



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

**Connecting Microsoft Teams  
Direct Routing using AudioCodes  
Mediant Virtual Edition (VE) and  
Cisco UCM 10.5**

**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 10.5 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 help service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

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

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

## 2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representation of Crestron UC-PHONE and UC-PHONE-PLUS connected O365 Cloud with Microsoft Teams Direct Routing to Cisco UCM 10.5 environment using AudioCodes Mediant VE SBC and PSTN Gateway for PSTN connectivity. Media bypass enables Configured teams side used in this topology.

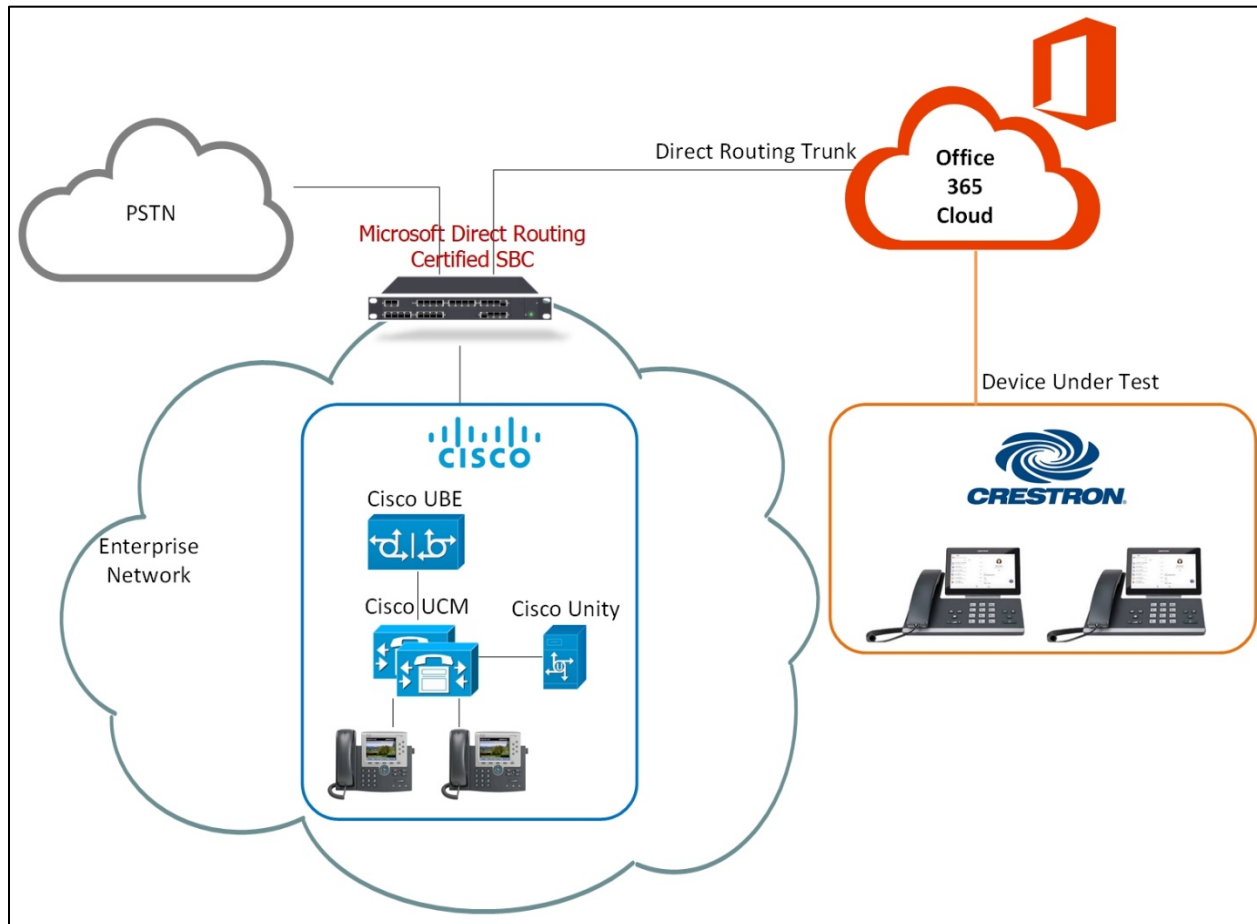


Figure 1 Network Topology

### 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 v10.5.2.18900-15
- Cisco Unity Connection v10.5.2.17900-13
- 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	<a href="#">Section 4.3</a>
Step 2	AudioCodes VE SBC Configuration	<a href="#">Section 4.4</a>
Step 3	Cisco UBE Configuration	<a href="#">Section 4.5</a>
Step 4	Cisco UCM Configuration	<a href="#">Section 4.6</a>

### 4.2 IP Address Worksheet

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

*Table 2 – IP Addresses*

Component	Lab Value
<b>AudioCodes</b>	
LAN IP Address	10.64.3.10
LAN Subnet Mask	255.255.255.0
WAN IP Address	192.XX.XX.XX
WAN Subnet Mask	255.255.255.128
<b>Cisco UCM</b>	
IP Address	10.64.3.83
Subnet Mask	255.255.255.0
<b>Cisco UBE</b>	
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
<b>Cisco Unity</b>	
LAN IP Address	172.16.27.73

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.

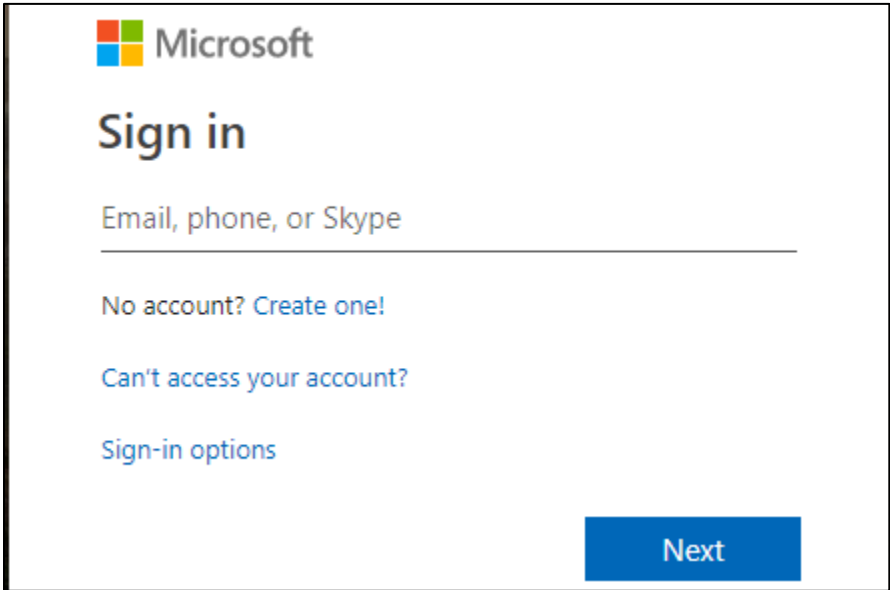


Figure 2: Office 365 Portal Login

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

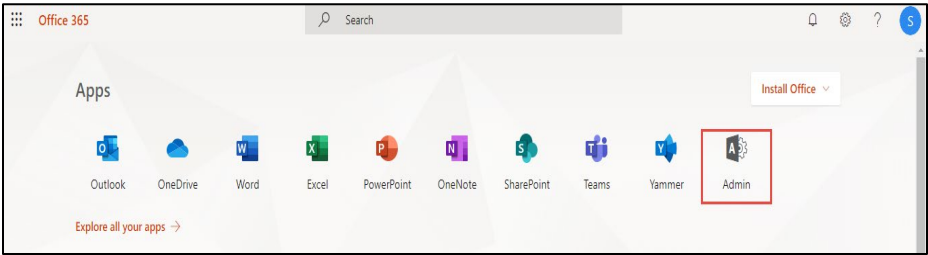


Figure 3: Office 365 Portal Login

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

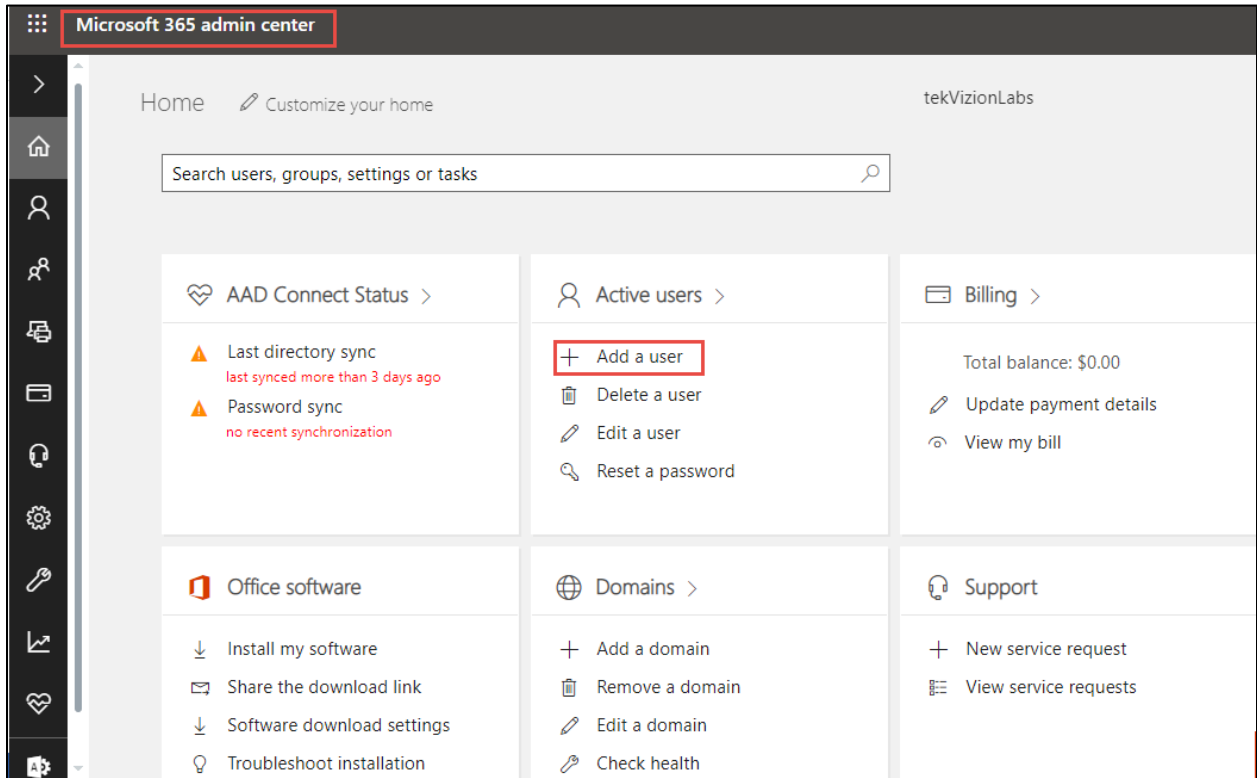


Figure 4: Teams User Creation

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

**Add user**

Basics

Product licenses

Optional settings

Finish

First name: crestron

Last name: teams5

Display name \*: crestronteam5

Username \*: crestroncrestronteam5@tekvisionlabs.com

Password settings

Auto-generate password

Let me create the password

Require this user to change their password when they first sign in

Send password in email upon completion

**Next**

Figure 5: Teams User Creation – Contd.

**Add user**

Basics

Product licenses

Optional settings

Finish

Select location \*: United States

Licenses (1) \*

Assign user a product license

Communications Credits  
Unlimited licenses available

Domestic Calling Plan  
3 of 5 licenses available

Intune  
95 of 100 licenses available

Microsoft Teams Commercial Cloud (User Initiated)  
Unlimited licenses available

Microsoft Teams Trial  
Unlimited licenses available

Office 365 E5  
6 of 13 licenses available

Office 365 E5 without Audio Conferencing  
26 of 100 licenses available

Create user without product license (not recommended)  
They may have limited or no access to Office 365 until you assign a product license.

Back **Next**

Figure 6: Teams User Creation – Contd.

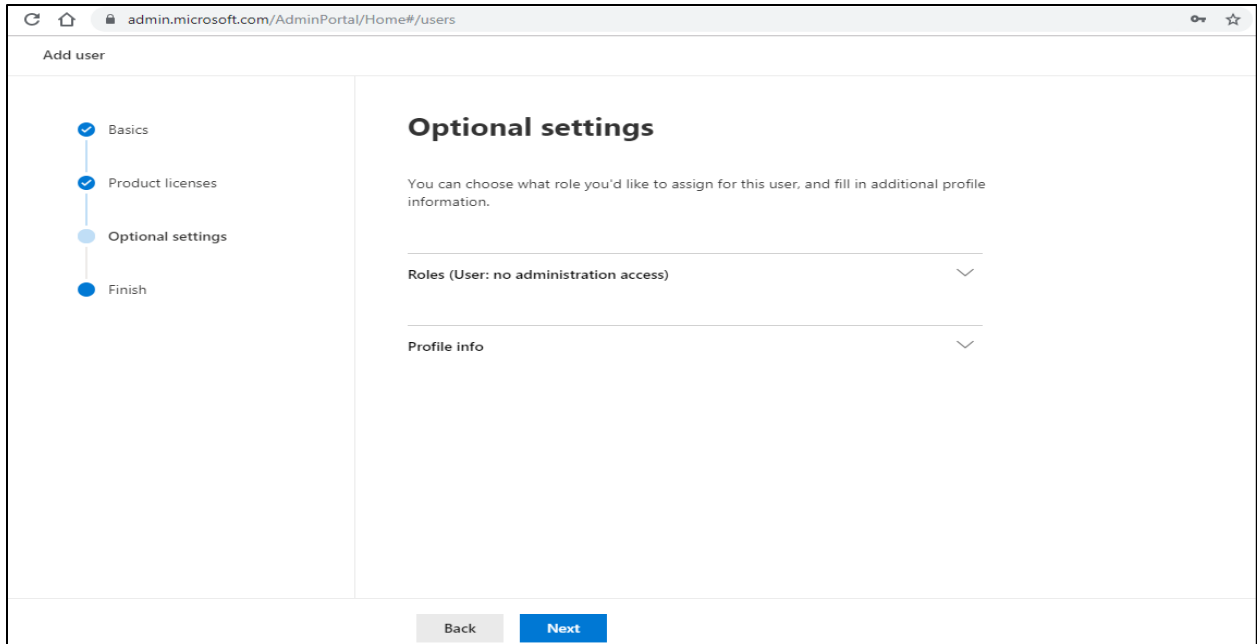


Figure 7: Teams User Creation – Contd.

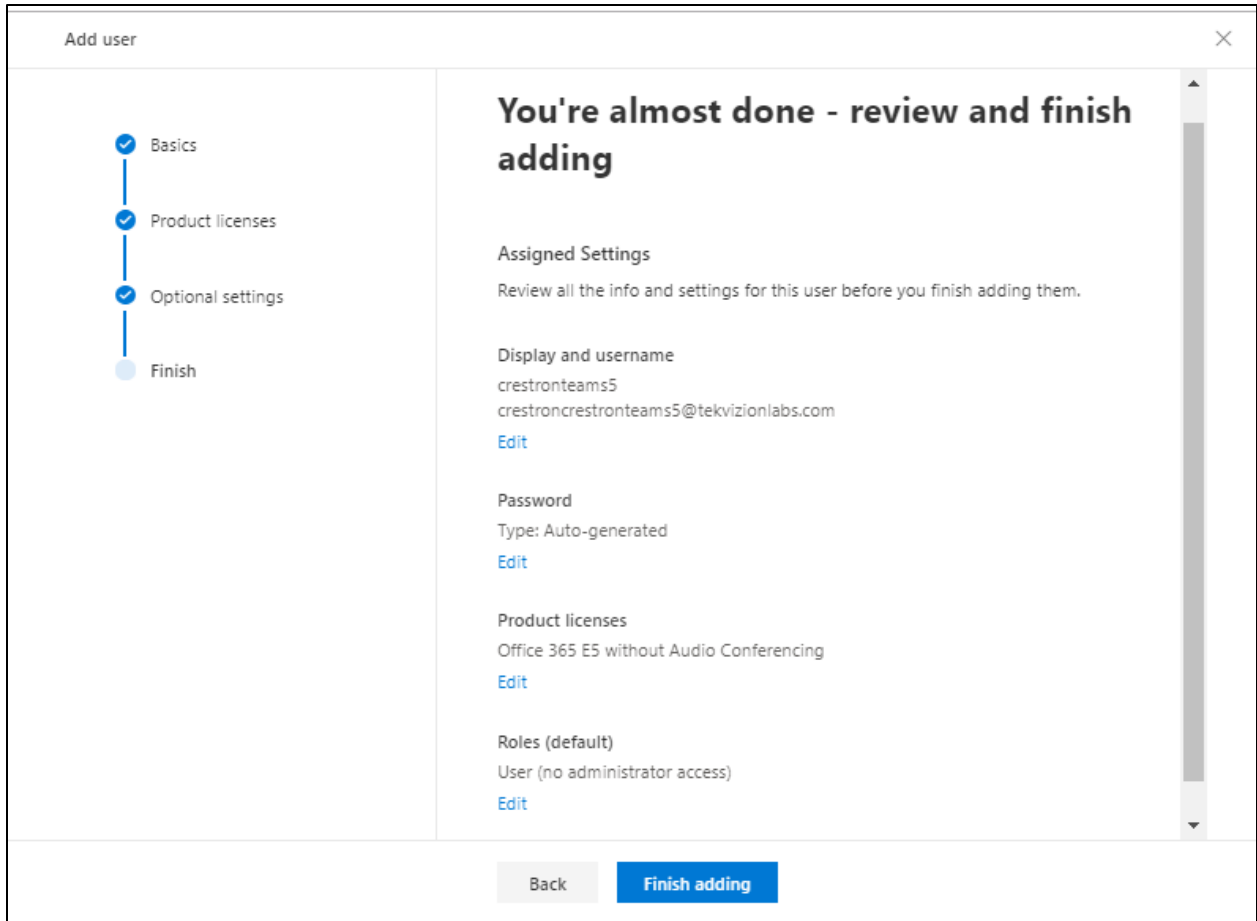
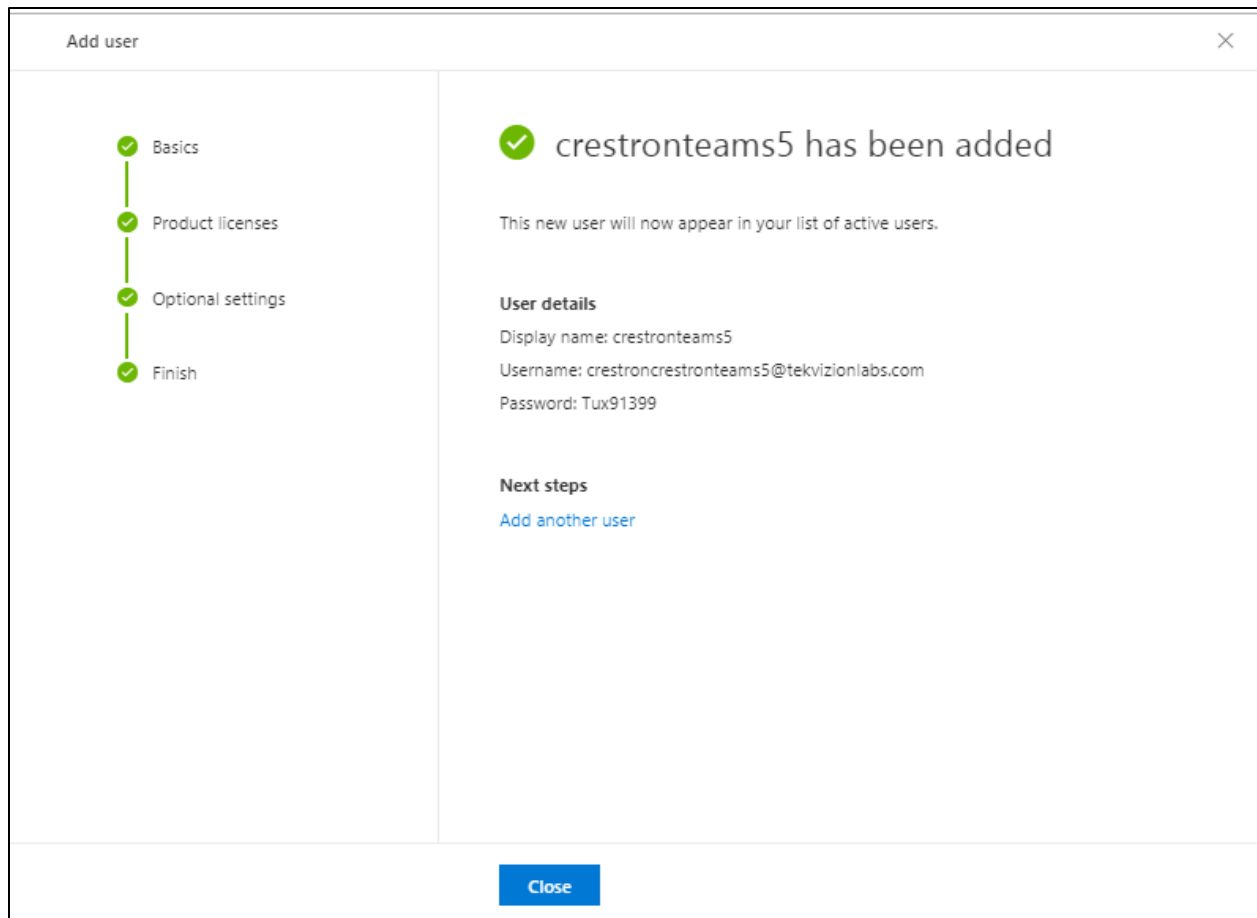


Figure 8: Teams User Creation – Contd.





*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.

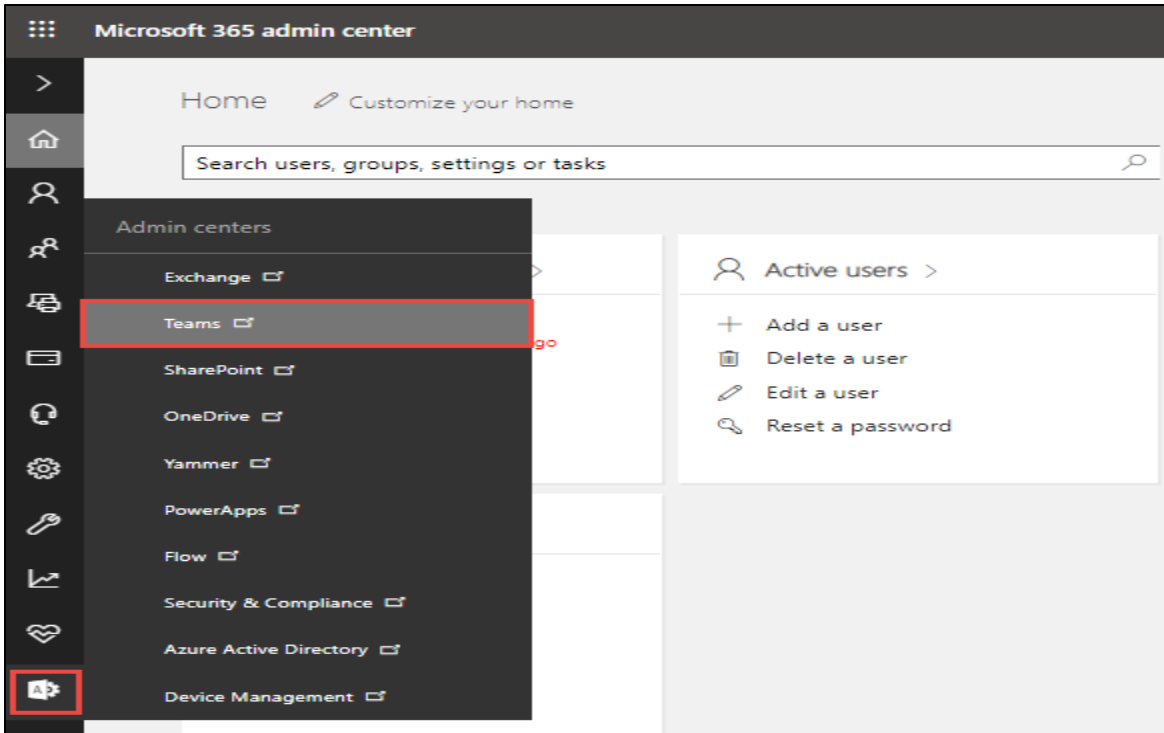


Figure 10: Microsoft O365 admin

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

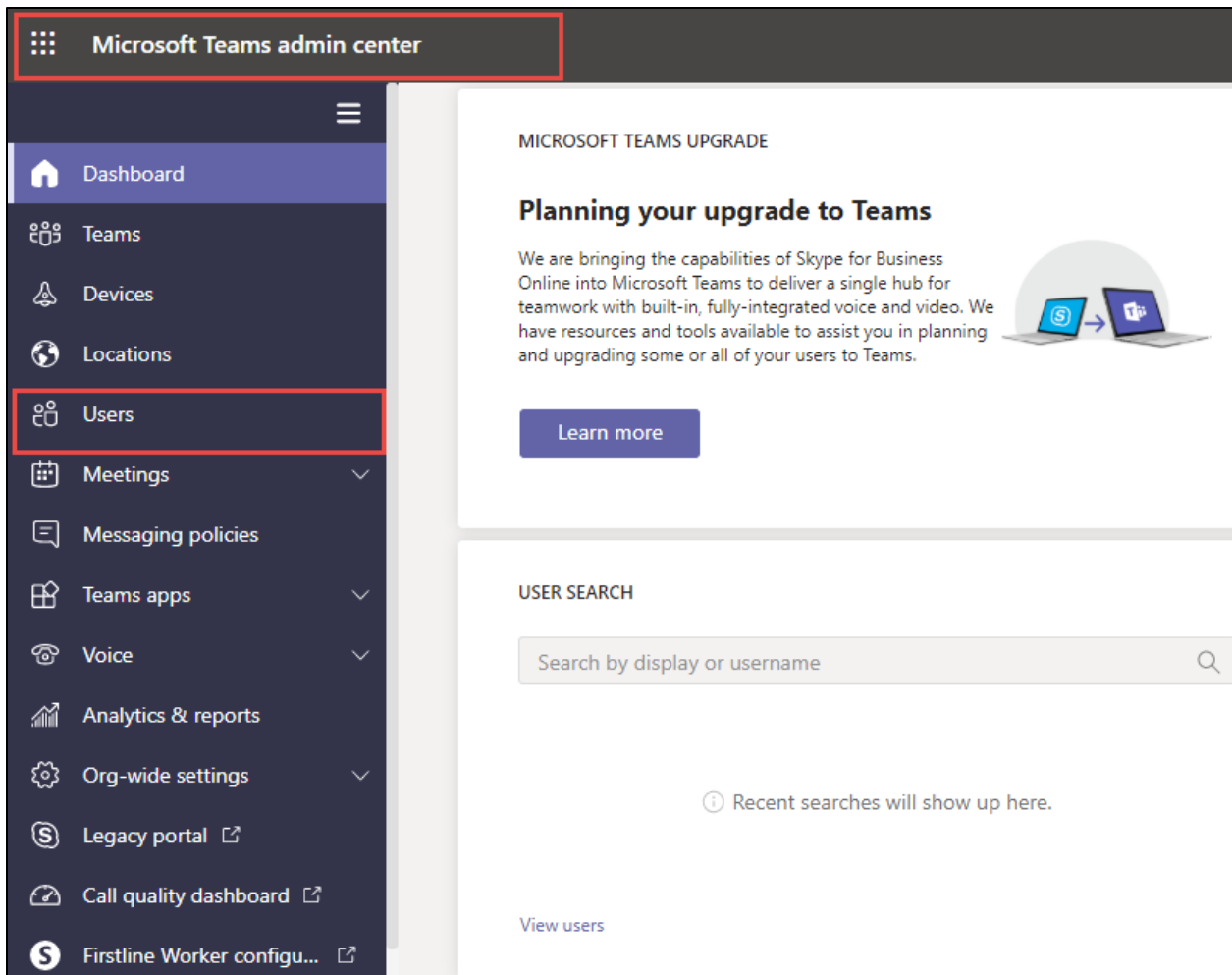


Figure 11: Microsoft O365 admin

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

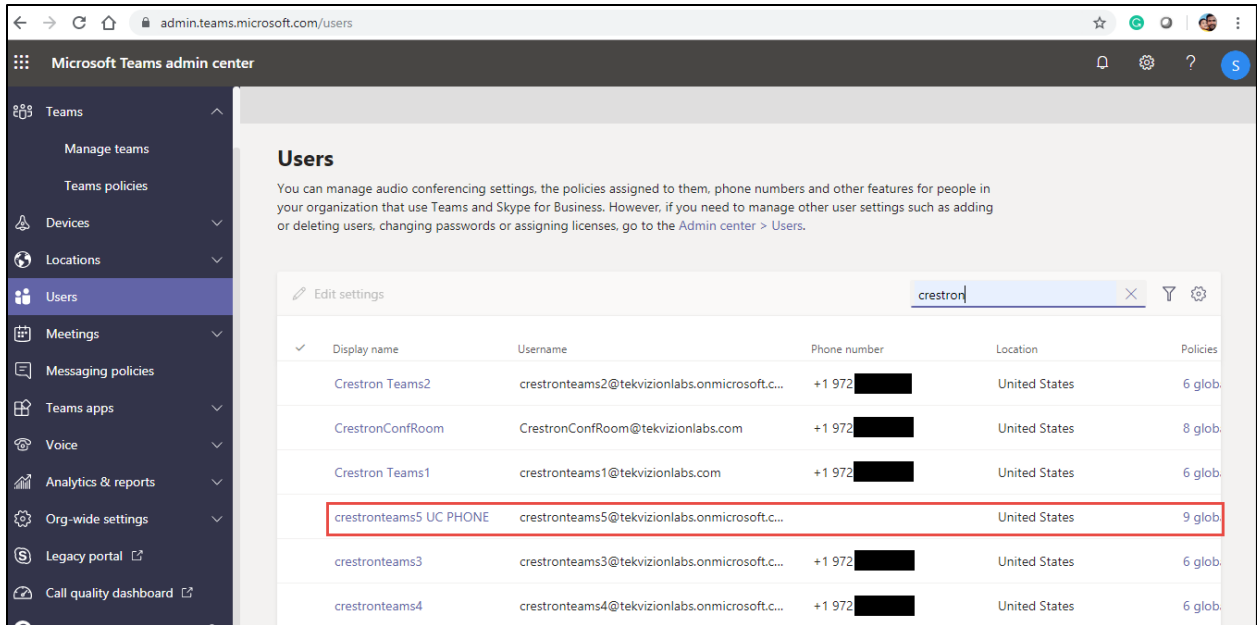


Figure 12: Microsoft O365 admin

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

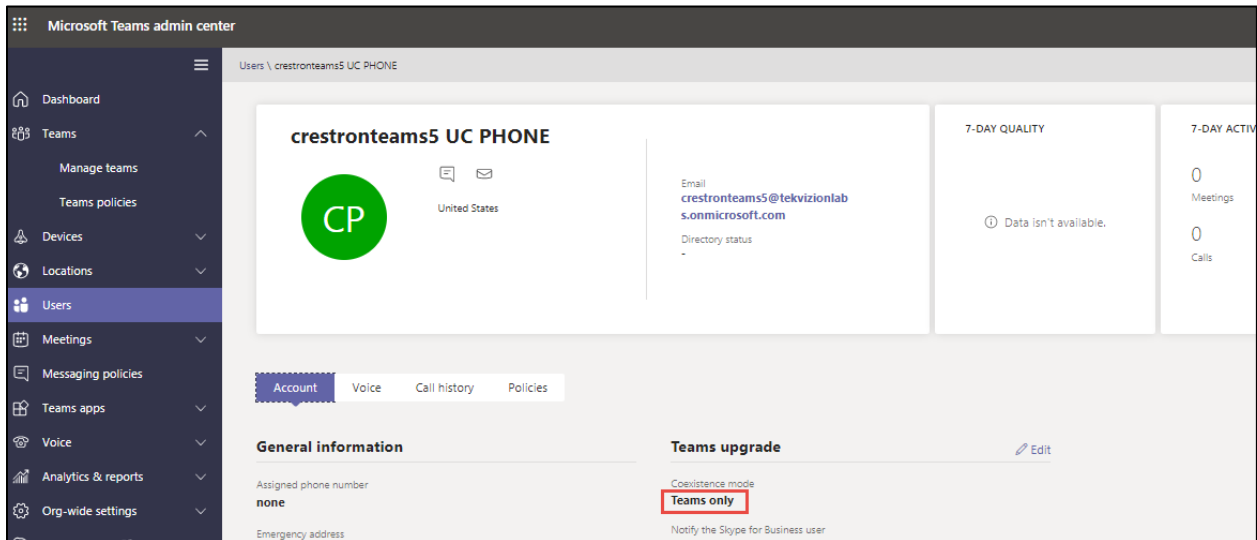


Figure 13: Teams User

### 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.

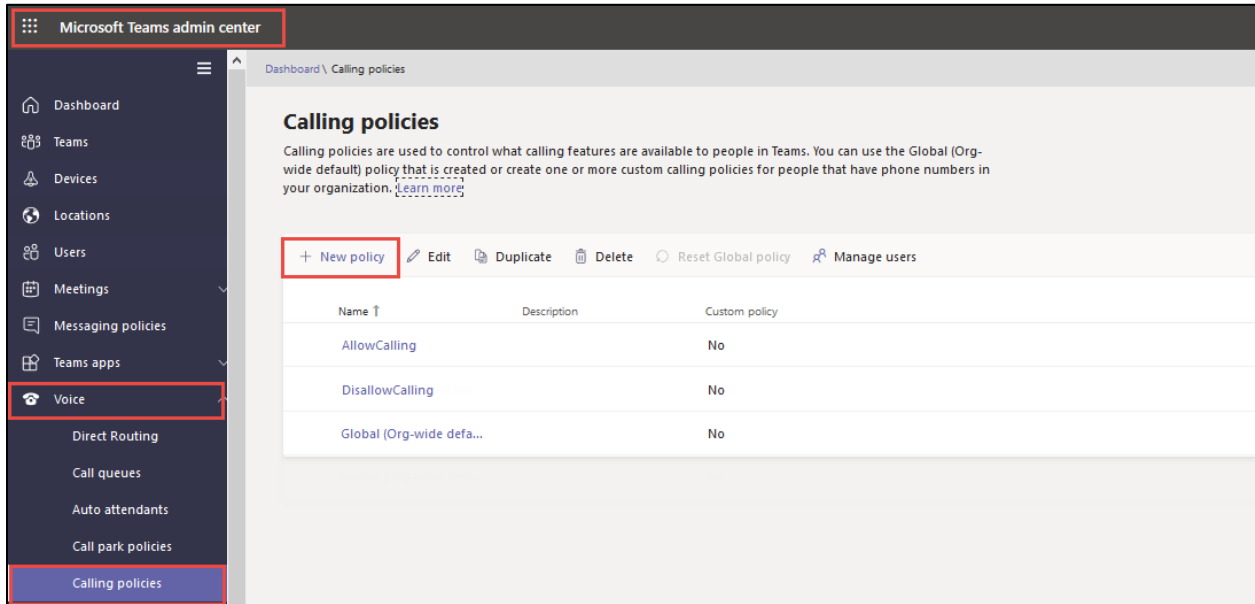


Figure 14 – Calling Policy

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

Dashboard \ Calling policies \ New policy

### Busy on Busy Enabled

Description

Make private calls  On

Call forwarding and simultaneous ringing to people in your organization  On

Call forwarding and simultaneous ringing to external phone numbers  On

Voicemail is available for routing inbound calls

Inbound calls can be routed to call groups  On

Allow delegation for inbound and outbound calls  On

Prevent toll bypass and send calls through the PSTN  Off

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

Figure 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 "crestronteam5@tekvizionlabs.com" -EnterpriseVoiceEnabled $true -HostedVoicemail $true
```

```
Set-CsUser -identity "crestronteam5@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 "crestronteam5" -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 crestronteam5 -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 "crestrontteams5@tekvizionlabs.com" - PolicyName anonymous\_policy**

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

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

## 4.4 AudioCodes VE SBC Configuration

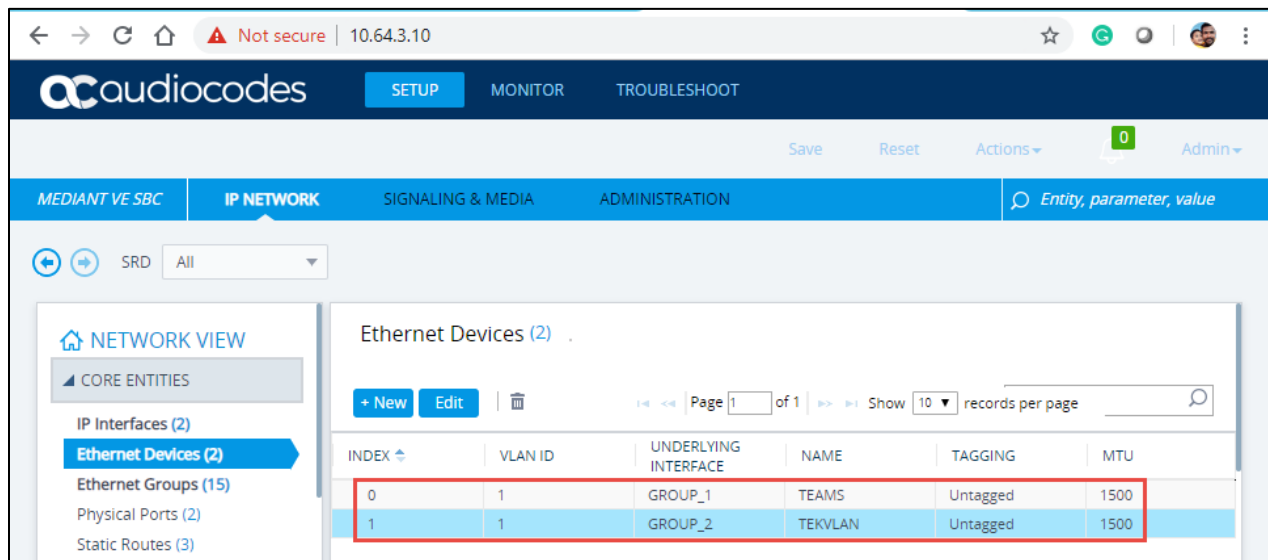
### 4.4.1 General

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.



The screenshot shows the AudioCodes Mediant VE SBC configuration interface. The 'IP NETWORK' tab is selected, and the 'Ethernet Devices' section is active. The table below shows two entries for Ethernet Devices, with the second entry highlighted in red.

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

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:

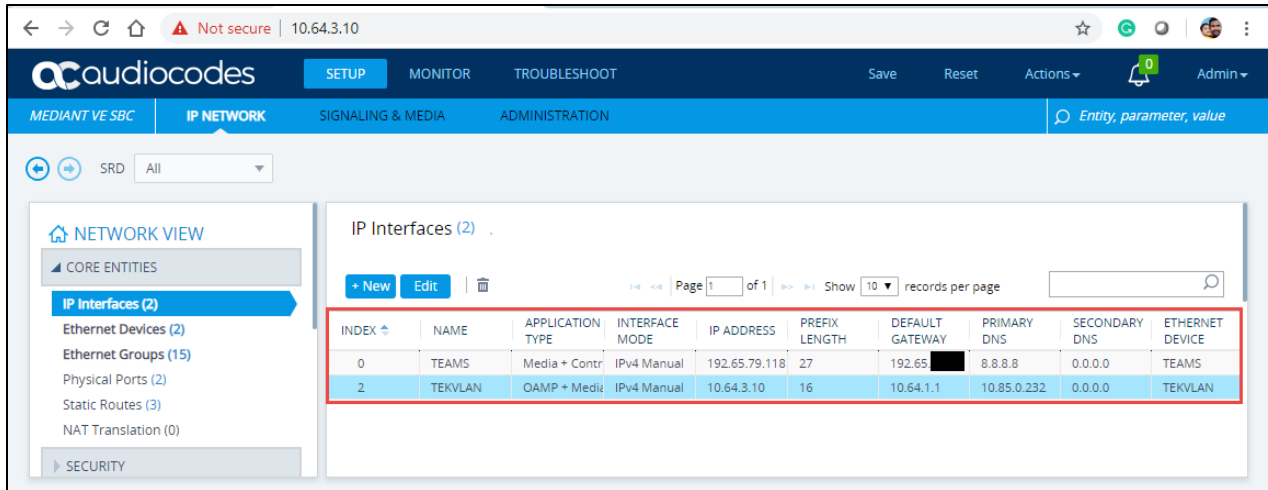


Figure 27 – IP interface Devices

### IP interface TEAMS

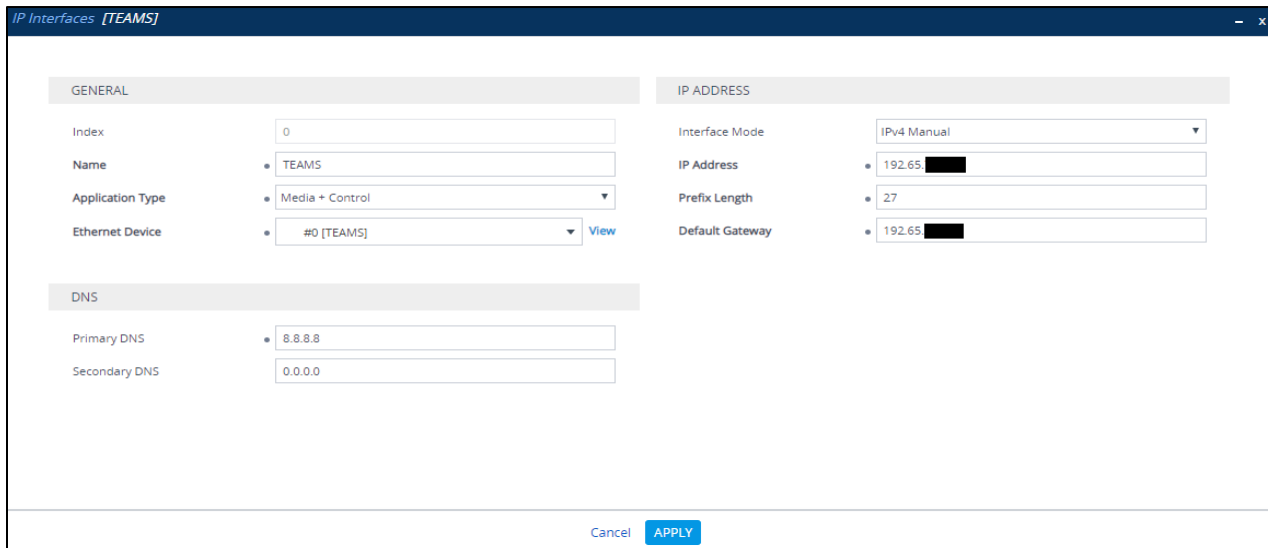


Figure 28 – IP interface Devices

## 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

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

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 Security**. Set the parameter 'Media Security' to Enable; configure the other parameters as shown below

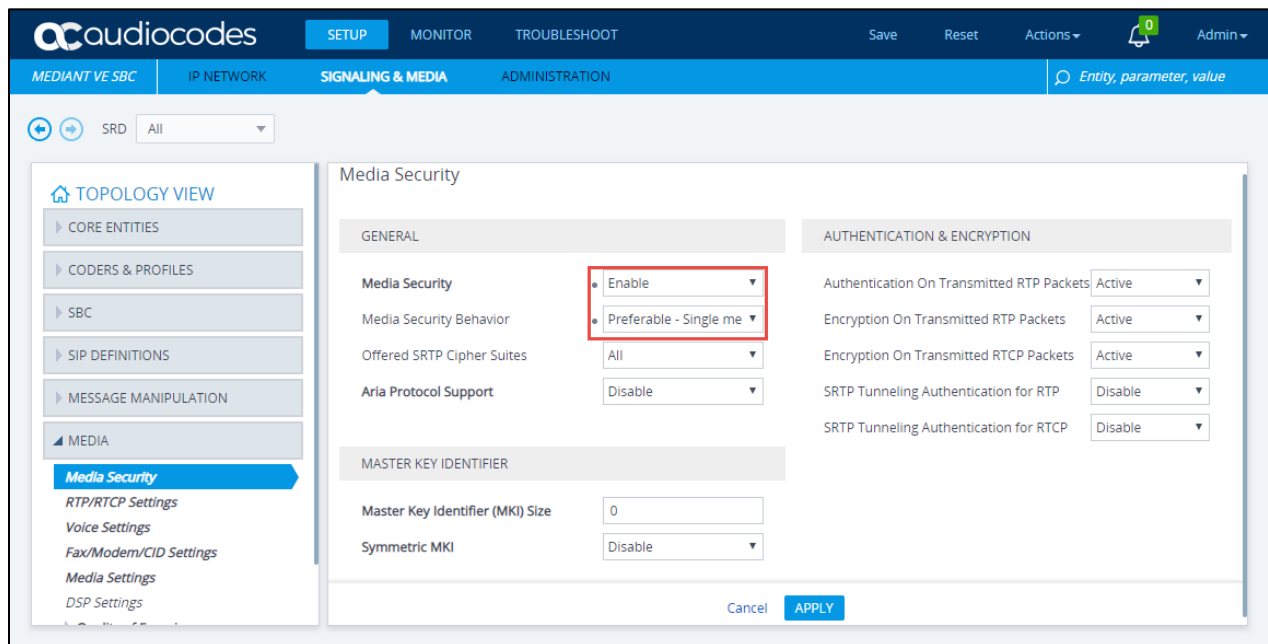


Figure 31 – Media Security

#### 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.

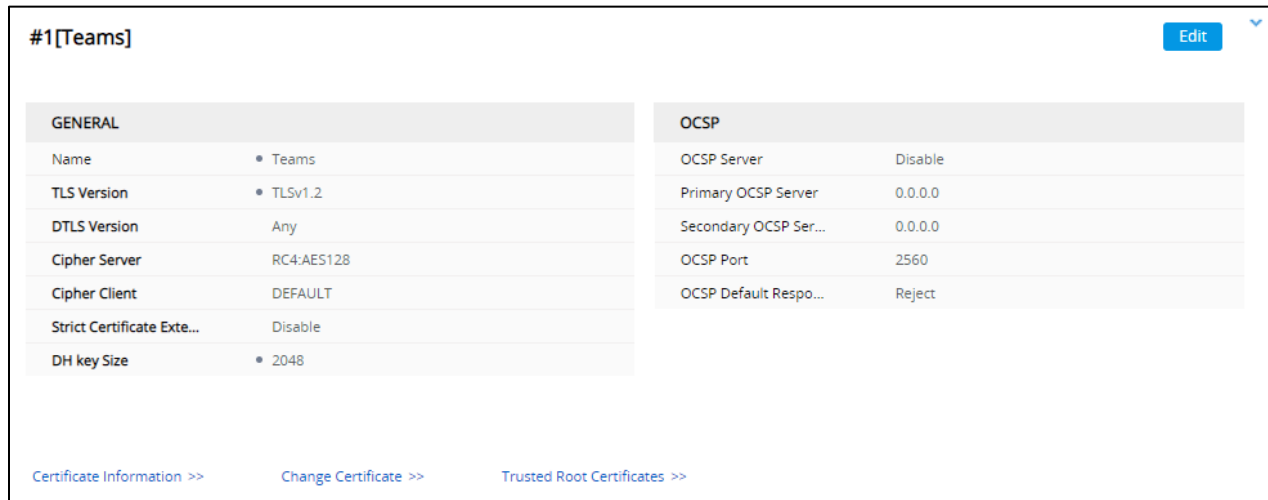


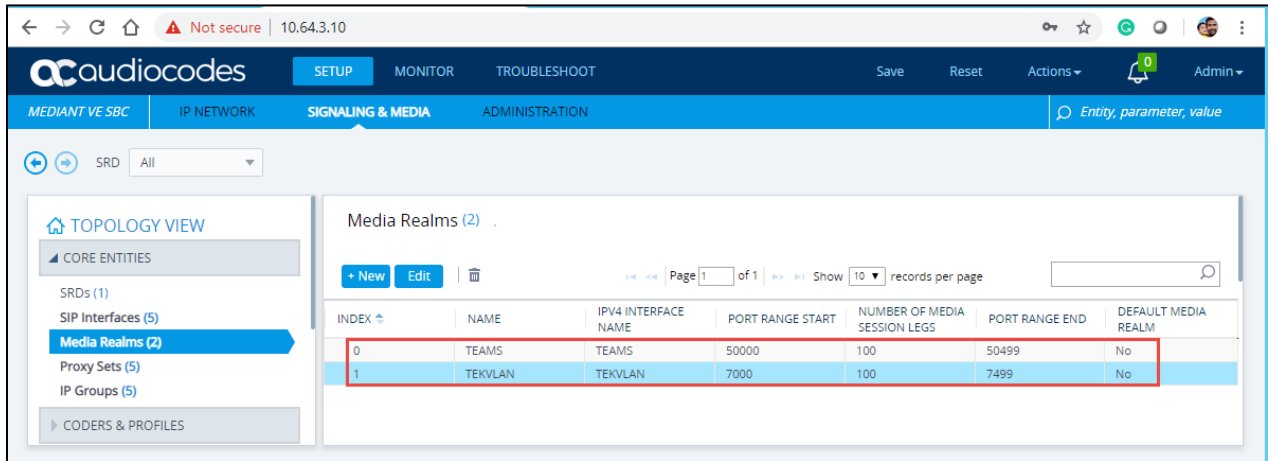
Figure 32 – Teams TLS

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**.

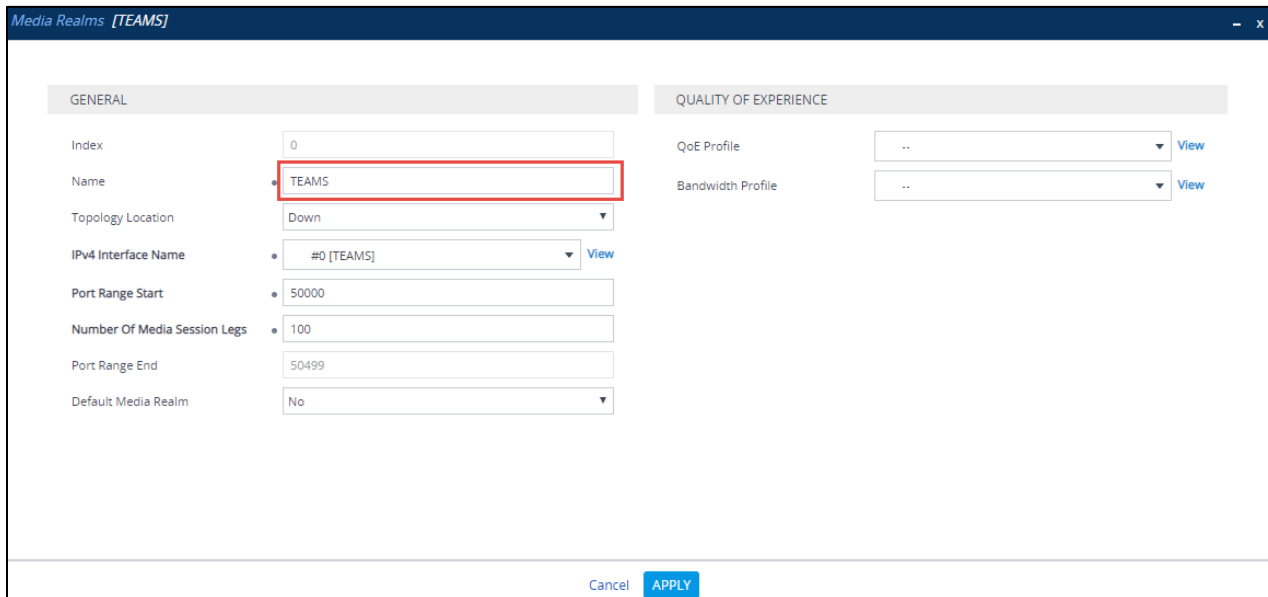


The screenshot shows the Audiocodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main navigation tabs are 'MEDIANT VE SBC', 'IP NETWORK', 'SIGNALING & MEDIA', and 'ADMINISTRATION'. The 'SIGNALING & MEDIA' tab is active, and the 'CORE ENTITIES' menu is expanded to show 'Media Realms (2)'. A table displays the configuration for two Media Realms:

INDEX	NAME	IPV4 INTERFACE NAME	PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	PORT RANGE END	DEFAULT MEDIA REALM
0	TEAMS	TEAMS	50000	100	50499	No
1	TEKVLAN	TEKVLAN	7000	100	7499	No

Figure 33 – Media Realms

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



The screenshot shows the configuration details for the 'TEAMS' Media Realm. The 'GENERAL' tab is active, and the 'Name' field is highlighted with a red box. The configuration includes the following fields:

Field	Value
Index	0
Name	TEAMS
Topology Location	Down
IPv4 Interface Name	#0 [TEAMS]
Port Range Start	50000
Number Of Media Session Legs	100
Port Range End	50499
Default Media Realm	No

Figure 34 – Teams



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

GENERAL		QUALITY OF EXPERIENCE	
Index	1	QoS Profile	.. View
Name	TEKVLAN	Bandwidth Profile	.. View
Topology Location	Up		
IPv4 Interface Name	#2 [TEKVLAN] View		
Port Range Start	7000		
Number Of Media Session Legs	100		
Port Range End	7499		
Default Media Realm	No		

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.

#0[DefaultSRD]
Edit
▼

GENERAL		REGISTRATION	
Name	• DefaultSRD	Max. Number o...	-1
Sharing Policy	Shared	User Security M...	Accept All
SBC Operation ...	B2BUA	Enable Un-Auth...	Enable
SBC Routing Pol...	• # [Default_SBCRoutingPolicy] <a href="#" style="color: #007bff; font-size: 0.8em;">View</a>		
Used By Routin...	Not Used		
Dial Plan	# [-] <a href="#" style="color: #007bff; font-size: 0.8em;">View</a>		
CAC Profile	# [-] <a href="#" style="color: #007bff; font-size: 0.8em;">View</a>		

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.

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

Figure 37 – Teams

The screenshot shows the bottom section of the configuration page. It includes fields for Enable TCP Keepalive (Enable), Used By Routing Server (Not Used), Pre-Parsing Manipulation Set (..), CAC Profile (..), and a CLASSIFICATION section with Classification Failure Response Type (0), Pre-classification Manipulation Set ID (-1), and Call Setup Rules Set ID (-1). At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 38 – Teams

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

SIP Interfaces [PSTNGW]

SRD #0 [DefaultSRD]

**GENERAL**

Index: 1

Name: PSTNGW

Topology Location: Up

Network Interface: #2 [TEKVLAN]

Application Type: SBC

UDP Port: 5060

TCP Port: 0

TLS Port: 0

Additional UDP Ports:

Additional UDP Ports Mode: Always Open

Encapsulating Protocol: No encapsulation

**MEDIA**

Media Realm: #1 [TEKVLAN]

Direct Media: Disable

**SECURITY**

TLS Context Name: ..

TLS Mutual Authentication:

Message Policy: ..

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

Figure 39 – PSTN

Enable TCP Keepalive: Disable

Used By Routing Server: Not Used

Pre-Parsing Manipulation Set: ..

CAC Profile: ..

**CLASSIFICATION**

Classification Failure Response Type: 500

Pre-classification Manipulation Set ID: -1

Call Setup Rules Set ID: -1

Cancel APPLY

Figure 40 – PSTN

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

SIP Interfaces [CISCO]

SRD #0 [DefaultSRD]

**GENERAL**

Index: 2

Name: CISCO

Topology Location: Down

Network Interface: #2 [TEKVLAN]

Application Type: SBC

UDP Port: 5062

TCP Port: 0

TLS Port: 0

Additional UDP Ports:

Additional UDP Ports Mode: Always Open

Encapsulating Protocol: No encapsulation

**MEDIA**

Media Realm: #1 [TEKVLAN]

Direct Media: Disable

**SECURITY**

TLS Context Name: #0 [default]

TLS Mutual Authentication:

Message Policy: --

User Security Mode: Not Configured

Enable Un-Authenticated Registrations: Not configured

Max. Number of Registered Users: -1

Figure 41 – Cisco

Enable TCP Keepalive: Disable

Used By Routing Server: Not Used

Pre-Parsing Manipulation Set: --

CAC Profile: --

**CLASSIFICATION**

Classification Failure Response Type: 500

Pre-classification Manipulation Set ID: -1

Call Setup Rules Set ID: -1

Cancel APPLY

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.

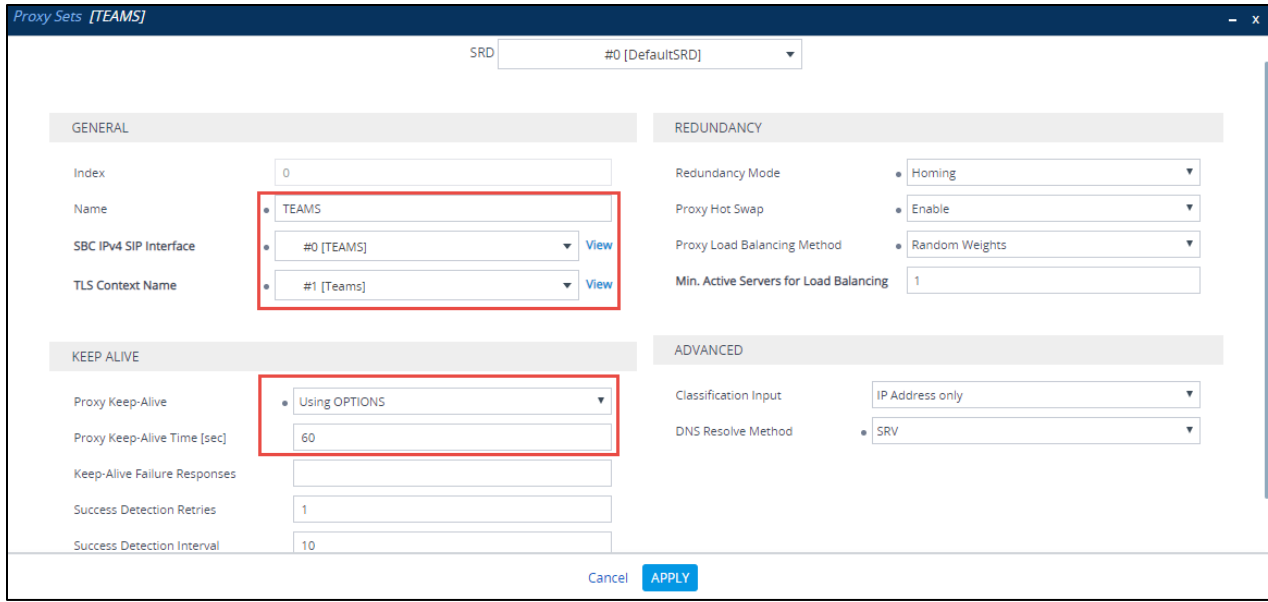


Figure 43 – Teams

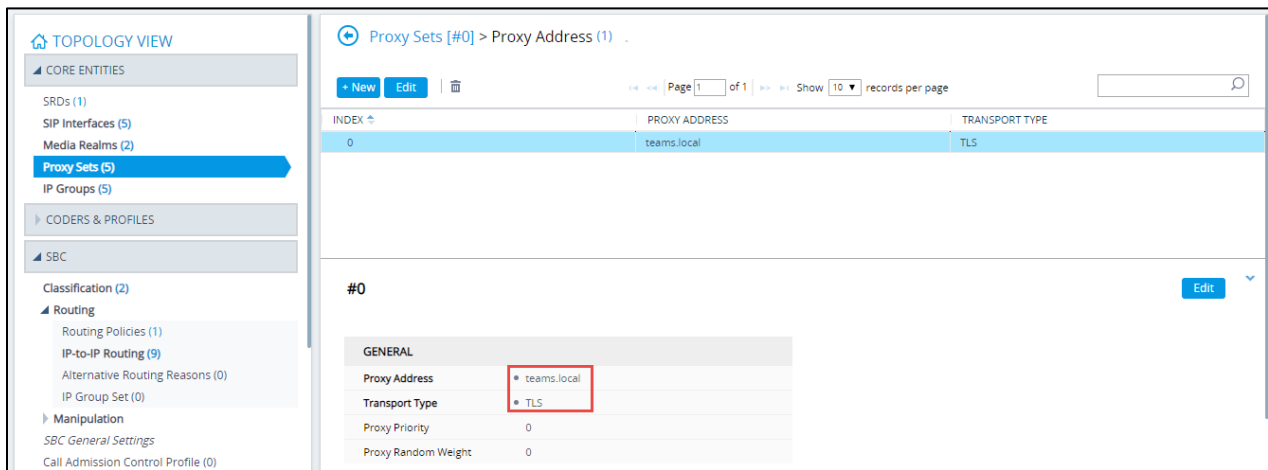


Figure 44 – Teams

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

Proxy Sets [PSTNGW] SRD #0 [DefaultSRD]

GENERAL		REDUNDANCY	
Index	1	Redundancy Mode	
Name	• PSTNGW	Proxy Hot Swap	Disable
SBC IPv4 SIP Interface	• #1 [PSTNGW] <a href="#">View</a>	Proxy Load Balancing Method	Disable
TLS Context Name	-- <a href="#">View</a>	Min. Active Servers for Load Balancing	1
KEEP ALIVE		ADVANCED	
Proxy Keep-Alive	• Using OPTIONS	Classification Input	IP Address only
Proxy Keep-Alive Time [sec]	60	DNS Resolve Method	
Keep-Alive Failure Responses			
Success Detection Retries	1		
Success Detection Interval	10		

Figure 45 – PSTN Gateway

Keep-Alive Failure Responses	
Success Detection Retries	1
Success Detection Interval	10
<a href="#">Cancel</a> <a href="#">APPLY</a>	

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

The screenshot shows the configuration page for an IP Group named 'TEAMS'. The 'GENERAL' section includes fields for Index (0), Name (TEAMS), Topology Location (Down), Type (Server), Proxy Set (#0 [TEAMS]), IP Profile (#1 [TEAMS\_Profile]), Media Realm (#0 [TEAMS]), Contact User, SIP Group Name (sb4.tekvizionlabs.com), Created By Routing Server (No), and Used By Routing Server (Not Used). The 'QUALITY OF EXPERIENCE' section includes QoE Profile and Bandwidth Profile dropdowns. The 'MESSAGE MANIPULATION' section includes Inbound and Outbound Message Manipulation Set dropdowns (set to 1 and 2 respectively), Message Manipulation User-Defined String 1 and 2, and Proxy Keep-Alive using IP Group settings (Enable). The 'SBC REGISTRATION AND AUTHENTICATION' section is partially visible at the bottom.

Figure 49 – IP Group – Teams – Contd.

The screenshot shows the configuration page for an IP Group named 'TEAMS'. The 'SBC GENERAL' section includes Proxy Set Connectivity (Connected), Classify By Proxy Set (Disable), SBC Operation Mode (Not Configured), SBC Client Forking Mode (Sequential), and CAC Profile (..). The 'ADVANCED' section includes Local Host Name (sb4.tekvizionlabs.com), UII Format (Disable), and Always Use Src Address (No). The 'GW GROUP STATUS' section includes Max. Number of Registered Users (-1), Registration Mode (User Initiates Registration), User Stickiness (Disable), User UDP Port Assignment (Disable), Authentication Mode (User Authenticates), Authentication Method List, SBC Server Authentication Type (According to Global Parameter), OAuth HTTP Service, Username (Admin), and Password (\*\*\*\*\*). The 'GW Group Registered IP Address' and 'GW Group Registered Status' (Not Registered) fields are also visible.

Figure 50 – IP Group – Teams – Contd.

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

Cancel [APPLY](#)

Figure 51 – IP Group – Teams

Configure an IP Group for PSTN Gateway as shown below

IP Groups [PSTNGW] SRD #0 [DefaultSRD]

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

Figure 52 – IP Group – PSTN – Contd.

IP Groups [PSTNGW]

Proxy Set Connectivity: Connected

Max. Number of Registered Users: -1

**SBC GENERAL**

Classify By Proxy Set: Enable

Registration Mode: User Initiates Registration

SBC Operation Mode: Not Configured

User Stickiness: Disable

SBC Client Forking Mode: Sequential

User UDP Port Assignment: Disable

CAC Profile: .. View

Authentication Mode: User Authenticates

Authentication Method List:

SBC Server Authentication Type: According to Global Parameter

OAuth HTTP Service: .. View

Username: Admin

Password: .....

**ADVANCED**

Local Host Name:

UUI Format: Disable

Always Use Src Address: No

**GW GROUP STATUS**

GW Group Registered IP Address:

GW Group Registered Status: Not Registered

Figure 53 – IP Group – PSTN – Contd.

**SBC ADVANCED**

Source URI Input:

Destination URI Input:

SIP Connect: No

SBC PSAP Mode: Disable

Route Using Request URI Port: Disable

DTLS Context: #0 [default] View

Keep Original Call-ID: No

Dial Plan: .. View

Call Setup Rules Set ID: -1

Tags:

Cancel APPLY

Figure 54 – IP Group

Configure an IP Group for Cisco UCM as shown below

The screenshot shows the configuration page for an IP Group in Cisco UCM. The page is titled "IP Groups [CISCO]". At the top, there is a dropdown menu for "SRD" set to "#0 [DefaultSRD]". The configuration is divided into several sections:

- GENERAL:**
  - Index: 2
  - Name: CISCO
  - Topology Location: Down
  - Type: Server
  - Proxy Set: #2 [CISCO]
  - IP Profile: #3 [CISCO\_Profile]
  - Media Realm: #1 [TEKVLAN]
  - Contact User: (empty)
  - SIP Group Name: 10.70.69.70
  - Created By Routing Server: No
- QUALITY OF EXPERIENCE:**
  - QoE Profile: ..
  - Bandwidth Profile: ..
- MESSAGE MANIPULATION:**
  - Inbound Message Manipulation Set: 4
  - Outbound Message Manipulation Set: 5
  - Message Manipulation User-Defined String 1: (empty)
  - Message Manipulation User-Defined String 2: (empty)
  - Proxy Keep-Alive using IP Group settings: Disable

Figure 55 – IP Group – Cisco – Contd.

The screenshot shows the configuration page for an IP Group in Cisco UCM, continuing from the previous figure. The page is titled "IP Groups [CISCO]". At the top, there is a dropdown menu for "SRD" set to "#0 [DefaultSRD]". The configuration is divided into several sections:

- GENERAL:**
  - Used By Routing Server: Not Used
  - Proxy Set Connectivity: NA
- SBC GENERAL:**
  - Classify By Proxy Set: Enable
  - SBC Operation Mode: Not Configured
  - SBC Client Forking Mode: Sequential
  - CAC Profile: ..
- ADVANCED:**
  - Local Host Name: (empty)
  - UUI Format: Disable
  - Always Use Src Address: No
- SBC REGISTRATION AND AUTHENTICATION:**
  - Max. Number of Registered Users: -1
  - Registration Mode: User Initiates Registration
  - User Stickiness: Disable
  - User UDP Port Assignment: Disable
  - Authentication Mode: User Authenticates
  - Authentication Method List: (empty)
  - SBC Server Authentication Type: According to Global Parameter
  - OAuth HTTP Service: ..
  - Username: Admin
  - Password: \*\*\*\*\*
- GW GROUP STATUS:**
  - GW Group Registered IP Address: (empty)

Figure 56 – IP Group – Cisco – Contd.

SBC ADVANCED

Source URI Input

Destination URI Input

SIP Connect: No

SBC PSAP Mode: Disable

Route Using Request URI Port: Disable

DTLS Context: #0 [default] View

Keep Original Call-ID: No

Dial Plan: -- View

Call Setup Rules Set ID: -1

Tags

GW Group Registered Status: Not Registered

Cancel APPLY

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.

IP Profiles [TEAMS\_Profile]

GENERAL

Index: 1

Name: TEAMS\_Profile

Created by Routing Server: No

MEDIA SECURITY

SBC Media Security Mode: SRTP

Symmetric MKI: Disable

MKI Size: 1

SBC Enforce MKI Size: Don't enforce

SBC Media Security Method: SDES

Reset SRTP Upon Re-key: Disable

Generate SRTP Keys Mode: Always

SBC SIGNALING

PRACK Mode: Optional

P-Asserted-Identity Header Mode: As Is

Diversion Header Mode: As Is

History-Info Header Mode: As Is

Session Expires Mode: Transparent

Remote Update Support: Not Supported

Remote re-INVITE: Not Supported

Remote Delayed Offer Support: Not Supported

Remote Representation Mode: According to Operation Mode

Keep Incoming Via Headers: According to Operation Mode

Keep Incoming Routing Headers: According to Operation Mode

Keep User-Agent Header: According to Operation Mode

Figure 58 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the IP Profile [TEAMS\_Profile]. The 'SBC EARLY MEDIA' section includes the following settings:

- Remote Early Media: Supported
- Remote Multiple 18x: Supported
- Remote Early Media Response Type: Transparent
- Remote Multiple Early Dialogs: According to Operation Mode
- Remote Multiple Answers Mode: Disable
- Remote Early Media RTP Detection Mode: By Media
- Remote RFC 3960 Support: Not Supported
- Remote Can Play Ringback: No
- Generate RTP: None

The 'SBC REGISTRATION' section includes the following settings:

- Handle X-Detect: No
- ISUP Body Handling: Transparent
- ISUP Variant: itu92
- Max Call Duration [min]: 0
- User Registration Time: 0
- NAT UDP Registration Time: -1
- NAT TCP Registration Time: -1

The 'SBC FORWARD AND TRANSFER' section includes the following settings:

- Remote REFER Mode: Regular
- Remote Replaces Mode: Standard
- Play RBT To Transferee: Yes

Figure 59 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the IP Profile [TEAMS\_Profile]. The 'SBC MEDIA' section includes the following settings:

- Mediation Mode: RTP Mediation
- Extension Coders Group: #0 [AudioCodersGroups\_0]
- Allowed Audio Coders: #0 [AllowedAudioCodersGroup\_TEAMS]
- Allowed Coders Mode: Preference
- Allowed Video Coders: ..
- Allowed Media Types:
- Direct Media Tag:
- RFC 2833 Mode: As Is
- RFC 2833 DTMF Payload Type: 101
- Alternative DTMF Method: As Is
- Send Multiple DTMF Methods: Disable
- Adapt RFC2833 BW to Voice coder BW: Disabled
- SDP Ptime Answer: Preferred Value

The 'SBC HOLD' section includes the following settings:

- Remote 3xx Mode: Handle Locally
- Remote Hold Format: Inactive
- Reliable Held Tone Source: Yes
- Play Held Tone: No

The 'SBC FAX' section includes the following settings:

- Fax Coders Group: ..
- Fax Mode: As Is
- Fax Offer Mode: All coders
- Fax Answer Mode: Single coder
- Remote Renegotiate on Fax Detection: Transparent
- Fax Rerouting Mode: Disable

Figure 60 – IP Profile – Teams – Contd.

IP Profiles [TEAMS\_Profile]

Preferred PTime	20	
Use Silence Suppression	Add	▼
RTP Redundancy Mode	As Is	▼
RTCP Mode	Generate Always	▼
Jitter Compensation	Disable	▼
ICE Mode	Lite	▼
SDP Handle RTCP	Don't Care	▼
RTCP Mux	Supported	▼
RTCP Feedback	Feedback Off	▼
Voice Quality Enhancement	Disable	▼
Max Opus Bandwidth	0	
Generate No-op	No	▼
Enhanced PLC	Disable	▼

<b>MEDIA</b>	
Broken Connection Mode	Disconnect
Media IP Version Preference	Only IPv4
RTP Redundancy Depth	Disable
<b>GATEWAY</b>	
Coders Group	#0 [AudioCodersGroups_0]
<b>LOCAL TONES</b>	
Local RingBack Tone Index	-1
Local Held Tone Index	-1

Figure 61 – IP Profile – Teams – Contd.

IP Profiles [TEAMS\_Profile]

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

Cancel **APPLY**

Figure 62 – IP Profile – Teams – Contd.

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

The screenshot shows the configuration for the IP Profile [PSTNGW\_Profile]. The settings are as follows:

Section	Parameter	Value
GENERAL	Index	2
	Name	PSTNGW_Profile
	Created by Routing Server	No
MEDIA SECURITY	SBC Media Security Mode	RTP
	Symmetric MKI	Disable
	MKI Size	0
	SBC Enforce MKI Size	Don't enforce
	SBC Media Security Method	SDES
	Reset SRTP Upon Re-key	Disable
	Generate SRTP Keys Mode	Only If Required
	SBC SIGNALING	PRACK Mode
P-Asserted-Identity Header Mode		As Is
Diversion Header Mode		As Is
History-Info Header Mode		As Is
Session Expires Mode		Supported
Remote Update Support		Supported Only After Connect
Remote re-INVITE		Supported only with SDP
Remote Delayed Offer Support		Not Supported
Remote Representation Mode		According to Operation Mode
Keep Incoming Via Headers		According to Operation Mode
Keep Incoming Routing Headers	According to Operation Mode	
Keep User-Agent Header	According to Operation Mode	

Figure 63 – IP Profile – PSTN Gateway – Contd.

The screenshot shows the configuration for the IP Profile [PSTNGW\_Profile]. The settings are as follows:

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

Figure 64 – IP Profile – PSTN Gateway – Contd.



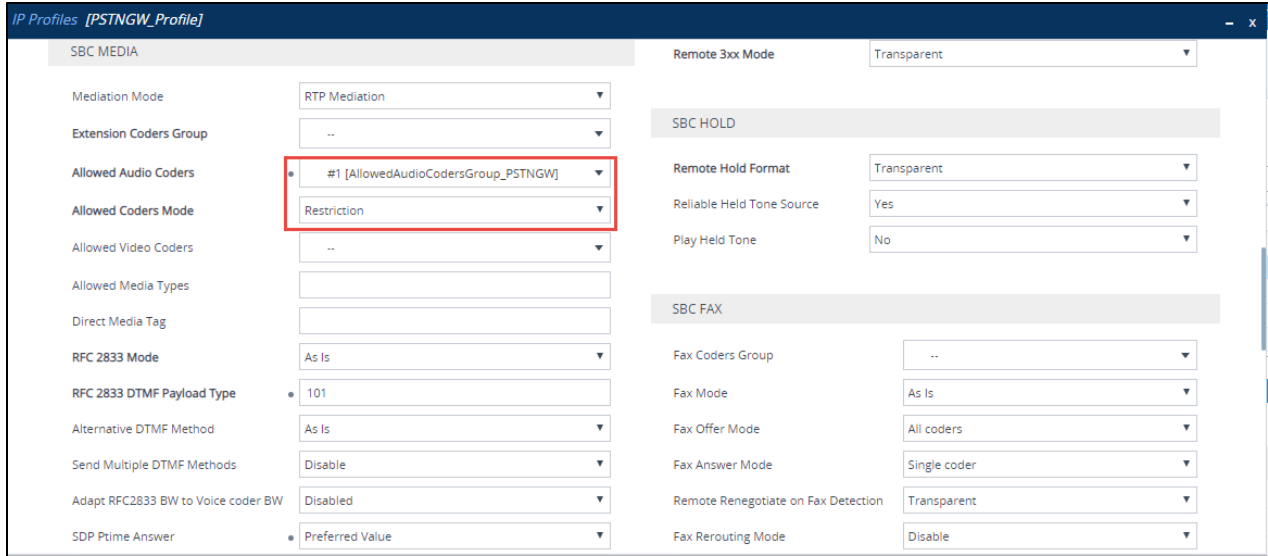


Figure 65 – IP Profile – PSTN Gateway – Contd.

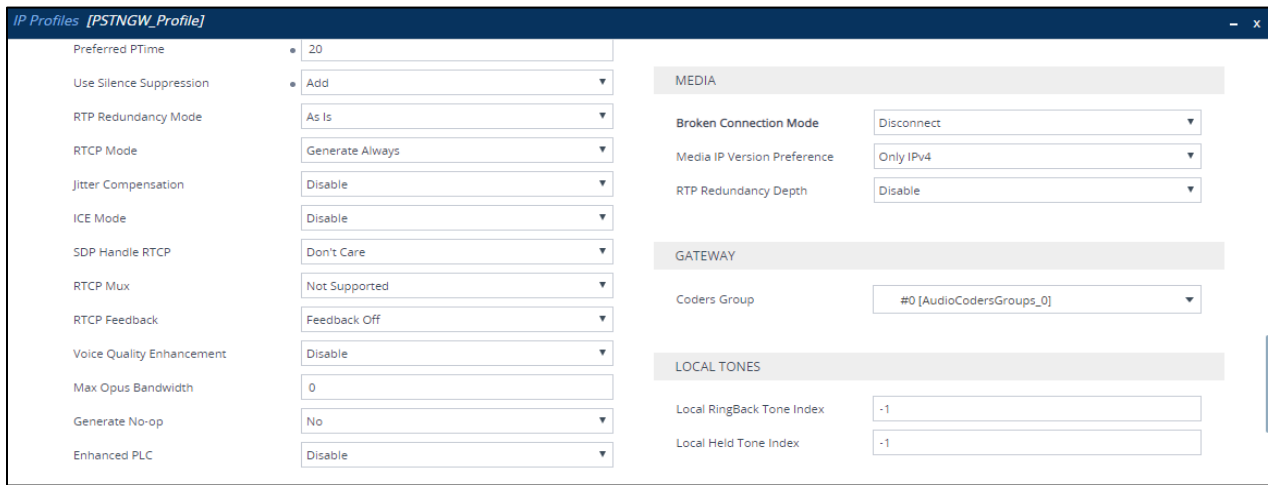


Figure 66 – IP Profile – PSTN Gateway – Contd.

**QUALITY OF SERVICE**

RTP IP DiffServ: 46

Signaling DiffServ: 24

**JITTER BUFFER**

Dynamic Jitter Buffer Minimum Delay [msec]: 10

Dynamic Jitter Buffer Optimization Factor: 10

Jitter Buffer Max Delay [msec]: 300

**VOICE**

Echo Canceled: Line

Input Gain (-32 to 31 dB): 0

Voice Volume (-32 to 31 dB): 0

Buttons: Cancel, APPLY

Figure 67 – IP Profile – PSTN Gateway

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

**GENERAL**

Index: 3

Name: CISCO\_Profile

Created by Routing Server: No

**MEDIA SECURITY**

SBC Media Security Mode: RTP

Symmetric MKI: Disable

MKI Size: 0

SBC Enforce MKI Size: Don't enforce

SBC Media Security Method: SDES

Reset SRTP Upon Re-key: Disable

Generate SRTP Keys Mode: Only if Required

**SBC SIGNALING**

PRACK Mode: Transparent

P-Asserted-Identity Header Mode: As Is

Diversion Header Mode: As Is

History-Info Header Mode: As Is

Session Expires Mode: Supported

Remote Update Support: Supported Only After Connect

Remote re-INVITE: Supported only with SDP

Remote Delayed Offer Support: Not Supported

Remote Representation Mode: According to Operation Mode

Keep Incoming Via Headers: According to Operation Mode

Keep Incoming Routing Headers: According to Operation Mode

Keep User-Agent Header: According to Operation Mode

Figure 68 – IP Profile – Cisco – Contd.

IP Profiles [CISCO\_Profile]

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

Figure 69 – IP Profile – Cisco – Contd.

IP Profiles [CISCO\_Profile]

<b>SBC MEDIA</b>		Remote 3xx Mode	Transparent
Mediation Mode	RTP Mediation	<b>SBC HOLD</b>	
Extension Coders Group	..	Remote Hold Format	Transparent
Allowed Audio Coders	#1 [AllowedAudioCodersGroup_PSTNGW]	Reliable Held Tone Source	Yes
Allowed Coders Mode	Restriction	Play Held Tone	No
Allowed Video Coders	..	<b>SBC FAX</b>	
Allowed Media Types		Fax Coders Group	..
Direct Media Tag		Fax Mode	As Is
RFC 2833 Mode	As Is	Fax Offer Mode	All coders
RFC 2833 DTMF Payload Type	0	Fax Answer Mode	Single coder
Alternative DTMF Method	As Is	Remote Renegotiate on Fax Detection	Transparent
Send Multiple DTMF Methods	Disable	Fax Rerouting Mode	Disable
Adapt RFC2833 BW to Voice coder BW	Disabled		
SDP Ptime Answer	Remote Answer		

Figure 70 – IP Profile – Cisco – Contd.

The screenshot shows the 'IP Profiles [CISCO\_Profile]' configuration window. It is divided into several sections:

- Left Column:** A list of settings with dropdown menus:
  - Use Silence Suppression: Transparent
  - RTP Redundancy Mode: As Is
  - RTCP Mode: Transparent
  - Jitter Compensation: Disable
  - ICE Mode: Disable
  - SDP Handle RTCP: Don't Care
  - RTCP Mux: Not Supported
  - RTCP Feedback: Feedback Off
  - Voice Quality Enhancement: Disable
  - Max Opus Bandwidth: 0
  - Generate No-op: No
  - Enhanced PLC: Disable
- MEDIA Section:**
  - Broken Connection Mode: Disconnect
  - Media IP Version Preference: Only IPv4
  - RTP Redundancy Depth: Disable
- GATEWAY Section:**
  - Coders Group: #0 [AudioCodersGroups\_0]
- LOCAL TONES Section:**
  - Local RingBack Tone Index: -1
  - Local Held Tone Index: -1

Figure 71 – IP Profile – Cisco – Contd.

The screenshot shows the 'IP Profiles [CISCO\_Profile]' configuration window, continuing from the previous one. It features three main sections:

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

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

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

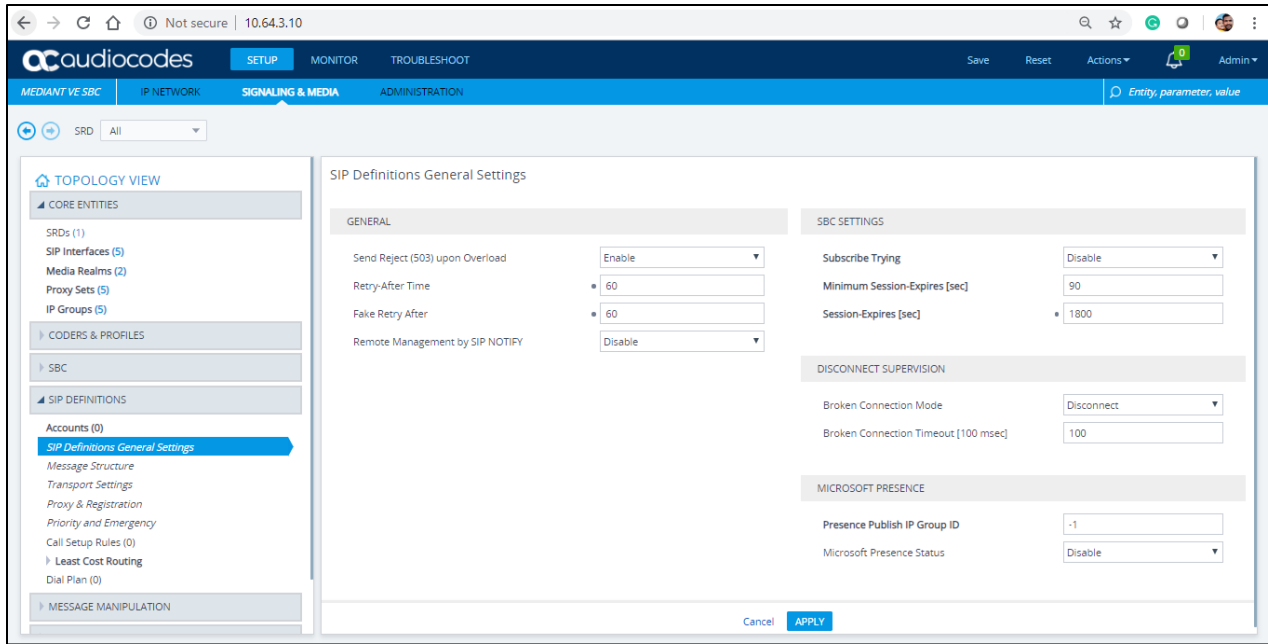


Figure 73 – SIP Definition

#### 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.

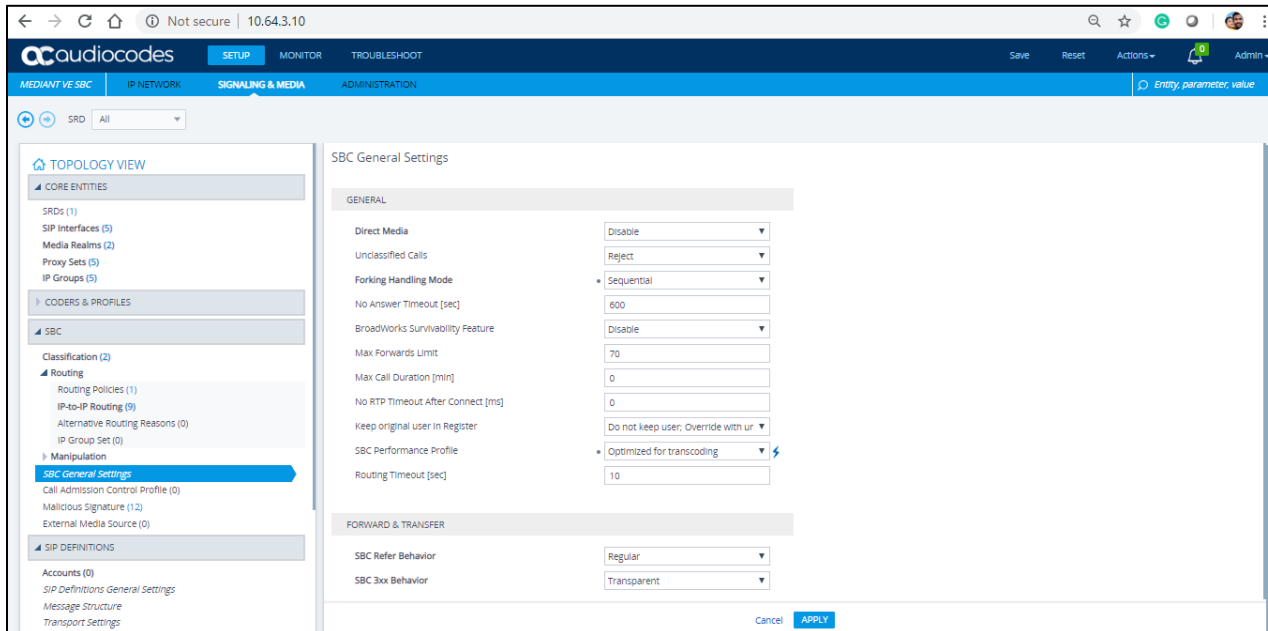


Figure 74 – SBC General Setting – Contd.

#### 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

The screenshot shows the configuration for a routing rule named "TEAMS -> PSTN". The "MATCH" section is configured with "Source IP Group" set to "#0 [TEAMS]". The "ACTION" section is configured with "Destination IP Group" and "Destination SIP interface" both set to "#1 [PSTNGW]".

Figure 75 – Teams to PSTN – Contd.

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

Figure 76 – Teams to PSTN

## Calls from PSTN Gateway to Teams

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

Routing Policy: #0 [Default\_SBCRoutingPolicy]

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

Figure 77 – PSTN to Teams – Contd.

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

Figure 78 – PSTN to Teams

## Calls from Teams to Cisco

IP-to-IP Routing [TEAMS to CISCO]

Routing Policy: #0 [Default\_SBCRoutingPolicy]

GENERAL	ACTION
Index: 1	Destination Type: IP Group
Name: <b>TEAMS to CISCO</b>	Destination IP Group: <b>#2 [CISCO]</b> <a href="#">View</a>
Alternative Route Options: Route Row	Destination SIP Interface: <b>#2 [CISCO]</b> <a href="#">View</a>
MATCH	
Source IP Group: <b>#0 [TEAMS]</b> <a href="#">View</a>	Destination Address:
Request Type: All	Destination Port: 5060
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: .. <a href="#">View</a>
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: .. <a href="#">View</a>

Figure 79 – Teams to Cisco – Contd.

Destination Username Pattern: <b>6</b>	Routing Tag Name: default
Destination Host: *	Internal Action: <a href="#">Editor</a>
Destination Tag:	
Message Condition: .. <a href="#">View</a>	
Call Trigger: Any	
ReRoute IP Group: Any <a href="#">View</a>	
<a href="#">Cancel</a> <a href="#">APPLY</a>	

Figure 80 – Teams to Cisco

IP-to-IP Routing [Cisco -> Teams Extn dialing]

Routing Policy: #0 [Default\_SBCRoutingPolicy]

GENERAL	ACTION
Index: 5	Destination Type: IP Group
Name: <b>Cisco -&gt; Teams Extn dialing</b>	Destination IP Group: <b>#0 [TEAMS]</b> <a href="#">View</a>
Alternative Route Options: Route Row	Destination SIP Interface: <b>#0 [TEAMS]</b> <a href="#">View</a>
MATCH	
Source IP Group: <b>#2 [CISCO]</b> <a href="#">View</a>	Destination Address:
Request Type: All	Destination Port: 0
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: .. <a href="#">View</a>
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: .. <a href="#">View</a>

Figure 81 – Cisco to Teams – Contd.

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

Figure 82 – Cisco to Teams



## 4.4.16 IP Group

### IP Group – Teams

The screenshot shows the configuration page for an IP Group named 'TEAMS'. The page is titled 'IP Groups [TEAMS]' and has a window control bar with a close button. At the top, there is a dropdown menu for 'SRD' set to '#0 [DefaultSRD]'. The configuration is organized into three main sections:

- GENERAL:** Contains fields for Index (0), Name (TEAMS), Topology Location (Down), Type (Server), Proxy Set (#0 [TEAMS]), IP Profile (#1 [TEAMS\_Profile]), Media Realm (#0 [TEAMS]), Contact User, SIP Group Name (sbc4.tekvizionlabs.com), and Created By Routing Server (No).
- QUALITY OF EXPERIENCE:** Contains QoE Profile (--) and Bandwidth Profile (--), both with 'View' links.
- MESSAGE MANIPULATION:** Contains Inbound Message Manipulation Set (1), Outbound Message Manipulation Set (2), Message Manipulation User-Defined String 1, Message Manipulation User-Defined String 2, and Proxy Keep-Alive using IP Group settings (Enable).

Red boxes highlight the following fields: Name (TEAMS), Proxy Set (#0 [TEAMS]), IP Profile (#1 [TEAMS\_Profile]), Media Realm (#0 [TEAMS]), Inbound Message Manipulation Set (1), and Outbound Message Manipulation Set (2).

Figure 83 – IP Groups Teams – Contd.

IP Groups [TEAMS]

Used By Routing Server: Not Used  
Proxy Set Connectivity: Connected

**SBC GENERAL**

Classify By Proxy Set: Disable  
SBC Operation Mode: Not Configured  
SBC Client Forking Mode: Sequential  
CAC Profile: .. View

**ADVANCED**

Local Host Name: sdc4.tekvisionlabs.com  
UII Format: Disable  
Always Use Src Address: No

**SBC REGISTRATION AND AUTHENTICATION**

Max. Number of Registered Users: -1  
Registration Mode: User Initiates Registration  
User Stickiness: Disable  
User UDP Port Assignment: Disable  
Authentication Mode: User Authenticates  
Authentication Method List:  
SBC Server Authentication Type: According to Global Parameter  
OAuth HTTP Service: .. View  
Username: Admin  
Password: .....

**GW GROUP STATUS**

GW Group Registered IP Address:

Figure 84 – IP Groups Teams – Contd.

**SBC ADVANCED**

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

Cancel APPLY

Figure 85 – IP Groups Teams

## IP Group – PSTN Gateway

IP Groups [PSTNGW]

SRD: #0 [DefaultSRD]

**GENERAL**

Index: 1  
Name: PSTNGW  
Topology Location: Up  
Type: Server  
Proxy Set: #1 [PSTNGW] View  
IP Profile: #2 [PSTNGW\_Profile] View  
Media Realm: #1 [TEKVLAN] View  
Contact User:  
SIP Group Name: 10.64.1.72  
Created By Routing Server: No

**QUALITY OF EXPERIENCE**

QoE Profile: .. View  
Bandwidth Profile: .. View

**MESSAGE MANIPULATION**

Inbound Message Manipulation Set: 0  
Outbound Message Manipulation Set: 3  
Message Manipulation User-Defined String 1:  
Message Manipulation User-Defined String 2:  
Proxy Keep-Alive using IP Group settings: Enable

Figure 86 – IP Groups PSTN – Contd.

IP Groups [PSTNGW]

Used By Routing Server: Not Used

Proxy Set Connectivity: Connected

**SBC GENERAL**

Classify By Proxy Set: Enable

SBC Operation Mode: Not Configured

SBC Client Forking Mode: Sequential

CAC Profile: .. [View](#)

**ADVANCED**

Local Host Name:

UI Format: Disable

Always Use Src Address: No

**SBC REGISTRATION AND AUTHENTICATION**

Max. Number of Registered Users: -1

Registration Mode: User Initiates Registration

User Stickiness: Disable

User UDP Port Assignment: Disable

Authentication Mode: User Authenticates

Authentication Method List:

SBC Server Authentication Type: According to Global Parameter

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

Username: Admin

Password: .....

**GW GROUP STATUS**

GW Group Registered IP Address:

Figure 87 – IP Groups PSTN – Contd.

**SBC ADVANCED**

Source URI Input:

Destination URI Input:

SIP Connect: No

SBC PSAP Mode: Disable

Route Using Request URI Port: Disable

DTLS Context: #0 [default] [View](#)

Keep Original Call-ID: No

Dial Plan: .. [View](#)

Call Setup Rules Set ID: -1

Tags:

Cancel [APPLY](#)

Figure 88 – IP Groups PSTN

IP Group – Cisco

IP Groups [CISCO] - x

SRD

GENERAL		QUALITY OF EXPERIENCE	
Index	<input type="text" value="2"/>	QoE Profile	<input type="text" value="--"/> <a href="#">View</a>
Name	<input type="text" value="CISCO"/>	Bandwidth Profile	<input type="text" value="--"/> <a href="#">View</a>
Topology Location	<input type="text" value="Down"/>	<b>MESSAGE MANIPULATION</b>	
Type	<input type="text" value="Server"/>	Inbound Message Manipulation Set	<input type="text" value="4"/>
Proxy Set	<input type="text" value="#2 [CISCO]"/> <a href="#">View</a>	Outbound Message Manipulation Set	<input type="text" value="5"/>
IP Profile	<input type="text" value="#3 [CISCO_Profile]"/> <a href="#">View</a>	Message Manipulation User-Defined String 1	<input type="text"/>
Media Realm	<input type="text" value="#1 [TEKVLAN]"/> <a href="#">View</a>	Message Manipulation User-Defined String 2	<input type="text"/>
Contact User	<input type="text"/>	Proxy Keep-Alive using IP Group settings	<input type="text" value="Disable"/>
SIP Group Name	<input type="text" value="10.70.69.70"/>		
Created By Routing Server	<input type="text" value="No"/>		

Figure 89 – IP Groups Cisco – Contd.

IP Groups [CISCO] - x

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

Figure 90 – IP Groups Cisco – Contd.

**SBC ADVANCED**

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

Cancel [APPLY](#)

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:

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

Figure 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

Message Manipulations [Filter Privacy ID except for Anonymous]

GENERAL		ACTION	
Index	<input type="text" value="28"/>	Action Subject	<input type="text" value="header:privacy"/> Editor
Name	<input type="text" value="Filter Privacy ID except for Anonymous"/>	Action Type	Remove
Manipulation Set ID	<input type="text" value="1"/>	Action Value	<input type="text"/> Editor
Row Role	Use Current Condition		
MATCH			
Message Type	<input type="text" value="invite:Request"/> Editor		
Condition	<input type="text" value="Header.From.URL.Host contains '.com'"/> Editor		

Cancel APPLY

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

The screenshot shows the 'Message Manipulations' configuration window for a rule named '[modify pai host towards teams]'. The window is divided into 'GENERAL' and 'MATCH' sections. In the 'GENERAL' section, the 'Index' is 21, the 'Name' is 'modify pai host towards teams', and the 'Manipulation Set ID' is 2. In the 'MATCH' section, the 'Message Type' is 'Invite'. The 'ACTION' section on the right shows the configuration for the action: 'Action Subject' is 'header.P-Asserted-Identity.URL.Host', 'Action Type' is 'Modify', and 'Action Value' is 'SDC4.tekvizionlabs.com'. Red boxes highlight the Name, Manipulation Set ID, Message Type, and the entire ACTION configuration area.

Figure 94 – SIP Message Manipulation – PAI

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

The screenshot shows the 'Message Manipulations' configuration window for a rule named '[modify to towards teams]'. The window is divided into 'GENERAL' and 'MATCH' sections. In the 'GENERAL' section, the 'Index' is 19, the 'Name' is 'modify to towards teams', and the 'Manipulation Set ID' is 2. In the 'MATCH' section, the 'Message Type' is 'Invite.request'. The 'ACTION' section on the right shows the configuration for the action: 'Action Subject' is 'header.to.url.host', 'Action Type' is 'Modify', and 'Action Value' is 'sip.pstnhub.microsoft.com'. Red boxes highlight the Name, Manipulation Set ID, Message Type, and the entire ACTION configuration area.

Figure 95 – SIP Message Manipulation - To

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

Message Manipulations [Towards Teams FROM]

**GENERAL**

Index: 0

Name: Towards Teams FROM

Manipulation Set ID: 2

Row Role: Use Current Condition

**MATCH**

Message Type: Options

Condition: param.message.address.dst.sipinterface==0

**ACTION**

Action Subject: Header.From.URL

Action Type: Modify

Action Value: sip.admin@sbc4.tekvizionlabs.com

Buttons: Cancel, APPLY

Figure 96 – SIP Message Manipulation - From

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

Message Manipulations [towards Teams Contact]

**GENERAL**

Index: 1

Name: towards Teams Contact

Manipulation Set ID: 2

Row Role: Use Current Condition

**MATCH**

Message Type: Options

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

**ACTION**

Action Subject: Header.Contact.URL.Host

Action Type: Modify

Action Value: sbc4.tekvizionlabs.com

Buttons: Cancel, APPLY

Figure 97 – SIP Message Manipulation - Contact

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



Message Manipulations [Towards Teams]

GENERAL		ACTION	
Index	2	Action Subject	Header.From.URL.host Editor
Name	Towards Teams Editor	Action Type	Modify
Manipulation Set ID	2 Editor	Action Value	'sbc4.tekvizionlabs.com' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Invite.Request Editor		
Condition	Editor		

Cancel APPLY

Figure 98 – SIP Message Manipulation - From

## Manipulation to PSTN

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

Message Manipulations [towards PSTNGW TO]

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

Cancel APPLY

Figure 99 – SIP Message Manipulation – To

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

Message Manipulations [Towards PSTNGW FROM]

**GENERAL**

Index: 4

Name: Towards PSTNGW FROM

Manipulation Set ID: 3

Row Role: Use Current Condition

**MATCH**

Message Type: Options

Condition: Param.Message.Address.dst.SIPInterface==1

**ACTION**

Action Subject: Header.From.url.host

Action Type: Modify

Action Value: '10.64.3.10'

Buttons: Cancel, APPLY

Figure 100 – SIP Message Manipulation – From

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

Message Manipulations [Referred-By to PSTNGW]

**GENERAL**

Index: 5

Name: Referred-By to PSTNGW

Manipulation Set ID: 3

Row Role: Use Current Condition

**MATCH**

Message Type: Invite

Condition: Header.Referred-By exists

**ACTION**

Action Subject: Header.Referred-By.url.host

Action Type: Modify

Action Value: '10.64.3.10'

Buttons: Cancel, APPLY

Figure 101 – SIP Message Manipulation – Referred – By

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

The screenshot shows the configuration for a SIP message manipulation. The window title is "Message Manipulations: [Towards PSTNGW Invite]".

**GENERAL**

- Index: 6
- Name: Towards PSTNGW Invite
- Manipulation Set ID: 3
- Row Role: Use Current Condition

**MATCH**

- Message Type: Invite Request
- Condition: (empty)

**ACTION**

- Action Subject: Header.From.URL.Host
- Action Type: Modify
- Action Value: '10.64.3.10'

Buttons: Cancel, APPLY

Figure 102 – SIP Message Manipulation – From

### Manipulation to Cisco

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

The screenshot shows the configuration for a SIP message manipulation. The window title is "Message Manipulations: [Filter Privacy ID except for Anonymous]".

**GENERAL**

- Index: 30
- Name: Filter Privacy ID except for Anonymous
- Manipulation Set ID: 5
- Row Role: Use Current Condition

**MATCH**

- Message Type: Invite Request
- Condition: Header.From.URL.Host regex \.com

**ACTION**

- Action Subject: header.privacy
- Action Type: Remove
- Action Value: (empty)

Buttons: Cancel, APPLY

Figure 103 – SIP Message Manipulation – Privacy

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

Message Manipulations [Modify SBC IP Teams -> Cisco]

GENERAL		ACTION	
Index	11	Action Subject	Header.From.URL.Host Editor
Name	Modify SBC IP Teams -> Cisco	Action Type	Modify
Manipulation Set ID	5	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Invite Request Editor		
Condition			

Cancel APPLY

Figure 104 – SIP Message Manipulation – From

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

Message Manipulations [Referred-By Teams -> Cisco]

GENERAL		ACTION	
Index	12	Action Subject	Header.Referred-By.url.host Editor
Name	Referred-By Teams -> Cisco	Action Type	Modify
Manipulation Set ID	5	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Invite Editor		
Condition	Header.Referred-By exists Editor		

Cancel APPLY

Figure 105 – SIP Message Manipulation – Referred By

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

The screenshot shows the 'Message Manipulations [From header Teams -> Cisco]' configuration window. It is divided into 'GENERAL' and 'MATCH' sections. In the 'GENERAL' section, the 'Name' field is set to 'From header Teams -> Cisco' and the 'Manipulation Set ID' is '5'. In the 'MATCH' section, the 'Message Type' is 'Options' and the 'Condition' is 'Param.Message.Address.dst.SIPInterface=="2"'. The 'ACTION' section is also visible, with 'Action Subject' set to 'Header.From.URL.host', 'Action Type' set to 'Modify', and 'Action Value' set to '10.64.3.10'. Red boxes highlight the 'Name', 'Manipulation Set ID', 'Message Type', 'Condition', 'Action Subject', 'Action Type', and 'Action Value' fields.

Figure 106 – SIP Message Manipulation – From

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

The screenshot shows the 'Message Manipulations [To header Teams -> Cisco]' configuration window. It is divided into 'GENERAL' and 'MATCH' sections. In the 'GENERAL' section, the 'Name' field is set to 'To header Teams -> Cisco' and the 'Manipulation Set ID' is '5'. In the 'MATCH' section, the 'Message Type' is 'Options' and the 'Condition' is 'Param.Message.Address.dst.SIPInterface=="2"'. The 'ACTION' section is also visible, with 'Action Subject' set to 'header.to.uri.host', 'Action Type' set to 'Modify', and 'Action Value' set to '10.64.3.10'. Red boxes highlight the 'Name', 'Manipulation Set ID', 'Message Type', 'Condition', 'Action Subject', 'Action Type', and 'Action Value' fields.

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-hiding1
mode border-element license capacity 202
allow-connections sip to sip3
fax protocol pass-through g711ulaw
sip
bind control source-interface GigabitEthernet0/1
bind media source-interface GigabitEthernet0/1
session refresh
asserted-id pa4i
early-offer forced
```

---

<sup>1</sup> Hide signaling and media peer addresses from endpoints other than gateway.

<sup>2</sup> 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

<sup>3</sup> This command enables Cisco UBE basic IP-to-IP voice communication feature.

<sup>4</sup> 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 passthru5
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
```

---

<sup>5</sup> 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.





```
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 voip8  
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  
!
```

---

<sup>8</sup> Inbound Dial-peer for Cisco UCM facing network

```
dial-peer voice 11 voip9
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 voip10
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
```

---

<sup>9</sup> Outbound Dial-peer towards AudioCodes

<sup>10</sup> Inbound Dial peer from AudioCodes

```
!  
dial-peer voice 13 voip11  
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
```

---

<sup>11</sup> 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
```

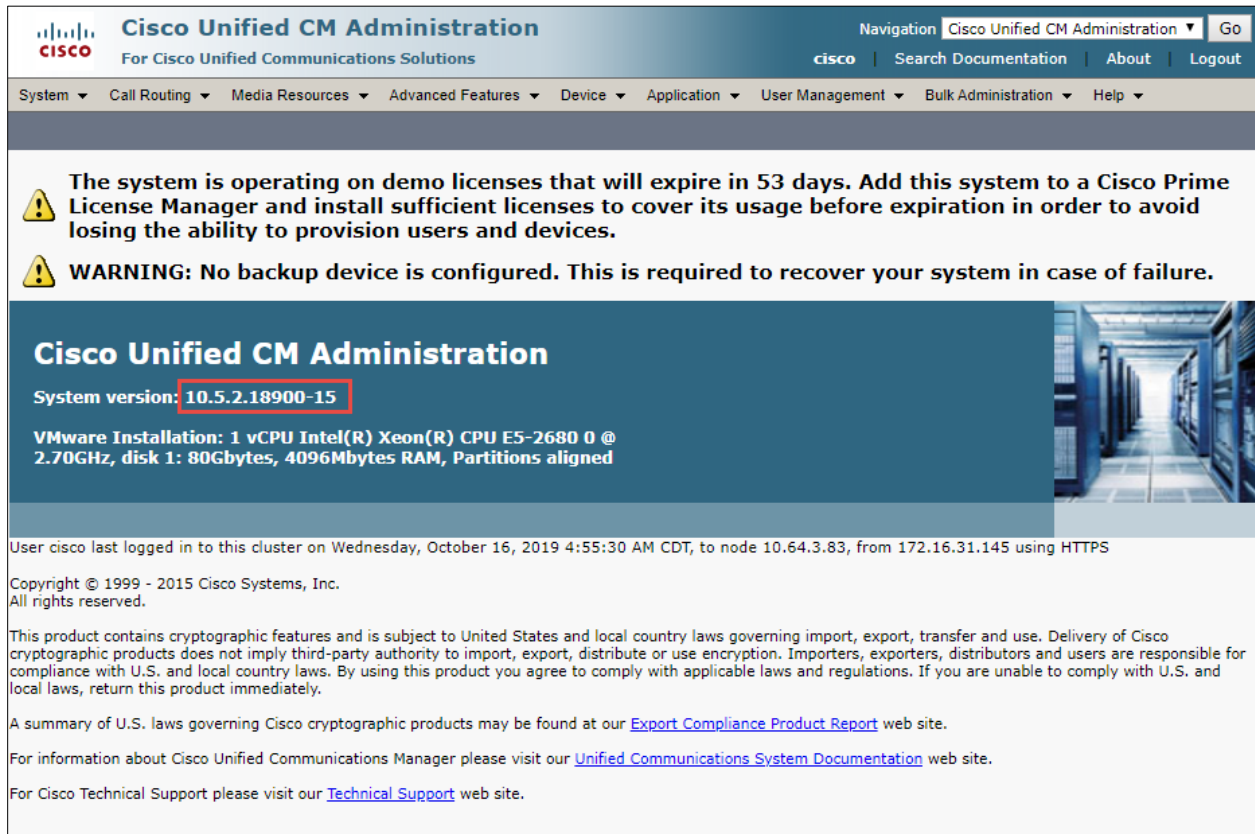
Crestron\_Teams#

## 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



The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and title "Cisco Unified CM Administration" are visible, along with navigation links for "Search Documentation", "About", and "Logout". A navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". Two warning messages are present: "The system is operating on demo licenses that will expire in 53 days. Add this system to a Cisco Prime License Manager and install sufficient licenses to cover its usage before expiration in order to avoid losing the ability to provision users and devices." and "WARNING: No backup device is configured. This is required to recover your system in case of failure." The main content area shows "Cisco Unified CM Administration" with the "System version: 10.5.2.18900-15" highlighted in a red box. Below this, it lists "VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned". A footer section contains copyright information, a disclaimer about cryptographic features, and links to "Export Compliance Product Report", "Unified Communications System Documentation", and "Technical Support" web sites.

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)

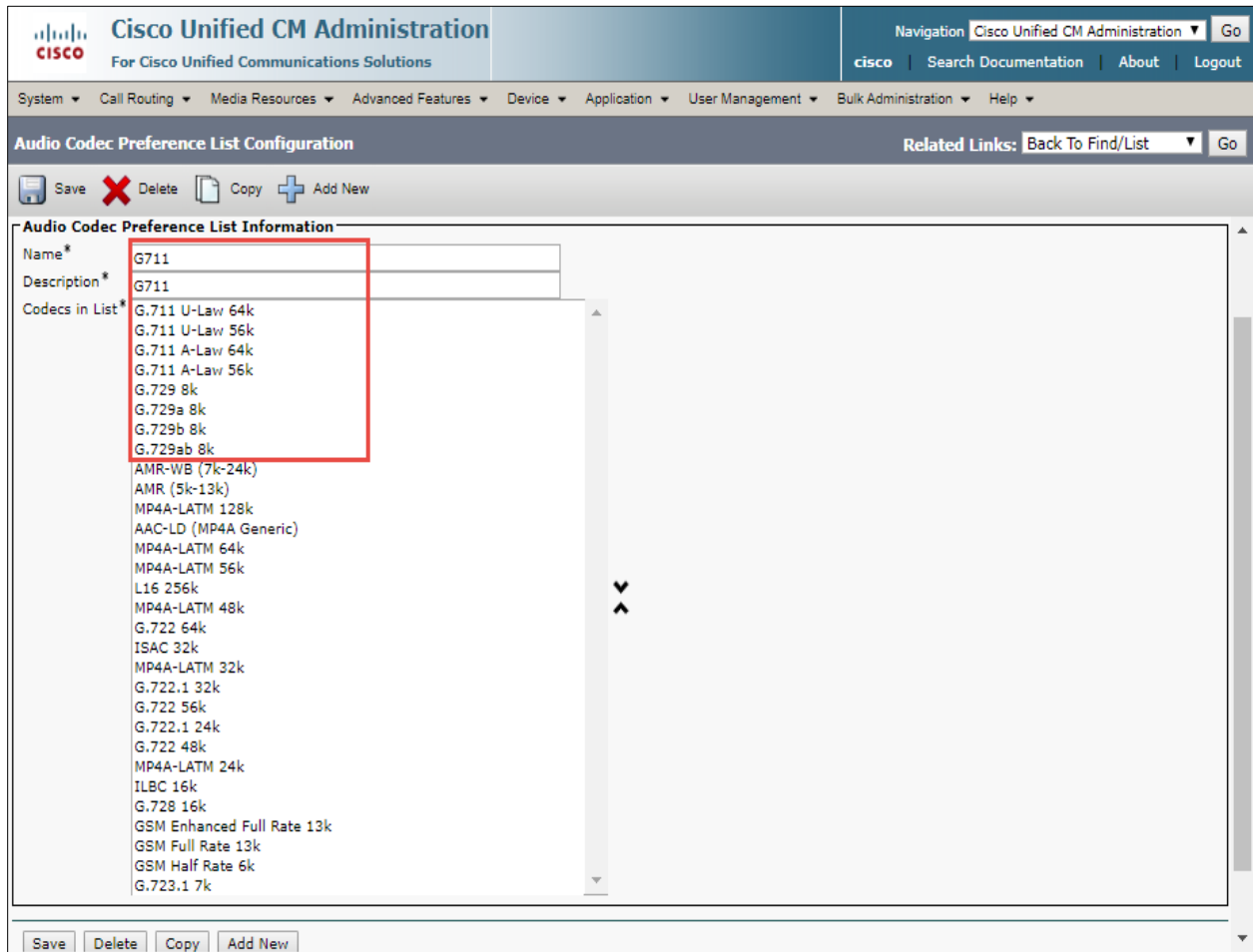


Figure 109 – Audio Codec Preference List



### 4.6.3 Cisco UCM Region Configuration

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

**Region Information**

Name\*

**Region Relationships**

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
g711_region	G711	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)

NOTE: Regions not displayed      Use System Default      Use System Default      Use System Default      Use System Default

**Modify Relationship to other Regions**

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<input type="checkbox"/> Default <input type="checkbox"/> g711_region <input type="checkbox"/> g729_region	<input type="radio"/> Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps	<input type="radio"/> Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps

Save Delete Reset Apply Config Add New

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

The screenshot shows the Cisco Unified CM Administration interface for configuring a Device Pool. The page title is "Device Pool Configuration" and the specific pool is "G711 (7 members\*\*)".

**Device Pool Information**

- Device Pool: G711 (7 members\*\*)

**Device Pool Settings**

- Device Pool Name\*: G711
- Cisco Unified Communications Manager Group\*: Default
- Calling Search Space for Auto-registration: < None >
- Adjunct CSS: < None >
- Reverted Call Focus Priority: Default
- Intercompany Media Services Enrolled Group: < None >

**Roaming Sensitive Settings**

- Date/Time Group\*: CMLocal
- Region\*: g711\_region
- Media Resource Group List: MGRL
- Location: < None >
- Network Locale: < None >
- SRST Reference\*: Disable
- Connection Monitor Duration\*\*\*: [Empty]
- Single Button Barge\*: Default
- Join Across Lines\*: Default
- Physical Location: < None >
- Device Mobility Group: < None >
- Wireless LAN Profile Group: < None > [View Details](#)

**Local Route Group Settings**

- Standard Local Route Group: < None >

**Device Mobility Related Information\*\*\*\***

- Device Mobility Calling Search Space: < None >
- AAR Calling Search Space: < None >
- AAR Group: < None >
- Calling Party Transformation CSS: < None >
- Called Party Transformation CSS: < None >

**Geolocation Configuration**

Figure 111 – Cisco UCM Device Pool – Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go  
 cisco | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Device Pool Configuration** Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Geolocation: < None >  
 Geolocation Filter: < None >

**Call Routing Information**

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS: < None >

**Connected Party Settings**

Connected Party Transformation CSS: < None >

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. This is used for this example

Set Device Pool\* = G711

The screenshot displays the Cisco Unified CM Administration interface for configuring an Annunciator. The page title is "Annunciator Configuration" and it includes a navigation menu at the top with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". Below the navigation, there are buttons for "Save", "Reset", and "Apply Config". A status message indicates "Update successful". The main configuration area, titled "Annunciator Information", contains the following fields:

Registration:	Registered with Cisco Unified Communications Manager nokiacum
IPv4 Address:	10.64.3.83
<input checked="" type="checkbox"/> Device is trusted	
Server*	[Redacted]ucm
Name*	ANN_2
Description	ANN
Device Pool*	G711
Location*	Hub_None
Use Trusted Relay Point*	Off

At the bottom, there are buttons for "Save", "Reset", and "Apply Config", and a note: "\*- indicates required item."

Figure 113 – Cisco UCM Annunciator

## 4.6.6 Cisco UCM Conference Bridge

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

Set Name\* = CFB\_2

Set Description = CFB. This is used for this example

Set Device Pool\* = G711

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with "Cisco Unified CM Administration" selected. Below this is a secondary menu with "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", and "Bulk Administration". The main content area is titled "Conference Bridge Configuration" and includes a "Related Links" section with "Back To Find/List". Below the title are "Save", "Reset", and "Apply Config" buttons. A "Status" section shows an "Update successful" message. The "Conference Bridge Information" section displays: "Conference Bridge : CFB\_2 (CFB)", "Registration: Registered with Cisco Unified Communications Manager nokiaucm", and "IPv4 Address: 10.64.3.83". The "Software Conference Bridge Info" section contains several fields: "Conference Bridge Type\*" is "Cisco Conference Bridge Software"; "Host Server" is partially obscured; a warning "Device is not trusted" is present; "Conference Bridge Name\*" is "CFB\_2" (highlighted with a red box); "Description" is "CFB"; "Device Pool\*" is "G711" (highlighted with a red box); "Common Device Configuration" is "< None >"; "Location\*" is "Hub\_None"; and "Use Trusted Relay Point\*" is "Default". At the bottom of the form are "Save", "Reset", and "Apply Config" buttons.

Figure 114 – Cisco UCM Conference Bridge

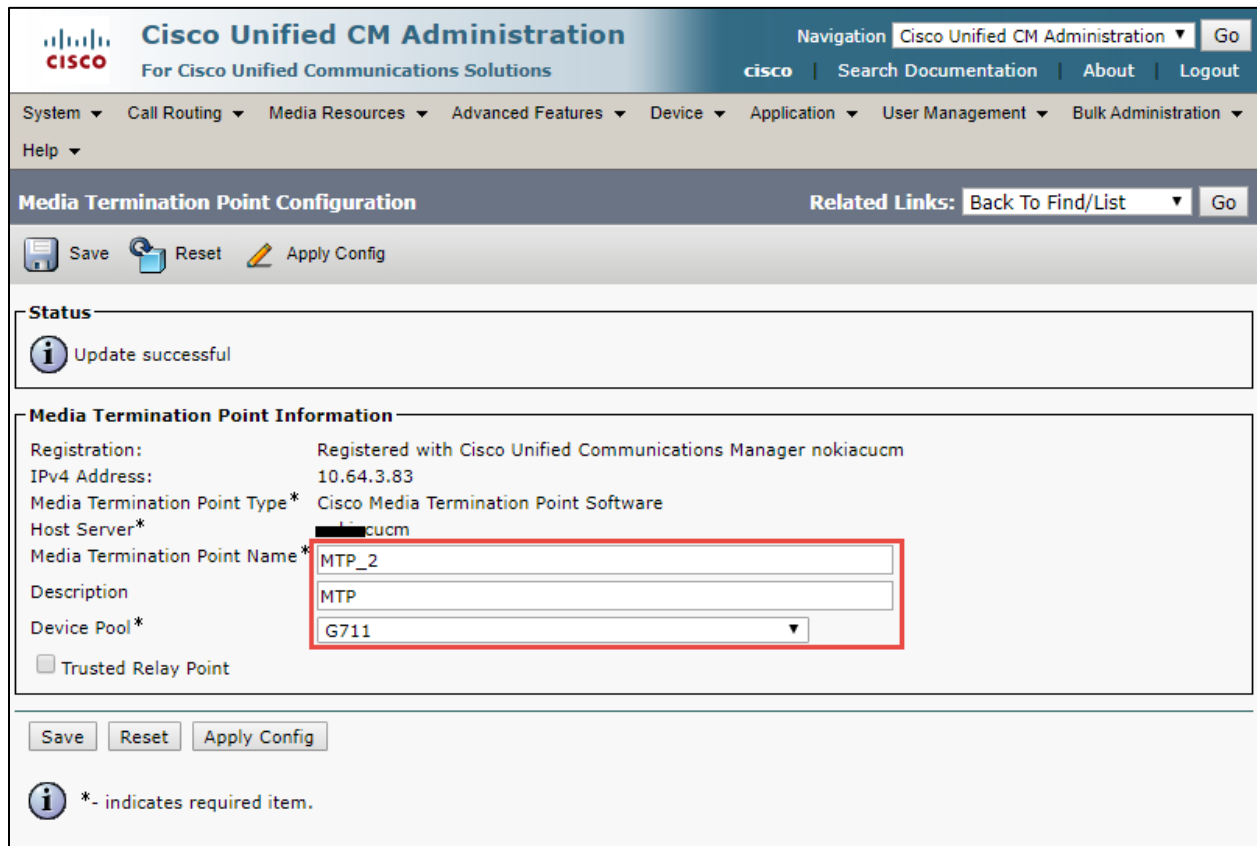
## 4.6.7 Cisco UCM MTP

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

Set Name\* = MTP\_2

Set Description = MTP. This is used for this example

Set Device Pool\* = G711



**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration ▾ Go  
cisco | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾  
Help ▾

**Media Termination Point Configuration** Related Links: Back To Find/List ▾ Go

Save Reset Apply Config

**Status**  
Update successful

**Media Termination Point Information**

Registration: Registered with Cisco Unified Communications Manager nokiacucm  
IPv4 Address: 10.64.3.83  
Media Termination Point Type\*: Cisco Media Termination Point Software  
Host Server\*: nokiacucm  
Media Termination Point Name\*: MTP\_2  
Description: MTP  
Device Pool\*: G711  
 Trusted Relay Point

Save Reset Apply Config

\*- indicates required item.

Figure 115 – Cisco UCM MTP

## 4.6.8 Cisco Media Resource Group

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

Set Name\* = MRG

Selected Media Resources Group as shown on below Screen used in this example

The screenshot shows the Cisco Unified CM Administration interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration". The status is "Ready". The Media Resource Group Name is "MRG". The "Selected Media Resources" list includes ANN\_2 (ANN), CFB\_2 (CFB), MOH\_2 (MOH), and MTP\_2 (MTP). The "Available Media Resources" list is empty. The "Use Multi-cast for MOH Audio" checkbox is unchecked.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go  
cisco | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Media Resource Group Configuration** Related Links: Back To Find/List | Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Media Resource Group Status**  
Media Resource Group: MRG (used by 0 devices)

**Media Resource Group Information**  
Name\*: MRG  
Description:

**Devices for this Group**  
Available Media Resources\*\*  
Selected Media Resources\*  
ANN\_2 (ANN)  
CFB\_2 (CFB)  
MOH\_2 (MOH)  
MTP\_2 (MTP)

Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save Delete Copy Add New

Figure 116 – Cisco UCM MTP

## 4.6.9 Cisco Media Resource Group List

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

Set Name\* = MRGL

Selected Media Resources Groups = MRG

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Media Resources" menu is expanded, showing "Media Resource Group List Configuration". The "Status" section indicates "Status: Ready". The "Media Resource Group List Status" section shows "Media Resource Group List: New". The "Media Resource Group List Information" section has a "Name\*" field containing "MRGL". The "Media Resource Groups for this List" section has two list boxes: "Available Media Resource Groups" (empty) and "Selected Media Resource Groups" (containing "MRG").

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
cisco | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Media Resource Group List Configuration** Related Links: Back To Find/List Go

Save

**Status**  
Status: Ready

**Media Resource Group List Status**  
Media Resource Group List: New

**Media Resource Group List Information**  
Name\* MRGL

**Media Resource Groups for this List**  
Available Media Resource Groups  
Selected Media Resource 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** = Cisco UBE. This is used for this example

Set **Device Pool\*** = G711. This is used for this example

Set **Media Resource Group List** = MRGL

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Cisco Unified CM Administration" and the breadcrumb navigation shows "Device -> Application -> User Management -> Bulk Administration -> Help". The main heading is "Trunk Configuration" with a "Related Links: Back To Find/List" button. Below the heading are icons for Save, Delete, Reset, and Add New. The "SIP Trunk Status" section shows "Service Status: Unknown - OPTIONS Ping not enabled" and "Duration: Unknown". The "Device Information" section contains the following fields:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type:	None(Default)
Device Name*:	Cube_Crestron_Teams
Description:	Cisco UBE
Device Pool*:	G711
Common Device Configuration:	< None >
Call Classification*:	Use System Default
Media Resource Group List:	MRGL
Location*:	Hub_None
AAR Group:	< None >
Tunneled Protocol*:	None
QSIG Variant*:	No Changes
ASN.1 ROSE OID Encoding*:	No Changes
Packet Capture Mode*:	None
Packet Capture Duration:	0

Below the table are several checkboxes and dropdown menus:

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support
- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
- Consider Traffic on This Trunk Secure\* (When using both sRTP and TLS)
- Route Class Signaling Enabled\* (Default)
- Use Trusted Relay Point\* (Default)
- PSTN Access
- Run On All Active Unified CM Nodes

Figure 118 – SIP Trunk – Cisco UBE – Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Intercompany Media Engine (IME)**  
E.164 Transformation Profile | < None >

**MLPP and Confidential Access Level Information**  
MLPP Domain | < None >  
Confidential Access Mode | < None >  
Confidential Access Level | < None >

**Call Routing Information**  
 Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\* | Default  
SIP Privacy\* | Default

**Inbound Calls**  
Significant Digits\* | All  
Connected Line ID Presentation\* | Default  
Connected Name Presentation\* | Default  
Calling Search Space | < None >  
AAR Calling Search Space | < None >  
Prefix DN |  
 Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**  
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.  
Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Figure 119 – SIP Trunk – Cisco UBE – Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Incoming Called Party Settings**  
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.  
Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**  
Connected Party Transformation CSS | < None >  
 Use Device Pool Connected Party Transformation CSS

**Outbound Calls**  
Called Party Transformation CSS | < None >  
 Use Device Pool Called Party Transformation CSS  
Calling Party Transformation CSS | < None >  
 Use Device Pool Calling Party Transformation CSS  
Calling Party Selection\* | Originator  
Calling Line ID Presentation\* | Default  
Calling Name Presentation\* | Default  
Calling and Connected Party Info Format\* | Deliver DN only in connected party  
 Redirecting Diversion Header Delivery - Outbound  
Redirecting Party Transformation CSS | < None >  
 Use Device Pool Redirecting Party Transformation CSS

**Caller Information**  
Caller ID DN |  
Caller Name |  
 Maintain Original Caller ID DN and Caller Name in Identity Headers

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\*** = Standard SIP Profile. This is used in this example.  
 Set **DTMF Signaling Method\*** = No Preference. This is used in this example.

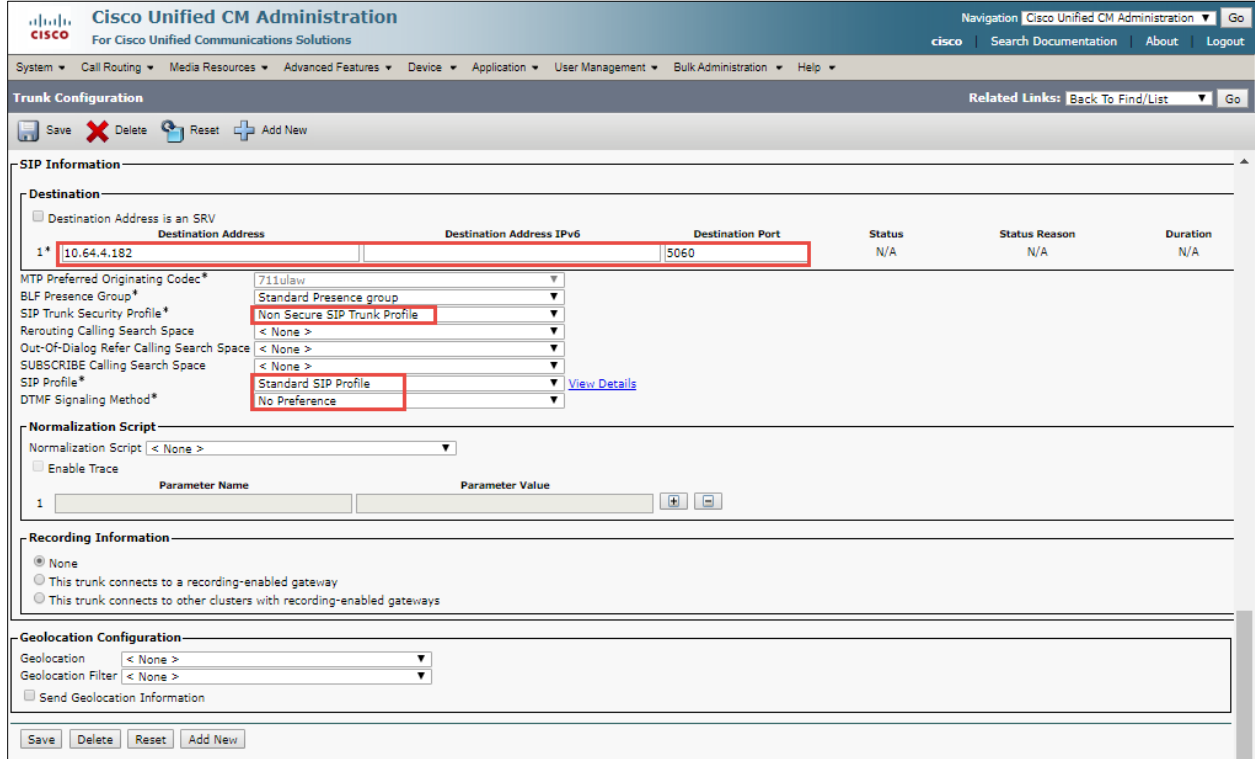


Figure 121 – SIP Trunk – Cisco UBE

#### 4.6.11 Cisco UCM SIP Trunk towards Cisco Unity

Set **Device Name\*** = Trunk-to-Unity. This is used for this example  
 Set **Description** = Trunk-to-Unity. This is used for this example  
 Set **Media Resource Group List** = MRGL

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
cisco Search Documentation About Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**SIP Trunk Status**  
Service Status: Unknown - OPTIONS Ping not enabled  
Duration: Unknown

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk-to-Unity
Description	Trunk-to-Unity
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MGRL
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input type="checkbox"/> PSTN Access	

Figure 122 – SIP Trunk – Cisco Unity – Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

Use Trusted Relay Point\* Default

PSTN Access

Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None >

**MLPP and Confidential Access Level Information**

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

**Call Routing Information**

Remote-Party-Id

Asserted-Identity

Asserted-Type\* Default

SIP Privacy\* Default

**Inbound Calls**

Significant Digits\* All

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Figure 123 – SIP Trunk – Cisco Unity – Contd.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, etc. The main content area is divided into several sections:

- Incoming Called Party Settings:** Contains a table for configuring incoming numbers. The table has columns for Number Type, Prefix, Strip Digits, Calling Search Space, and Use Device Pool CSS. The "Incoming Number" row shows "Default" for Prefix, "0" for Strip Digits, "< None >" for Calling Search Space, and a checked "Use Device Pool CSS" checkbox.
- Connected Party Settings:** Includes a dropdown for "Connected Party Transformation CSS" set to "< None >" and a checked "Use Device Pool Connected Party Transformation CSS" checkbox.
- Outbound Calls:** Contains multiple settings for outbound calls, including "Called Party Transformation CSS", "Use Device Pool Called Party Transformation CSS", "Calling Party Transformation CSS", "Use Device Pool Calling Party Transformation CSS", "Calling Party Selection", "Calling Line ID Presentation", "Calling Name Presentation", "Calling and Connected Party Info Format", "Redirecting Diversion Header Delivery - Outbound", "Redirecting Party Transformation CSS", and "Use Device Pool Redirecting Party Transformation CSS".
- Caller Information:** Includes fields for "Caller ID DN" and "Caller Name", and a checkbox for "Maintain Original Caller ID DN and Caller Name in Identity Headers".
- SIP Information:** This section is partially visible at the bottom of the page.

Figure 124 – SIP Trunk – Cisco Unity – Contd.

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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go  
 cisco | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**SIP Information**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	172.16.27.73		5060	N/A	N/A	N/A

MTP Preferred Originating Codec\* 711ulaw  
 BLF Presence Group\* Standard Presence group  
 SIP Trunk Security Profile\* Non Secure SIP Trunk Profile  
 Rerouting Calling Search Space < None >  
 Out-Of-Dialog Refer Calling Search Space < None >  
 SUBSCRIBE Calling Search Space < None >  
 SIP Profile\* Standard SIP Profile [View Details](#)  
 DTMF Signaling Method\* No Preference

**Normalization Script**

Normalization Script < None >  
 Enable Trace

	Parameter Name	Parameter Value
1		

**Recording Information**

None  
 This trunk connects to a recording-enabled gateway  
 This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation < None >  
 Geolocation Filter < None >  
 Send Geolocation Information

Save | Delete | Reset | Add New

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

The screenshot shows the Cisco Unity Connection Administration web interface. The left sidebar is expanded to 'Telephony Integrations' > 'Phone System'. The main content area is titled 'Phone System Basics (cisco\_crestron)'. The 'Phone System Name\*' field is highlighted with a red box and contains the text 'cisco\_crestron'. Below this are sections for 'Message Waiting Indicators', 'Call Loop Detection by Using DTMF', and 'Call Loop Detection by Using Extension'. The 'Call Loop Detection by Using Extension' section has a checked checkbox for 'Enable for Forwarded Message Notification Calls (by Using Extension)'. The 'Phone View Settings' section has an unchecked checkbox for 'Enable Phone View'. At the bottom, there are 'Save', 'Delete', 'Previous', and 'Next' buttons.

Figure 126 – SIP Trunk – Phone System – Contd.

The screenshot shows the 'Outgoing Call Restrictions' configuration page in the Cisco Unity Connection Administration interface. The left sidebar is expanded to 'Telephony Integrations' > 'Phone System'. The main content area has fields for 'Username', 'CTI Phone Access', and 'Password'. Below these are 'Outgoing Call Restrictions' options: 'Enable outgoing calls' (selected), 'Disable all outgoing calls immediately', and 'Disable all outgoing calls between'. The 'Disable all outgoing calls between' option has 'Beginning Time' and 'Ending Time' fields, both set to 12:00 AM. At the bottom, there are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (\*) are required.'

Figure 127 – SIP Trunk – Phone System – Contd.



## 4.7.2 Phone Group

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

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar is expanded to 'Telephony Integrations' > 'Port Group'. The main content area is titled 'Port Group Basics (CiscoUCM-10.5\_1)'. It includes a navigation bar with 'Save', 'Delete', 'Previous', and 'Next' buttons. A status message indicates: 'One or more port groups need to be reset.' The 'Port Group' section shows 'Display Name\*' as 'CiscoUCM-10.5\_1' (highlighted with a red box), 'Integration Method' as 'SIP', and 'Reset Status' as 'Reset Required' with a 'Reset' button. The 'Session Initiation Protocol (SIP) Settings' section has 'Register with SIP Server' checked, 'Authenticate with SIP Server' unchecked, and fields for 'Authentication Username', 'Authentication Password', and 'Contact Line Name'. The 'SIP Security Profile' is set to '5060' and 'SIP Transport Protocol' is 'TCP'. The 'Advertised Codec Settings' section has a 'Change Advertising' button and a table with columns 'Display Name' and 'Packet Size'.

Display Name	Packet Size
G.711 mu-law	20
G.729	20

Figure 128 –Phone Group – Contd.

The screenshot shows the 'Advertised Codec Settings' section of the configuration page. It features a 'Change Advertising' button and a table with columns 'Display Name' and 'Packet Size'. The table contains two rows: 'G.711 mu-law' with a packet size of 20, and 'G.729' with a packet size of 20. Below the table is another 'Change Advertising' button. The 'Message Waiting Indicator Settings' section has 'Enable Message Waiting Indicators' checked and fields for 'Delay between Requests' (0 milliseconds), 'Maximum Concurrent Requests' (0), 'Retries After Successful Attempt' (0), and 'Retry Interval After Successful Attempt' (5 milliseconds). At the bottom are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (\*) are required.'

Display Name	Packet Size
G.711 mu-law	20
G.729	20

Figure 129 –Phone Group

## 4.7.3 Port

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

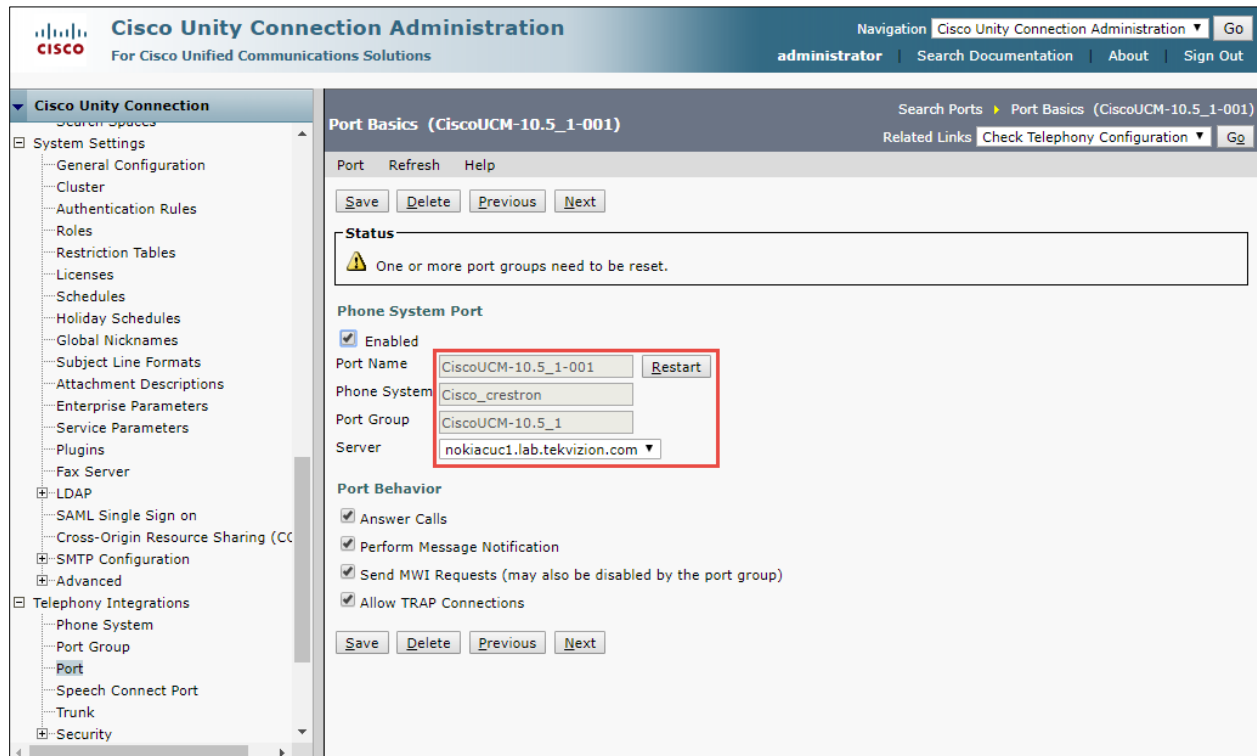


Figure 130 – Port

## 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** = **Crestron** - This is used to identify the User

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

All other values are default.

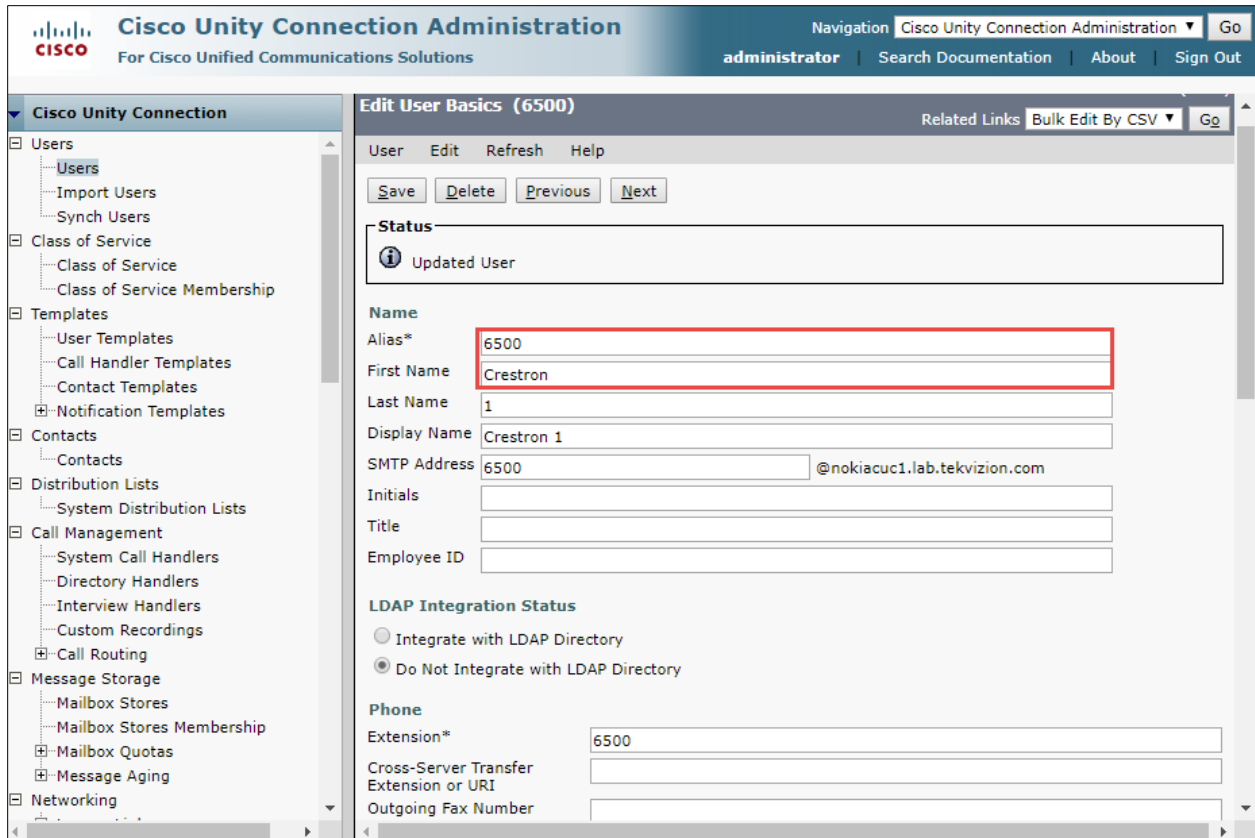


Figure 131 – User

Set **Users\*** = **Cisco\_crestron** - Phone system used in this example

**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unity Connection Administration Go  
administrator | Search Documentation | About | Sign Out

**Cisco Unity Connection**

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
  - Class of Service
  - Class of Service Membership
- Templates
  - User Templates
  - Call Handler Templates
  - Contact Templates
  - Notification Templates
- Contacts
  - Contacts
- Distribution Lists
  - System Distribution Lists
- Call Management
  - System Call Handlers
  - Directory Handlers
  - Interview Handlers
  - Custom Recordings
  - Call Routing
- Message Storage
  - Mailbox Stores
  - Mailbox Stores Membership
  - Mailbox Quotas
  - Message Aging
- Networking

Outgoing Fax Number:

Outgoing Fax Server: --- Not Selected ---

Partition: nokiauc1 Partition

Search Scope: nokiauc1 Search Space

Phone System: **Cisco\_crestron**

Class of Service: Voice Mail User COS

Active Schedule: Weekdays

Set for Self-enrollment at Next Sign-In

List in Directory

Send Non-Delivery Receipts on Failed Message Delivery

Skip PIN When Calling From a Known Extension  
**Caution!** Security risk. See Help for This Page for details.

Use Short Calendar Caching Poll Interval

Recorded Name:

**Location**

Address:

Building:

City:

State:

Postal Code:

Country: United States

Use System Default Time Zone

Time Zone: (GMT-06:00) America/Chicago

Language:  Use System Default Language

Figure 132 – User

- Contacts
  - Contacts
- Distribution Lists
  - System Distribution Lists
- Call Management
  - System Call Handlers
  - Directory Handlers
  - Interview Handlers
  - Custom Recordings
  - Call Routing
- Message Storage
  - Mailbox Stores
  - Mailbox Stores Membership
  - Mailbox Quotas
  - Message Aging
- Networking

Language:  Use System Default Language

English(United States)

Department:

Manager:

Billing ID:

Corporate Email Address:

Generate SMTP Proxy Address From Corporate Email Address

Directory URI:

Corporate Phone Number:

Fields marked with an asterisk (\*) are required.

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
<b>Cisco UBE</b>	<b>Cisco Unified Border Element</b>

## 6 Summary of Tests and Results

External ID	Title	Procedure	Expected Results	Status	Comments
1	Teams user Calls PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. PBX A user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	PASSED	
2	Teams user Calls PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. PBX B user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams
3	Teams user Calls PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		5. Teams user hangs up the call 6. Verify call is cleared successfully 7. Repeat steps 1 to 4 8. PSTN user hangs up the call 9. Verify call is cleared successfully	2. Call is disconnected		
4	Teams user Calls PBX A user and hangs up before answer	1. Make a voice call from Teams user to PBX A user 2. PBX A user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX A user is ringing 5. PBX A user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
5	Teams user Calls PBX B user and hangs up before answer	1. Make a voice call from Teams user to PBX B user 2. PBX B user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX B user is ringing 5. PBX B user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	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	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. PSTN user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Repeat steps 1 to 4</li> <li>8. Teams user hangs up the call</li> <li>9. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is connected with bi-directional audio, voice is clear, no echo</li> <li>2. Call is disconnected</li> </ol>	PASSED	
10	PBX A user Calls Teams user and hangs up before answer	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PBX A user hears Ring back Tone</li> <li>4. PBX A user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	PASSED	
11	PBX B user Calls Teams user and hangs up before answer	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PBX B user hears Ring back Tone</li> <li>4. PBX B user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. Teams user starts ringing</li> <li>3. PSTN user hears Ring back Tone</li> <li>4. PSTN user hangs up the call while Teams user is ringing</li> <li>5. Teams user stops ringing</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is disconnected before answer</li> </ol>	PASSED	
13	Teams user Calls PBX A user and performs hold/resume	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. Teams user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ol>	PASSED	
14	Teams user Calls PBX B user and performs hold/resume	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
		<ul style="list-style-type: none"> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. PBX A user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ul>			receiver or speaker button.
17	PBX B user Calls Teams user and Teams user performs hold/resume	<ul style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> <li>9. PBX B user hangs up the call</li> <li>10. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ul>	NOT APPLICABLE	PBX B is not tested with this cycle
18	PSTN user Calls Teams user and Teams performs hold/resume	<ul style="list-style-type: none"> <li>1. Make a voice call from PSTN user to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user initiates call hold</li> <li>6. Verify no audio is present while call is on hold</li> <li>7. Teams user resumes the call</li> <li>8. Verify two way audio is re-established between the two end points</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is placed on hold successfully</li> <li>2. No audio present during hold</li> <li>3. Call is resumed successfully</li> <li>4. Two way audio present after call is resumed</li> </ul>	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.

External ID	Title	Procedure	Expected Results	Status	Comments
		9. PSTN user hangs up the call 10. Verify call is cleared successfully			
19	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX A user	1. Make a voice call from Teams user to PBX A user 1 2. Teams user hears Ring back Tone 3. PBX A user 1 answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX A user 2 6. Verify PBX A user 1 is placed on hold 7. PBX A user 2 answers the call 8. Verify two way audio 9. Teams user completes the transfer 10. Verify two way audio between PBX A user 1 and PBX A user 2 11. PBX A user 1 hangs up the call 12. Verify call is cleared successfully	1. Call is transferred successfully 2. Two way audio present after call is transferred	PASSED	
20	Teams user Calls PBX A user, Teams user performs Attended Transfer to PBX B user	1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user places a consultation call to PBX B user 6. Verify PBX A user is placed on hold 7. PBX B user answers the call 8. Verify two way audio 9. Teams user completes the transfer	1. Call is transferred successfully 2. Two way audio present after call is transferred	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
		<ul style="list-style-type: none"> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user 1 and PBX B user 2</li> <li>11. PBX B user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>			
23	Teams user Calls PBX B user, Teams user performs Attended Transfer to PBX A user	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PBX A user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PBX A user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	NOT APPLICABLE	This testing is for only one PBX with Teams
24	Teams user Calls PBX B user, Teams user performs Attended	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
	Transfer to PSTN user	<ul style="list-style-type: none"> <li>6. Verify PBX B user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PSTN user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>			
25	Teams user Calls PSTN user, Teams user performs Attended Transfer to PBX B user	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user</li> <li>6. Verify PSTN user is placed on hold</li> <li>7. PBX B user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PSTN user and PBX B user</li> <li>11. PSTN user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	NOT APPLICABLE	This testing is for only one PBX with Teams
26	Teams user Calls PSTN user, Teams user performs	<ul style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> </ul>	<ul style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ul>	PASSED	



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

External ID	Title	Procedure	Expected Results	Status	Comments
	user performs Attended Transfer to PBX A user	<ol style="list-style-type: none"> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user 2</li> <li>6. Verify PBX A user 1 is placed on hold</li> <li>7. PBX A user 2 answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user 1 and PBX A user 2</li> <li>11. PBX A user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	present after call is transferred		
29	PBX A user Calls Teams user, Teams user performs Attended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user</li> <li>6. Verify PBX A user is placed on hold</li> <li>7. PBX B user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user and PBX B user</li> <li>11. PBX A user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> <li>6. Verify PBX A user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX A user and PSTN user</li> <li>11. PBX A user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
31	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user 1 to Teams user</li> <li>2. PBX B user 1 hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX B user 2</li> <li>6. Verify PBX B user 1 is placed on hold</li> <li>7. PBX B user 2 answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user 1 and PBX B user 2</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
		<ol style="list-style-type: none"> <li>11. PBX B user 1 hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>			
32	PBX B user Calls Teams user, Teams user performs Attended Transfer to PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PBX A user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PBX A user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> <li>10. Verify two way audio between PBX B user and PBX A user</li> <li>11. PBX B user hangs up the call</li> <li>12. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user places a consultation call to PSTN user</li> <li>6. Verify PBX B user is placed on hold</li> <li>7. PSTN user answers the call</li> <li>8. Verify two way audio</li> <li>9. Teams user completes the transfer</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX A user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX A user and PSTN user</li> <li>10. PBX A user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	
40	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user 1</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user 1 answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user 2</li> <li>6. PBX B user 2 starts ringing</li> <li>7. PBX B user 1 hears Ring back Tone</li> <li>8. PBX B user 2 answers the call</li> <li>9. Verify two way audio between PBX B user 1 and PBX B user 2</li> <li>10. PBX B user 1 hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PBX B user and PBX A user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams
42	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PSTN user</li> <li>6. PSTN user starts ringing</li> <li>7. PBX B user hears Ring back Tone</li> <li>8. PSTN user answers the call</li> <li>9. Verify two way audio between PBX B user and PSTN user</li> <li>10. PBX B user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX B user</li> <li>6. PBX B user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX B user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX B user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user transfers the call to PBX A user</li> <li>6. PBX A user starts ringing</li> <li>7. PSTN user hears Ring back Tone</li> <li>8. PBX A user answers the call</li> <li>9. Verify two way audio between PSTN user and PBX A user</li> <li>10. PSTN user hangs up the call</li> <li>11. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Call is transferred successfully</li> <li>2. Two way audio present after call is transferred</li> </ol>	PASSED	

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

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

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

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



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

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

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

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



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

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

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

External ID	Title	Procedure	Expected Results	Status	Comments
75	PBX A user Calls Teams user, Teams user CFA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user 2</li> <li>2. Make a voice call from PBX A user 1 to Teams user</li> <li>3. PBX A user 2 starts ringing</li> <li>4. PBX A user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
76	PBX A user Calls Teams user, Teams user CFA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
77	PBX A user Calls Teams user, Teams user CFA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user</li> <li>2. Make a voice call from PBX A user to Teams user</li> <li>3. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX A user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
78	PBX B user Calls Teams user, Teams user CFA to PBX B user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user 2</li> <li>2. Make a voice call from PBX B user 1 to Teams user</li> <li>3. PBX B user 2 starts ringing</li> <li>4. PBX B user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	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	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
80	PBX B user Calls Teams user, Teams user CFA to PSTN user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user</li> <li>2. Make a voice call from PBX B user to Teams user</li> <li>3. PSTN user starts ringing</li> <li>4. PSTN user answers the call</li> <li>5. Verify two way audio</li> <li>6. PBX B user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	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	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX B user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. PBX B user starts ringing</li> <li>4. PBX B user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	NOT APPLICABLE	This testing is for only one PBX with Teams
82	PSTN user Calls Teams user, Teams user CFA to PBX A user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PBX A user</li> <li>2. Make a voice call from PSTN user to Teams user</li> <li>3. PBX A user starts ringing</li> <li>4. PBX A user answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	
83	PSTN 1 user Calls Teams user, Teams user CFA to PSTN 2 user	<ol style="list-style-type: none"> <li>1. Teams user sets call forwarding all to PSTN user 2</li> <li>2. Make a voice call from PSTN user 1 to Teams user</li> <li>3. PSTN user 2 starts ringing</li> <li>4. PSTN user 2 answers the call</li> <li>5. Verify two way audio</li> <li>6. PSTN user 1 hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>	1. Teams user is able to forward the incoming call to correct destination	PASSED	

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

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

External ID	Title	Procedure	Expected Results	Status	Comments
94	Teams user with restricted Caller ID Calls PBX A user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PBX A user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX A user starts ringing</li> <li>4. Verify caller ID displayed on PBX A user is Unavailable/Private/Anonymous</li> <li>5. PBX A user answers the call</li> <li>6. Verify two way audio</li> <li>7. Teams user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
95	Teams user with restricted Caller ID Calls PBX B user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PBX B user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PBX B user starts ringing</li> <li>4. Verify caller ID displayed on PBX B user is Unavailable/Private/Anonymous</li> <li>5. PBX B user answers the call</li> <li>6. Verify two way audio</li> <li>7. Teams user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams
96	Teams user with restricted Caller ID Calls PSTN user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user with restricted caller ID to PSTN user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. PSTN user starts ringing</li> <li>4. Verify caller ID displayed on PSTN user is Unavailable/Private/Anonymous</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to dial an outbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<ol style="list-style-type: none"> <li>5. PSTN user answers the call</li> <li>6. Verify two way audio</li> <li>7. Teams user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>			
97	PBX A user with restricted Caller ID Calls Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user with restricted caller ID to Teams user</li> <li>2. PBX A user hears Ring back Tone</li> <li>3. Teams user starts ringing</li> <li>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous</li> <li>5. Teams user answers the call</li> <li>6. Verify two way audio</li> <li>7. PBX A user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive an inbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
98	PBX B user with restricted Caller ID Calls Teams user	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user with restricted caller ID to Teams user</li> <li>2. PBX B user hears Ring back Tone</li> <li>3. Teams user starts ringing</li> <li>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous</li> <li>5. Teams user answers the call</li> <li>6. Verify two way audio</li> <li>7. PBX B user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive an inbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from PSTN user with restricted caller ID to Teams user</li> <li>2. PSTN user hears Ring back Tone</li> <li>3. Teams user starts ringing</li> <li>4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous</li> <li>5. Teams user answers the call</li> <li>6. Verify two way audio</li> <li>7. PSTN user hangs up the call</li> <li>8. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive an inbound call with restricted caller ID</li> <li>2. Call is successful with two way audio</li> </ol>	PASSED	
100	PBX A user Calls Teams user and leaves voicemail	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user</li> <li>2. Teams user does not answer the call</li> <li>3. Allow the call to get forwarded to voicemail</li> <li>4. PBX A user successfully leaves voicemail</li> <li>5. Teams user receives voicemail notification</li> <li>6. Teams user successfully retrieves voicemail</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive and retrieve voicemail successfully</li> </ol>	PASSED	
101	PBX B user Calls Teams user and	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user</li> <li>2. Teams user does not answer the call</li> <li>3. Allow the call to get forwarded to</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to receive and retrieve voicemail successfully</li> </ol>	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	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user</li> <li>2. PBX B user does not answer the call</li> <li>3. Allow the call to get forwarded to voicemail</li> <li>4. Teams user successfully leaves voicemail and navigates voicemail menu using DTMF</li> </ol>	1. Teams user is able to leave voicemail and navigate voice mail menu using DTMF successfully	NOT APPLICABLE	This testing is for only one PBX with Teams
105	Teams user Calls PBX A user, PBX A returns call failure response	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user</li> <li>2. PBX A returns 486 Busy</li> <li>3. Verify Teams user gets appropriate notification or announcement and the call is cleared</li> <li>4. Repeat steps 1 to 3 where PBX A returns 480, 404, 503 SIP responses</li> <li>5. Document the observation on Teams user side</li> </ol>	1. Teams user handles the failure response successfully	PASSED	
106	Teams user Calls PBX A user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX A user using SIP URI</li> <li>2. PBX A user starts ringing</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	NOT TESTED	SIP URI Not tested for this PBX
107	Teams user Calls PBX B	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to PBX B user using SIP URI</li> <li>2. PBX B user starts ringing</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
	user using SIP URI	<ol style="list-style-type: none"> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	with two way audio successfully		
108	PBX A user Calls Teams user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX A user to Teams user using SIP URI</li> <li>2. PBX A user starts ringing</li> <li>3. PBX A user answers the call</li> <li>4. Verify two way audio</li> <li>5. PBX A user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	NOT TESTED	SIP URI Not tested for this PBX
109	PBX B user Calls Teams user using SIP URI	<ol style="list-style-type: none"> <li>1. Make a voice call from PBX B user to Teams user using SIP URI</li> <li>2. PBX B user starts ringing</li> <li>3. PBX B user answers the call</li> <li>4. Verify two way audio</li> <li>5. PBX B user hangs up the call</li> <li>6. Verify call is cleared successfully</li> </ol>	<ol style="list-style-type: none"> <li>1. Teams user is able to call using SIP URI</li> <li>2. Call is connected with two way audio successfully</li> </ol>	NOT APPLICABLE	This testing is for only one PBX with Teams
110	Teams user calls Skype for Business user	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to Skype for Business user</li> <li>2. Teams user hears Ring back Tone</li> <li>3. Skype for Business user answers the call</li> <li>4. Verify two way audio</li> <li>5. Teams user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Verify the same scenario where</li> </ol>		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	<ol style="list-style-type: none"> <li>1. Make a voice call from Skype for Business user to Teams user</li> <li>2. Skype for Business user hears Ring back Tone</li> <li>3. Teams user answers the call</li> <li>4. Verify two way audio</li> <li>5. Skype for Business user hangs up the call</li> <li>6. Verify call is cleared successfully</li> <li>7. Verify the same scenario where Skype for Business user is internal and external</li> </ol>		NOT APPLICABLE	Not applicable for this topology
112	Teams user calls Skype for Business External Mobile user	<ol style="list-style-type: none"> <li>1. Skype for business user is an External Mobile user</li> <li>2. Make a voice call from Teams user to Skype for Business user</li> <li>3. Teams user hears Ring back Tone</li> <li>4. Skype for Business user answers the call</li> <li>5. Verify two way audio</li> <li>6. Teams user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>		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	<ol style="list-style-type: none"> <li>1. Skype for business user is an External Mobile user</li> <li>2. Make a voice call from Skype for Business user to Teams user</li> <li>3. Skype for Business user hears Ring back Tone</li> <li>4. Teams user answers the call</li> <li>5. Verify two way audio</li> <li>6. Skype for Business user hangs up the call</li> <li>7. Verify call is cleared successfully</li> </ol>		NOT APPLICABLE	Not applicable for this topology
114	Teams user call other tenant users	<ol style="list-style-type: none"> <li>1. Make a voice call from Teams user to another tenant users (Teams desktop client user, Teams mobile user, Skype for Business Online user)</li> <li>2. Verify call is successful</li> <li>3. Make one call to each different user one by one</li> </ol>		NOT APPLICABLE	Not applicable for this topology
115	Teams users joins a meeting scheduled by Skype for business On-premises user	<ol style="list-style-type: none"> <li>1. Skype for business user schedules a meeting and invites Teams user 1 and Teams user 2</li> <li>2. Teams user 1 joins the meeting using the Join button</li> <li>3. Teams user 2 joins the meeting using the dial-in conferencing number</li> <li>4. Verify Teams users are able to join the meeting successfully</li> <li>5. Verify all three users are able to hear</li> </ol>		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