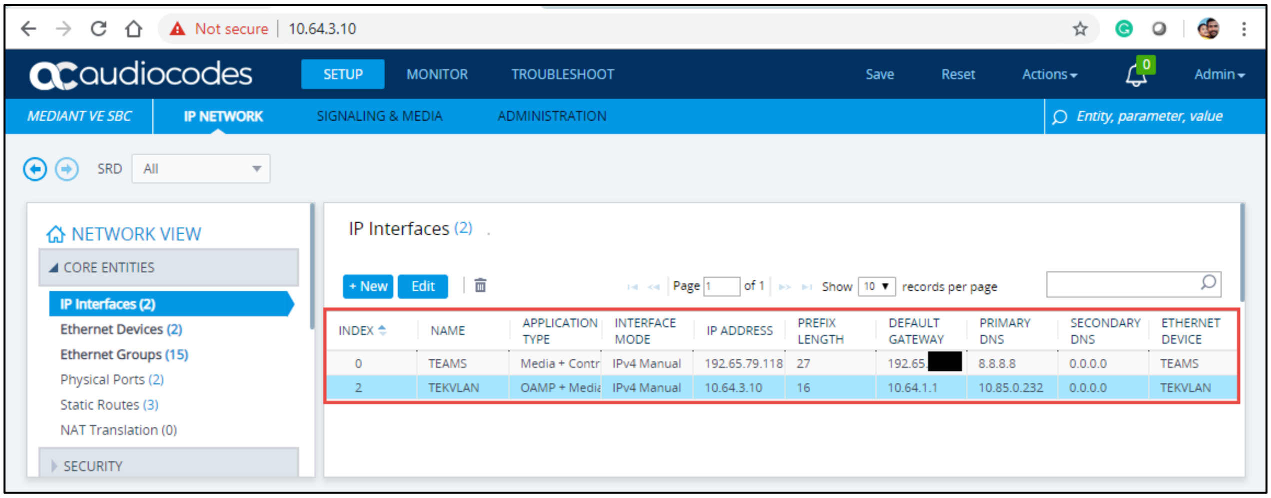

Figure 27 – IP interface Devices

IP interface TEAMS

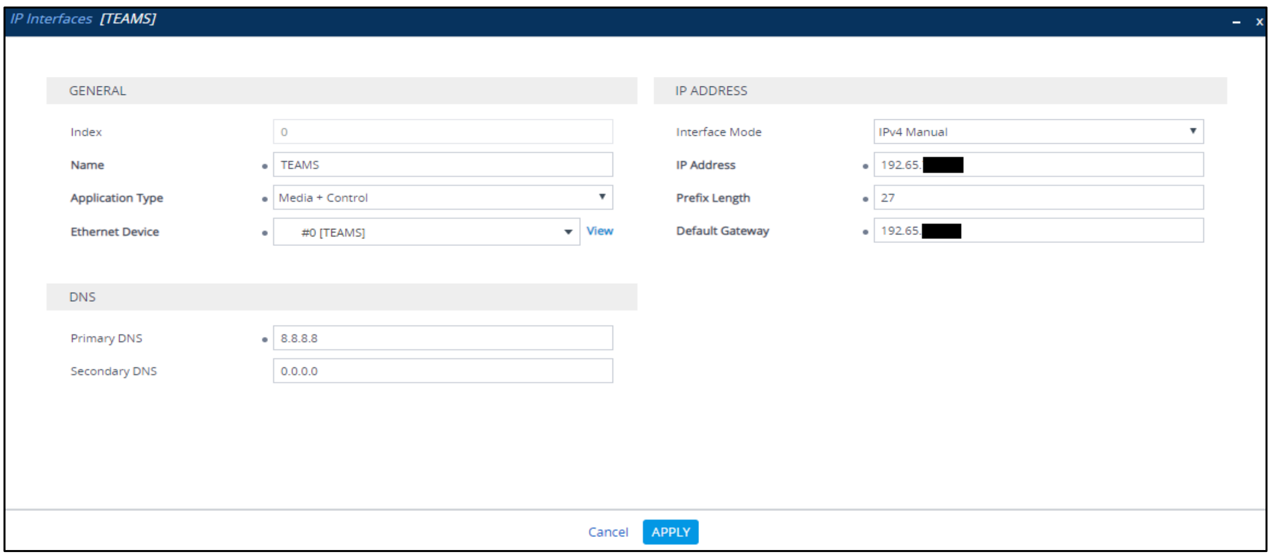

Figure 28 – IP interface Devices

IP Interfaces – TEKVLAN

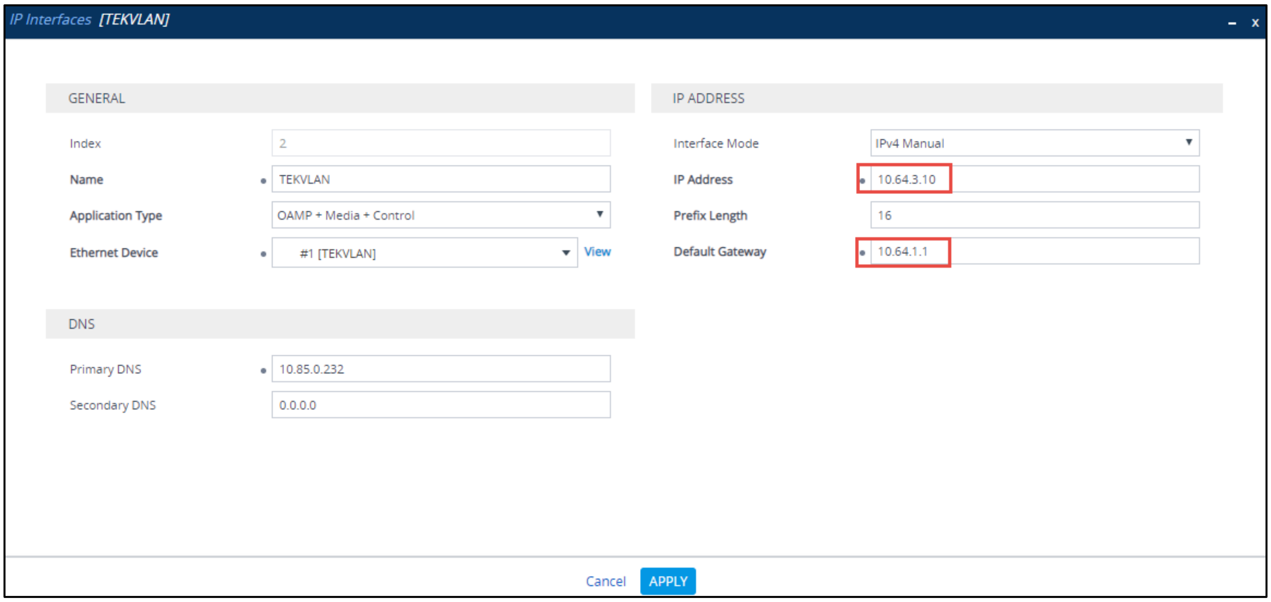

Figure 29 – IP interface Devices

4.4.4 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

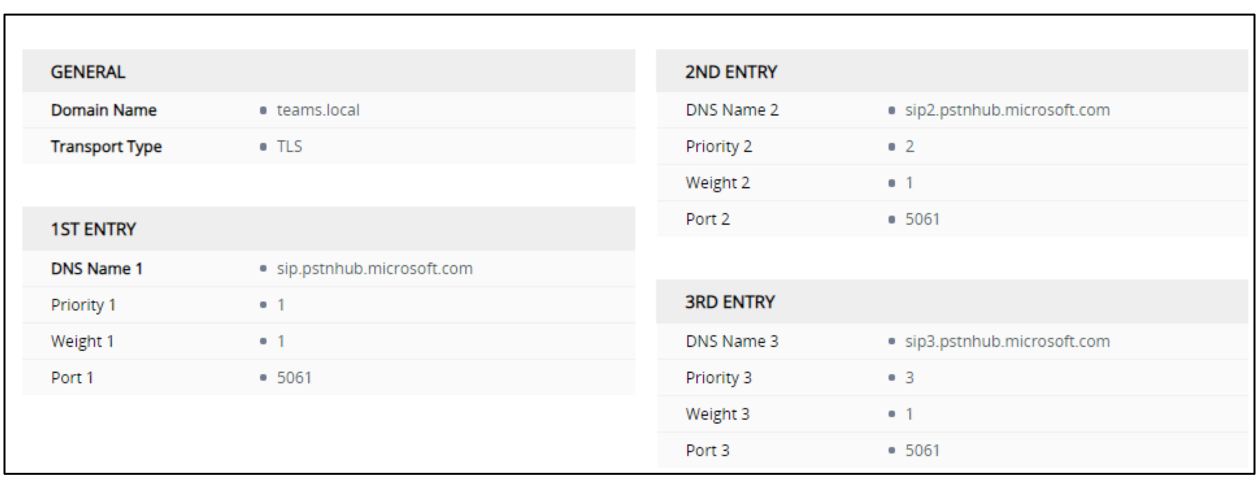

Figure 30 – DNS SRV Records

4.4.5 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

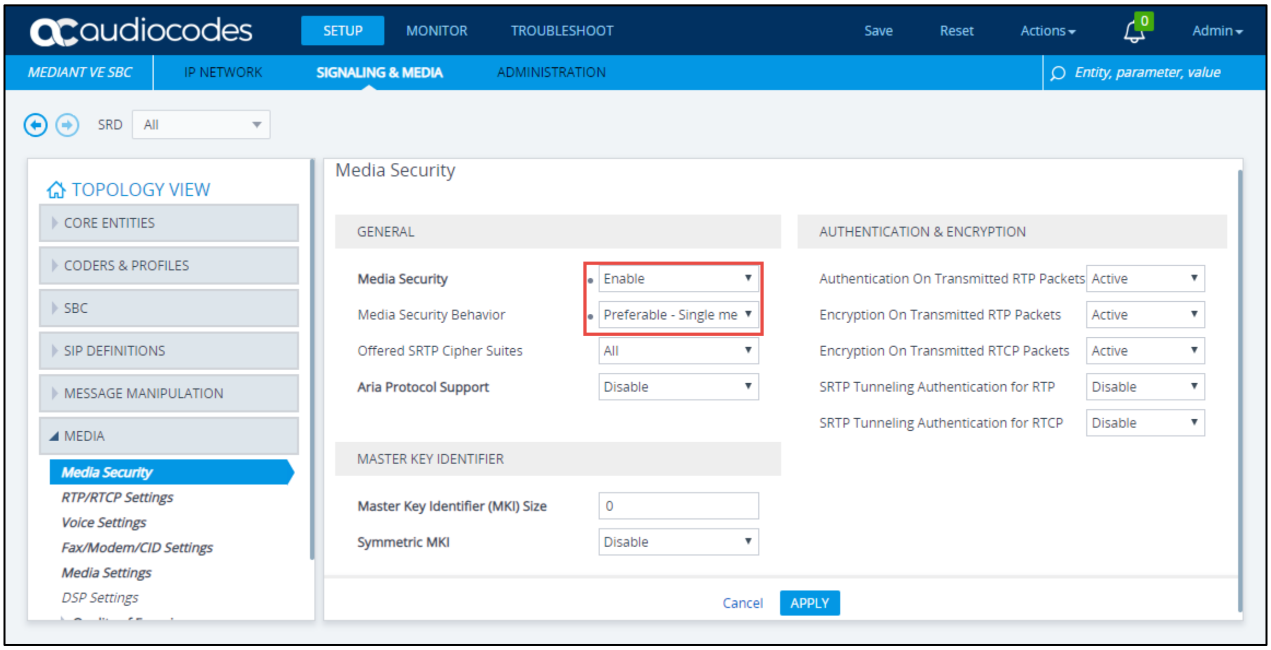

Figure 31 – Media Security

4.4.6 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.

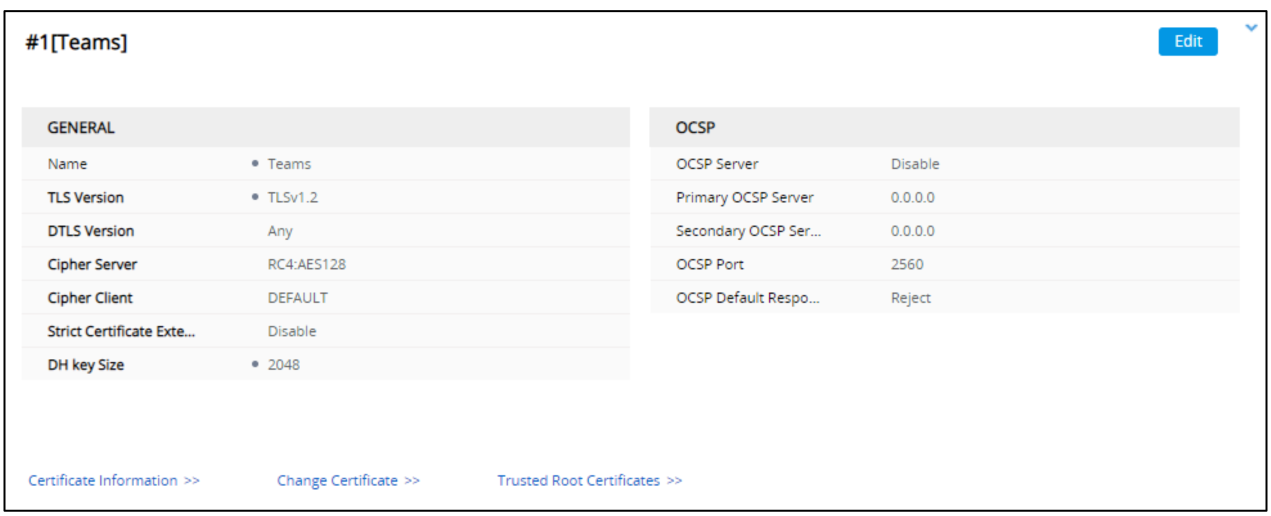

Figure 32 – Teams TLS

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.

4.4.7 Configure Media Realms

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

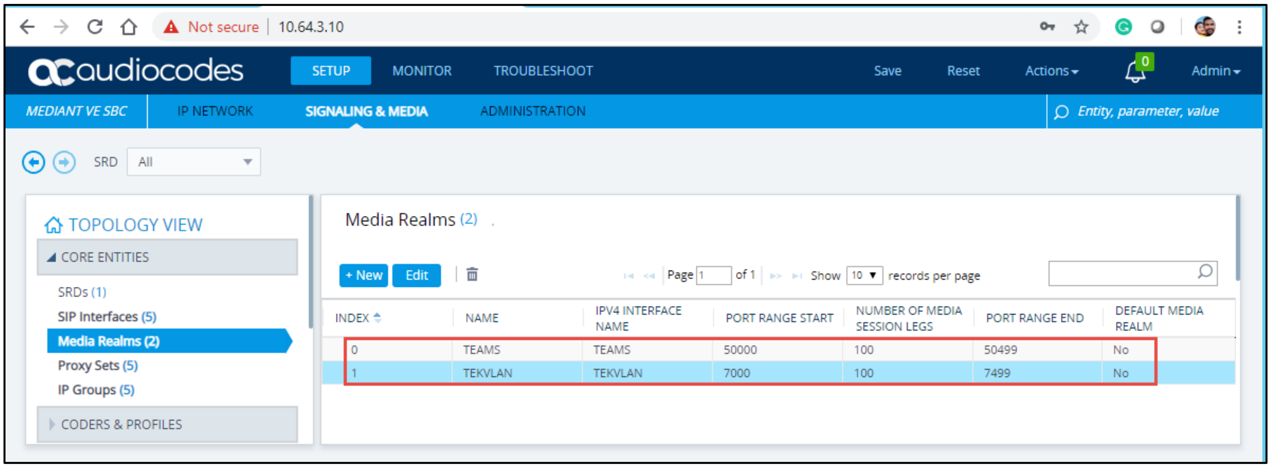

Figure 33 – Media Realms

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

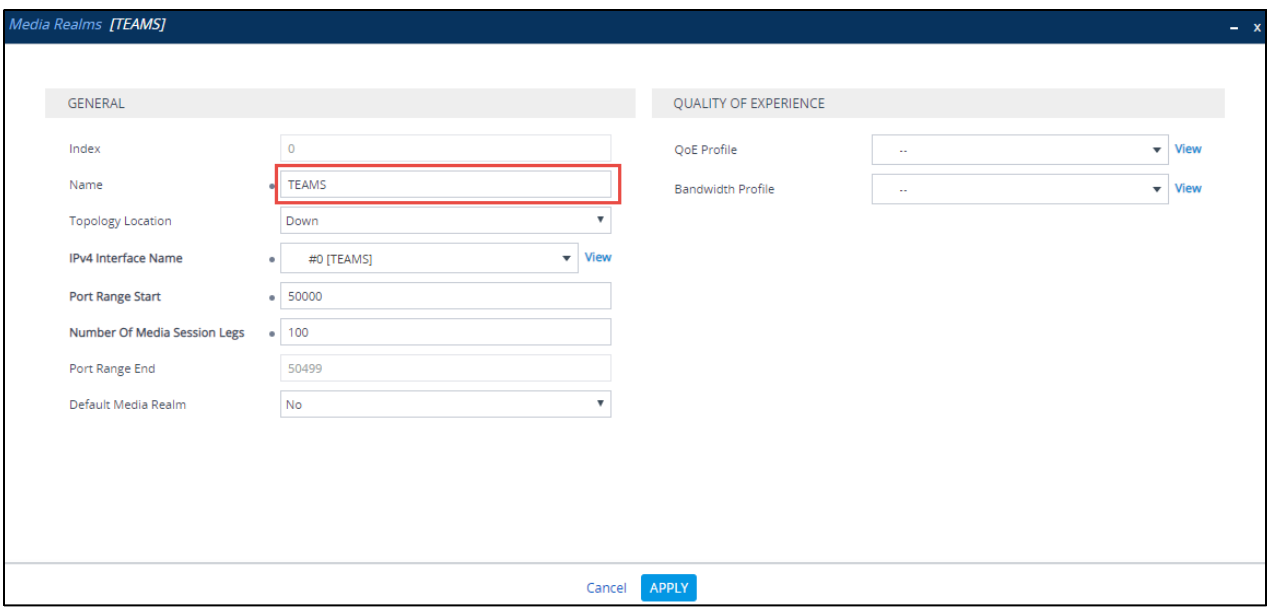

Figure 34 – Teams

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

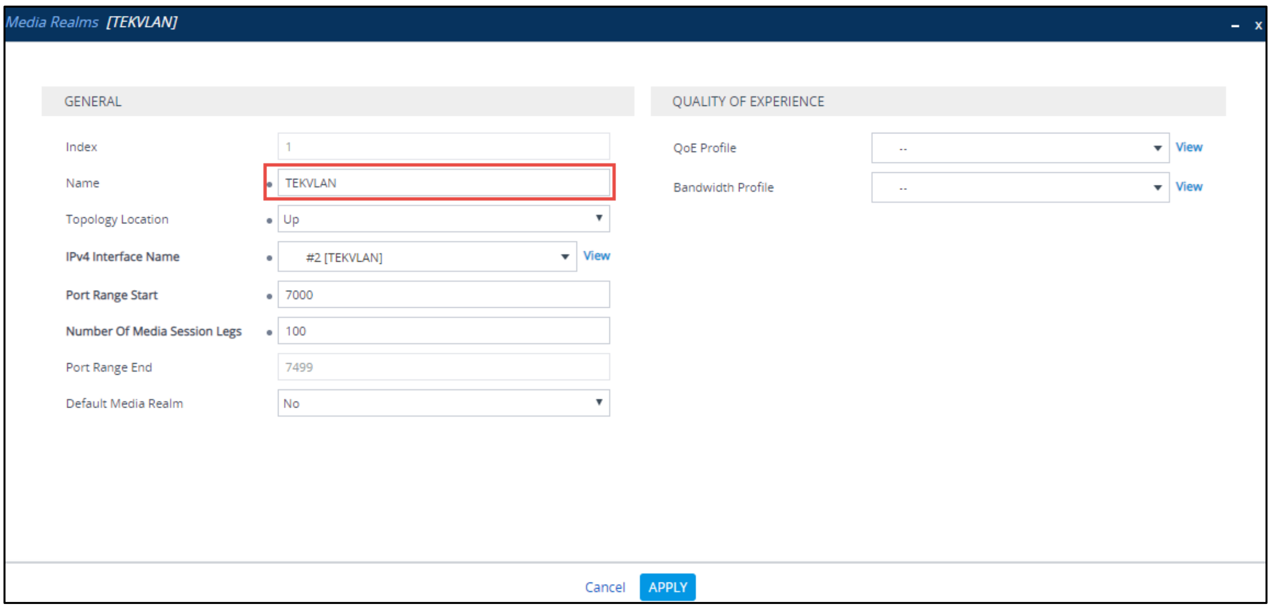

Figure 35 – LAN LAB

4.4.8 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.

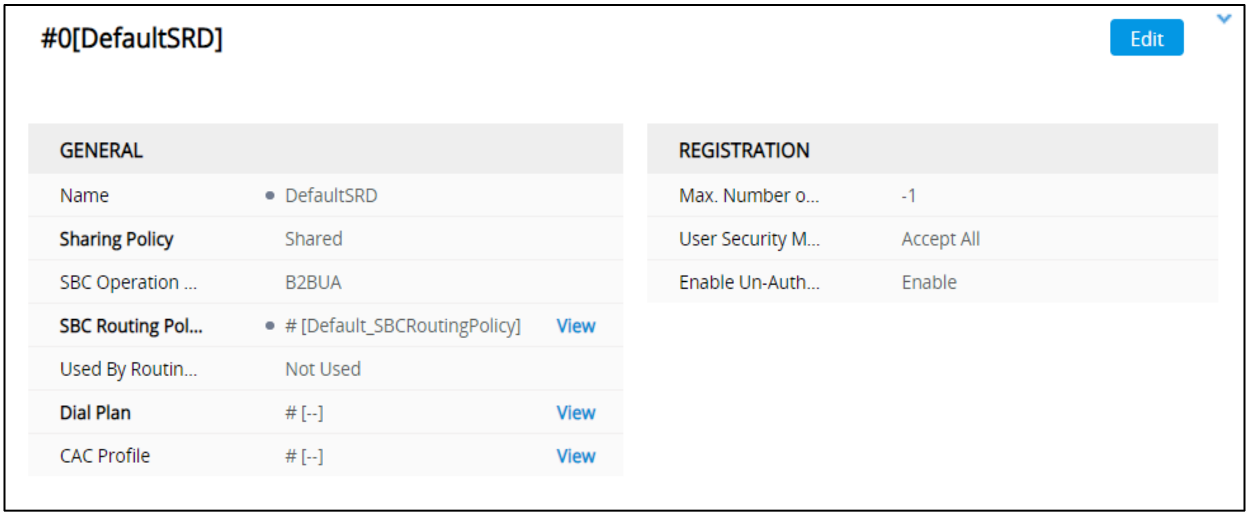

Figure 36 – Default SRD

4.4.9 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.

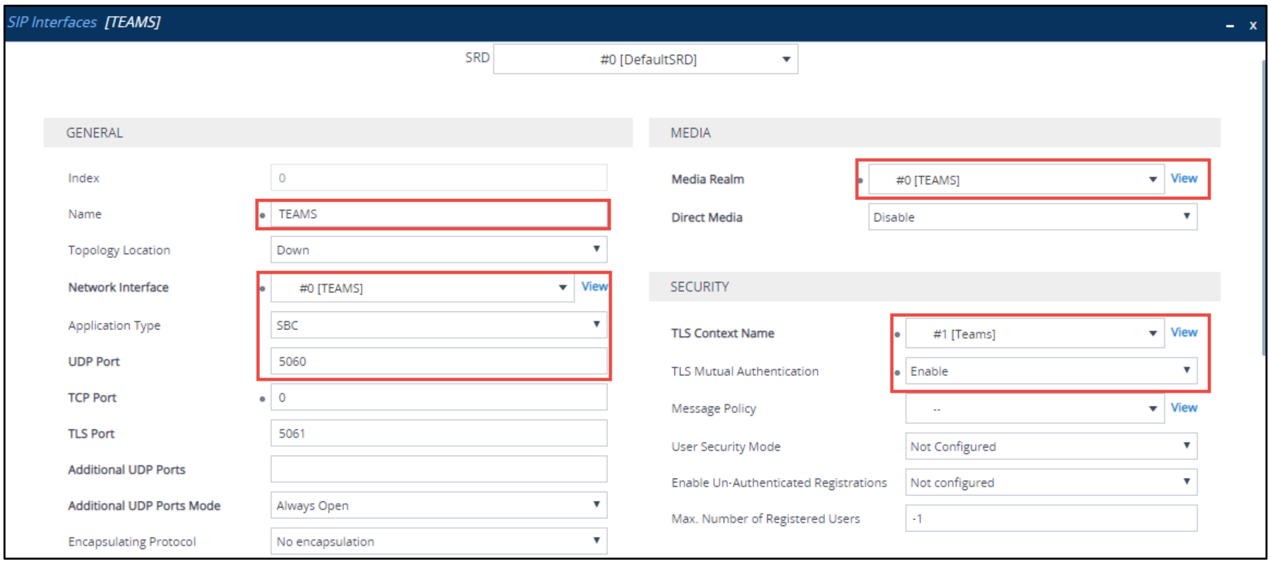

Figure 37 – Teams

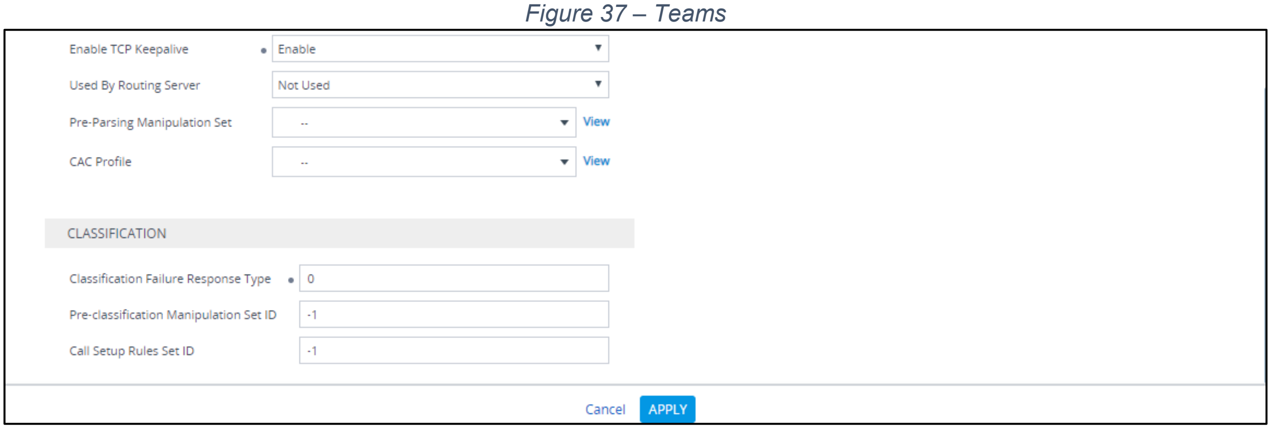

Figure 38 – Teams

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

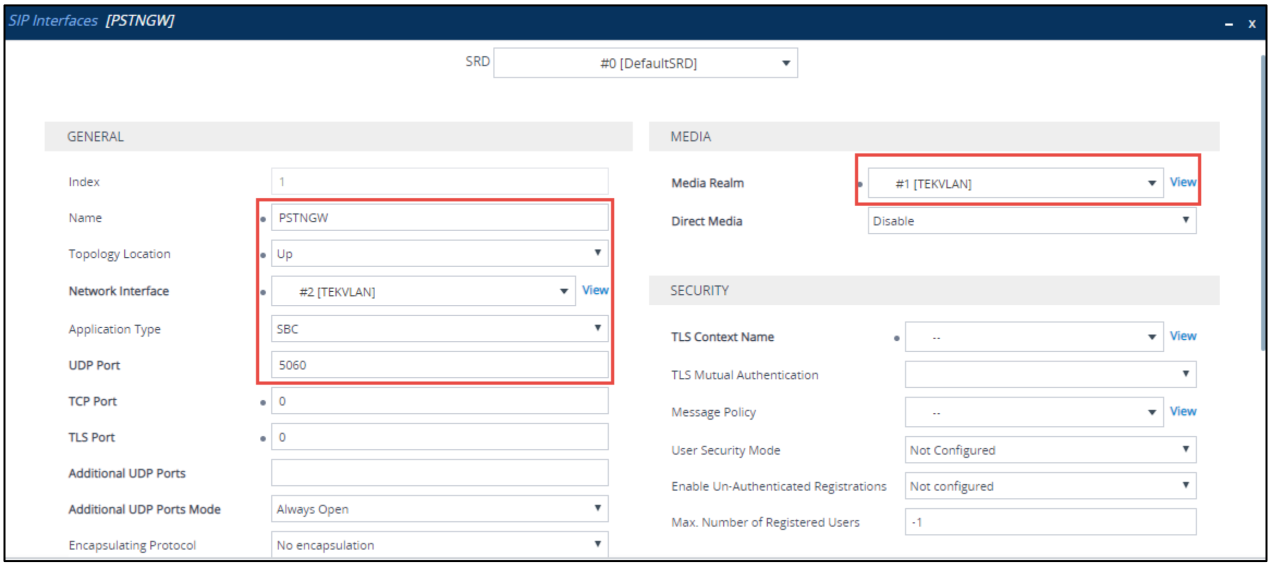

Figure 39 – PSTN

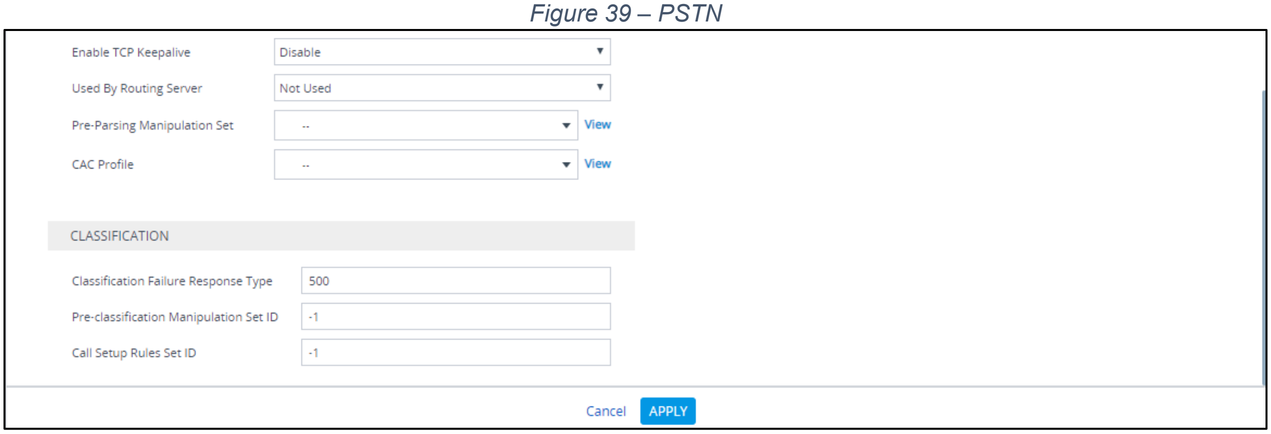

Figure 40 – PSTN

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

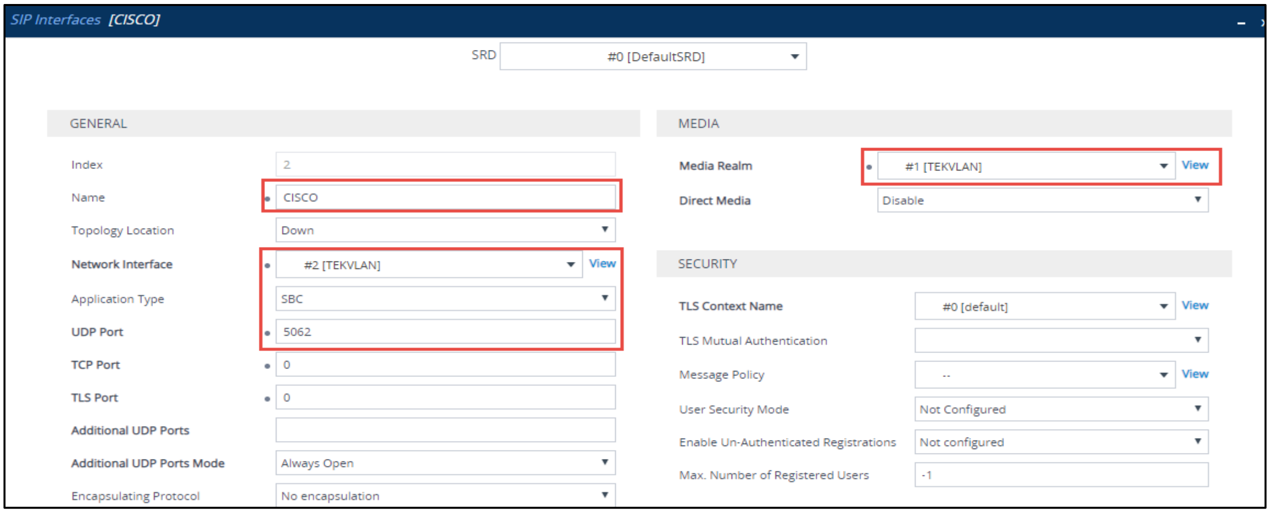

Figure 41 – Cisco

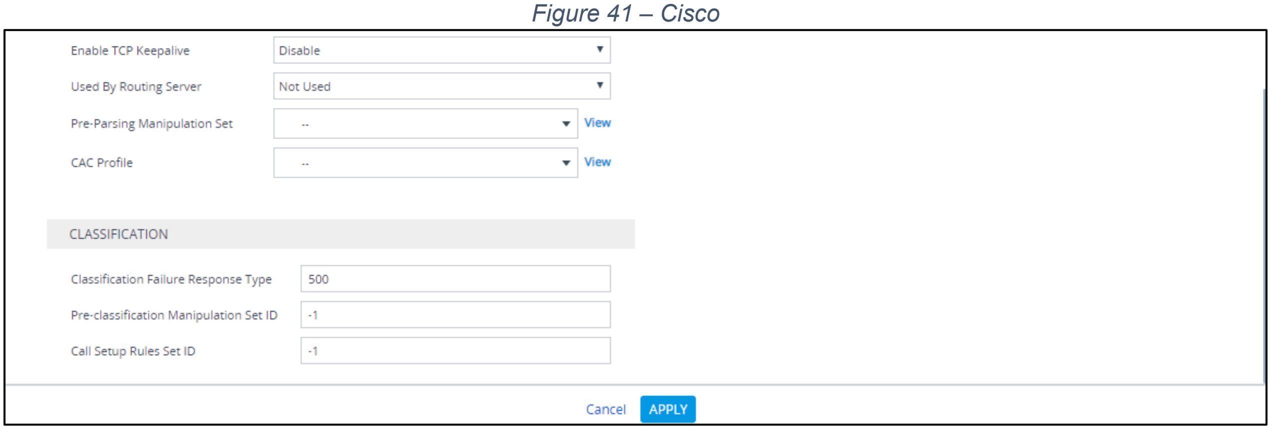

Figure 42 – Cisco

4.4.10 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.

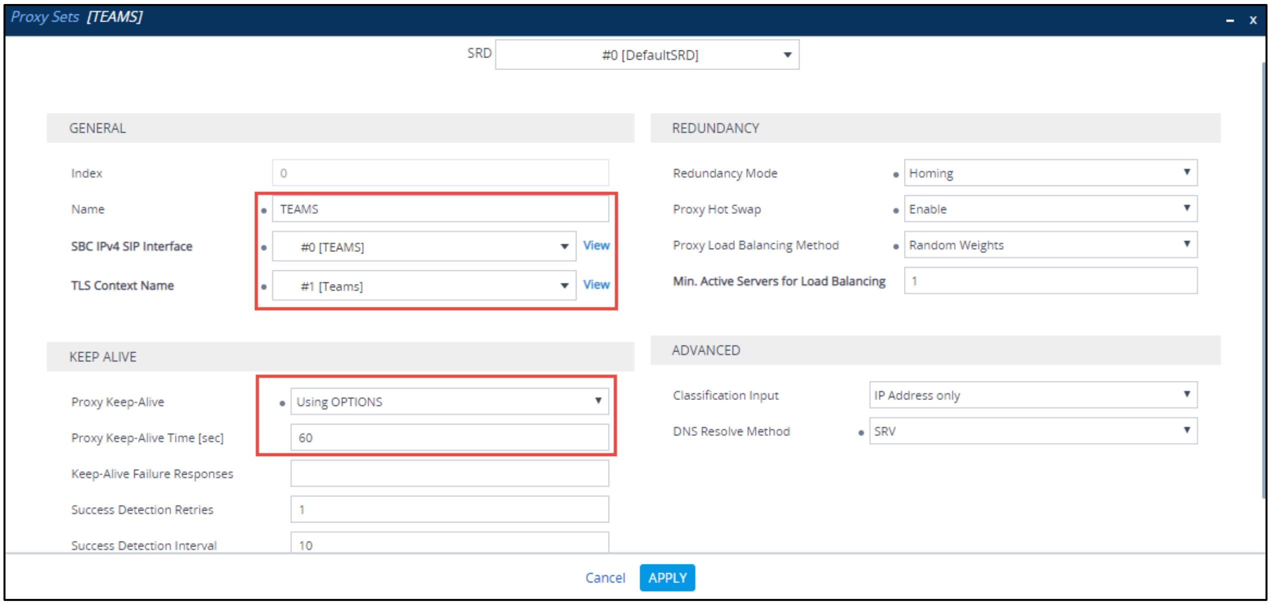

Figure 43 – Teams

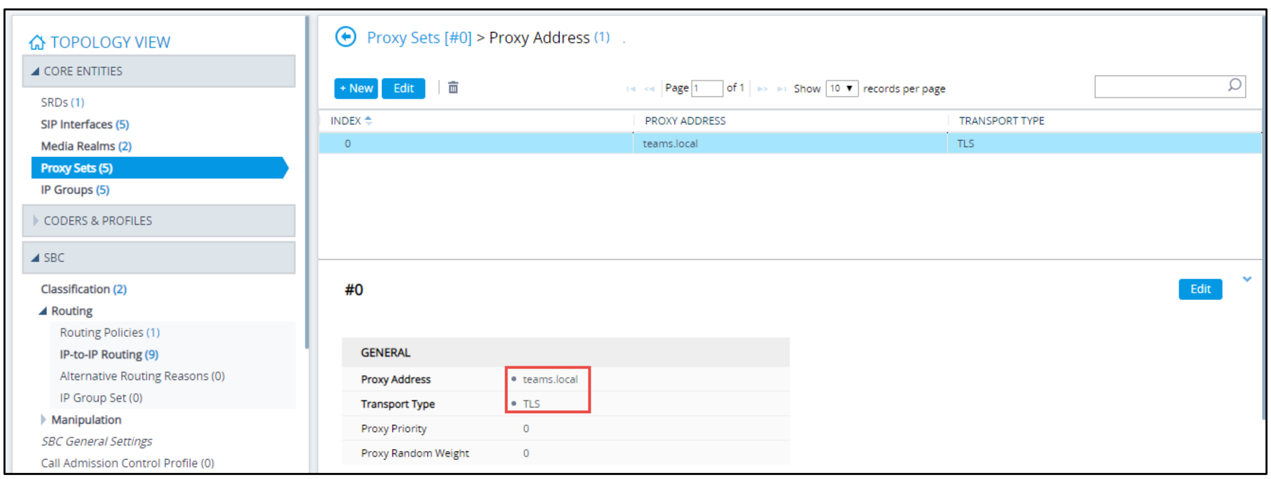

Figure 44 – Teams

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

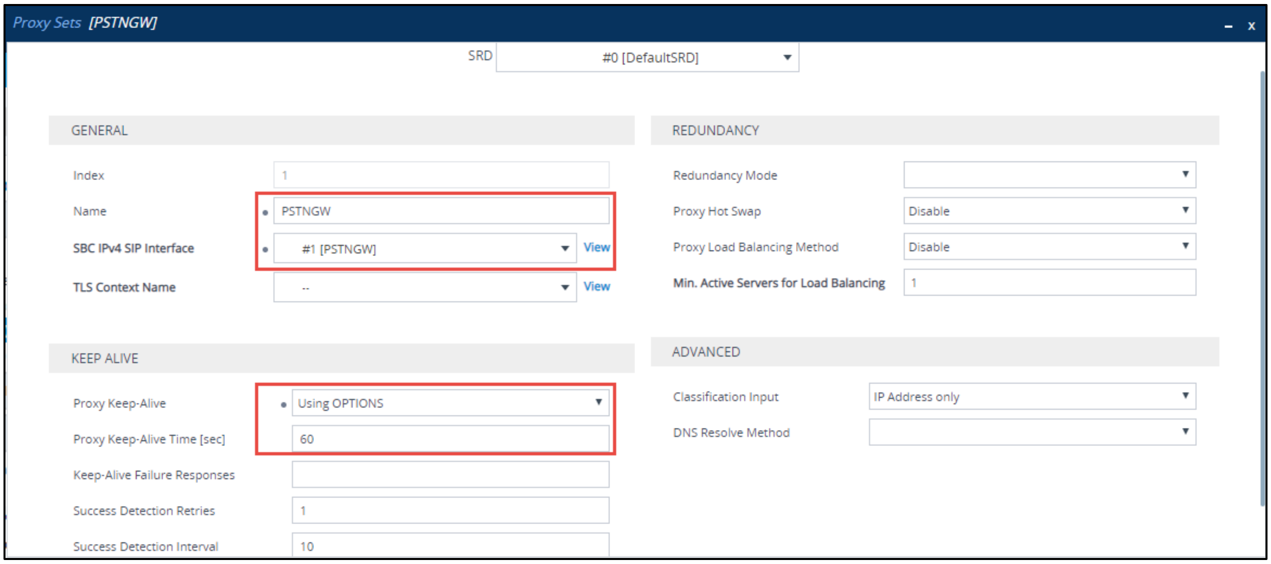

Figure 45 – PSTN Gateway

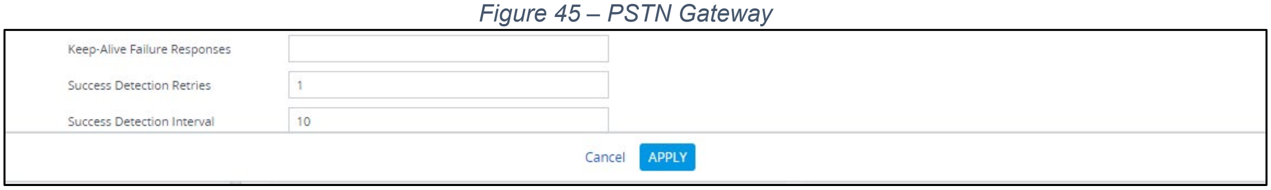

Figure 46 – PSTN Gateway

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

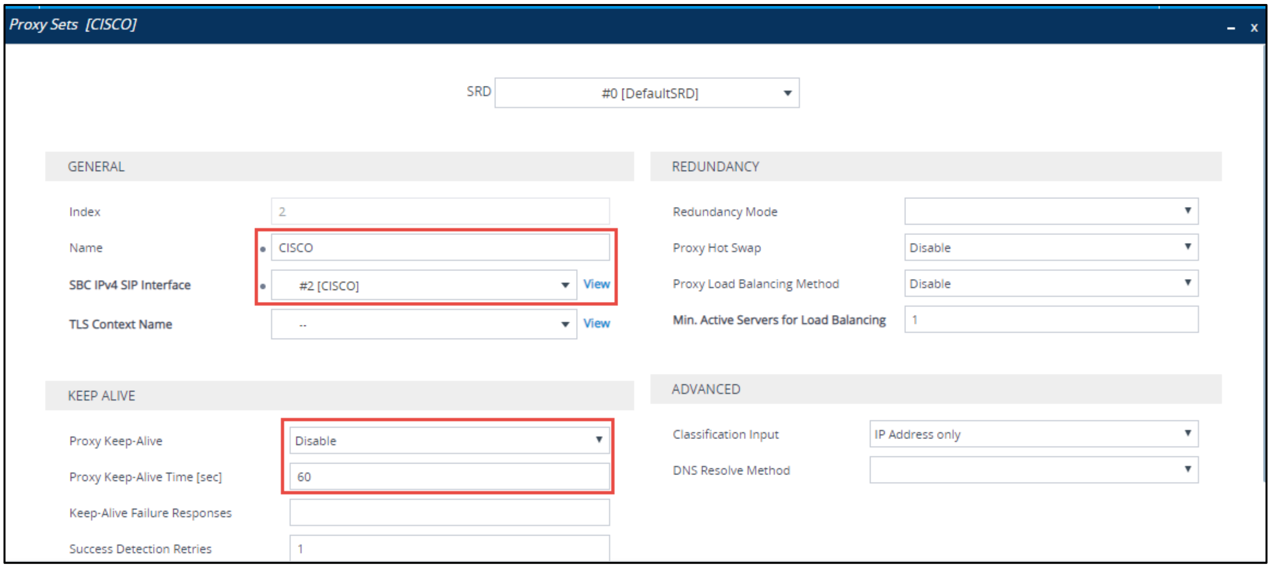

Figure 47 – Cisco UCM

Figure 48 – Cisco UCM

4.4.11 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

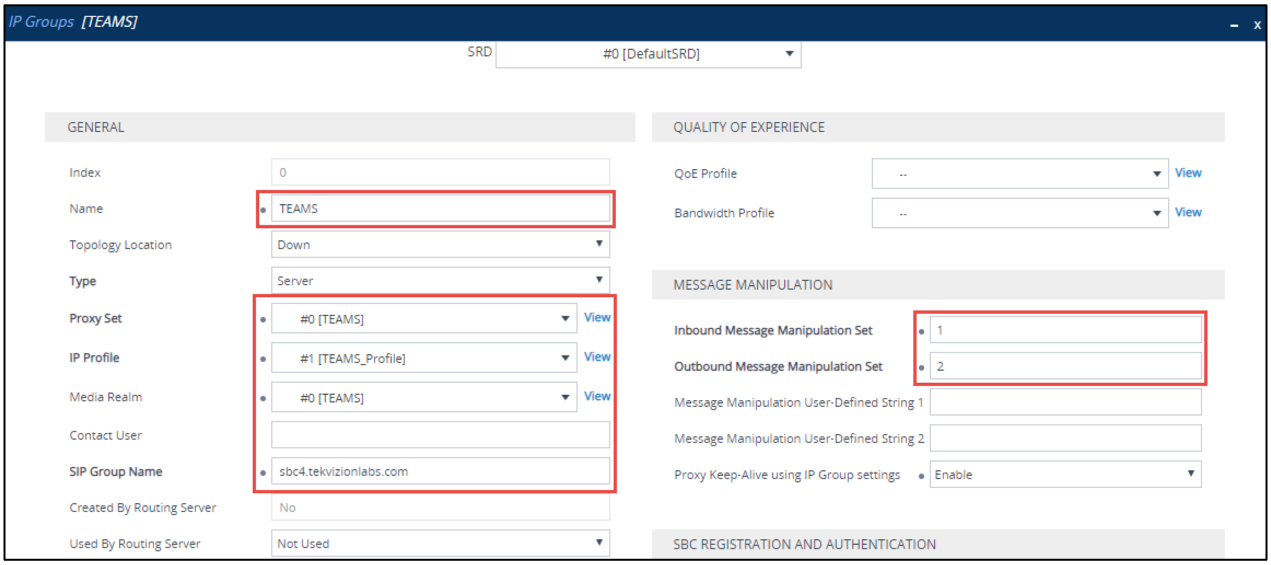

Figure 49 – IP Group – Teams – Contd.

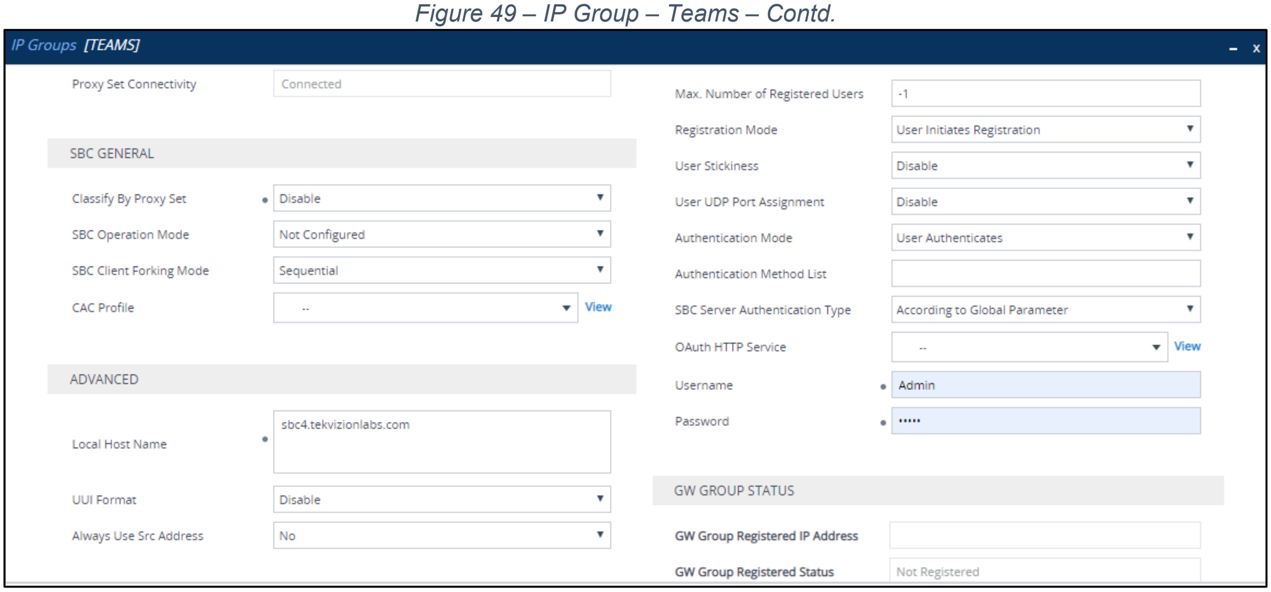

Figure 50 – IP Group – Teams – Contd.

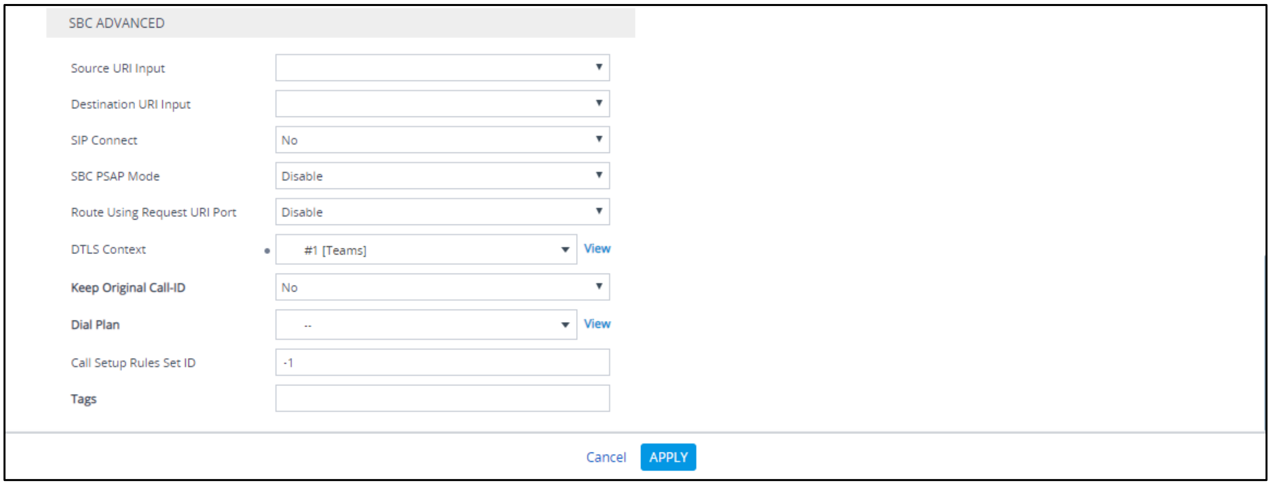

Figure 51 – IP Group – Teams

Configure an IP Group for PSTN Gateway as shown below

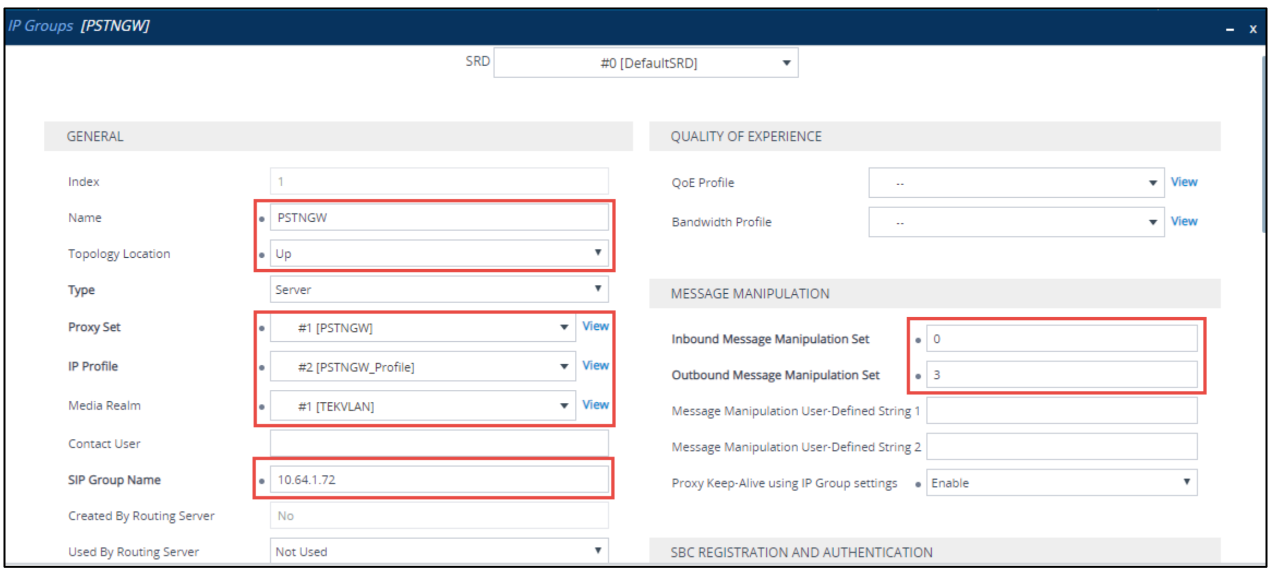

Figure 52 – IP Group – PSTN – Contd.

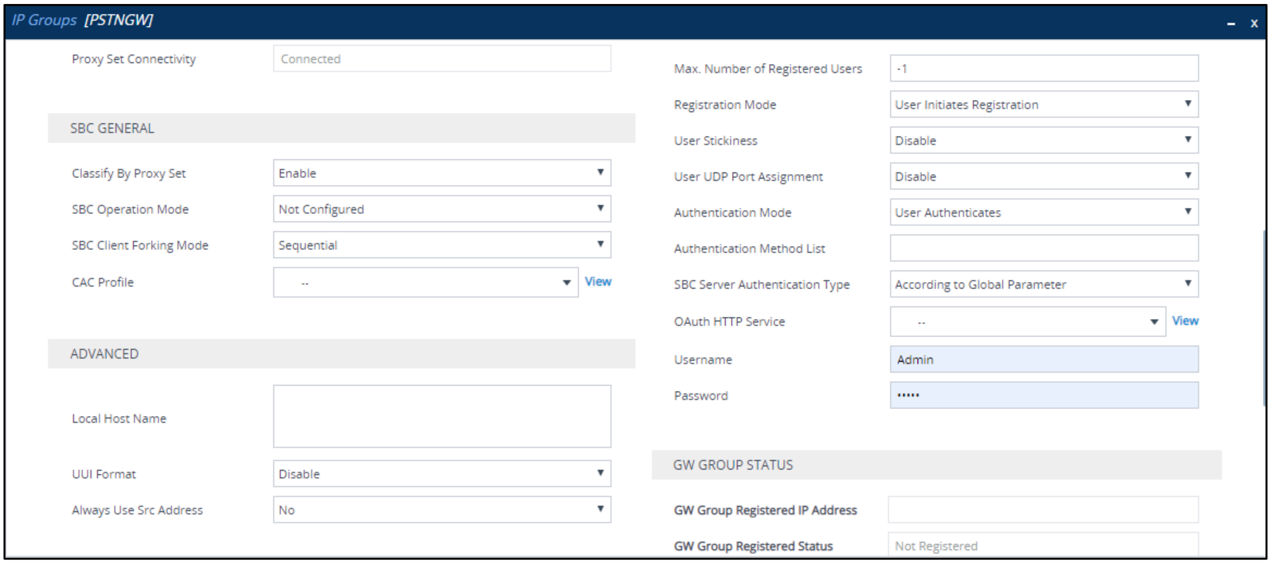

Figure 53 – IP Group – PSTN – Contd.

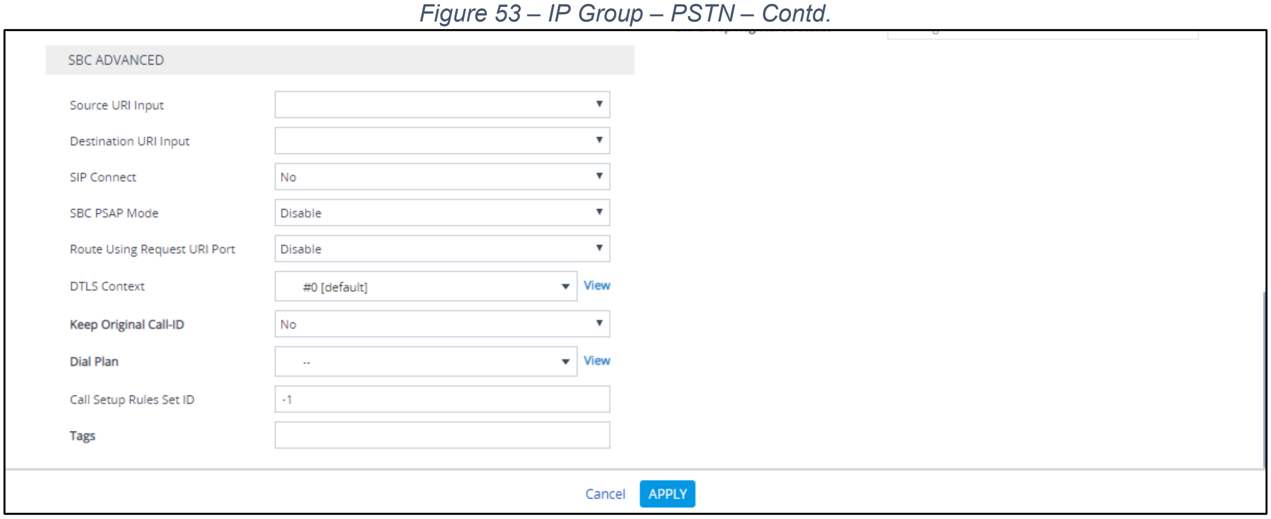

Figure 54 – IP Group

Configure an IP Group for Cisco UCM as shown below

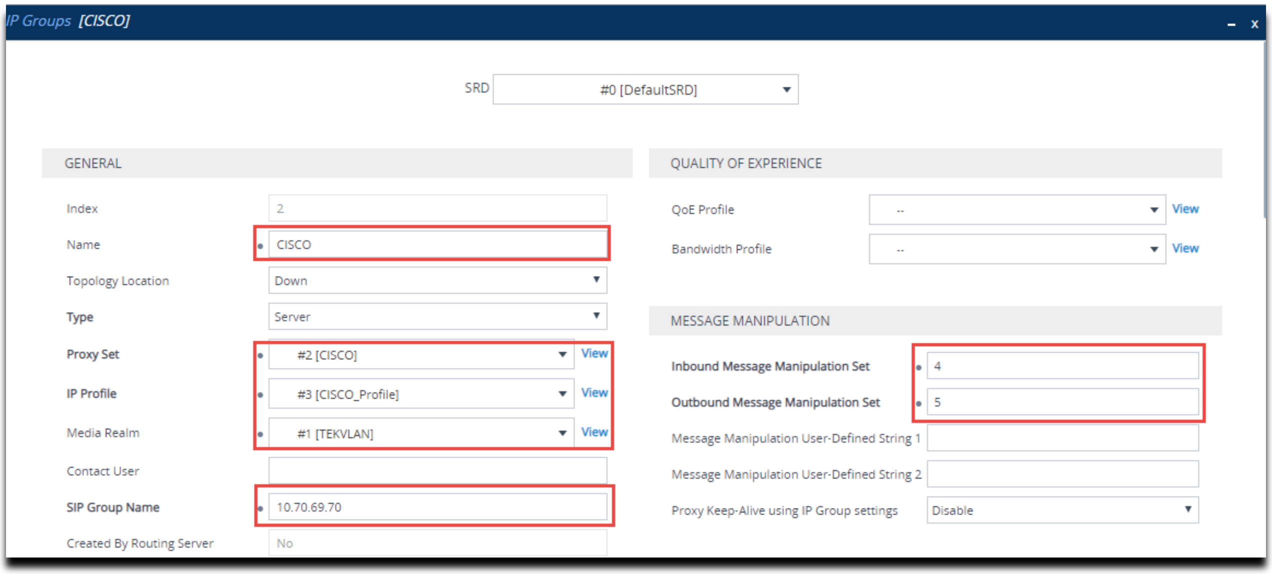

Figure 55 – IP Group – Cisco – Contd.

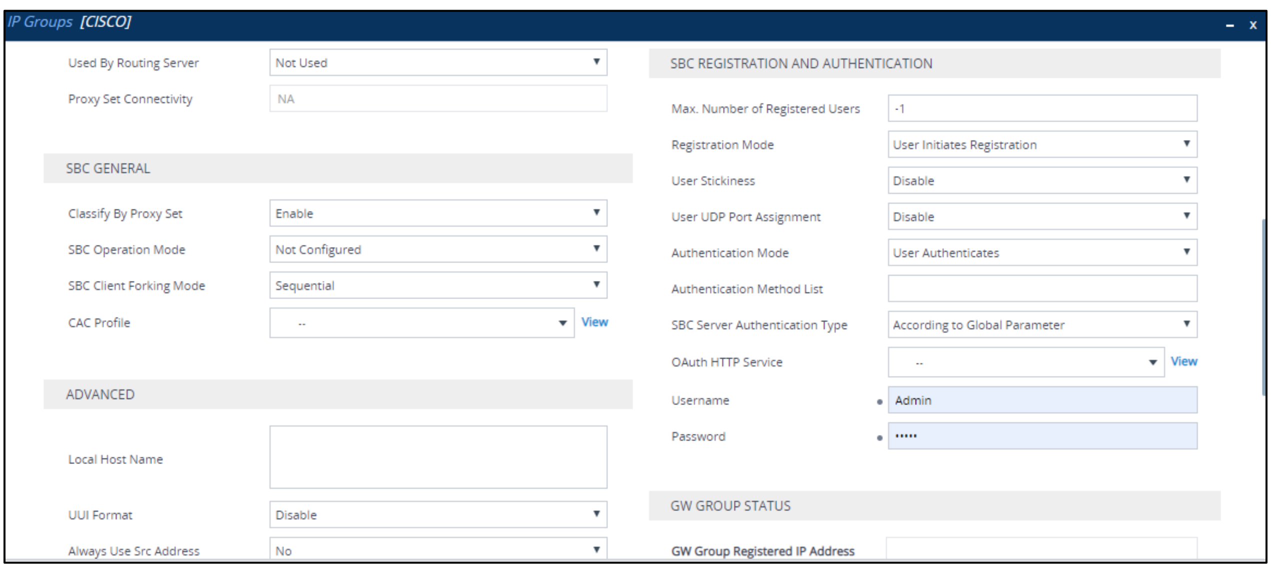

Figure 56 – IP Group – Cisco – Contd.

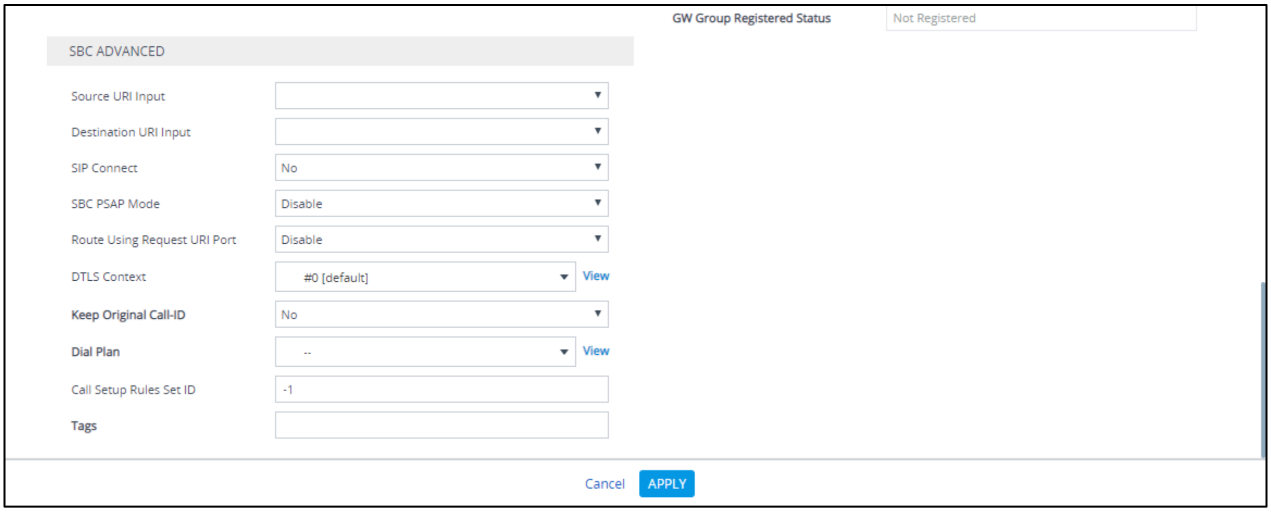

Figure 57 – IP Group

4.4.12 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.

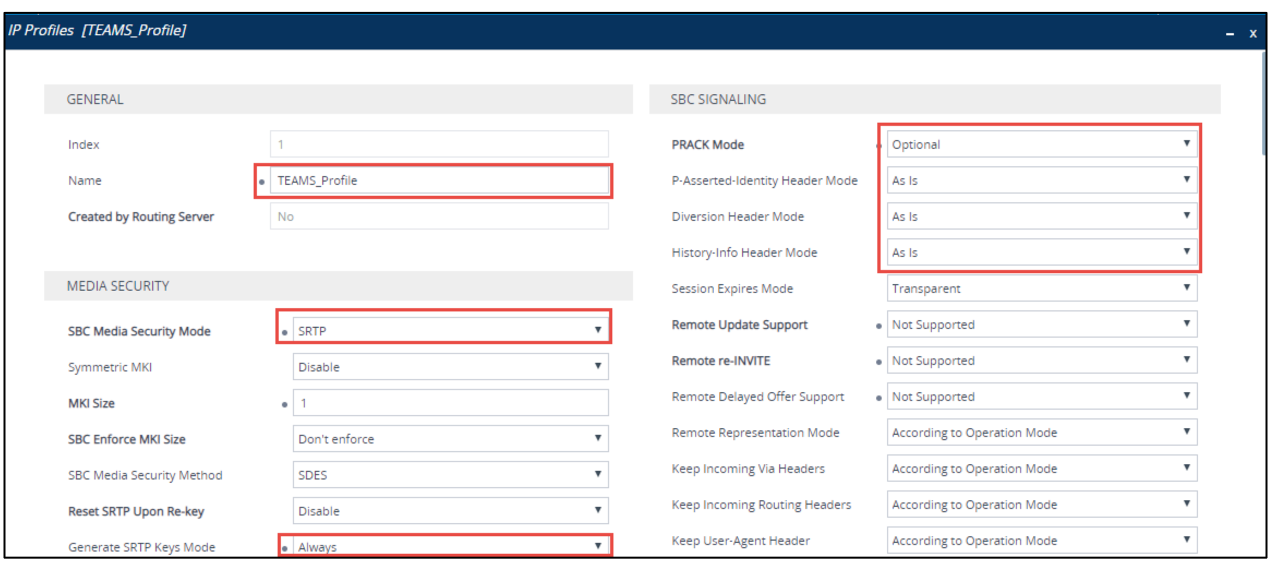

Figure 58 – IP Profile – Teams – Contd.

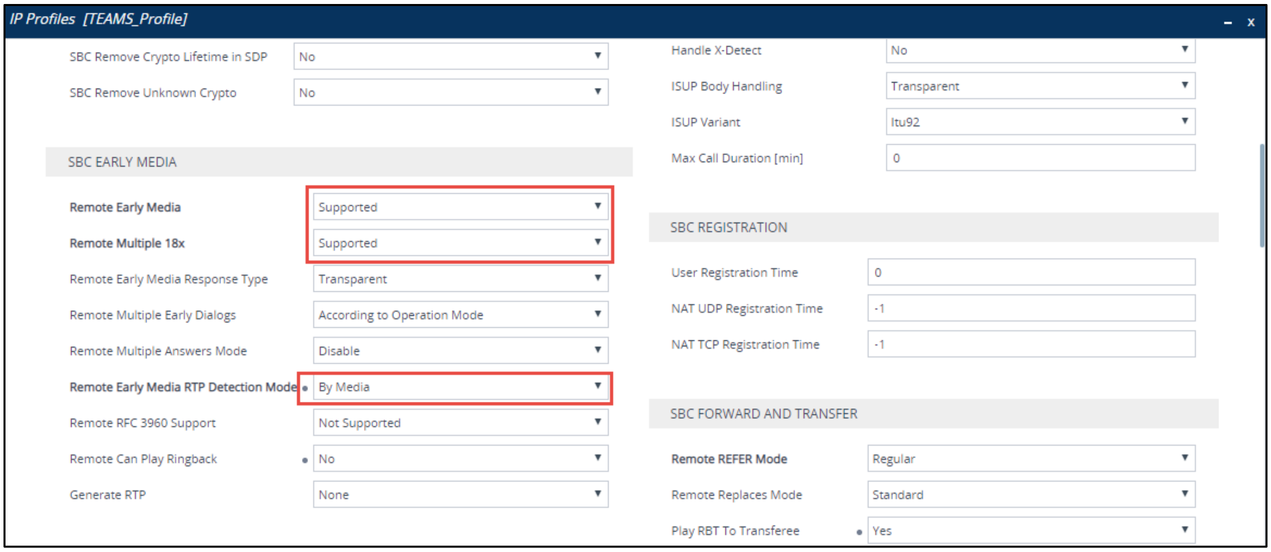

Figure 59 – IP Profile – Teams – Contd.

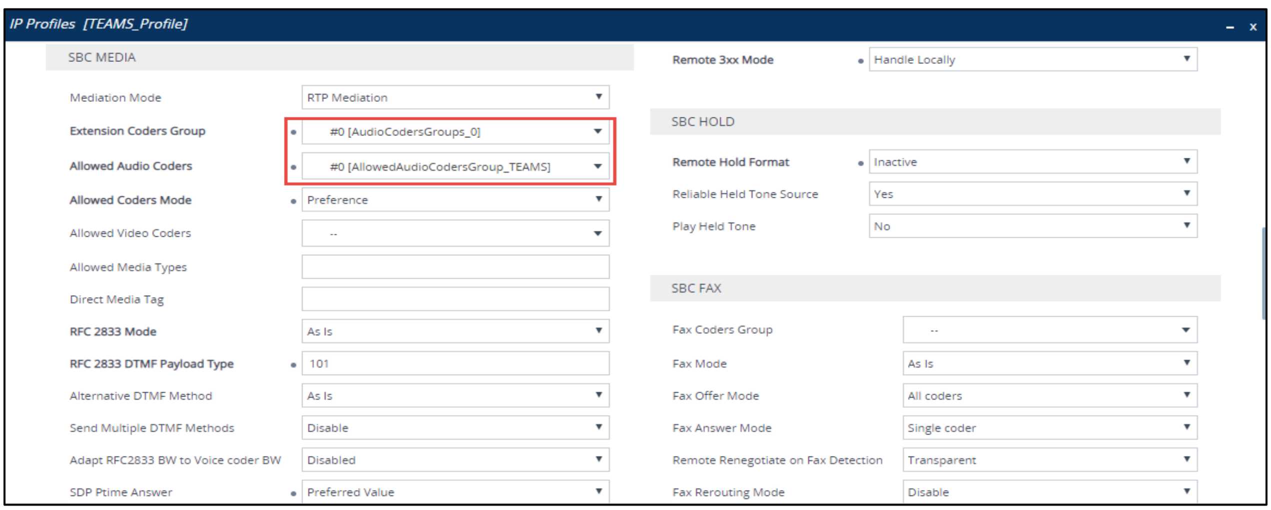

Figure 60 – IP Profile – Teams – Contd.

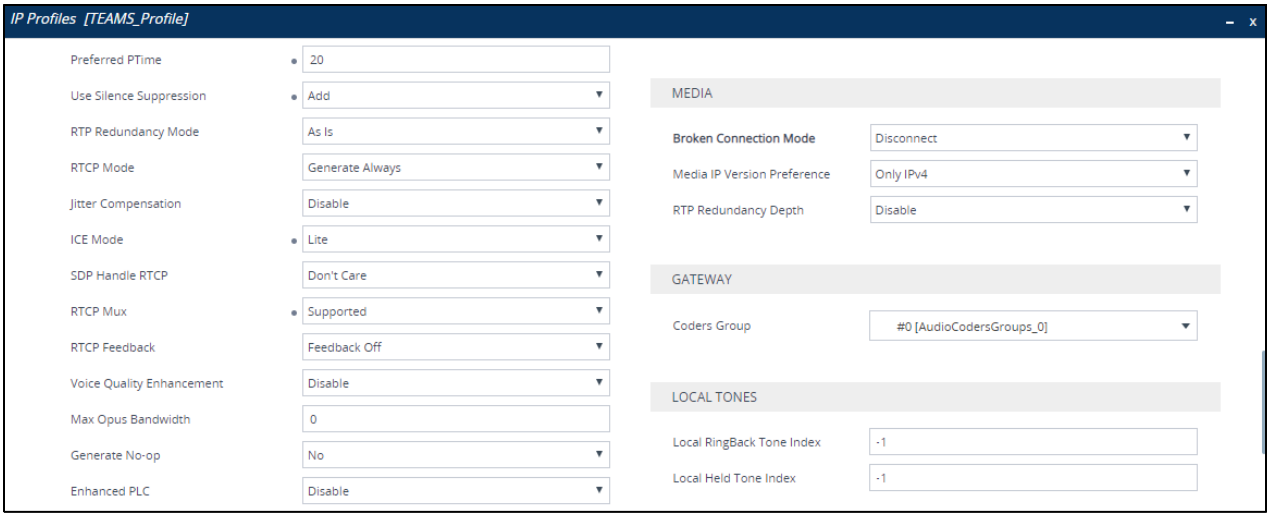

Figure 61 – IP Profile – Teams – Contd.

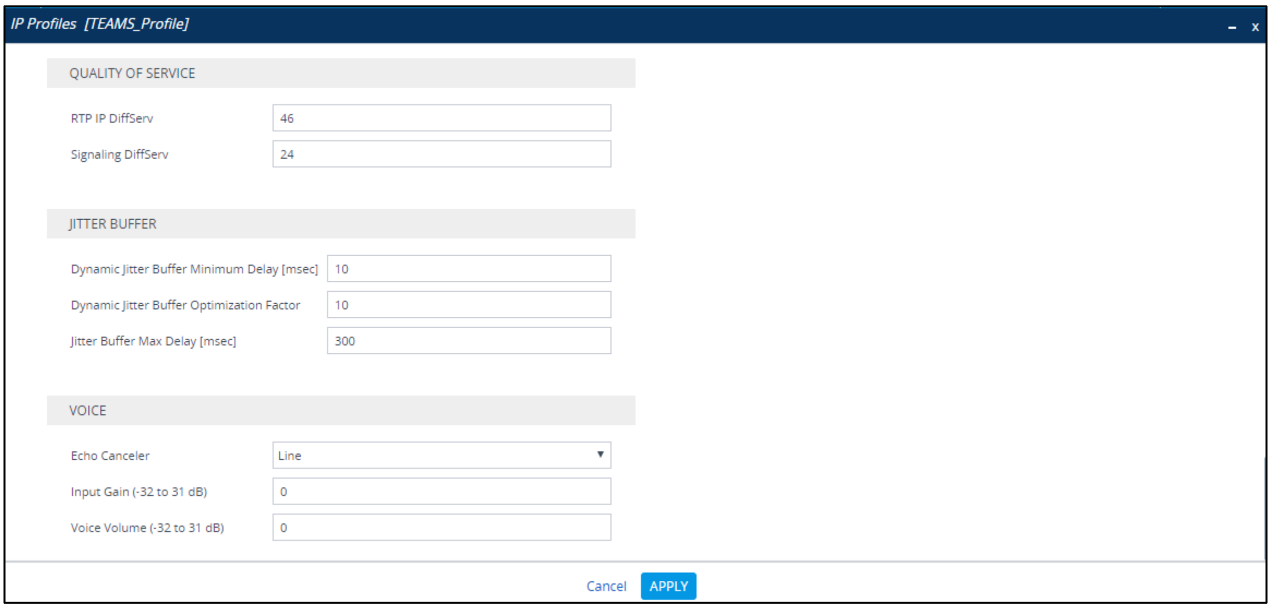

Figure 62 – IP Profile – Teams – Contd.

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

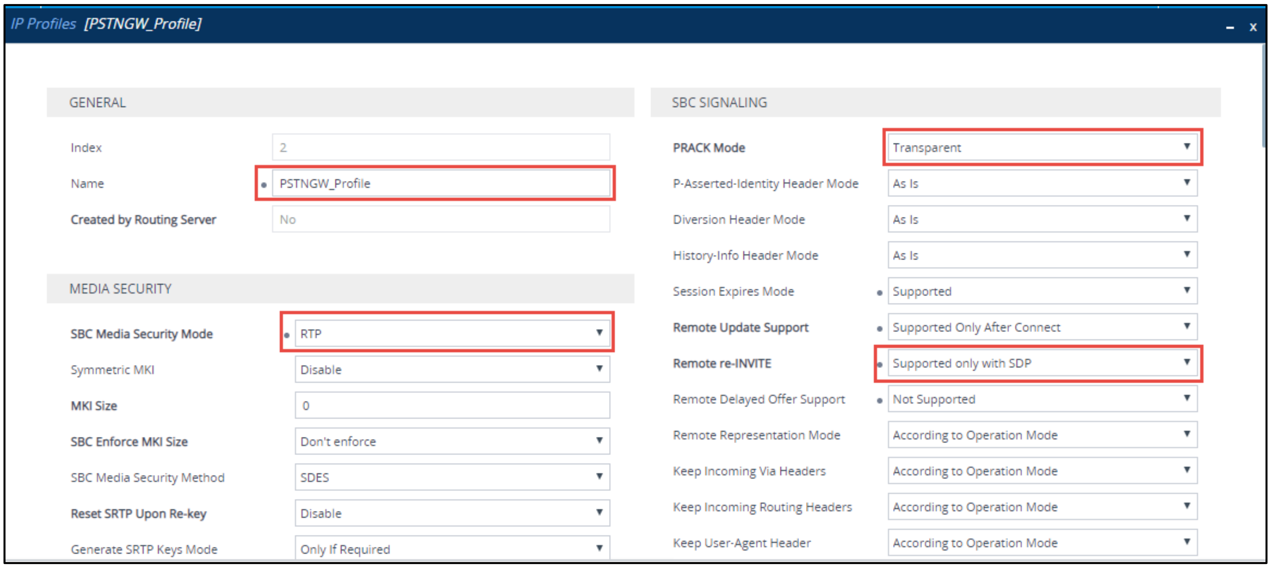

Figure 63 – IP Profile – PSTN Gateway – Contd.

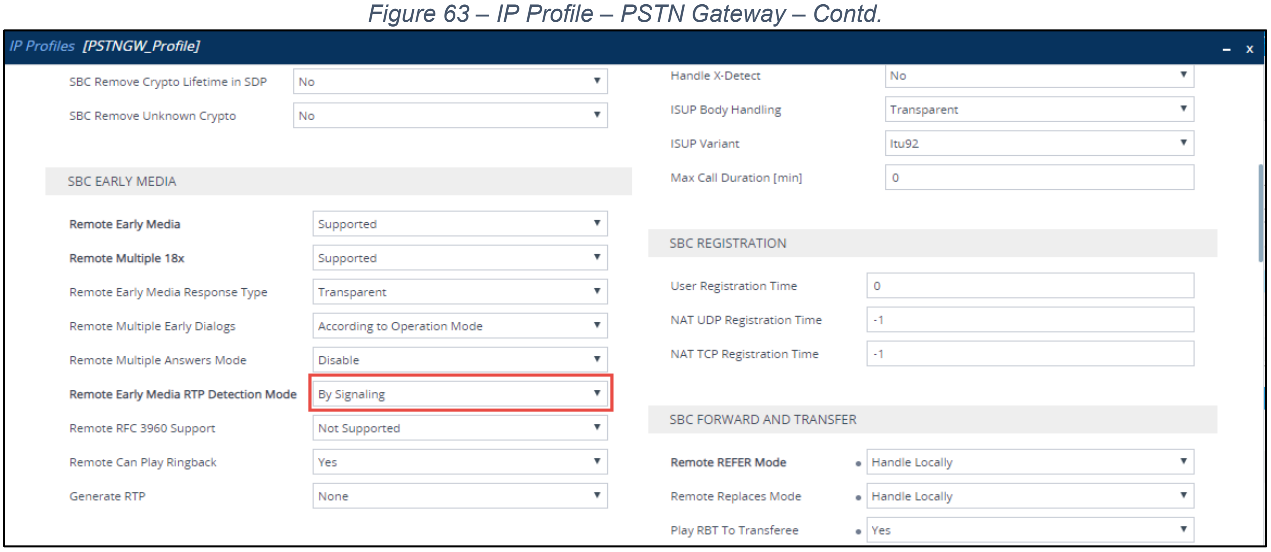

Figure 64 – IP Profile – PSTN Gateway – Contd.

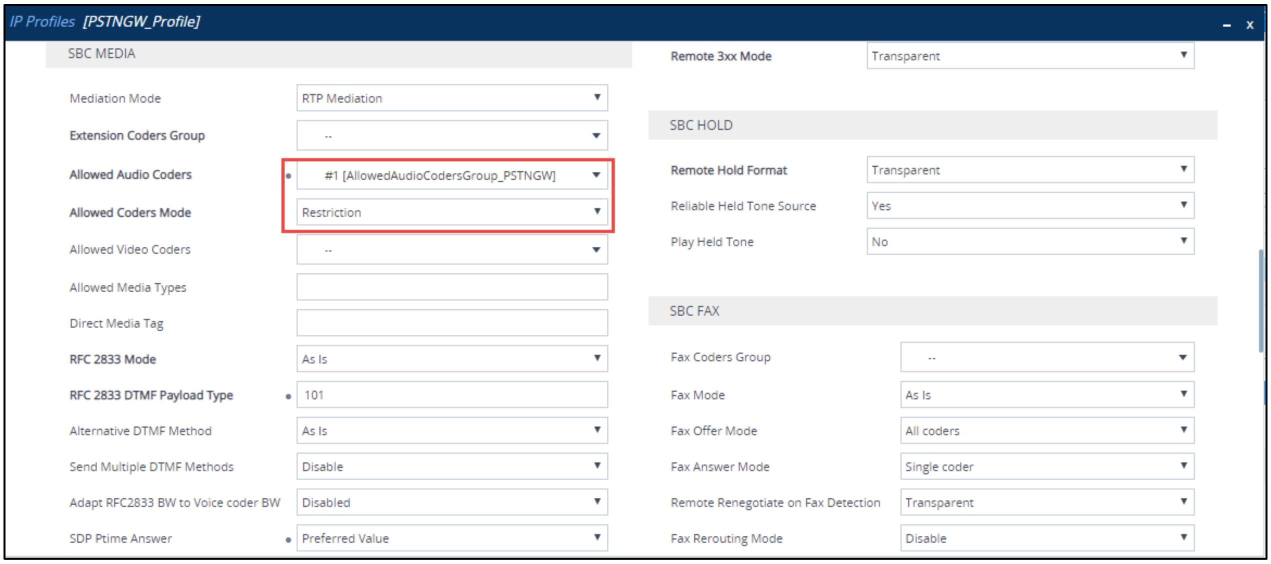

Figure 65 – IP Profile – PSTN Gateway – Contd.

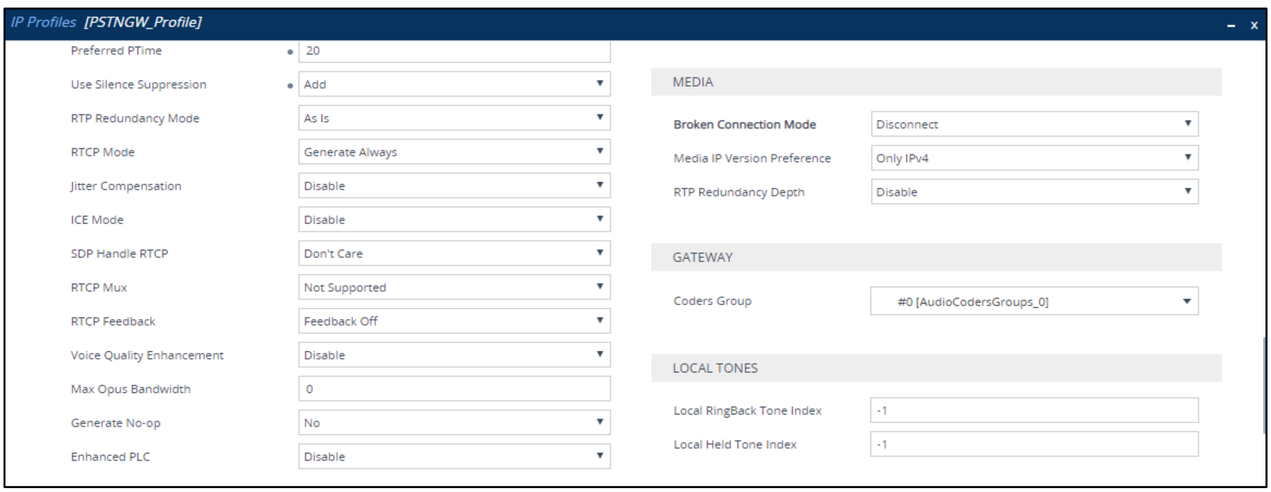

Figure 66 – IP Profile – PSTN Gateway – Contd.

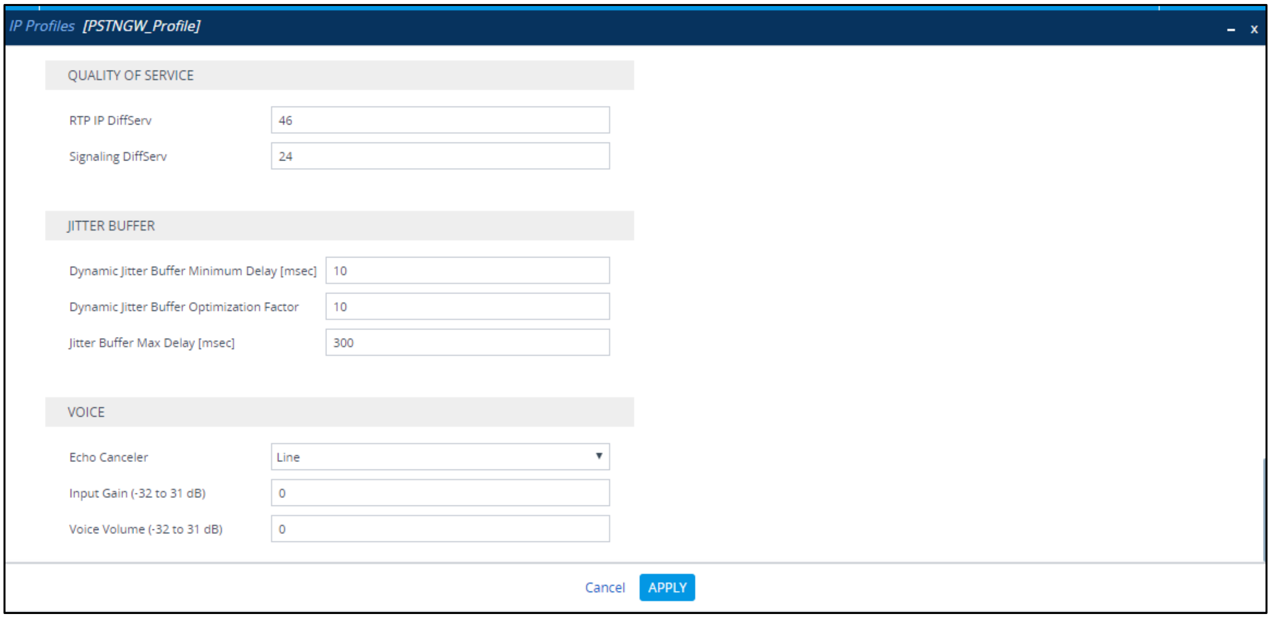

Figure 67 – IP Profile – PSTN Gateway

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

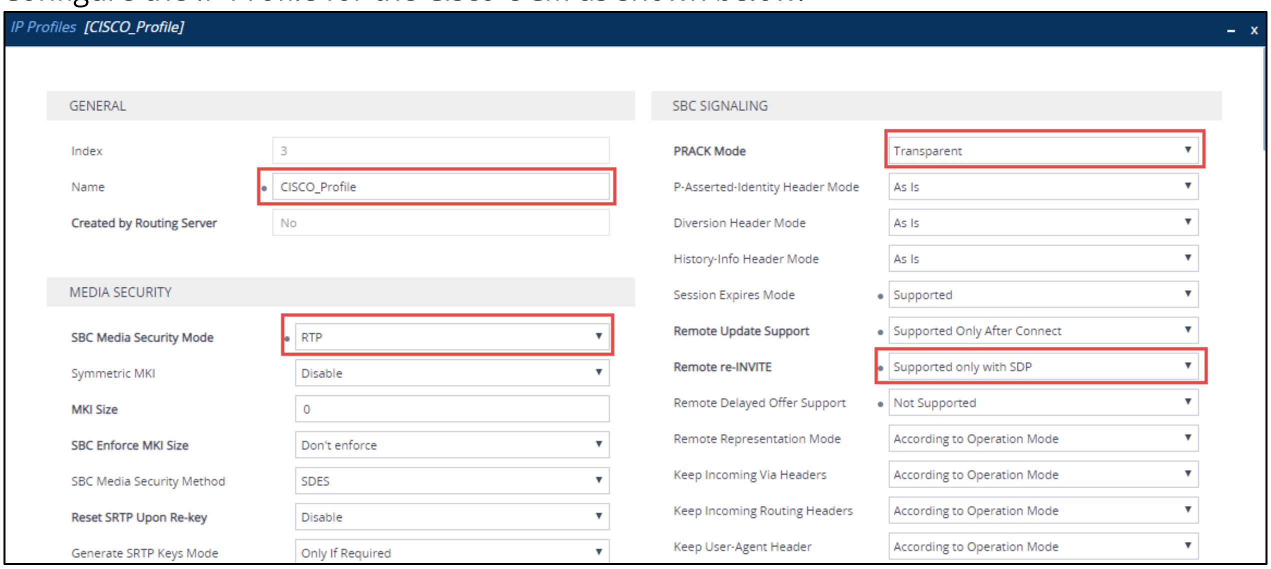

Figure 68 – IP Profile – Cisco – Contd.

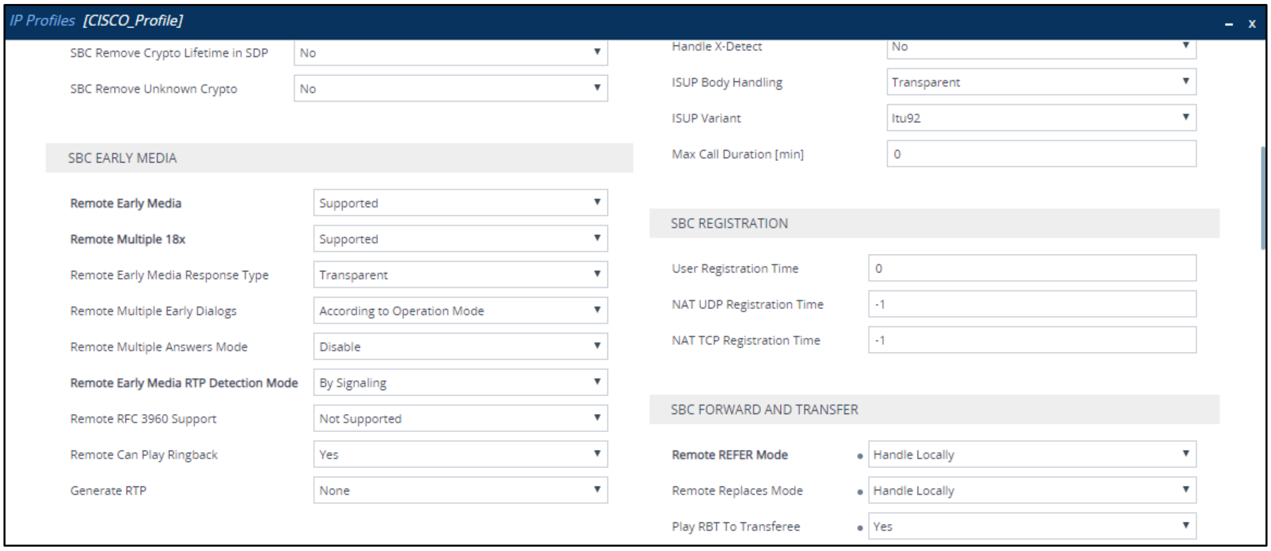

Figure 69 – IP Profile – Cisco – Contd.

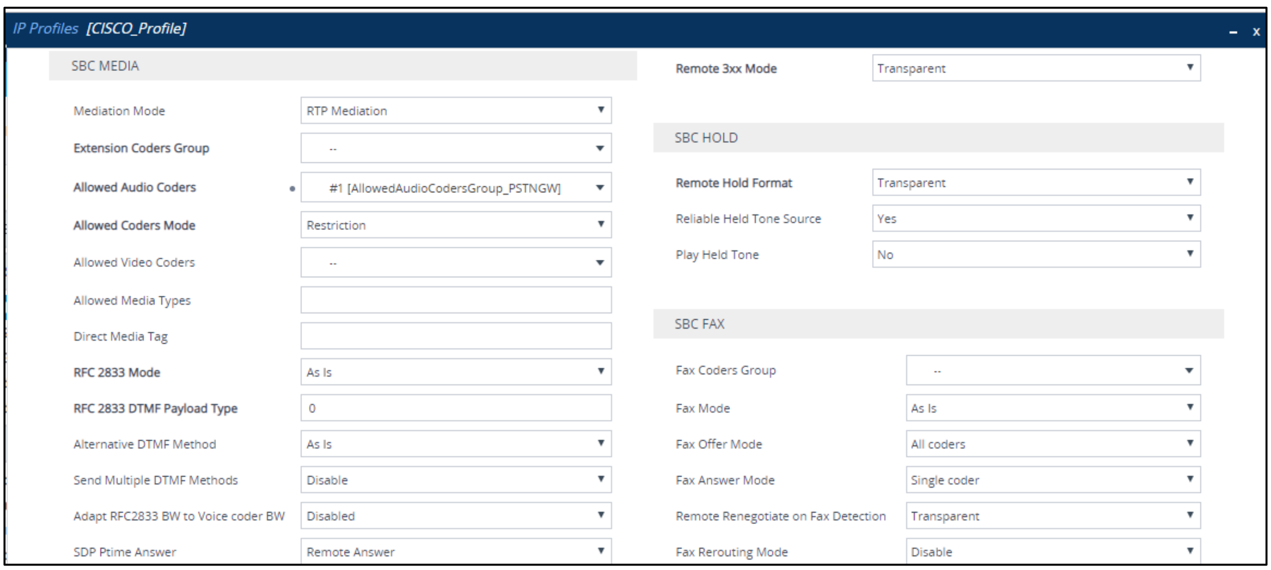

Figure 70 – IP Profile – Cisco – Contd.

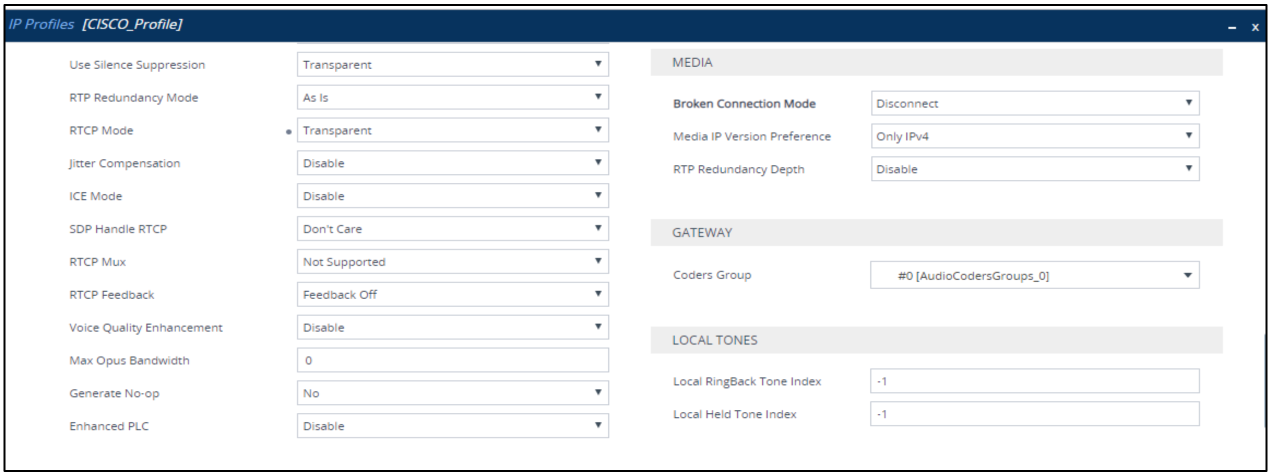

Figure 71 – IP Profile – Cisco – Contd.

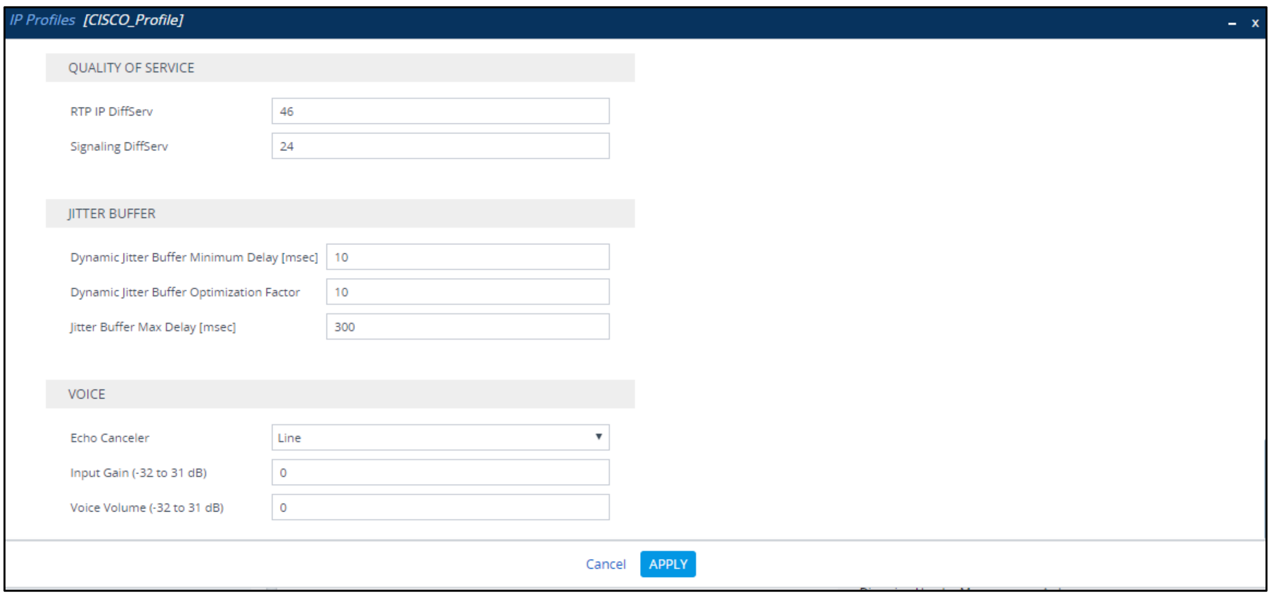

Figure 72 – IP Profile – Cisco

4.4.13 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

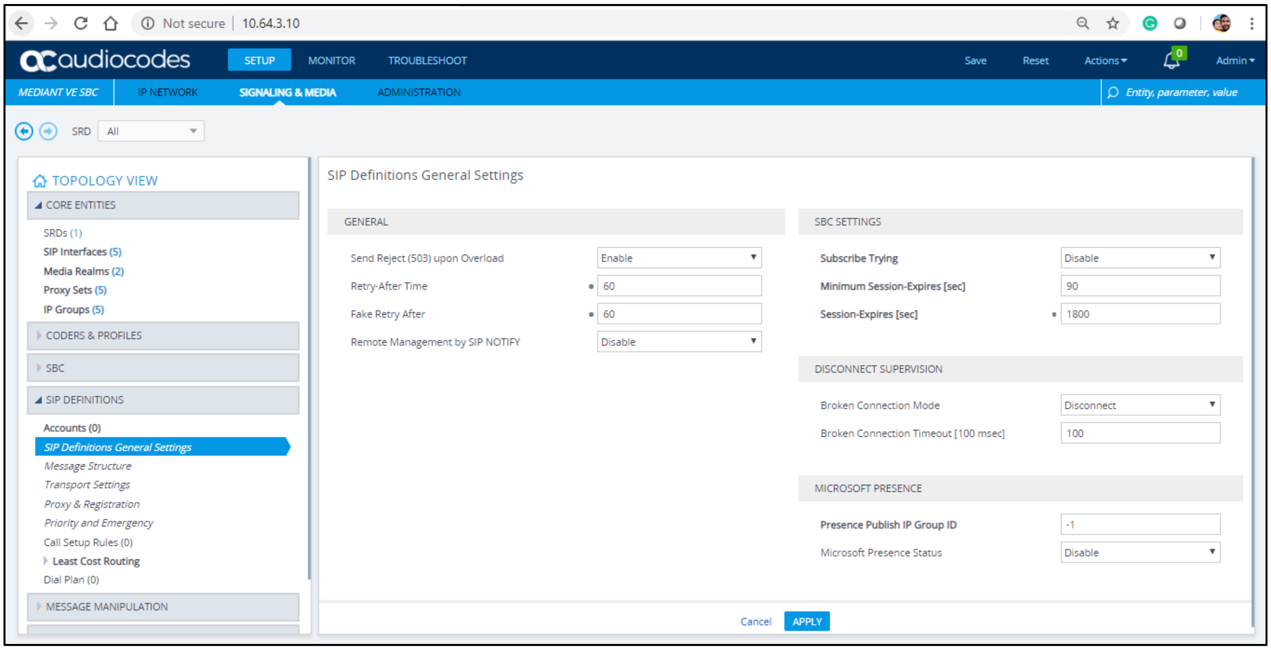

Figure 73 – SIP Definition

4.4.14 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.

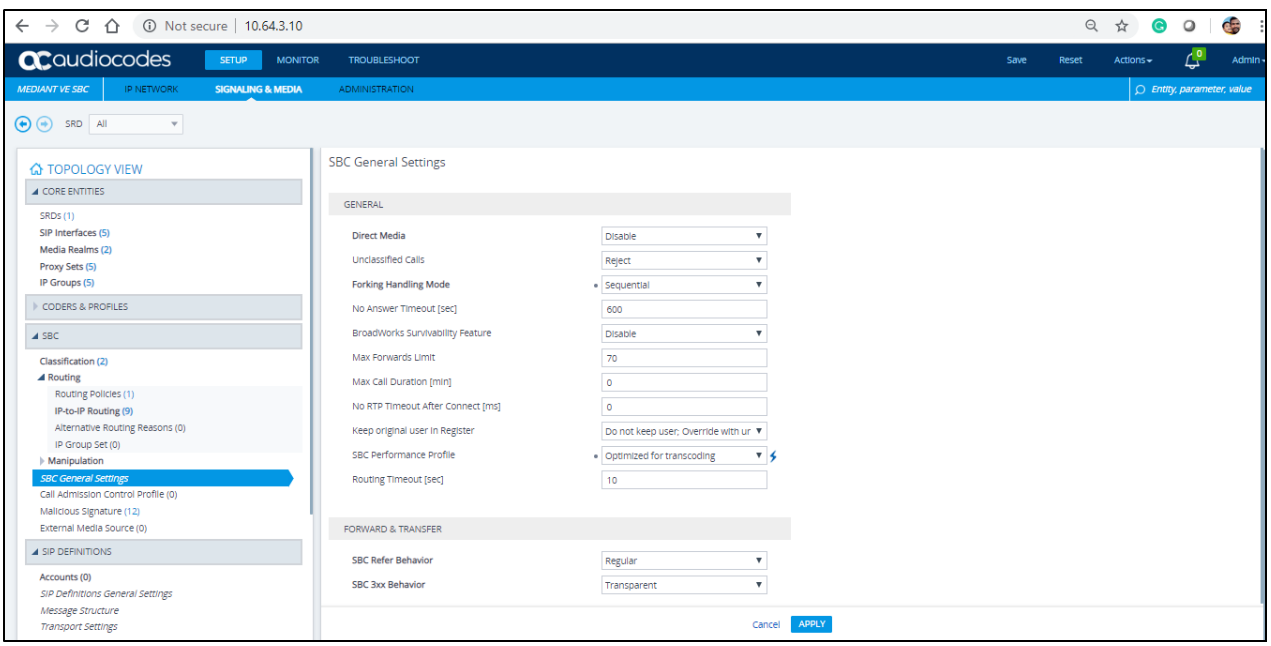

Figure 74 – SBC General Setting – Contd.

4.4.15 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

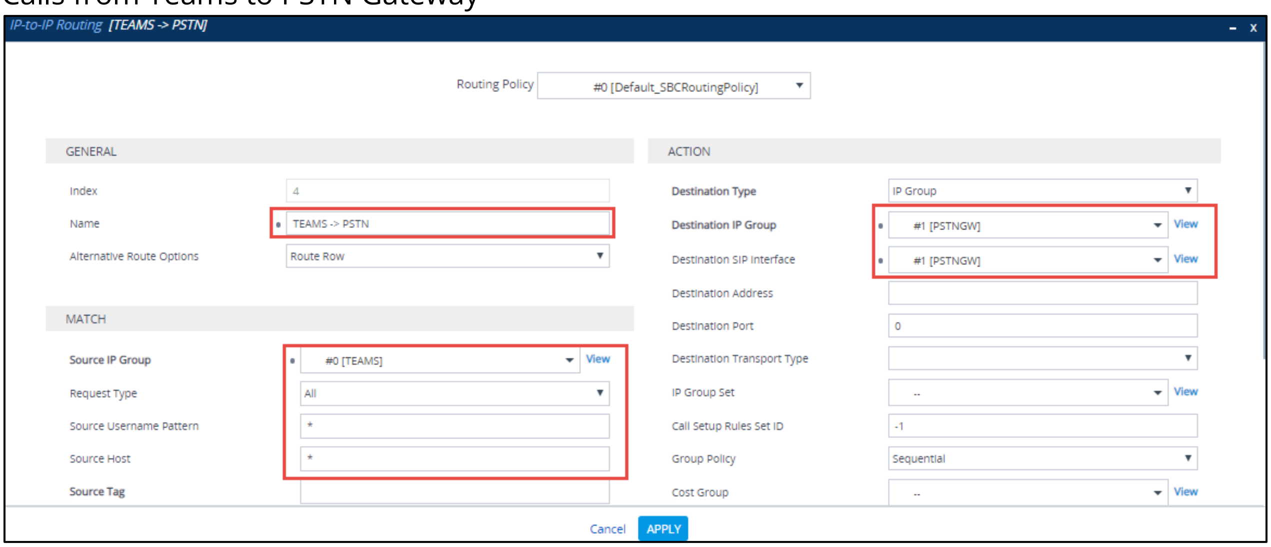

Figure 75 – Teams to PSTN – Contd.

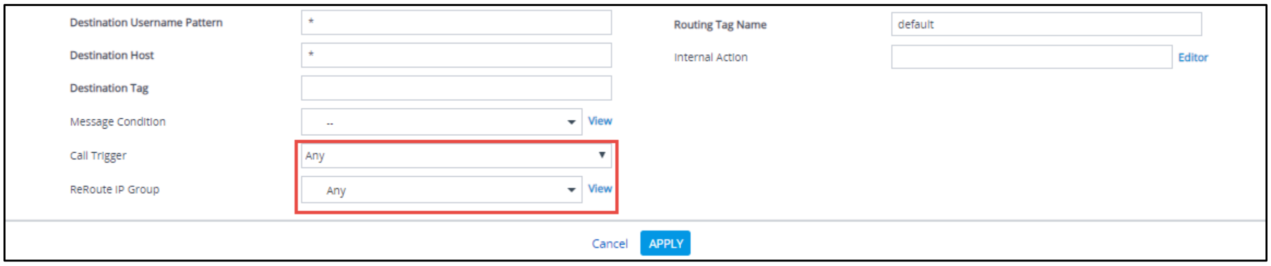

Figure 76 – Teams to PSTN

Calls from PSTN Gateway to Teams

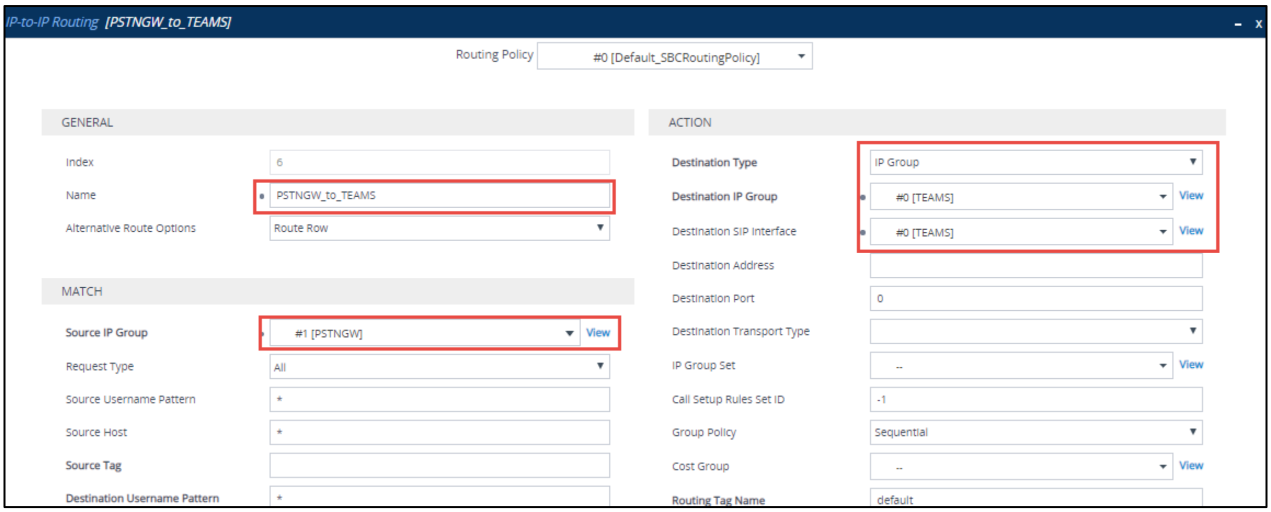

Figure 77 – PSTN to Teams – Contd.

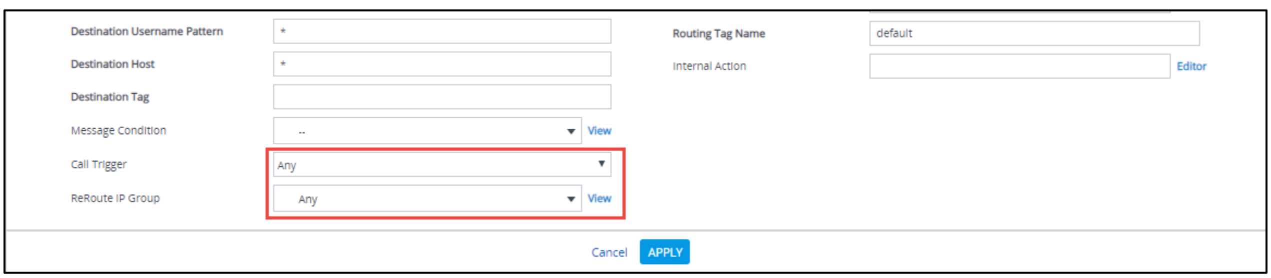

Figure 78 – PSTN to Teams

Calls from Teams to Cisco

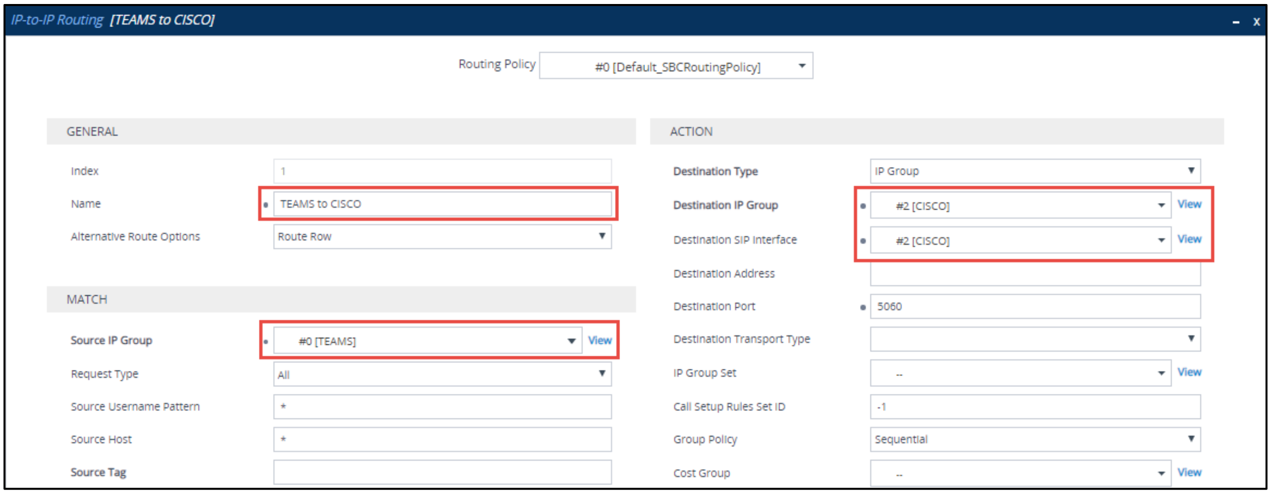

Figure 79 –Teams to Cisco – Contd.

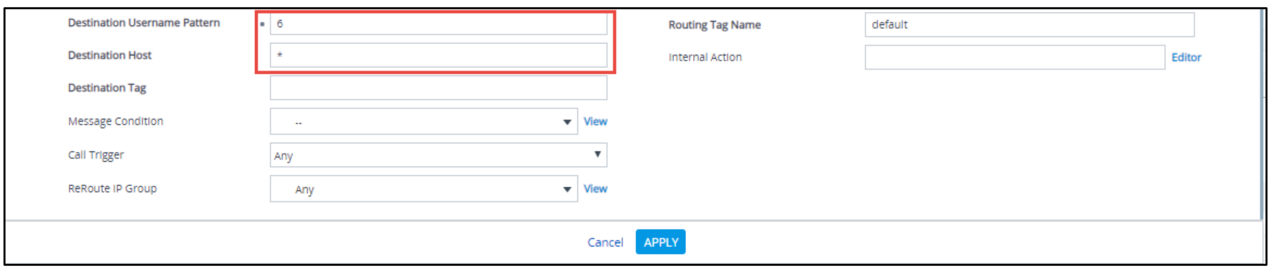

Figure 80 –Teams to Cisco

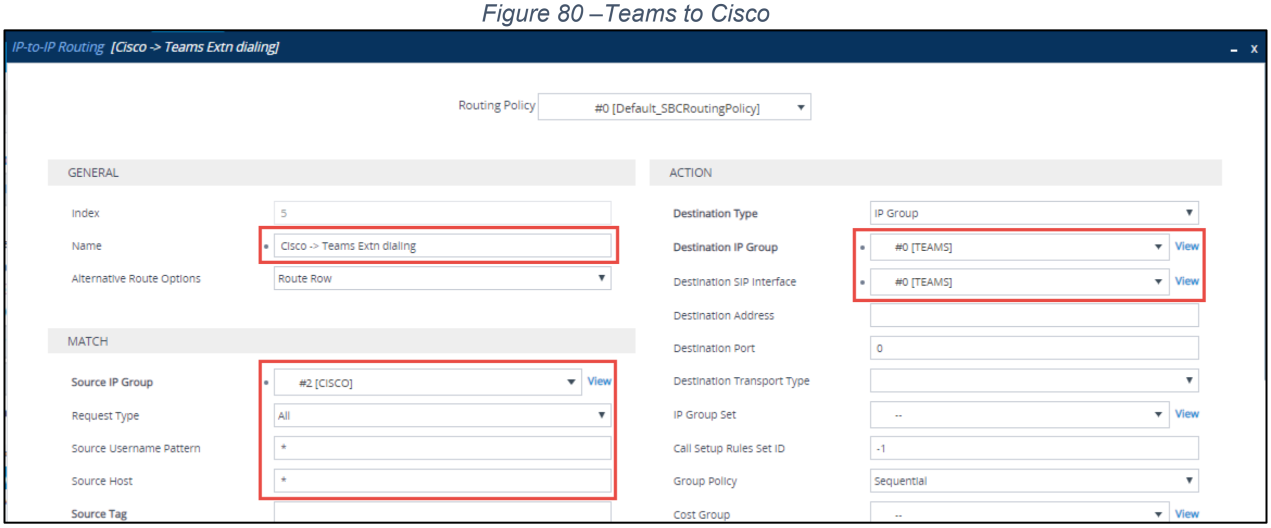

Figure 81 –Cisco to Teams – Contd.

Figure 82 –Cisco to Teams

4.4.16 IP Group

IP Group – Teams

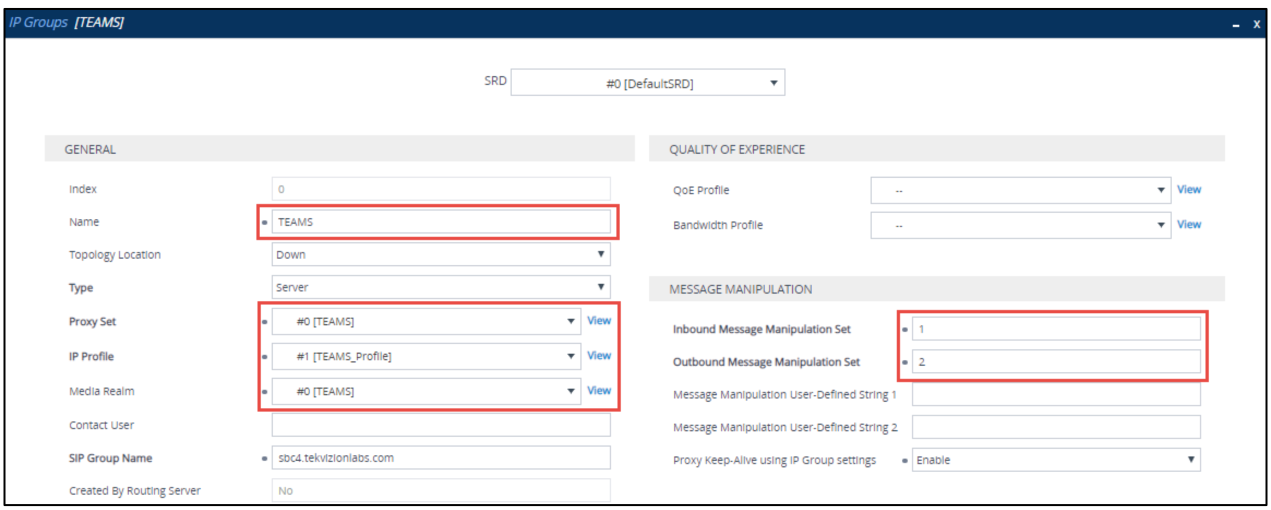

Figure 83 – IP Groups Teams – Contd.

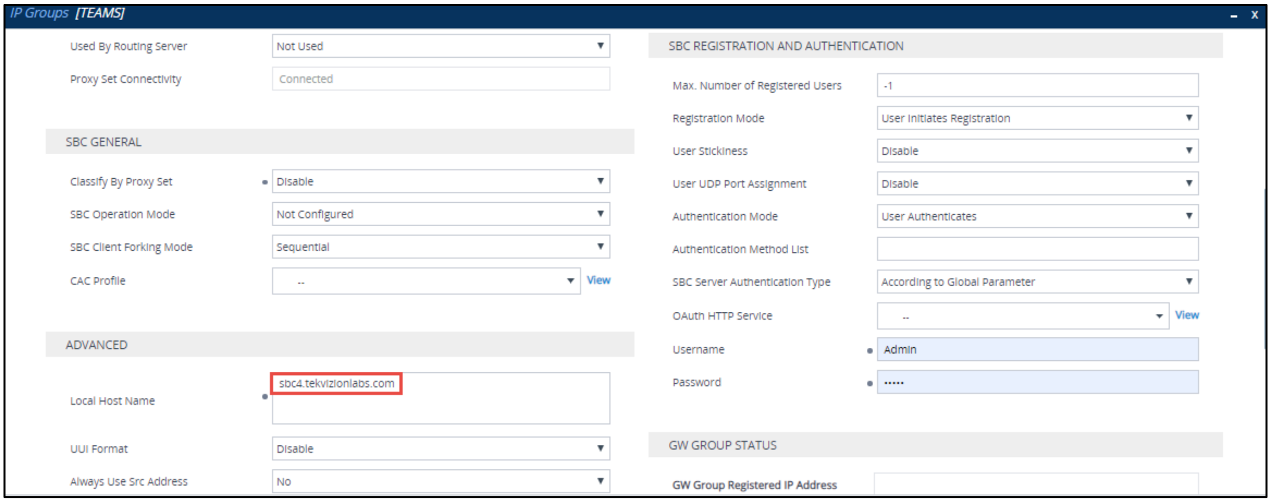

Figure 84 – IP Groups Teams – Contd.

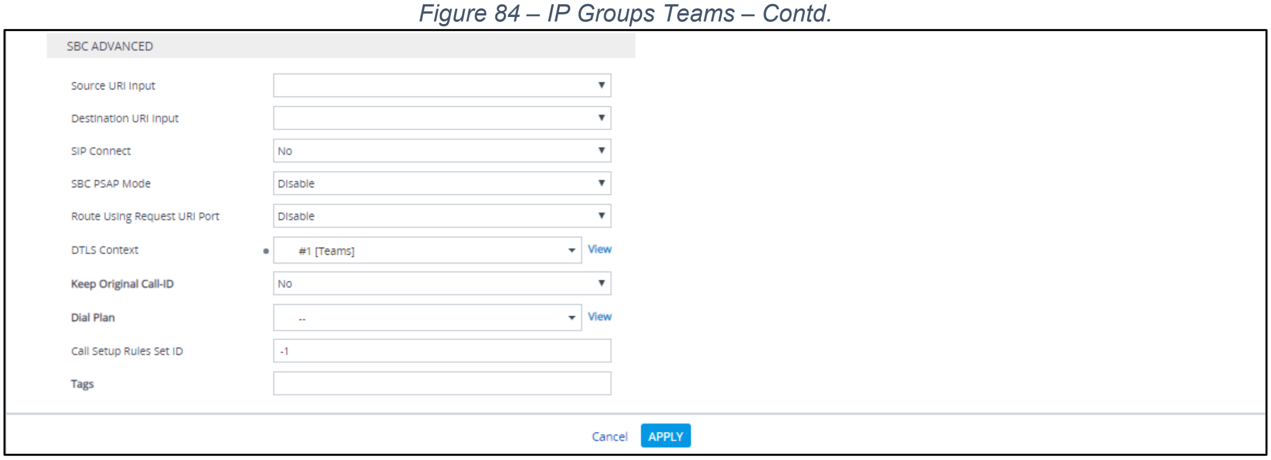

Figure 85 – IP Groups Teams

IP Group – PSTN Gateway

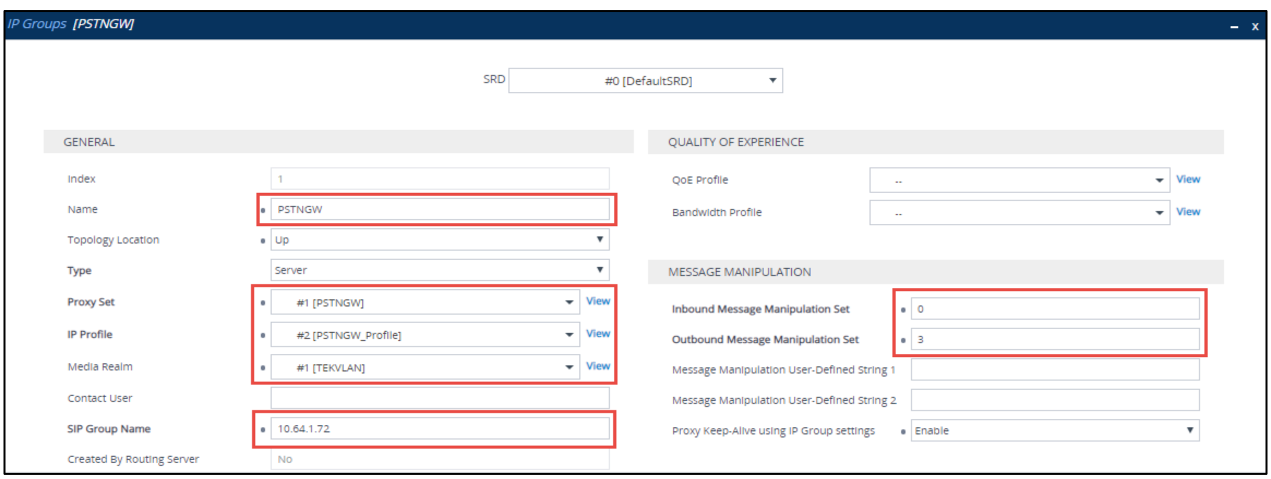

Figure 86 – IP Groups PSTN – Contd.

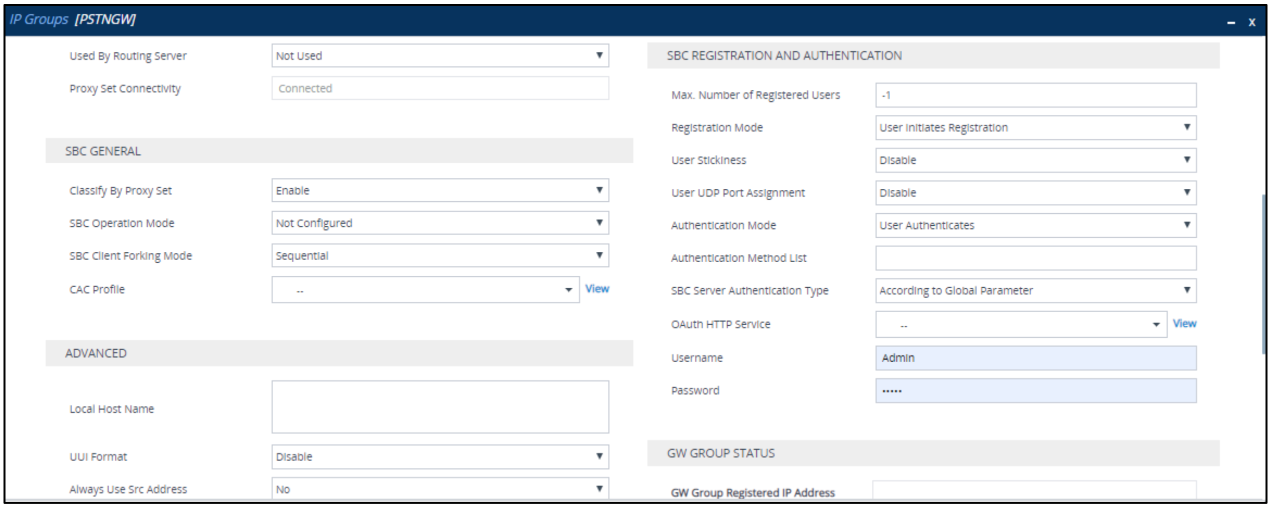

Figure 87 – IP Groups PSTN – Contd.

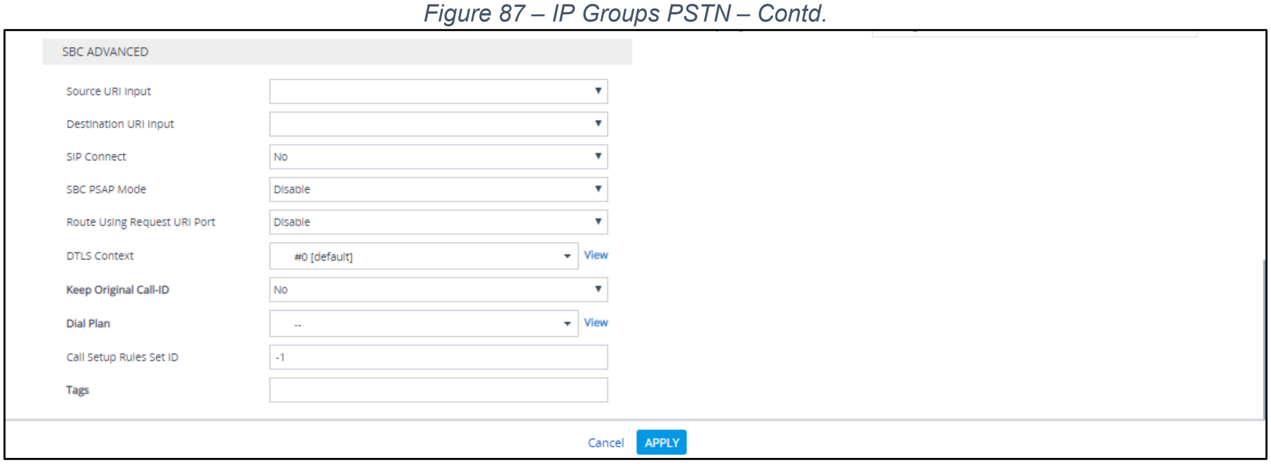

Figure 88 – IP Groups PSTN

IP Group – Cisco

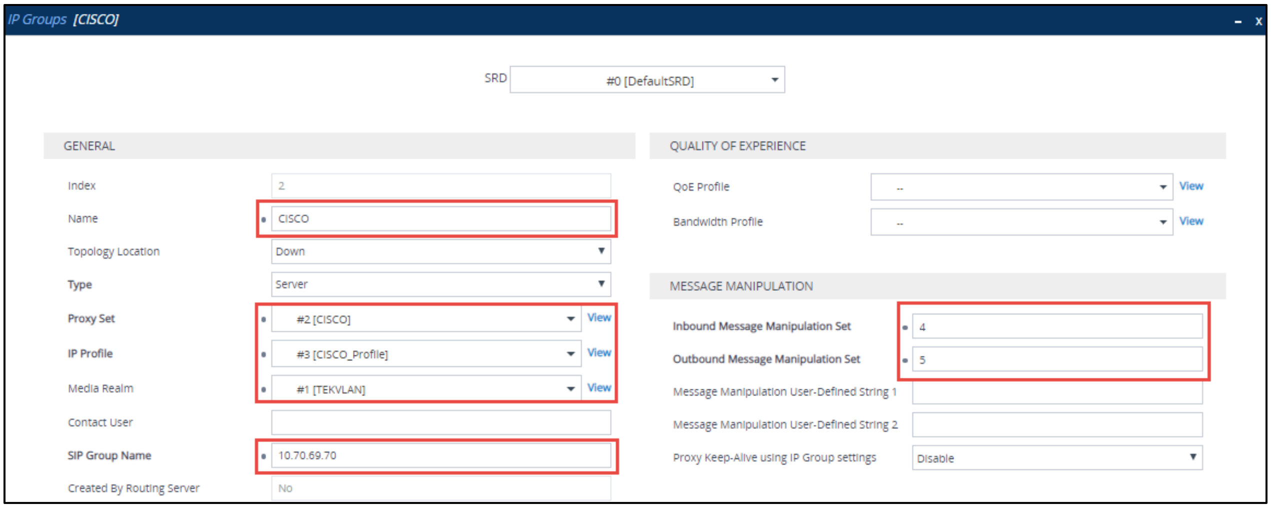

Figure 89 – IP Groups Cisco – Contd.

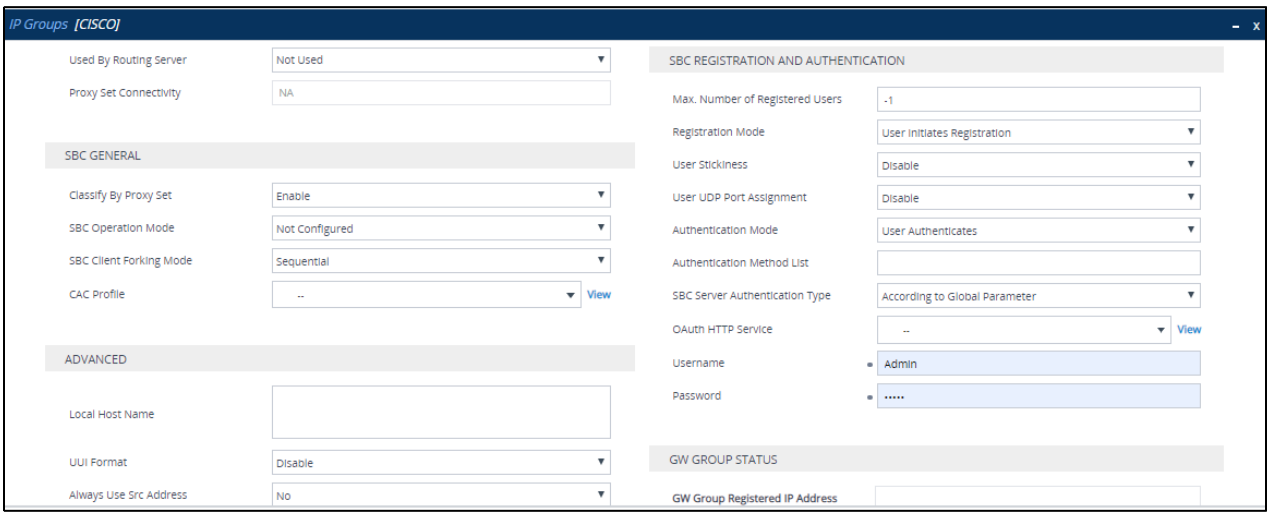

Figure 90 – IP Groups Cisco – Contd.

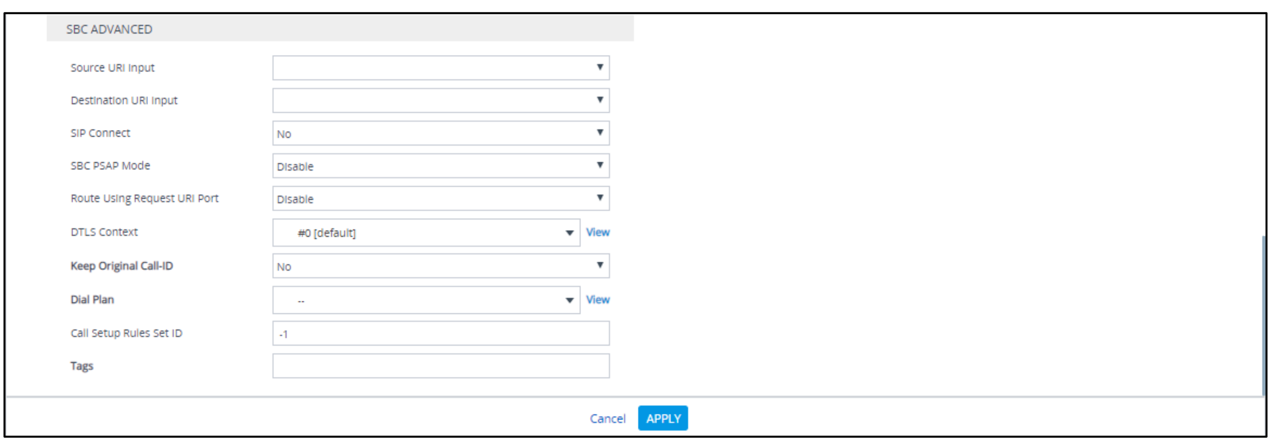

Figure 91 – IP Groups Cisco

4.4.17 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:

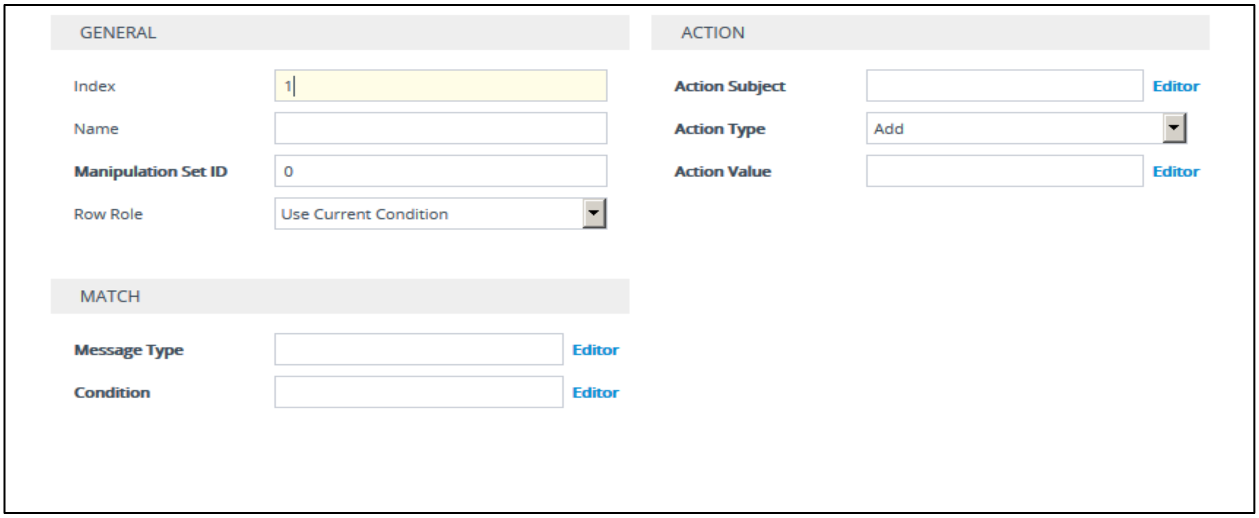

Figure 92 – SIP Message Manipulation

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

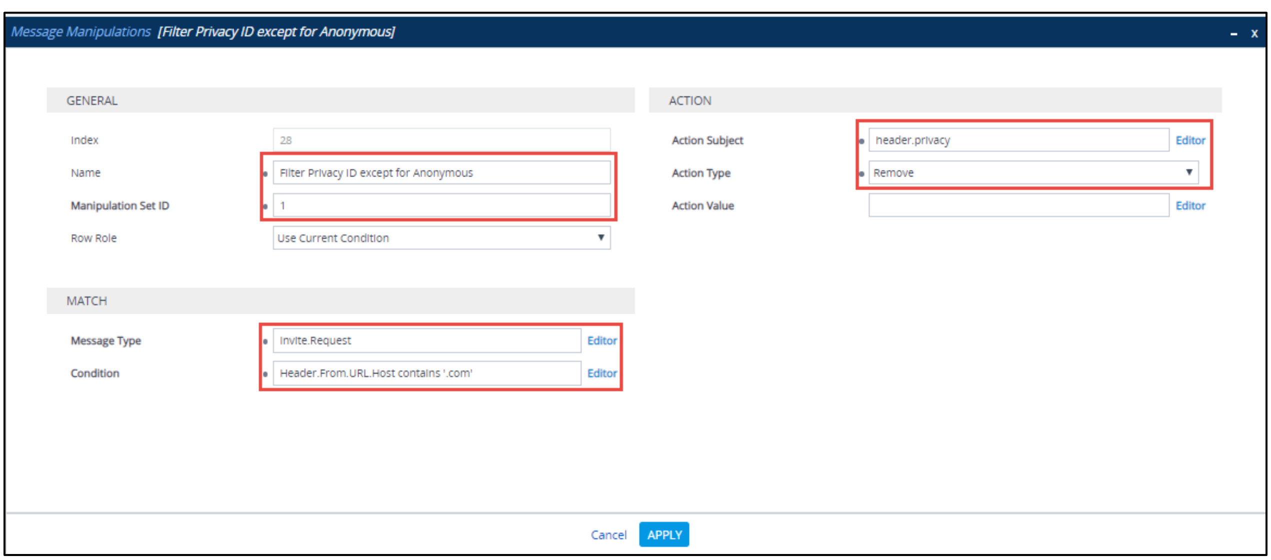

Figure 93 – SIP Message Manipulation - Privacy

Manipulation to Teams

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

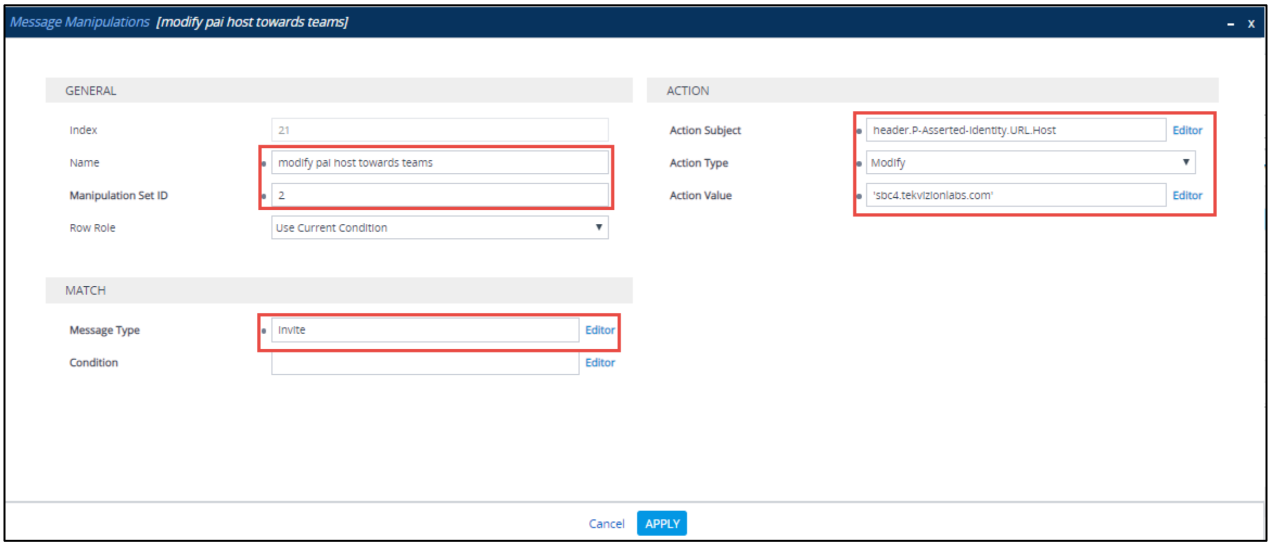

Figure 94 – SIP Message Manipulation – PAI

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

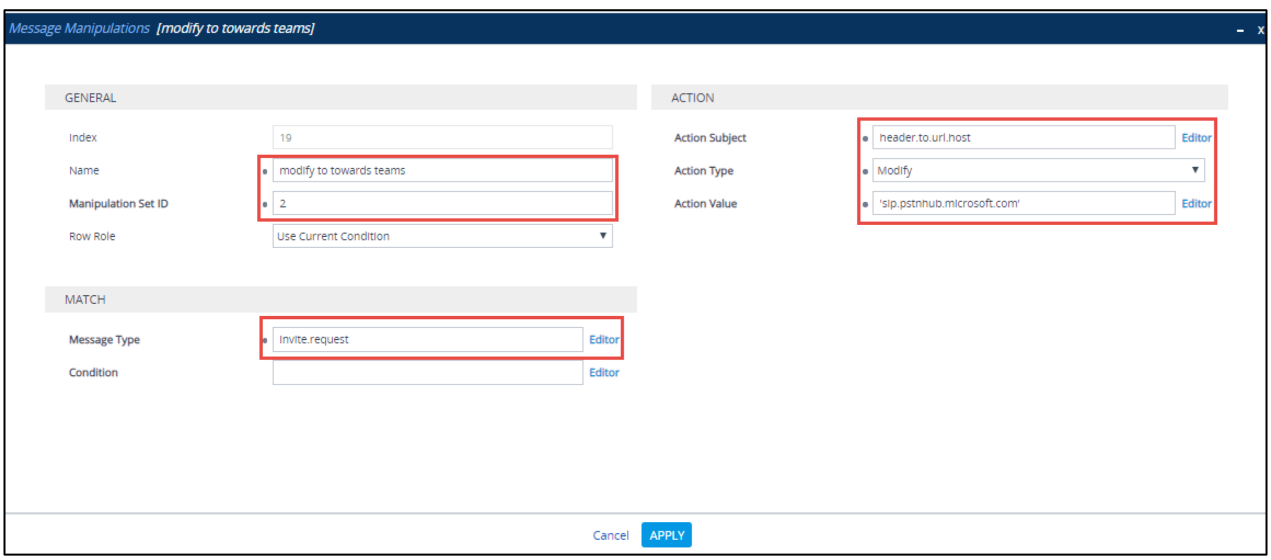

Figure 95 – SIP Message Manipulation - To

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

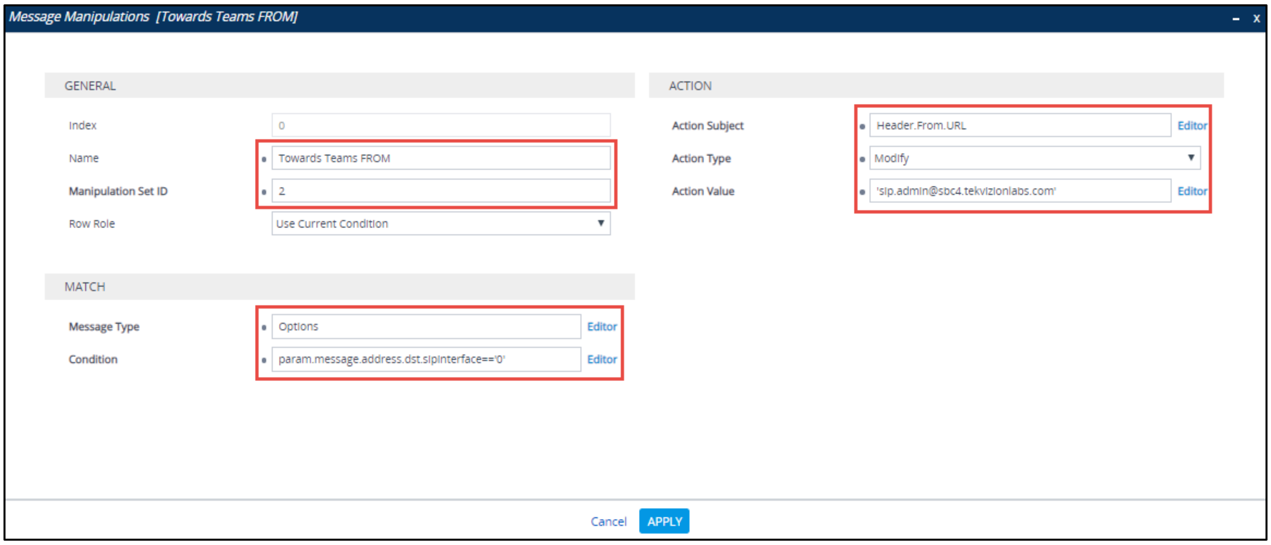

Figure 96 – SIP Message Manipulation - From

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

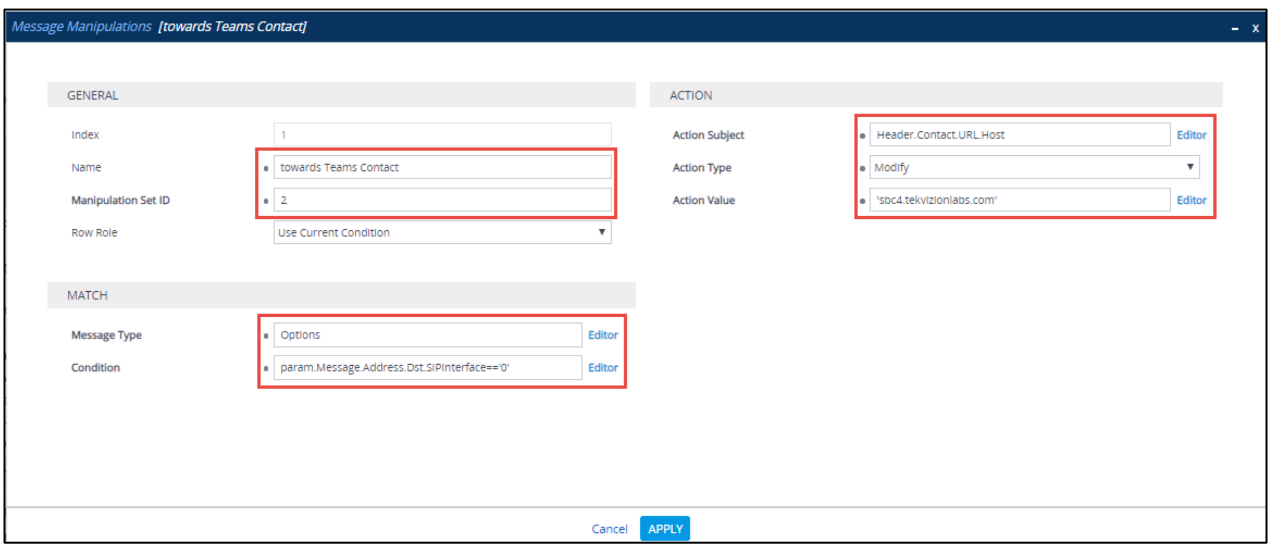

Figure 97 – SIP Message Manipulation - Contact

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

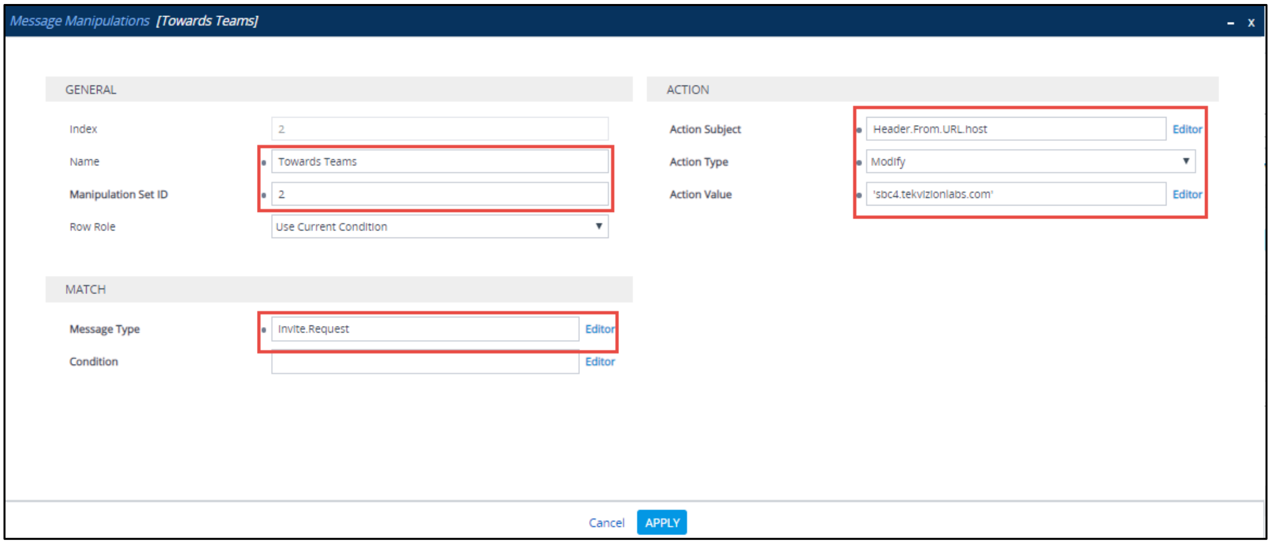

Figure 98 – SIP Message Manipulation - From

Manipulation to PSTN

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

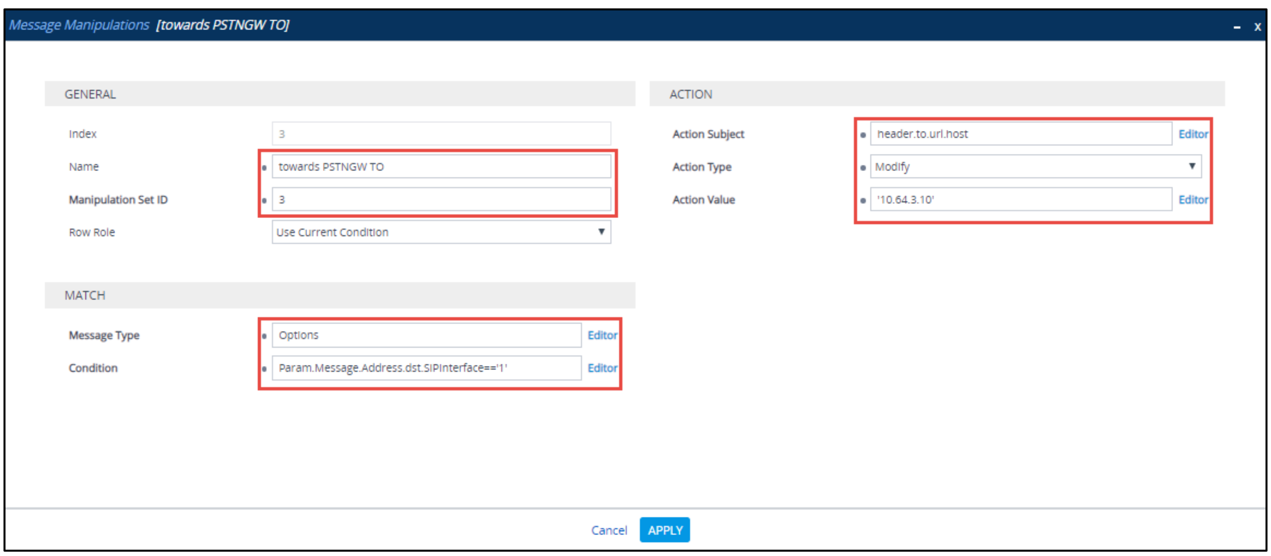

Figure 99 – SIP Message Manipulation – To

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

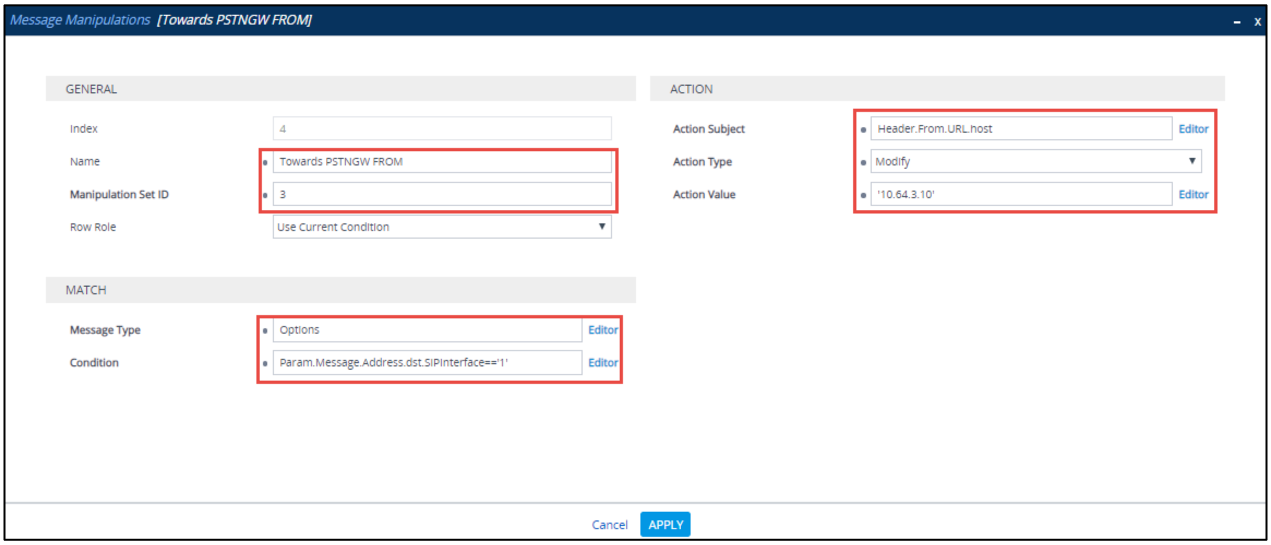

Figure 100 – SIP Message Manipulation – From

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

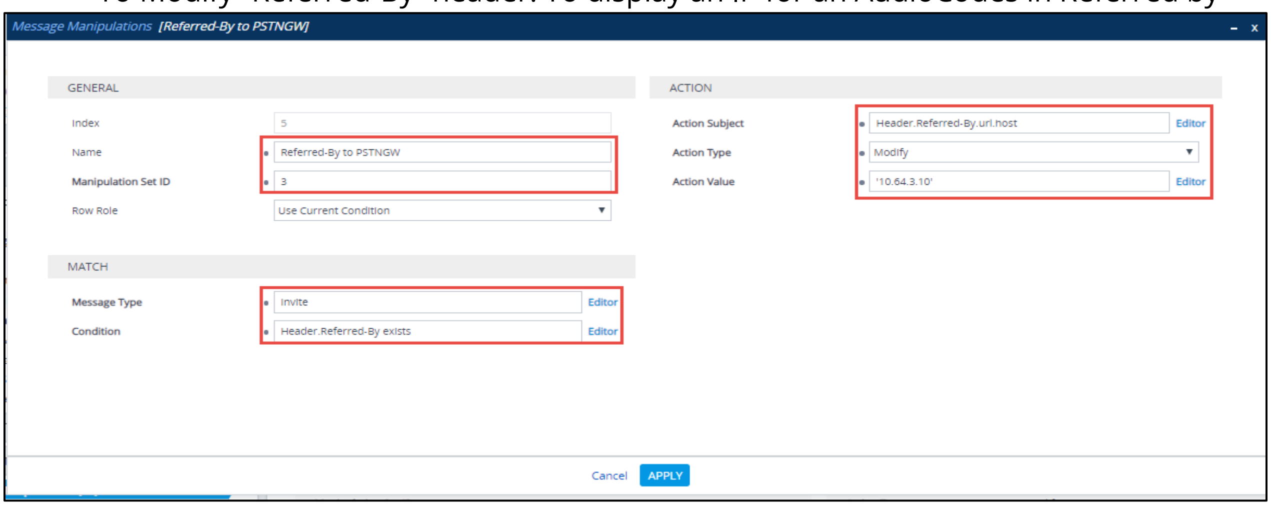

Figure 101 – SIP Message Manipulation – Referred – By

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

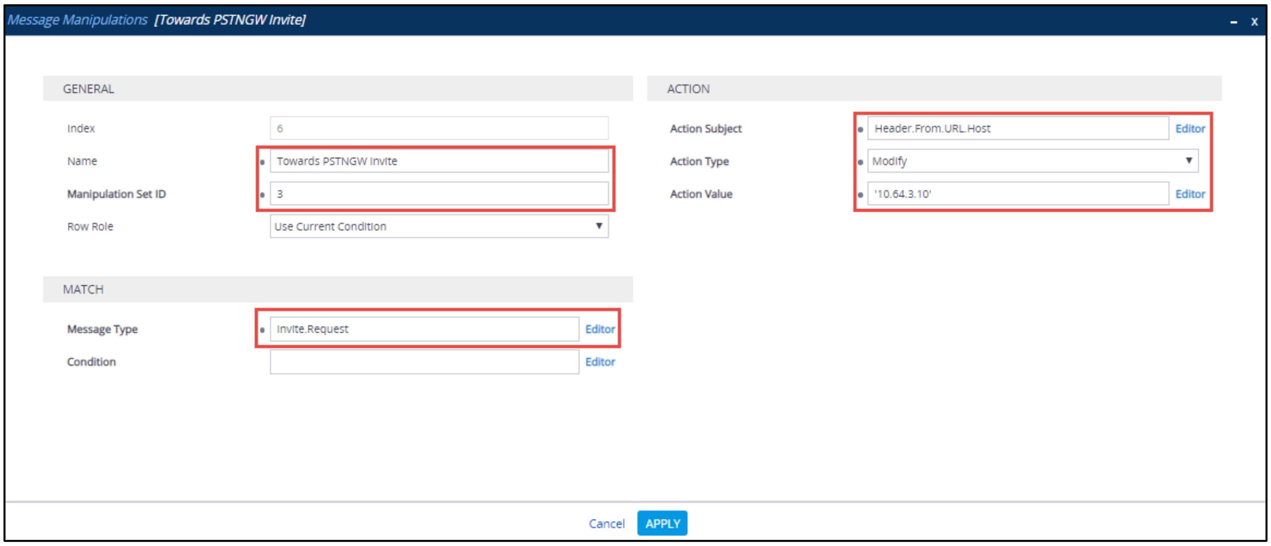

Figure 102 – SIP Message Manipulation – From

Manipulation to Cisco

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

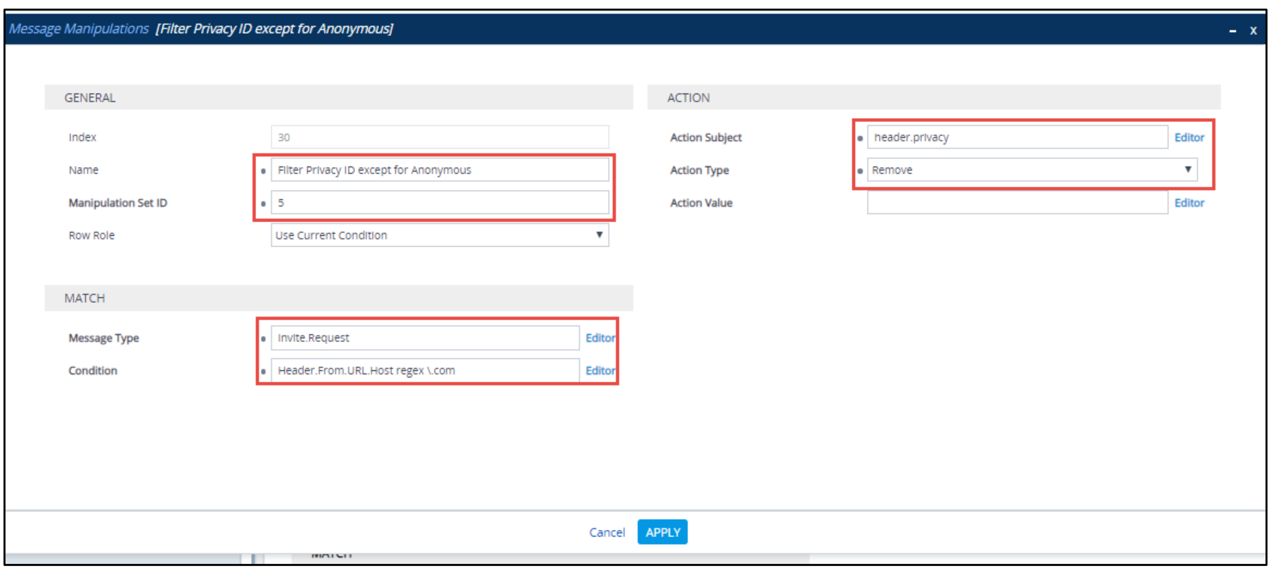

Figure 103 – SIP Message Manipulation – Privacy

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

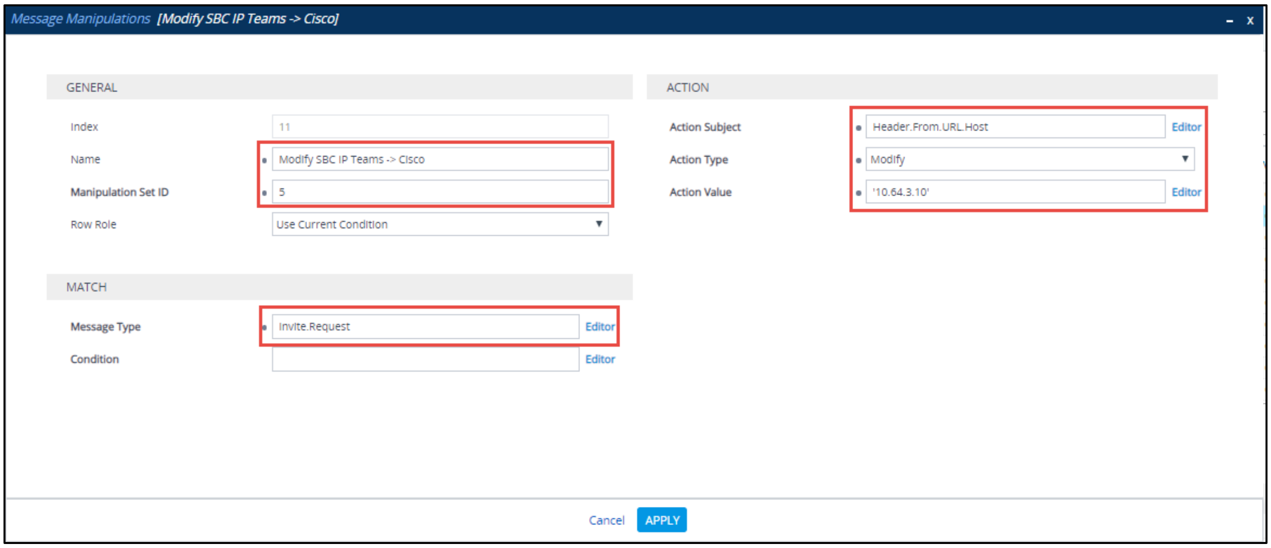

Figure 104 – SIP Message Manipulation – From

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

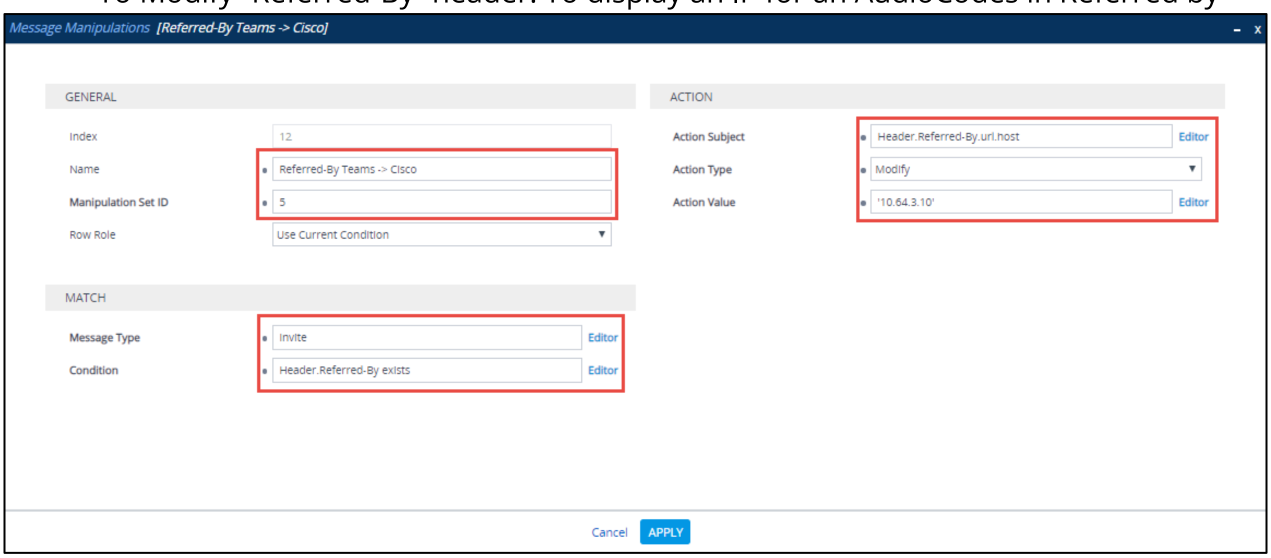

Figure 105 – SIP Message Manipulation – Referred By

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

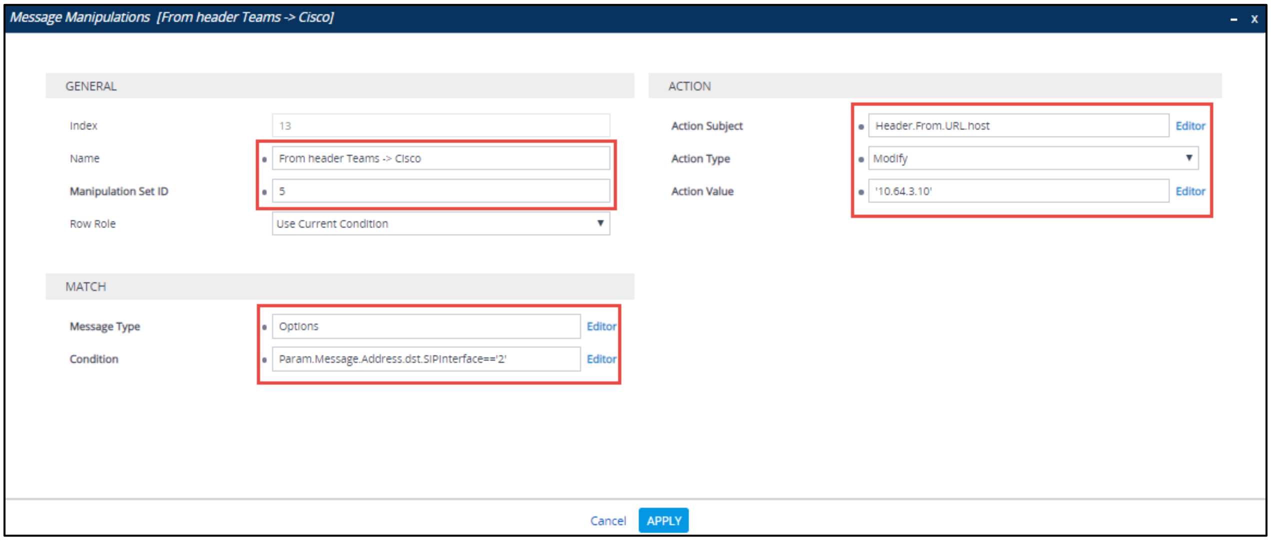

Figure 106 – SIP Message Manipulation – From

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

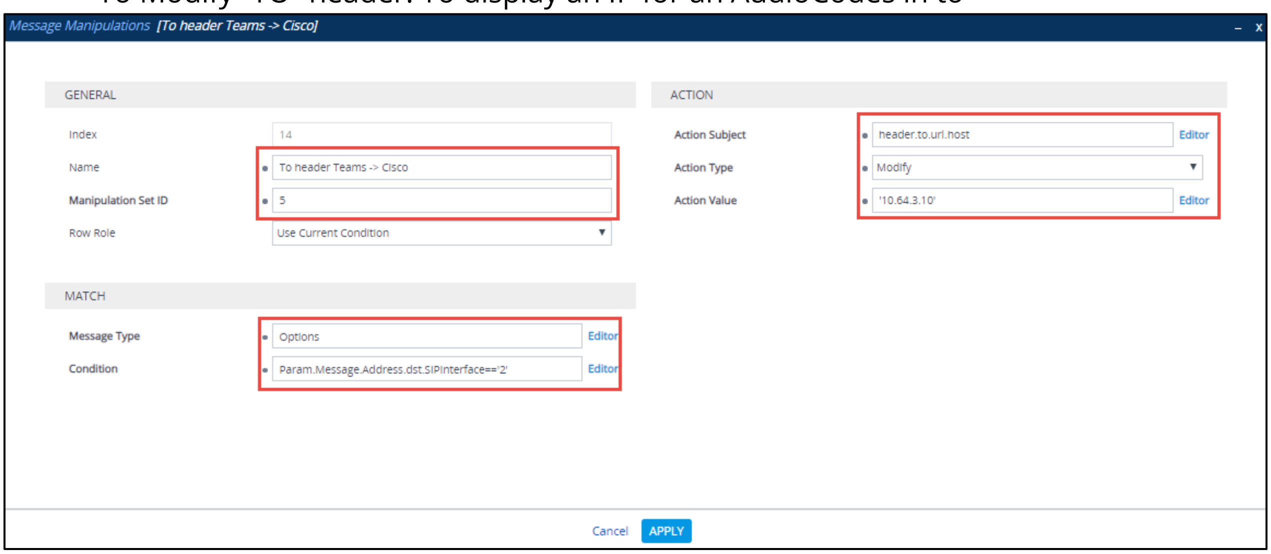

Figure 107 – SIP Message Manipulation – to

4.5 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-hiding[1]
 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#
```

4.6 Cisco UCM Configuration

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

4.6.1 Version

Cisco UCM version

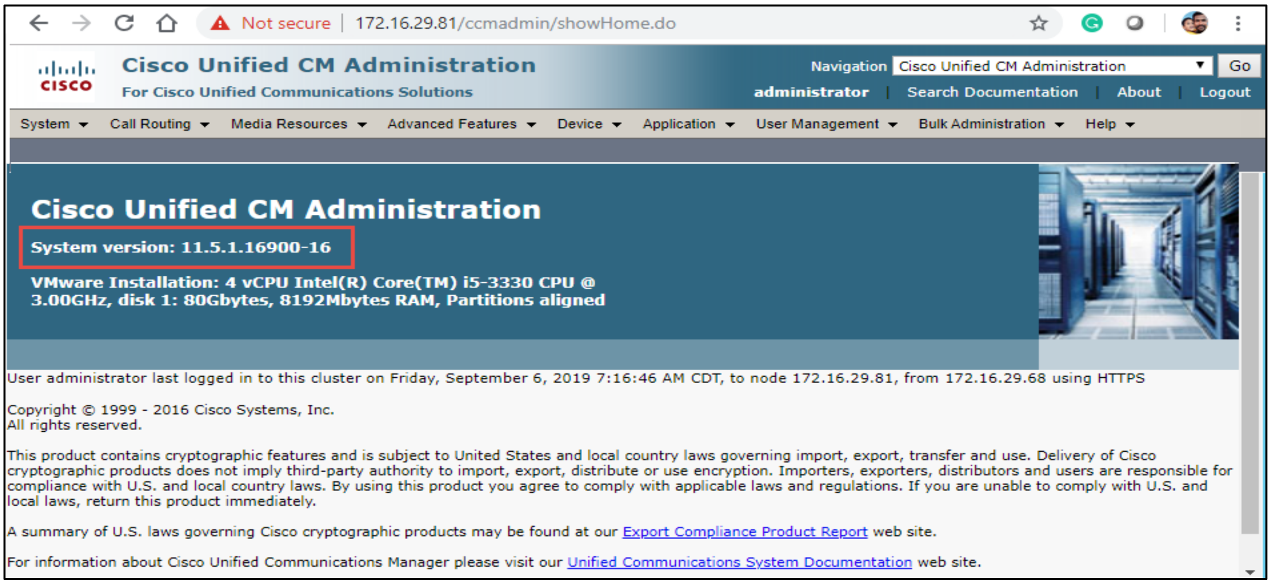

Figure 108 – Cisco UCM Version

4.6.2 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)

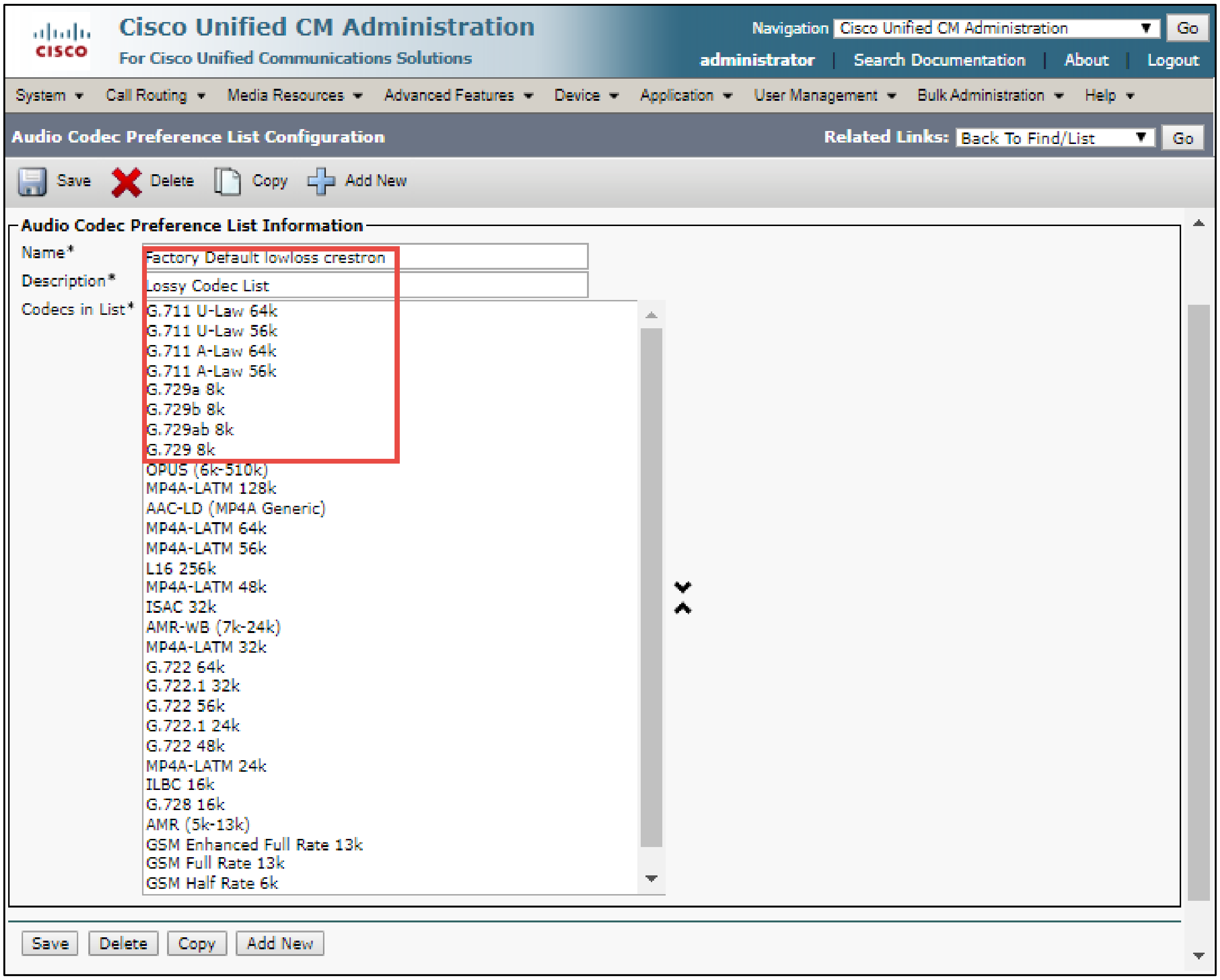

Figure 109 – Audio Codec Preference List

4.6.3 Cisco UCM Region Configuration

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

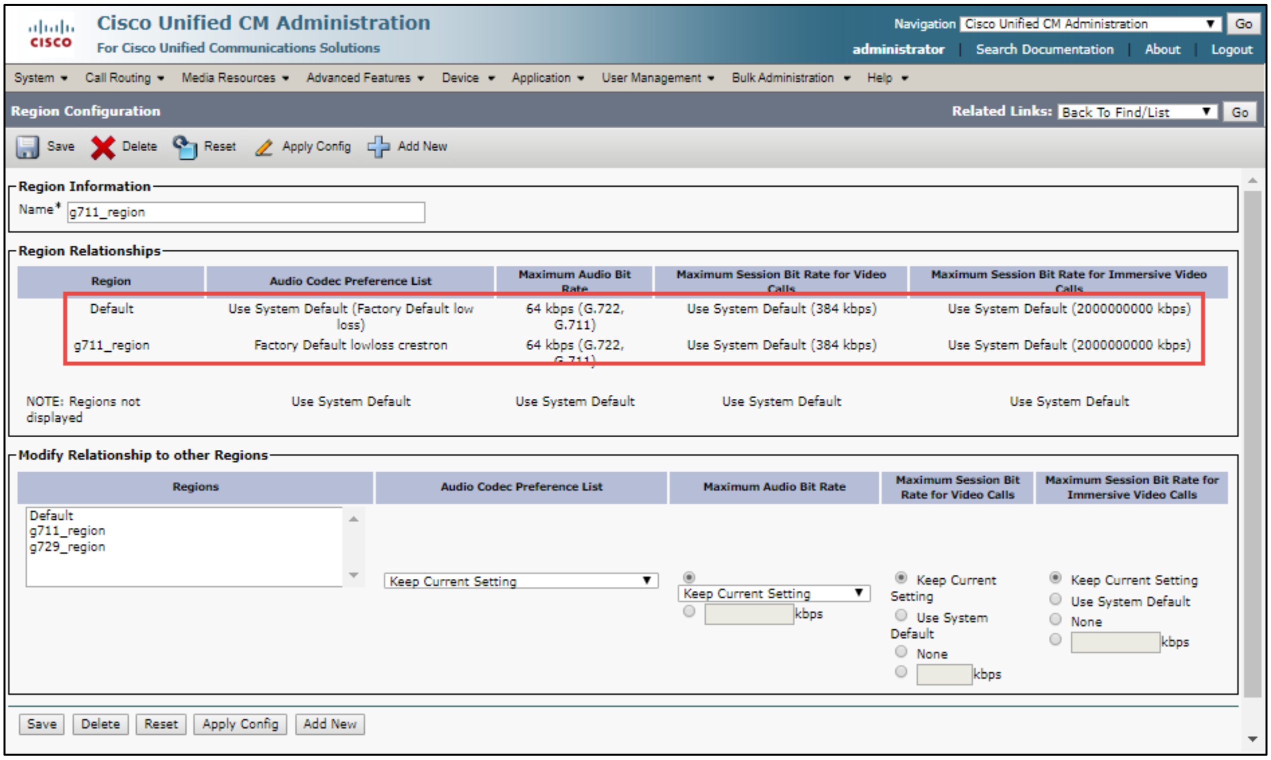

Figure 110 – Cisco UCM Region

4.6.4 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

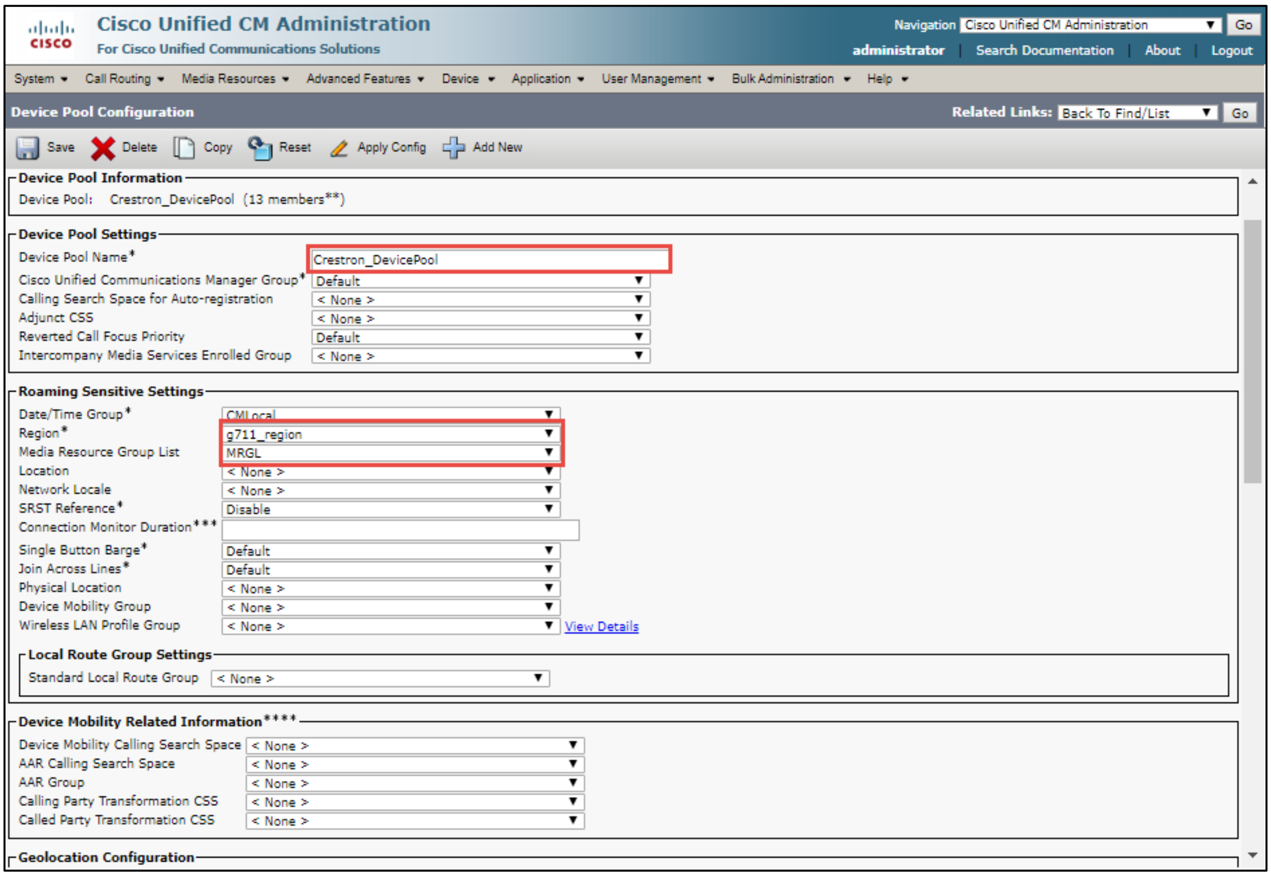

Figure 111 – Cisco UCM Device Pool – Contd.

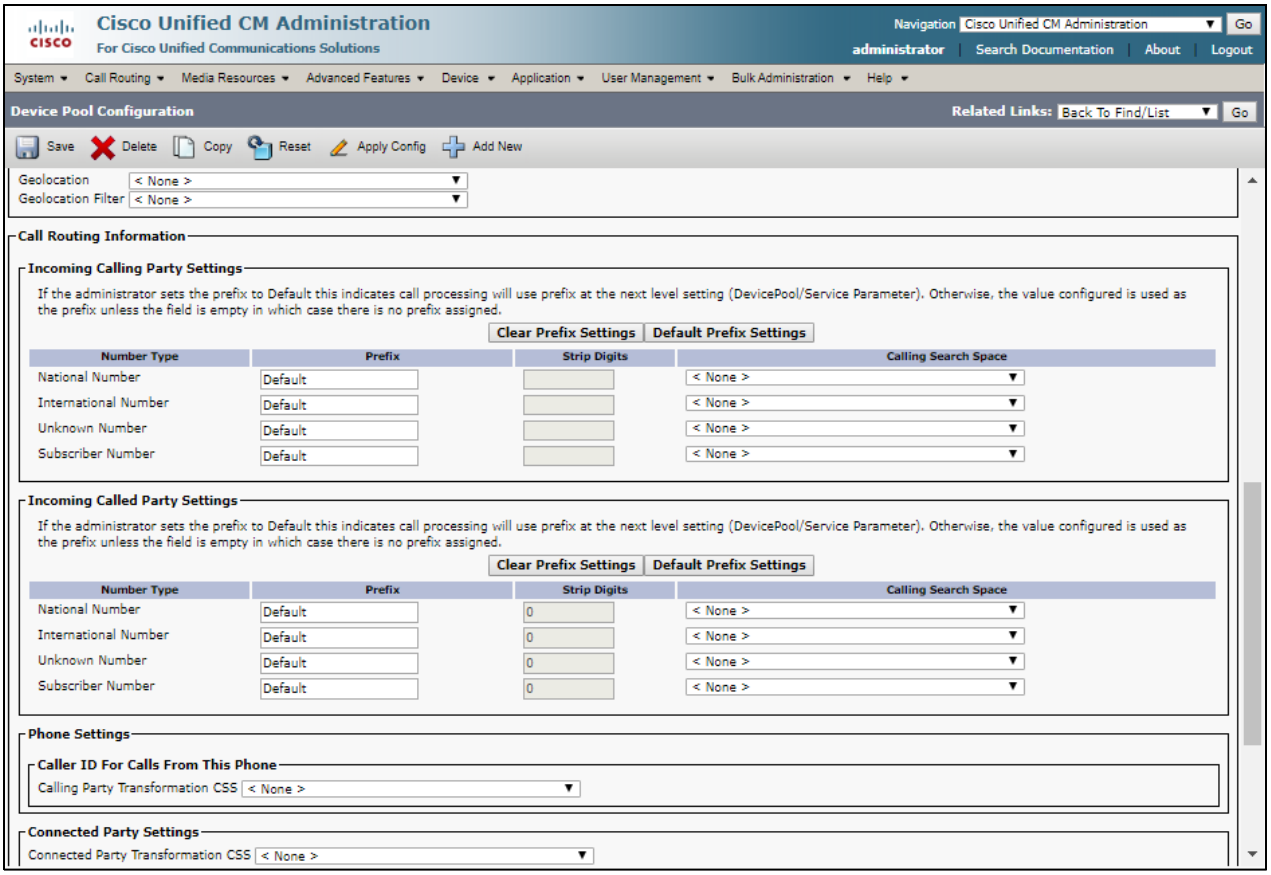

Figure 112 – Cisco UCM Device Pool

4.6.5 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

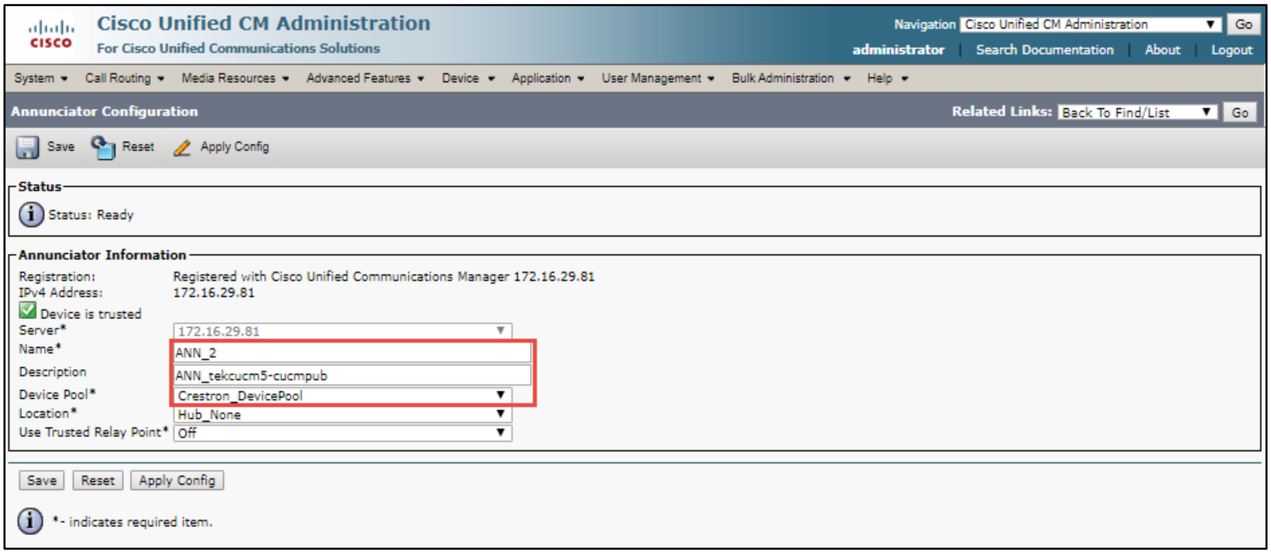

Figure 113 – Cisco UCM Annunciator

4.6.6 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

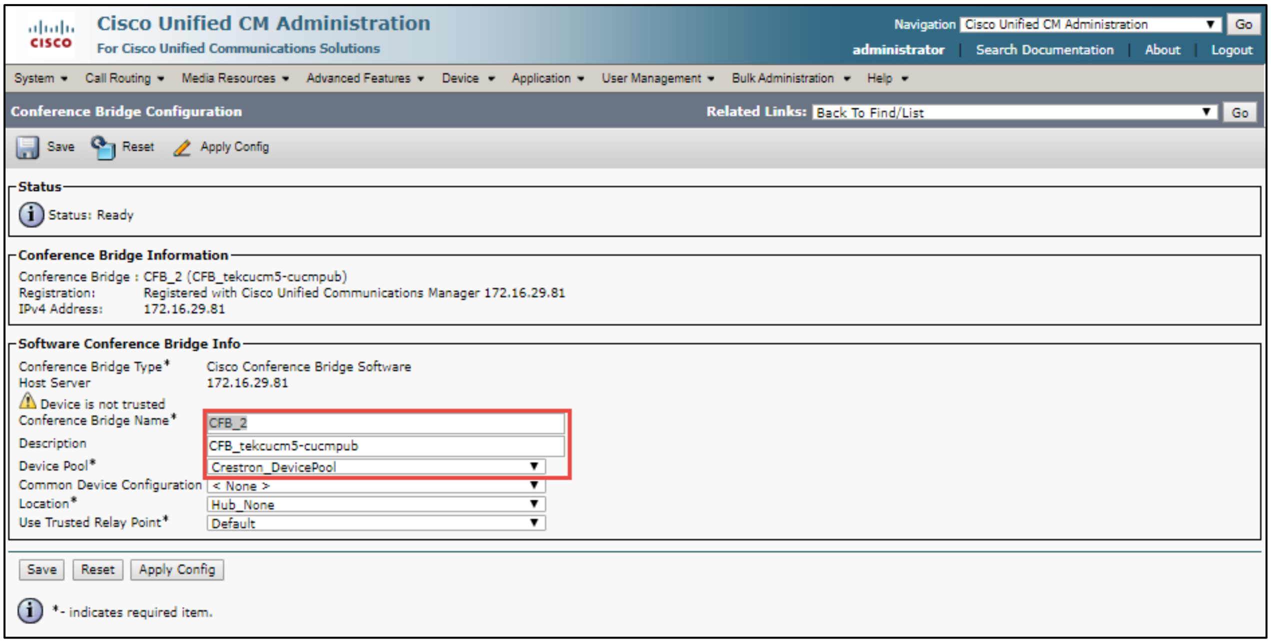

Figure 114 – Cisco UCM Conference Bridge

4.6.7 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

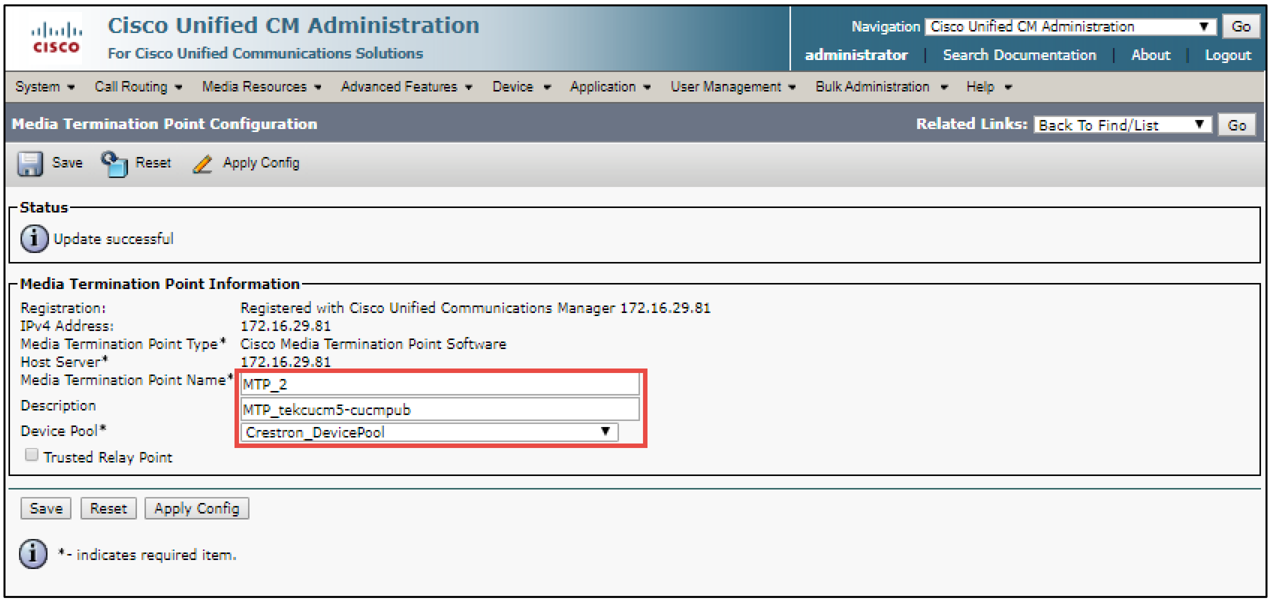

Figure 115 – Cisco UCM MTP

4.6.8 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

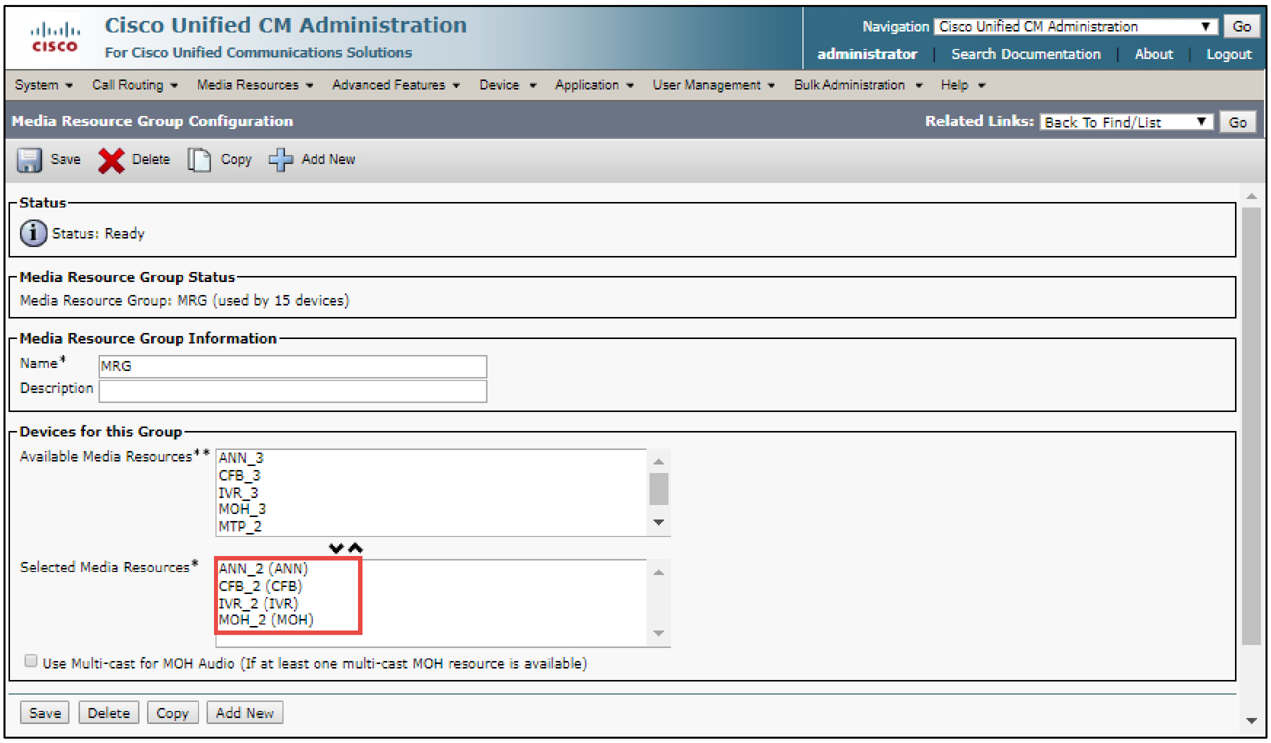

Figure 116 – Cisco UCM MTP

4.6.9 Cisco Media Resource Group List

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

Set Name* = MRGL
Selected Media Resources Groups = MRG

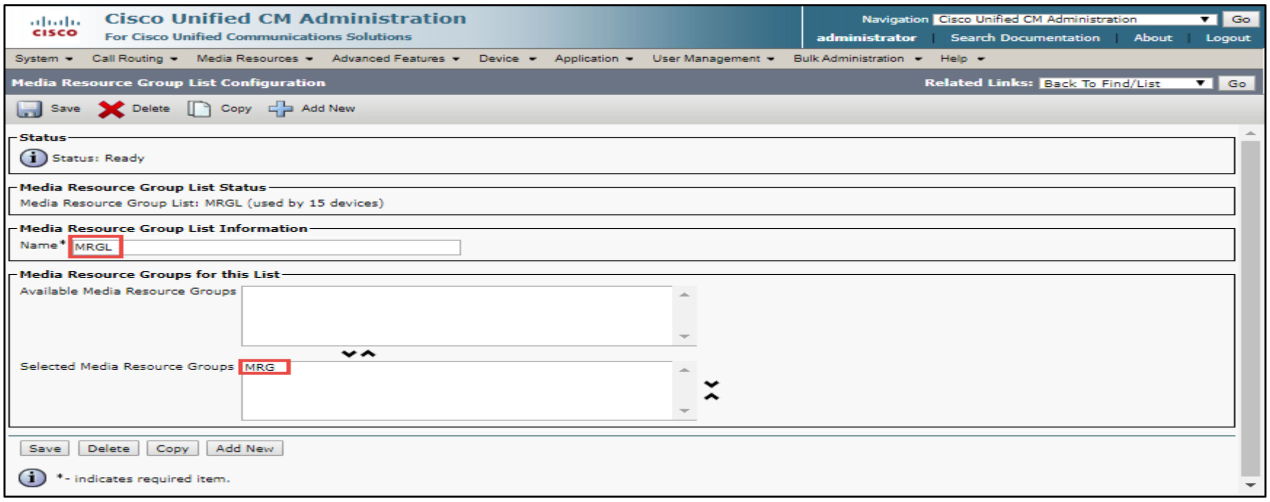

Figure 117 – Cisco UCM MRGL

4.6.10 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

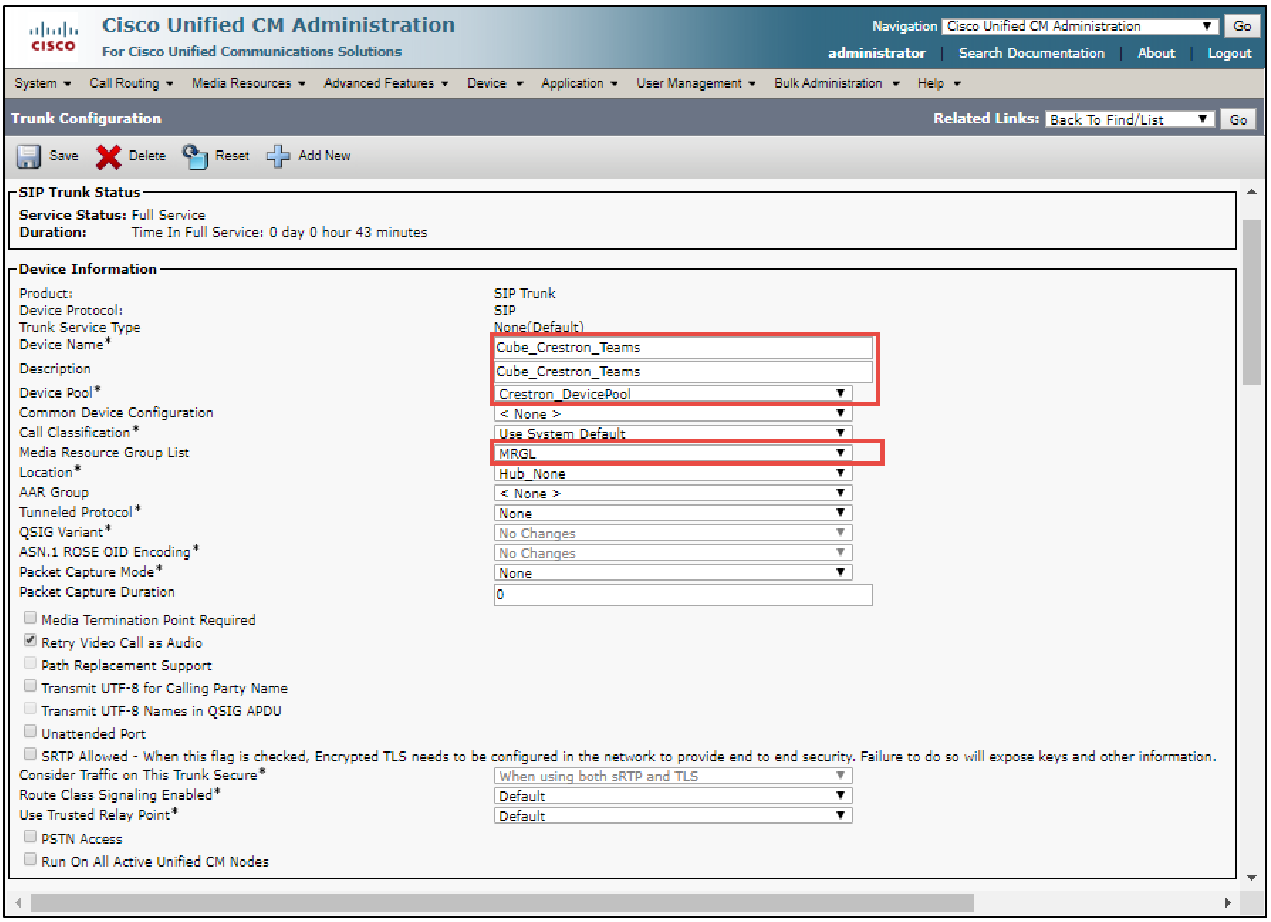

Figure 118 – SIP Trunk – Cisco UBE – Contd.

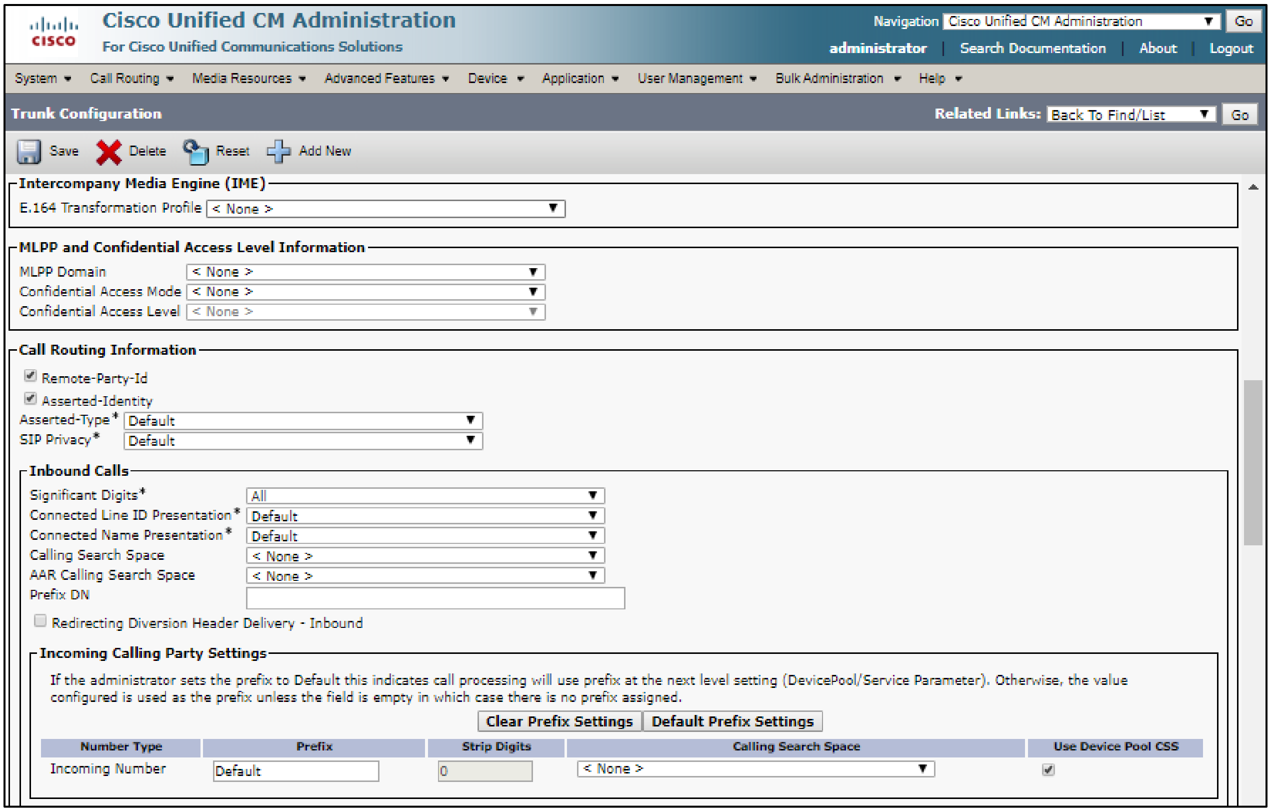

Figure 119 – SIP Trunk – Cisco UBE – Contd.

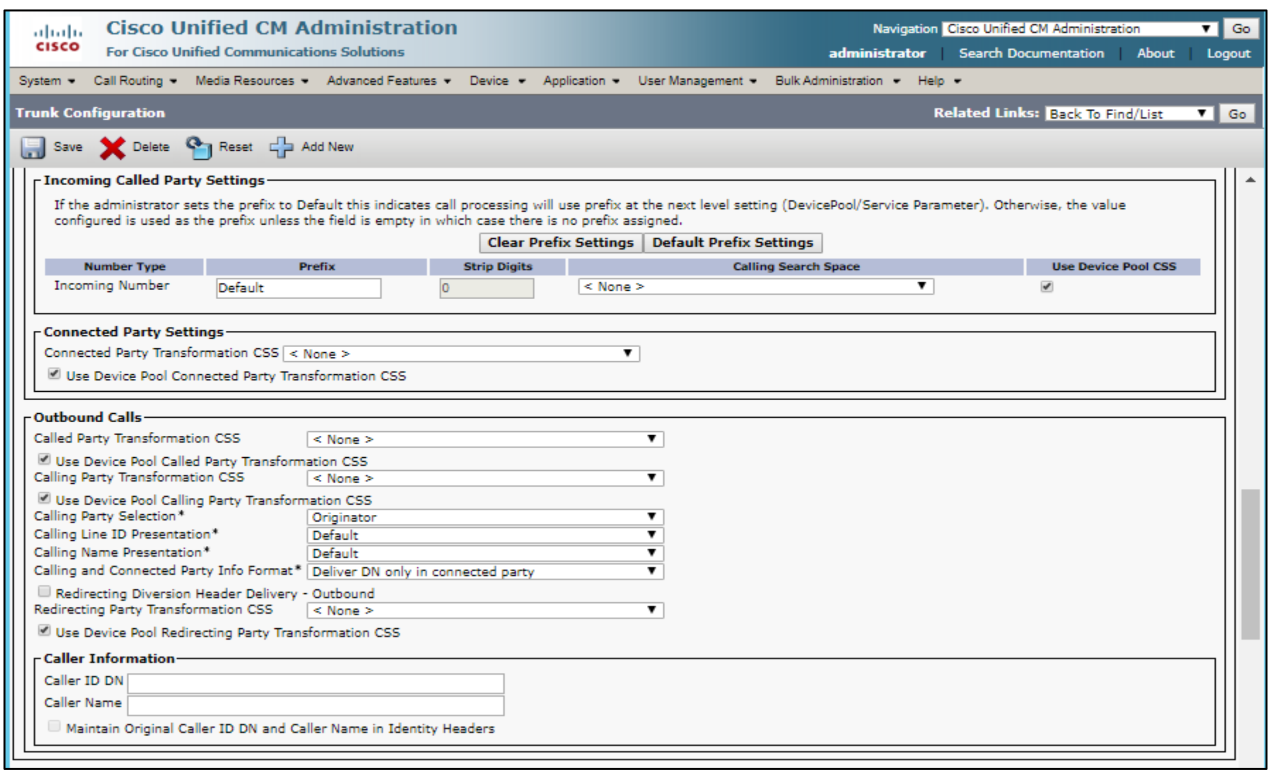

Figure 120 – SIP Trunk – Cisco UBE – Contd.

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.

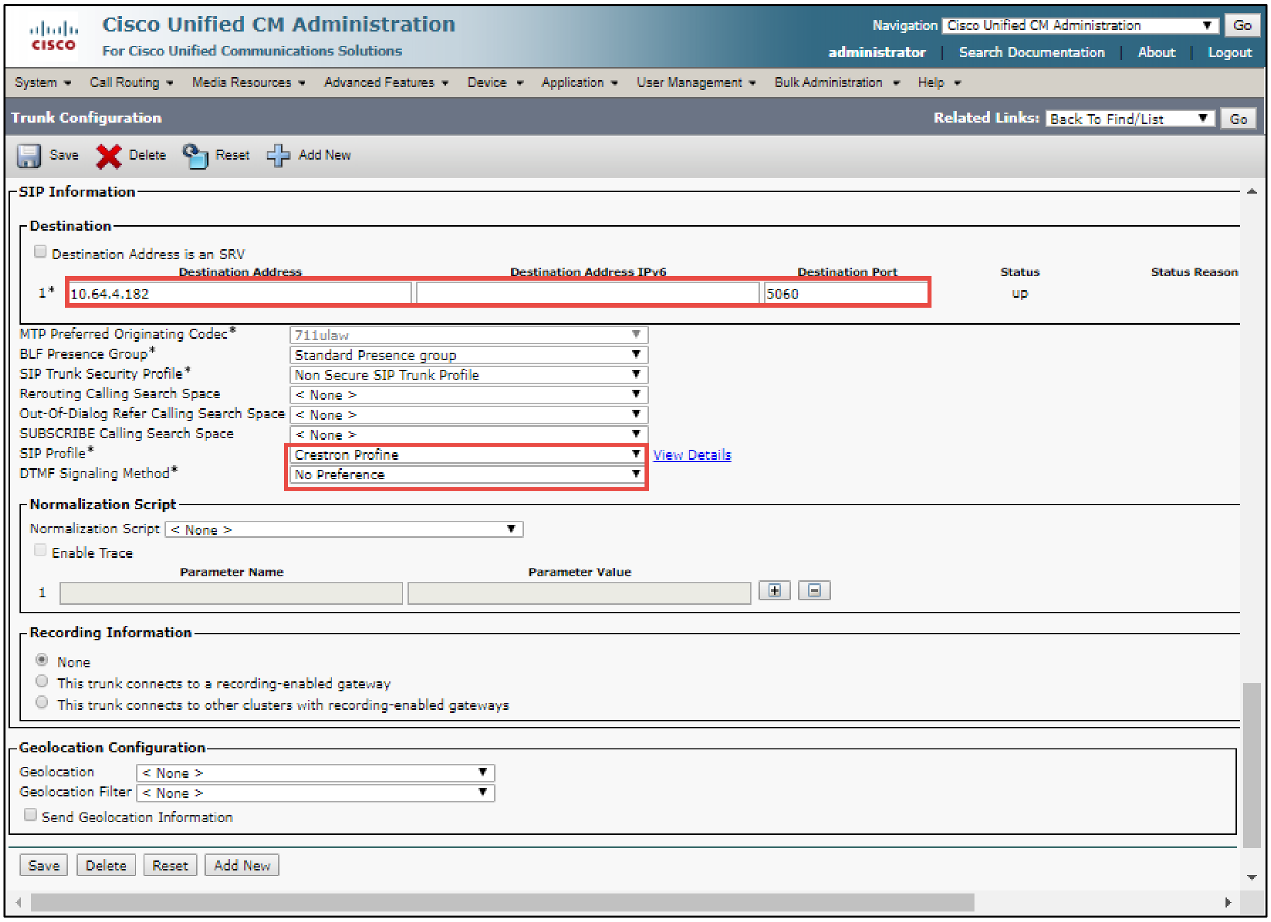

Figure 121 – SIP Trunk – Cisco UBE

4.6.11 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

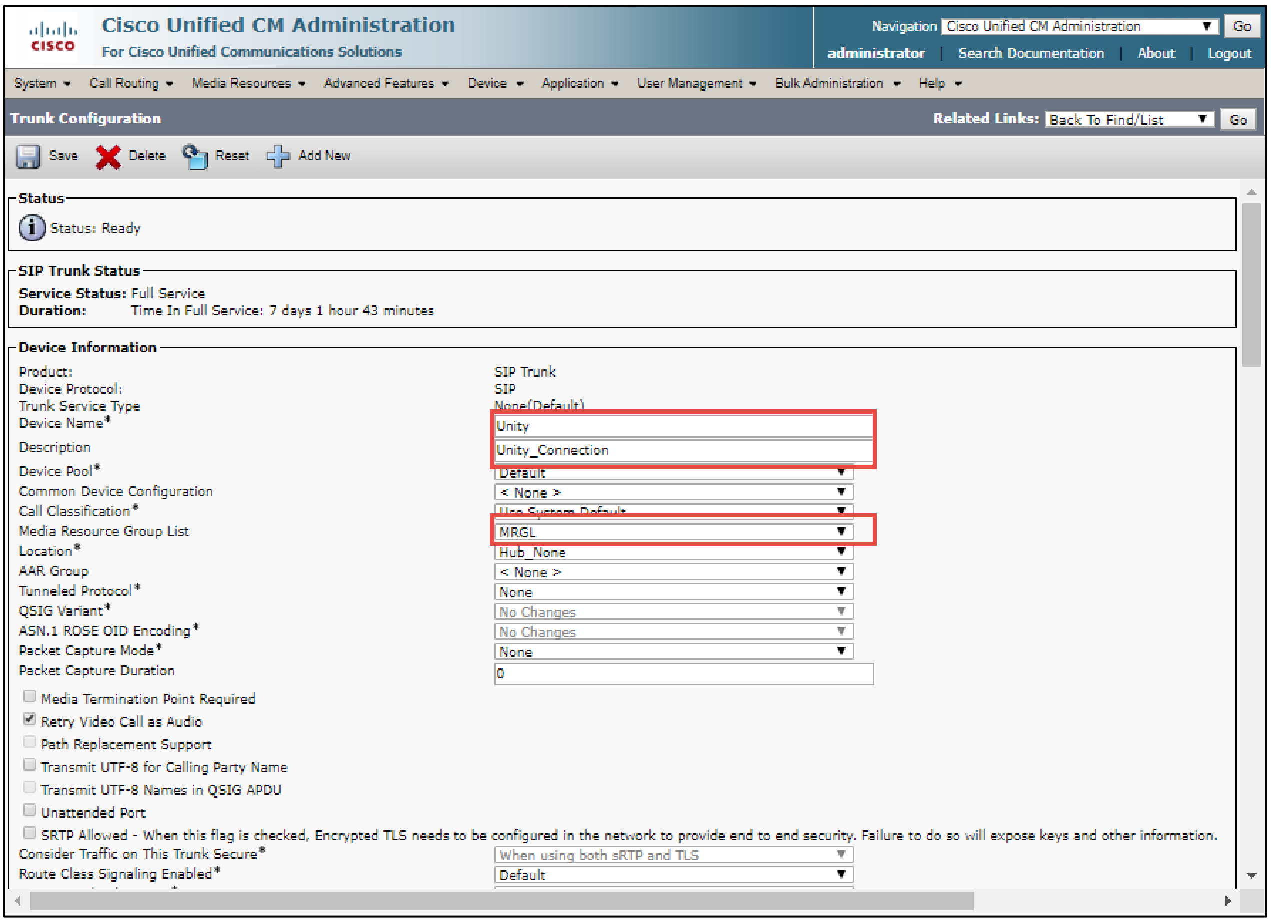

Figure 122 – SIP Trunk – Cisco Unity – Contd.

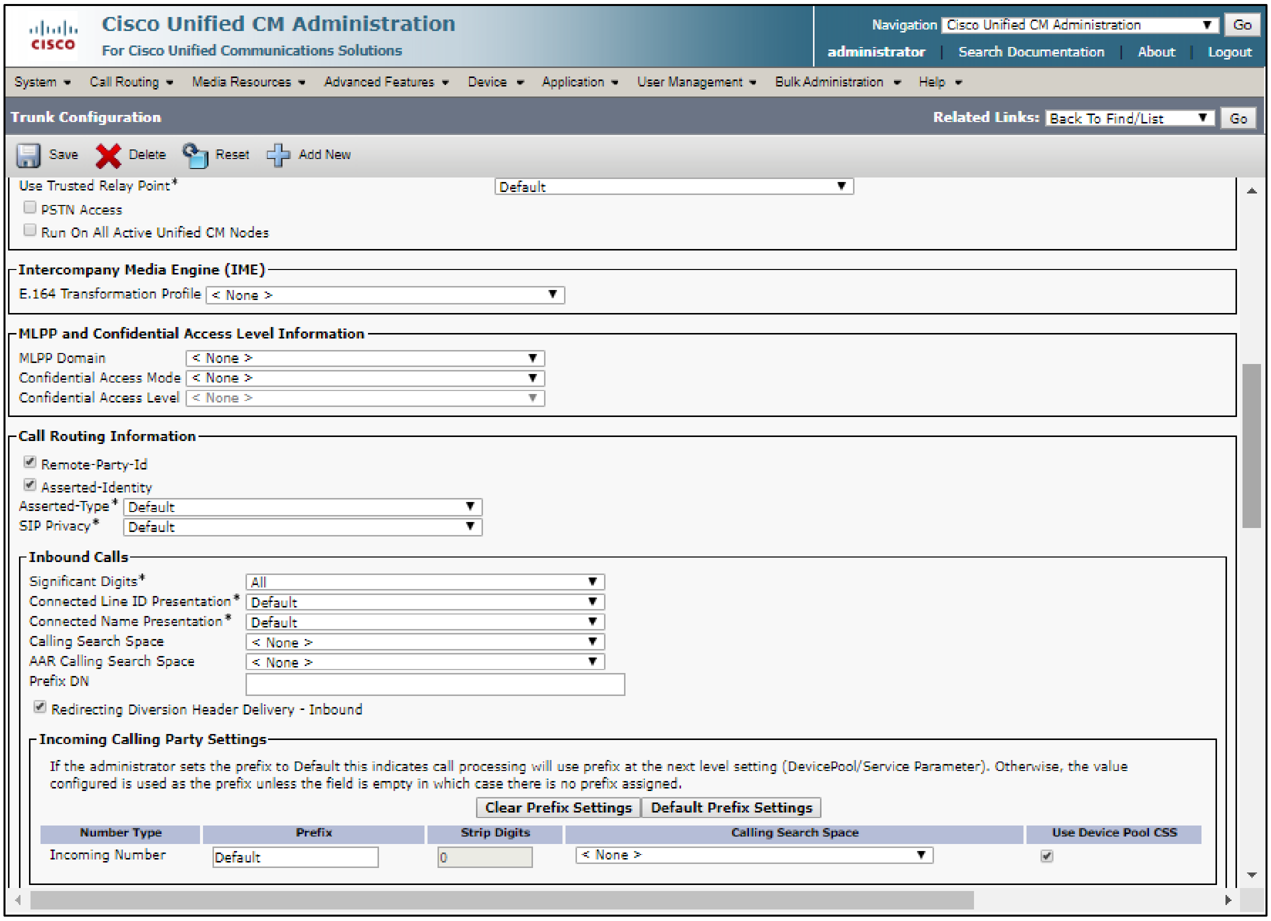

Figure 123 – SIP Trunk – Cisco Unity – Contd.

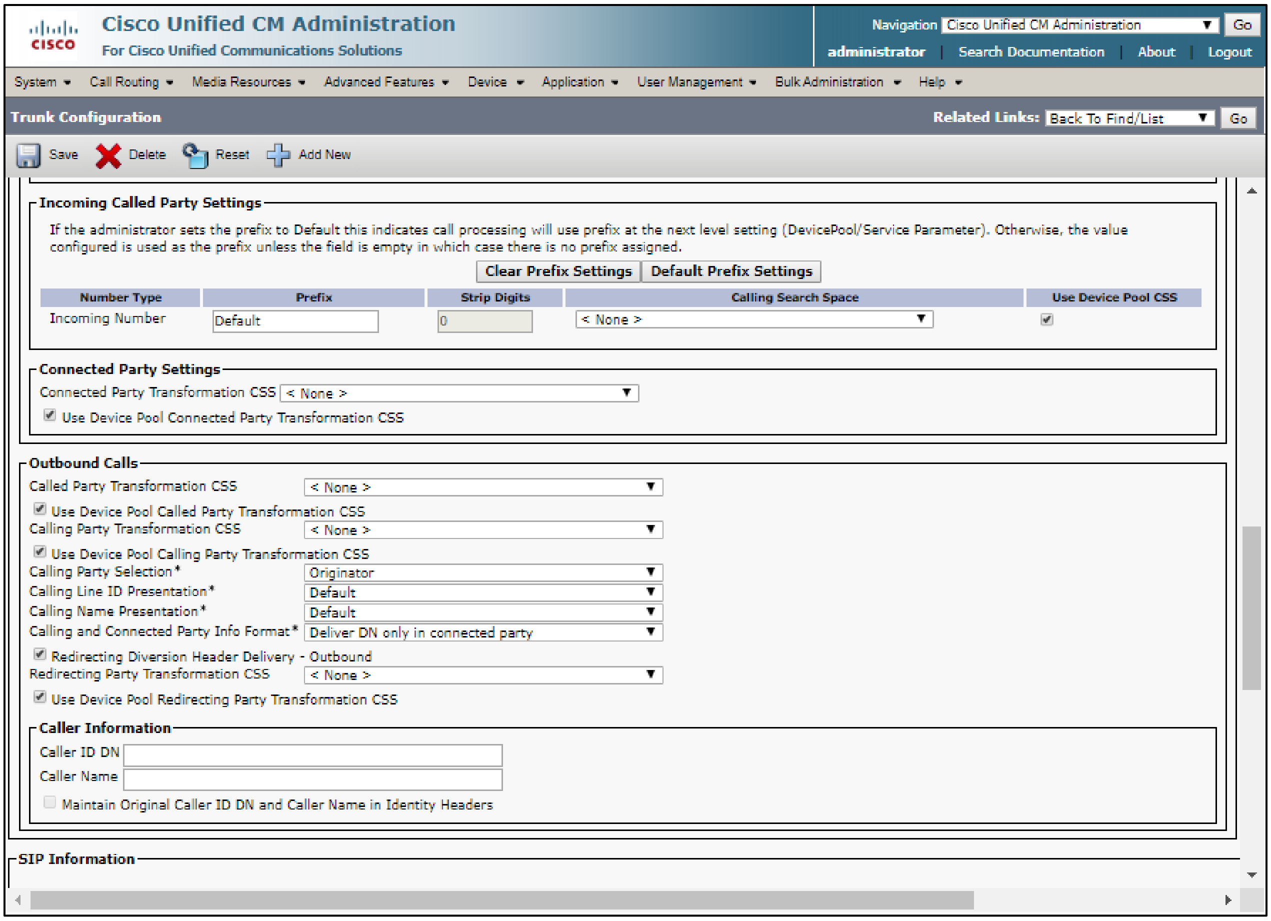

Figure 124 – SIP Trunk – Cisco Unity – Contd.

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.

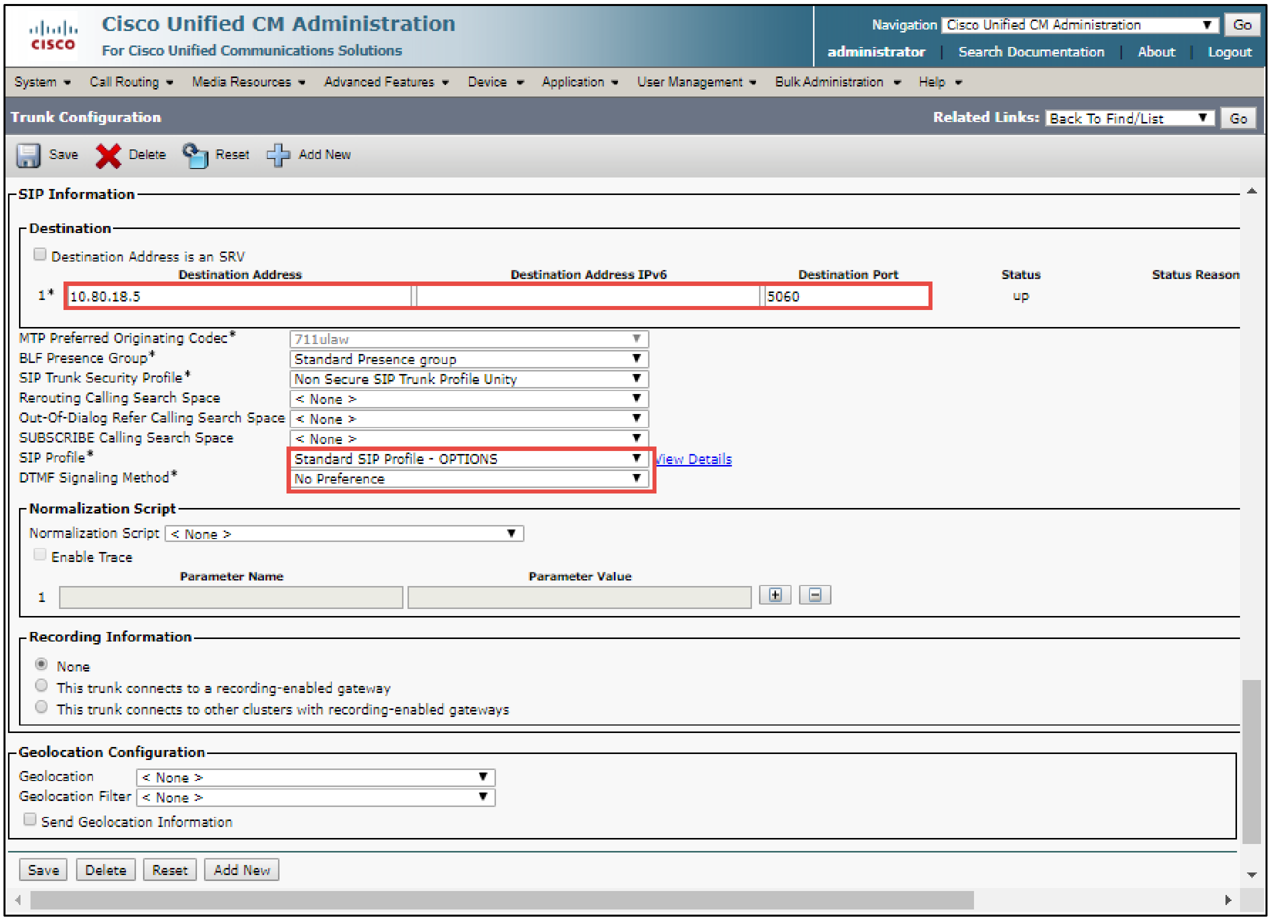

Figure 125 – SIP Trunk – Cisco Unity

4.7 Cisco Unity Connection (CUC)

4.7.1 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

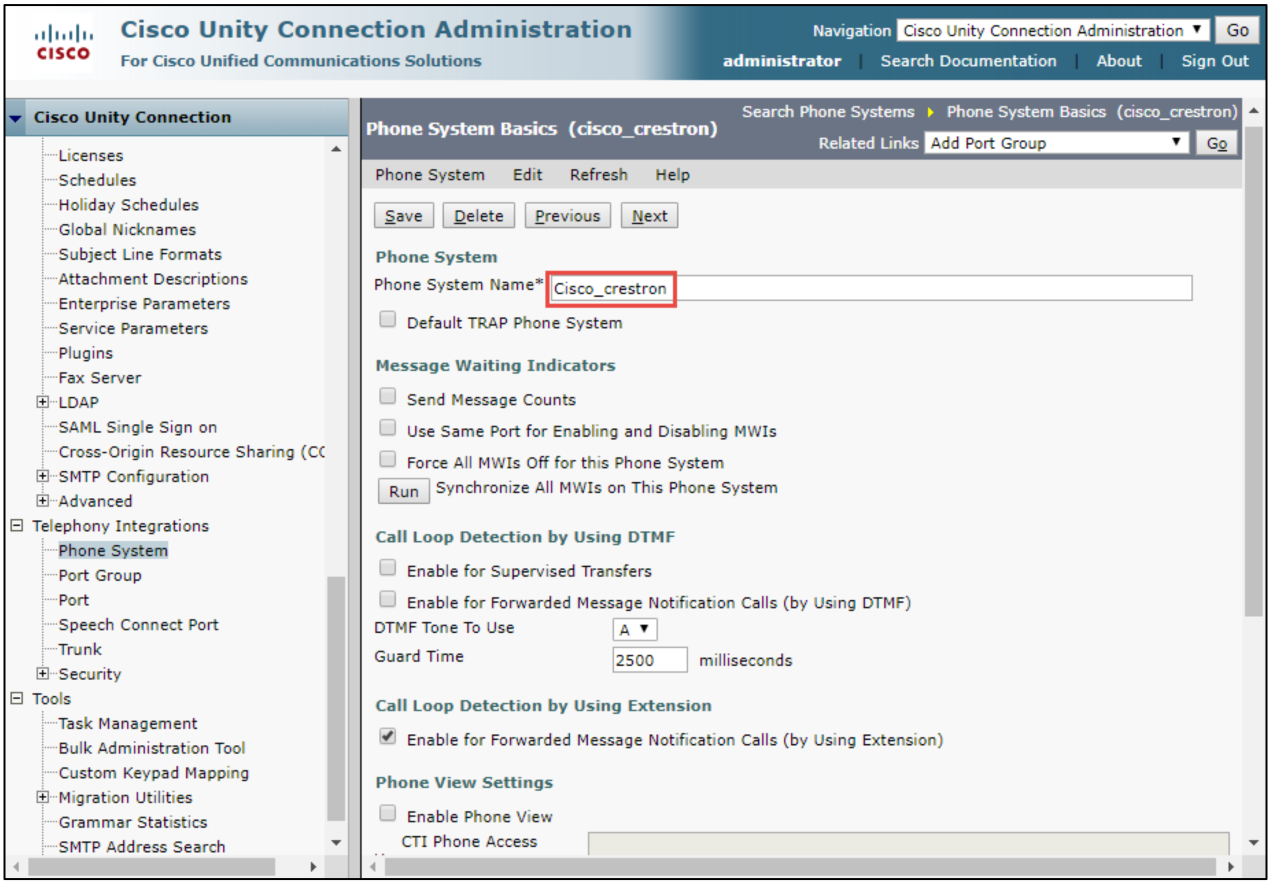

Figure 126 – SIP Trunk – Phone System – Contd.

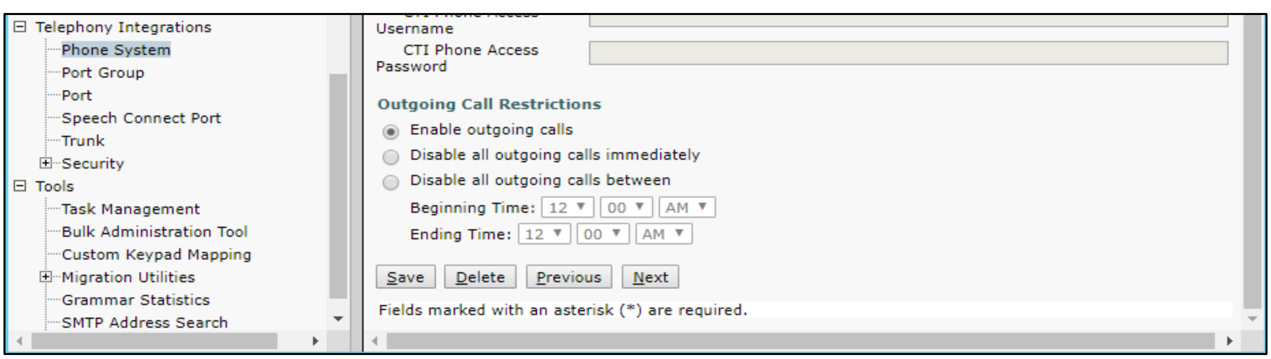

Figure 127 – SIP Trunk – Phone System – Contd.

4.7.2 Phone Group

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

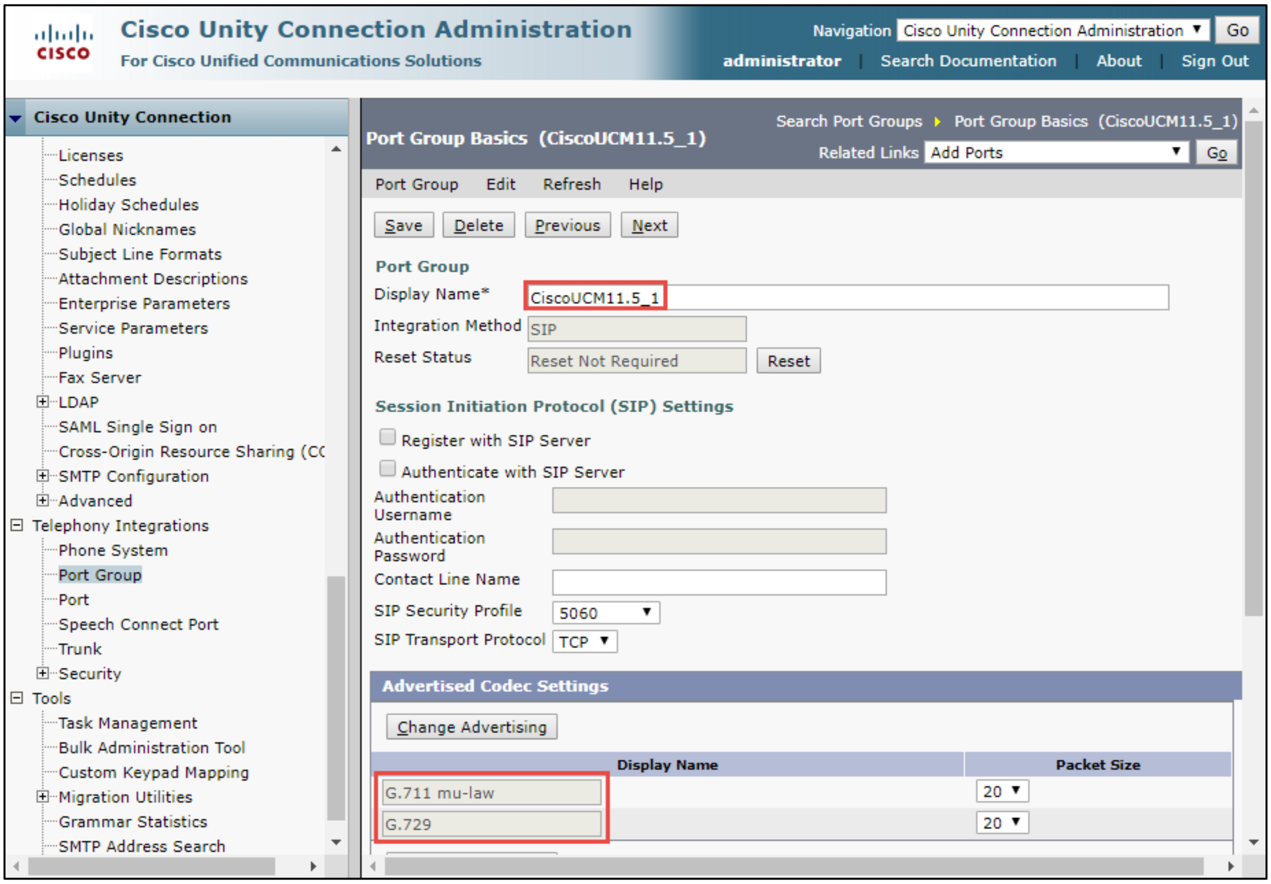

Figure 128 –Phone Group – Contd.

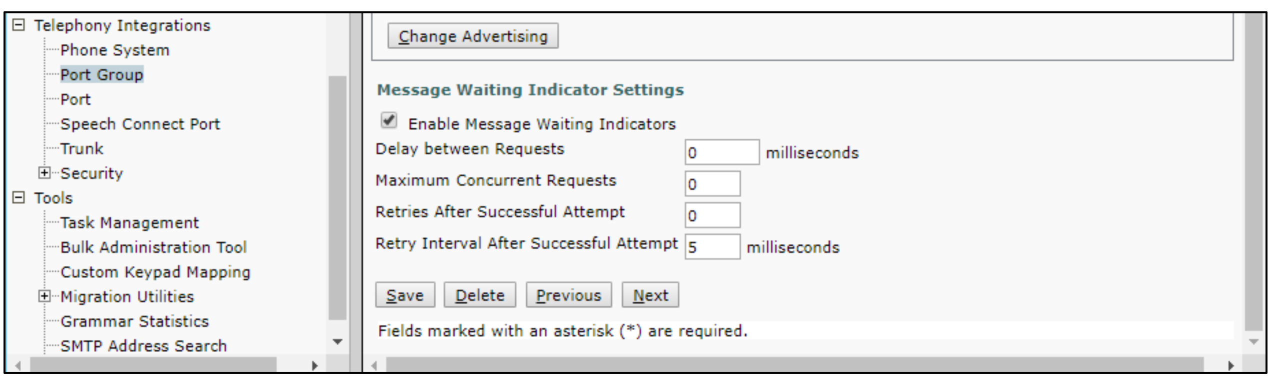

Figure 129 –Phone Group

4.7.3 Port

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

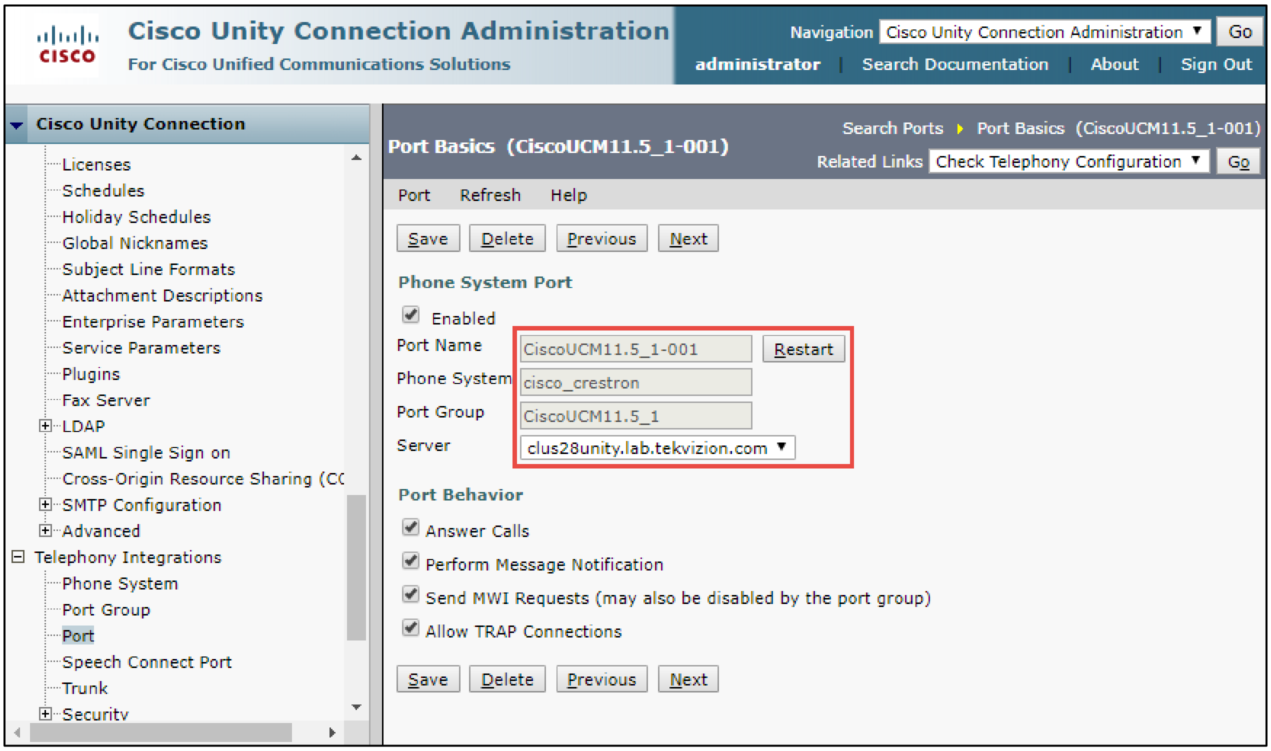

Figure 130 – Port

4.7.4 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.

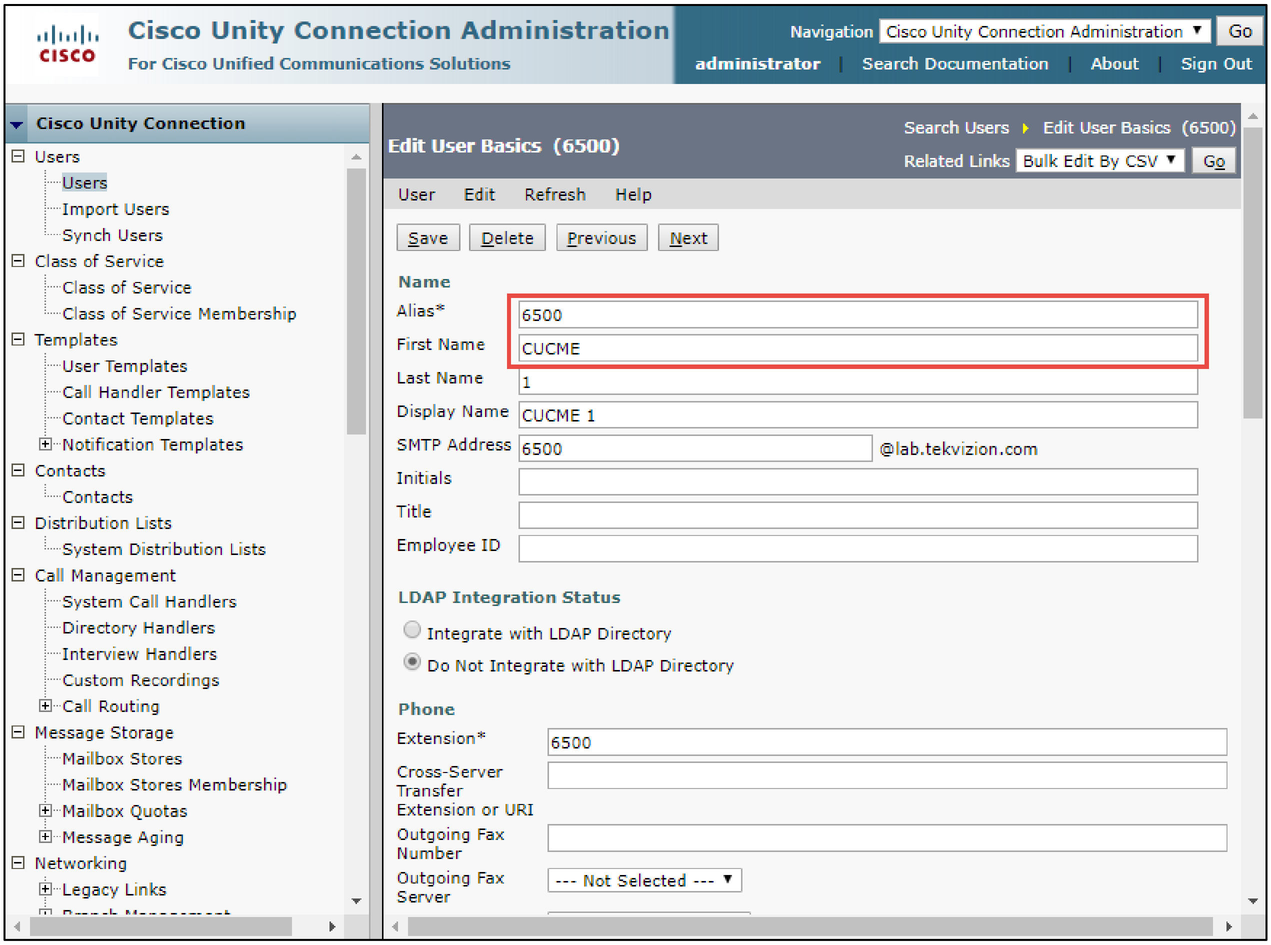

Figure 131 – User

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

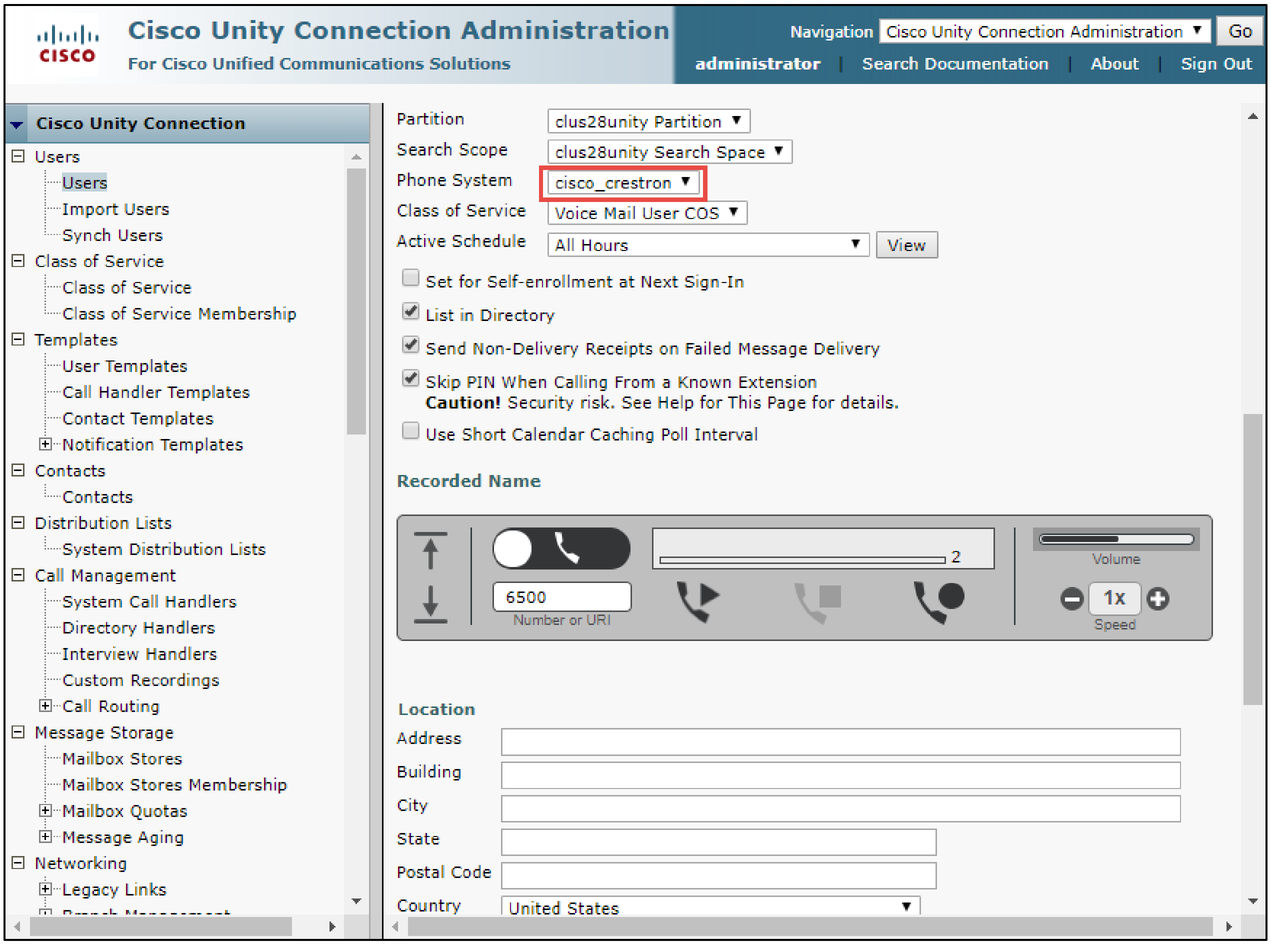

Figure 132 – User

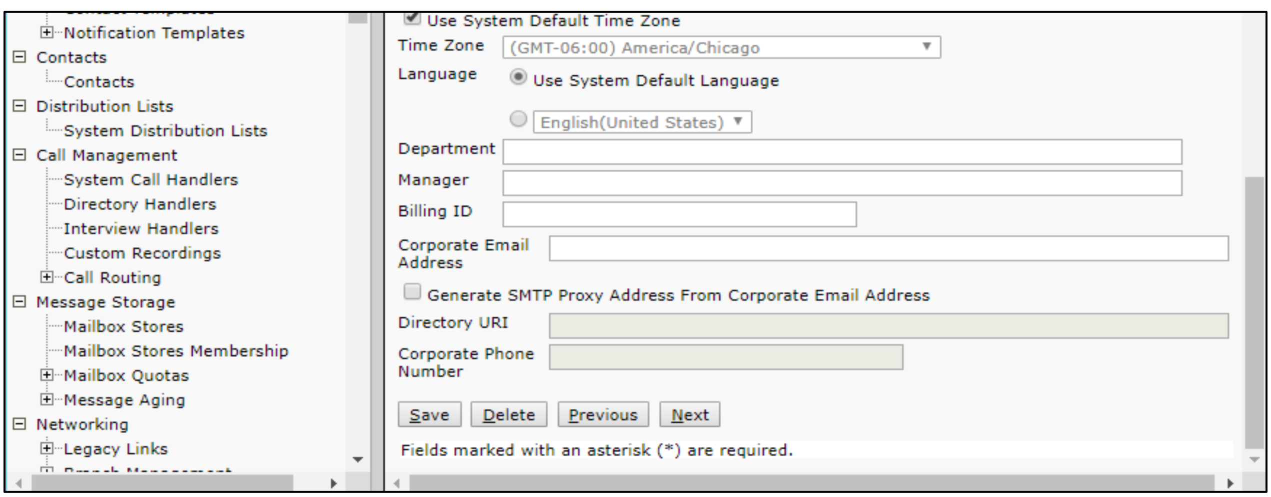

Figure 133 – User

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 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PBX A user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	PASSED	
2	Teams user Calls PBX B user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PBX B user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	NOT APPLICABLE	This testing is for only one PBX with Teams
3	Teams user Calls PSTN user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio	1. Call is connected with bi-directional audio, voice is clear, no echo	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PSTN user hangs up the call 9. Verify call is cleared successfully	2. Call is disconnected		
4	Teams user Calls PBX A user and hangs up before answer	1. Make a voice call from Teams user to PBX A user 2. PBX A user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX A user is ringing 5. PBX A user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
5	Teams user Calls PBX B user and hangs up before answer	1. Make a voice call from Teams user to PBX B user 2. PBX B user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX B user is ringing 5. PBX B user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	NOT APPLICABLE	This testing is for only one PBX with Teams
6	Teams user Calls PSTN user and hangs up before answer	1. Make a voice call from Teams user to PSTN user 2. PSTN user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PSTN user is ringing	1. Call is disconnected before answer	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		5. PSTN user stops ringing 6. Verify call is cleared successfully			
7	PBX A user Calls Teams user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. PBX A user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. Teams user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	PASSED	
8	PBX B user Calls Teams user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. PBX B user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. Teams user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
9	PSTN user Calls Teams user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. PSTN user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. Teams user hangs up the call 9. Verify call is cleared successfully	1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected	PASSED	
10	PBX A user Calls Teams user and hangs up before answer	1. Make a voice call from PBX A user to Teams user 2. Teams user starts ringing 3. PBX A user hears Ring back Tone 4. PBX A user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
11	PBX B user Calls Teams user and hangs up before answer	1. Make a voice call from PBX B user to Teams user 2. Teams user starts ringing 3. PBX B user hears Ring back Tone 4. PBX B user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
12	PSTN user Calls Teams user and hangs up before answer	1. Make a voice call from PSTN user to Teams user 2. Teams user starts ringing 3. PSTN user hears Ring back Tone 4. PSTN user hangs up the call while Teams user is ringing 5. Teams user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
13	Teams user Calls PBX A user and performs hold/resume	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. Teams user hangs up the call 10. Verify call is cleared successfully	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	PASSED	
14	Teams user Calls PBX B user and performs hold/resume	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
		8. Verify two way audio is re-established between the two end points 9. PBX A user hangs up the call 10. Verify call is cleared successfully			receiver or speaker button.
17	PBX B user Calls Teams user and Teams user performs hold/resume	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points 9. PBX B user hangs up the call 10. Verify call is cleared successfully	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	NOT APPLICABLE	PBX B is not tested with this cycle
18	PSTN user Calls Teams user and Teams performs hold/resume	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user initiates call hold 6. Verify no audio is present while call is on hold 7. Teams user resumes the call 8. Verify two way audio is re-established between the two end points	1. Call is placed on hold successfully 2. No audio present during hold 3. Call is resumed successfully 4. Two way audio present after call is resumed	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.

External ID	Title	Procedure	Expected Results	Status	Comments
		9. PSTN user hangs up the call 10. Verify call is cleared successfully			
19	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from Teams user to PBX A user 1 2. Teams user hears Ring back Tone 3. PBX A user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 2 6. Verify PBX A user 1 is placed on hold 7. PBX A user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user 1 and PBX A user 2 11. PBX A user 1 hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
20	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PBX A user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
		10. Verify two way audio between PBX A user and PBX B user 11. PBX A user hangs up the call 12. Verify call is cleared successfully			
21	Teams user Calls PBX A user, Teams user performs Attended Transfer to PSTN user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 6. Verify PBX A user is placed on hold 7. PSTN user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user and PSTN user 11. PBX A user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
22	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PBX B user 1 2. Teams user hears Ring back Tone 3. PBX B user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 2 6. Verify PBX B user 1 is placed on hold 7. PBX B user 2 answers the call	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
		8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user 1 and PBX B user 2 11. PBX B user 1 hangs up the call 12. Verify call is cleared successfully			
23	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 6. Verify PBX B user is placed on hold 7. PBX A user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user and PBX A user 11. PBX B user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
24	Teams user Calls PBX B user, Teams user performs Attended	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
	Transfer to PSTN user	6. Verify PBX B user is placed on hold 7. PSTN user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user and PSTN user 11. PBX B user hangs up the call 12. Verify call is cleared successfully			
25	Teams user Calls PSTN user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PSTN user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user and PBX B user 11. PSTN user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
26	Teams user Calls PSTN user, Teams user performs	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
	user performs Attended Transfer to PBX A user	3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 2 6. Verify PBX A user 1 is placed on hold 7. PBX A user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user 1 and PBX A user 2 11. PBX A user 1 hangs up the call 12. Verify call is cleared successfully	present after call is transferred		
29	PBX A user Calls Teams user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PBX A user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user and PBX B user 11. PBX A user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
30	PBX A user Calls Teams user, Teams user performs Attended Transfer to PSTN user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 6. Verify PBX A user is placed on hold 7. PSTN user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user and PSTN user 11. PBX A user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
31	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 2 6. Verify PBX B user 1 is placed on hold 7. PBX B user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user 1 and PBX B user 2	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
		11. PBX B user 1 hangs up the call 12. Verify call is cleared successfully			
32	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 6. Verify PBX B user is placed on hold 7. PBX A user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX B user and PBX A user 11. PBX B user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
33	PBX B user Calls Teams user, Teams user performs Attended Transfer to PSTN user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PSTN user 6. Verify PBX B user is placed on hold 7. PSTN user answers the call 8. Verify two way audio 9. Teams user completes the transfer	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
		10. Verify two way audio between PBX B user and PSTN user 11. PBX B user hangs up the call 12. Verify call is cleared successfully			
34	PSTN user Calls Teams user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PSTN user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PSTN user and PBX B user 11. PSTN user hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
35	PSTN user Calls Teams user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 6. Verify PSTN user is placed on hold 7. PBX A user answers the call	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
	Transfer to PBX A user	6.PBX A user 2 starts ringing 7. PBX A user 1 hears Ring back Tone 8. PBX A user 2 answers the call 9. Verify two way audio between PBX A user 1 and PBX A user 2 10. PBX A user 1 hangs up the call 11. Verify call is cleared successfully			
38	Teams user Calls PBX A user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PBX A user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PBX A user and PBX B user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
39	Teams user Calls PBX A user, Teams user performs Unattended Transfer to PSTN user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX A user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX A user and PSTN user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
40	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from Teams user to PBX B user 1 2. Teams user hears Ring back Tone 3. PBX B user 1 answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 2 6.PBX B user 2 starts ringing 7. PBX B user 1 hears Ring back Tone 8. PBX B user 2 answers the call 9. Verify two way audio between PBX B user 1 and PBX B user 2 10. PBX B user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
41	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX A user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PBX B user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PBX B user and PBX A user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
42	Teams userCalls PBX B user, Teams user performs Unattended Transfer to PSTN user	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX B user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX B user and PSTN user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
43	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PSTN user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PSTN user and PBX B user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
44	Teams user Calls PSTN user, Teams user performs Unattended Transfer to PBX A user	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PSTN user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PSTN user and PBX A user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
47	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PBX A user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PBX A user and PBX B user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
48	PBX A user Calls Teams user, Teams user performs Unattended Transfer to PSTN user	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX A user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX A user and PSTN user 10. PBX A user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
49	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 2 6.PBX B user 2 starts ringing 7. PBX B user 1 hears Ring back Tone 8. PBX B user 2 answers the call 9. Verify two way audio between PBX B user 1 and PBX B user 2 10. PBX B user 1 hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
50	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PBX A user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX A user 6.PBX A user starts ringing 7. PBX B user hears Ring back Tone 8. PBX A user answers the call 9. Verify two way audio between PBX B user and PBX A user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
51	PBX B user Calls Teams user, Teams user performs Unattended Transfer to PSTN user	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PSTN user 6.PSTN user starts ringing 7. PBX B user hears Ring back Tone 8. PSTN user answers the call 9. Verify two way audio between PBX B user and PSTN user 10. PBX B user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams
52	PSTN user Calls Teams user, Teams user performs Unattended Transfer to PBX B user	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user transfers the call to PBX B user 6.PBX B user starts ringing 7. PSTN user hears Ring back Tone 8. PBX B user answers the call 9. Verify two way audio between PSTN user and PBX B user 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
55	PSTN user calls Teams user, Teams user performs Unattended Transfer to second Teams user	1. Make a voice call from PSTN user to Teams user 1 2. PSTN user hears Ring back Tone 3. Teams user 1 answers the call 4. Verify two way audio 5. Teams user 1 transfers the call to Teams user 2 6.Teams user 2 starts ringing 7. Teams user 2 answers the call 8. Verify two way audio between PSTN user and Teams user 2 10. PSTN user hangs up the call 11. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
56	Teams user Calls PBX A user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from Teams user to PBX A user 1 2. Teams user hears Ring back Tone 3. PBX A user 1 answers the call 4. Verify two way audio 5. Teams user adds PBX A user 2 to the ongoing call 6.PBX A user 2 starts ringing 7. PBX A user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
57	Teams user user Calls PBX A user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6.PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
58	Teams user user Calls PBX A user, Teams user adds PSTN user to the ongoing call	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6.PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via 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
59	Teams user user Calls PBX B user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from Teams user to PBX B user 1 2. Teams user hears Ring back Tone 3. PBX B user 1 answers the call 4. Verify two way audio 5. Teams user adds PBX B user 2 to the ongoing call 6.PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams
60	Teams user user Calls PBX B user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6.PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
61	Teams user user Calls PBX B user, Teams user adds PSTN user to the ongoing call	1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams
62	Teams user user Calls PSTN user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
63	Teams user user Calls PSTN user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
64	Teams user user Calls PSTN 1 user, Teams user adds PSTN 2 user to the ongoing call	1. Make a voice call from Teams user to PSTN user 1 2. Teams user hears Ring back Tone 3. PSTN user 1 answers the call 4. Verify two way audio 5. Teams user adds PSTN user 2 to the ongoing call 6. PSTN user 2 starts ringing 7. PSTN user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
65	PBX A user Calls Teams user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from PBX A user 1 to Teams user 2. PBX A user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX A user 2 to the ongoing call 6. PBX A user 2 starts ringing 7. PBX A user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
66	PBX A user Calls Teams user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBXB user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams
67	PBX A user Calls Teams user, Teams user adds PSTN user to the ongoing call	1. Make a voice call from PBX A user to Teams user 2. PBX A user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

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
68	PBX B user Calls Teams user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX B user 2 to the ongoing call 6. PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
69	PBX B user Calls Teams user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams
70	PBX B user Calls Teams user, Teams user adds PSTN user to the ongoing call	1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
71	PSTN user Calls Teams user, Teams user adds PBX B user to the ongoing call	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	NOT APPLICABLE	This testing is for only one PBX with Teams
72	PSTN user Calls Teams user, Teams user adds PBX A user to the ongoing call	1. Make a voice call from PSTN user to Teams user 2. PSTN user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

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

External ID	Title	Procedure	Expected Results	Status	Comments
73	PSTN 1 user Calls Teams user, Teams user adds PSTN 2 user to the ongoing call	1. Make a voice call from PSTN user 1 to Teams user 2. PSTN user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PSTN user 2 to the ongoing call 6. PSTN user 2 starts ringing 7. PSTN user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
74	PSTN user Calls Teams user, Teams user adds two or more users to the ongoing call	1. Make a voice call from PSTN user to Teams user 1 2. PSTN user hears Ring back Tone 3. Teams user 1 answers the call 4. Verify two way audio 5. Teams user 1 adds Teams user 2 to the ongoing call 6. Verify Teams user 2 is added successfully to the call 7. Teams user 1 adds PBX A user to the ongoing call 9. Verify PBX A user is added successfully to the call 10. Teams user 1 adds PBX B user to the ongoing call 11. Verify PBX B user is added successfully to the call 12. Verify all four users are able to hear each other 13. All the users hang up and call is cleared successfully for all the users	1. Third user is added to the call successfully 2. All three users are able to hear each other	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
75	PBX A user Calls Teams user, Teams user CFA to PBX A user	1. Teams user sets call forwarding all to PBX A user 2 2. Make a voice call from PBX A user 1 to Teams user 3. PBX A user 2 starts ringing 4. PBX A user 2 answers the call 5. Verify two way audio 6. PBX A user 1 hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	PASSED	
76	PBX A user Calls Teams user, Teams user CFA to PBX B user	1. Teams user sets call forwarding all to PBX B user 2. Make a voice call from PBX A user to Teams user 3. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
77	PBX A user Calls Teams user, Teams user CFA to PSTN user	1. Teams user sets call forwarding all to PSTN user 2. Make a voice call from PBX A user to Teams user 3. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
78	PBX B user Calls Teams user, Teams user CFA to PBX B user	1. Teams user sets call forwarding all to PBX B user 2 2. Make a voice call from PBX B user 1 to Teams user 3. PBX B user 2 starts ringing 4. PBX B user 2 answers the call 5. Verify two way audio 6. PBX B user 1 hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
79	PBX B user Calls Teams user, Teams user CFA to PBX A user	1. Teams user sets call forwarding all to PBX A user 2. Make a voice call from PBX B user to Teams user 3. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
80	PBX B user Calls Teams user, Teams user CFA to PSTN user	1. Teams user sets call forwarding all to PSTN user 2. Make a voice call from PBX B user to Teams user 3. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
81	PSTN user Calls Teams user, Teams user CFA to PBX B user	1. Teams user sets call forwarding all to PBX B user 2. Make a voice call from PSTN user to Teams user 3. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
82	PSTN user Calls Teams user, Teams user CFA to PBX A user	1. Teams user sets call forwarding all to PBX A user 2. Make a voice call from PSTN user to Teams user 3. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	PASSED	
83	PSTN 1 user Calls Teams user, Teams user CFA to PSTN 2 user	1. Teams user sets call forwarding all to PSTN user 2 2. Make a voice call from PSTN user 1 to Teams user 3. PSTN user 2 starts ringing 4. PSTN user 2 answers the call 5. Verify two way audio 6. PSTN user 1 hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call to correct destination	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
84	PBX A user Calls Teams user, Teams user CFNA to PBX A user	1. Teams user sets call forwarding no answer to PBX A user 2 2. Make a voice call from PBX A user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user 2 starts ringing 4. PBX A user 2 answers the call 5. Verify two way audio 6. PBX A user 1 hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
85	PBX A user Calls Teams user, Teams user CFNA to PBX B user	1. Teams user sets call forwarding no answer to PBX B user 2. Make a voice call from PBX A user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
86	PBX A user Calls Teams user, Teams user CFNA to PSTN user	1. Teams user sets call forwarding no answer to PSTN user 2. Make a voice call from PBX A user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX A user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	This testing is for only one PBX with Teams
87	PBX B user Calls Teams user, Teams user CFNA to PBX B user	1. Teams user sets call forwarding no answer to PBX B user 2 2. Make a voice call from PBX B user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user 2 starts ringing 4. PBX B user 2 answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
88	PBX B user Calls Teams user, Teams user CFNA to PBX A user	1. Teams user sets call forwarding no answer to PBX A user 2. Make a voice call from PBX B user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	NOT APPLICABLE	This testing is for only one PBX with Teams
89	PBX B user Calls Teams user, Teams user CFNA to PSTN user	1. Teams user sets call forwarding no answer to PSTN user 2. Make a voice call from PBX B user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user starts ringing 4. PSTN user answers the call 5. Verify two way audio 6. PBX B user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
90	PSTN user Calls Teams user, Teams user CFNA to PBX B user	1. Teams user sets call forwarding no answer to PBX B user 2. Make a voice call from PSTN user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX B user starts ringing 4. PBX B user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	NOT APPLICABLE	This testing is for only one PBX with Teams
91	PSTN user Calls Teams user, Teams user CFNA to PBX A user	1. Teams user sets call forwarding no answer to PBX A user 2. Make a voice call from PSTN user to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PBX A user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up the call 7. Verify call is cleared successfully	1. Teams user is able to forward the incoming call successfully on reaching the No answer timeout value	PASSED	
92	PSTN 1 user Calls Teams	1. Teams user sets call forwarding no answer to PSTN user 2	1. Teams user is able to forward the	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
	user, Teams user CFNA to PSTN 2 user	2. Make a voice call from PSTN user 1 to Teams user 3. Teams user starts ringing 4. Teams user does not answer the call 5. Call gets forwarded after the no answer timeout value is reached 6. PSTN user 2 starts ringing 4. PSTN user 2 answers the call 5. Verify two way audio 6. PSTN user 1 hangs up the call 7. Verify call is cleared successfully	incoming call successfully on reaching the No answer timeout value		
93	PSTN user calls Teams user, Teams user and users set for simultaneous ringing also rings	1. Teams user sets simultaneous ringing to PBX A user and PBX B user 2. Make a voice call from PSTN user to Teams user 3. Teams user, PBX A user and PBX B user starts ringing 4. PBX A user answers the call 5. Verify two way audio 6. PSTN user hangs up 7. Verify call is cleared successfully 8. Repeat steps 2 to 6 where PBX B user answers the call		PASSED	Tested only with PBX A

External ID	Title	Procedure	Expected Results	Status	Comments
94	Teams user with restricted Caller ID Calls PBX A user	1. Make a voice call from Teams user with restricted caller ID to PBX A user 2.Teams user hears Ring back Tone 3. PBX A user starts ringing 4. Verify caller ID displayed on PBX A user is Unavailable/Private/Anonymous 5. PBX A user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
95	Teams user with restricted Caller ID Calls PBX B user	1. Make a voice call from Teams user with restricted caller ID to PBX B user 2.Teams user hears Ring back Tone 3. PBX B user starts ringing 4. Verify caller ID displayed on PBX B user is Unavailable/Private/Anonymous 5. PBX B user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	NOT APPLICABLE	This testing is for only one PBX with Teams
96	Teams user with restricted Caller ID Calls PSTN user	1. Make a voice call from Teams user with restricted caller ID to PSTN user 2.Teams user hears Ring back Tone 3. PSTN user starts ringing 4. Verify caller ID displayed on PSTN user is Unavailable/Private/Anonymous	1. Teams user is able to dial an outbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
99	PSTN user with restricted Caller ID Calls Teams user	1. Make a voice call from PSTN user with restricted caller ID to Teams user 2. PSTN user hears Ring back Tone 3. Teams user starts ringing 4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous 5. Teams user answers the call 6. Verify two way audio 7. PSTN user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to receive an inbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
100	PBX A user Calls Teams user and leaves voicemail	1. Make a voice call from PBX A user to Teams user 2. Teams user does not answer the call 3. Allow the call to get forwarded to voicemail 4. PBX A user successfully leaves voicemail 5. Teams user receives voicemail notification 6. Teams user successfully retrieves voicemail	1. Teams user is able to receive and retrieve voicemail successfully	PASSED	
101	PBX B user Calls Teams user and	1. Make a voice call from PBX B user to Teams user 2. Teams user does not answer the call 3. Allow the call to get forwarded to	1. Teams user is able to receive and retrieve voicemail successfully	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
104	Teams user Calls PBX B user and leaves voicemail	1. Make a voice call from Teams user to PBX B user 2. PBX B user does not answer the call 3. Allow the call to get forwarded to voicemail 4. Teams user successfully leaves voicemail and navigates voicemail menu using DTMF	1. Teams user is able to leave voicemail and navigate voice mail menu using DTMF successfully	NOT APPLICABLE	This testing is for only one PBX with Teams
105	Teams user Calls PBX A user, PBX A returns call failure response	1. Make a voice call from Teams user to PBX A user 2. PBX A returns 486 Busy 3. Verify Teams user gets appropriate notification or announcement and the call is cleared 4. Repeat steps 1 to 3 where PBX A returns 480, 404, 503 SIP responses 5. Document the observation on Teams user side	1. Teams user handles the failure response successfully	PASSED	
106	Teams user Calls PBX A user using SIP URI	1. Make a voice call from Teams user to PBX A user using SIP URI 2. PBX A user starts ringing 3. PBX A user answers the call 4. Verify two way audio 5. Teams user hangs up the call 6. Verify call is cleared successfully	1. Teams user is able to call using SIP URI 2. Call is connected with two way audio successfully	NOT TESTED	SIP URI Not tested for this PBX
107	Teams user Calls PBX B	1. Make a voice call from Teams user to PBX B user using SIP URI 2. PBX B user starts ringing	1. Teams user is able to call using SIP URI 2. Call is connected	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
		Skype for Business user is internal and external			
111	Skype for Business user calls Teams user	1. Make a voice call from Skype for Business user to Teams user 2. Skype for Business user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Skype for Business user hangs up the call 6. Verify call is cleared successfully 7. Verify the same scenario where Skype for Business user is internal and external		NOT APPLICABLE	Not applicable for this topology
112	Teams user calls Skype for Business External Mobile user	1. Skype for business user is an External Mobile user 2. Make a voice call from Teams user to Skype for Business user 3. Teams user hears Ring back Tone 4. Skype for Business user answers the call 5. Verify two way audio 6. Teams user hangs up the call 7. Verify call is cleared successfully		NOT APPLICABLE	Not applicable for this topology

External ID	Title	Procedure	Expected Results	Status	Comments
113	Skype for Business External Mobile user calls Teams user	1. Skype for business user is an External Mobile user 2. Make a voice call from Skype for Business user to Teams user 3. Skype for Business user hears Ring back Tone 4. Teams user answers the call 5. Verify two way audio 6. Skype for Business user hangs up the call 7. Verify call is cleared successfully		NOT APPLICABLE	Not applicable for this topology
114	Teams user call other tenant users	1. Make a voice call from Teams user to another tenant users (Teams desktop client user, Teams mobile user, Skype for Business Online user) 2. Verify call is successful 3. Make one call to each different user one by one		NOT APPLICABLE	Not applicable for this topology
115	Teams users joins a meeting scheduled by Skype for business On-premises user	1. Skype for business user schedules a meeting and invites Teams user 1 and Teams user 2 2. Teams user 1 joins the meeting using the Join button 3. Teams user 2 joins the meeting using the dial-in conferencing number 4. Verify Teams users are able to join the meeting successfully 5. Verify all three users are able to hear		NOT APPLICABLE	Not applicable for this topology

External ID	Title	Procedure	Expected Results	Status	Comments
		each other 6. Skype for Business user ends the meeting			
116	Teams user invites Skype for business users for a meeting	1. Teams user schedules a meeting and invites Skype for Business user 1 and Skype for Business user 2 2.Skype for Business user 1 joins the meeting using the Meeting link 3.Skype for Business user 2 joins the meeting using the dial-in conferencing number 4. Verify all three users are able to hear each other 5.Teams user ends the meeting		NOT APPLICABLE	Not applicable for this topology