



Crestron UC-PHONE and UC-PHONE-PLUS

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

September 2019

Document History

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1 Audience

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

1.1 Crestron UC-PHONE and UC-PHONE-PLUS

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

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

1.2 tekVizion Labs

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

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

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

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

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

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

2 SIP Trunking Network Components

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

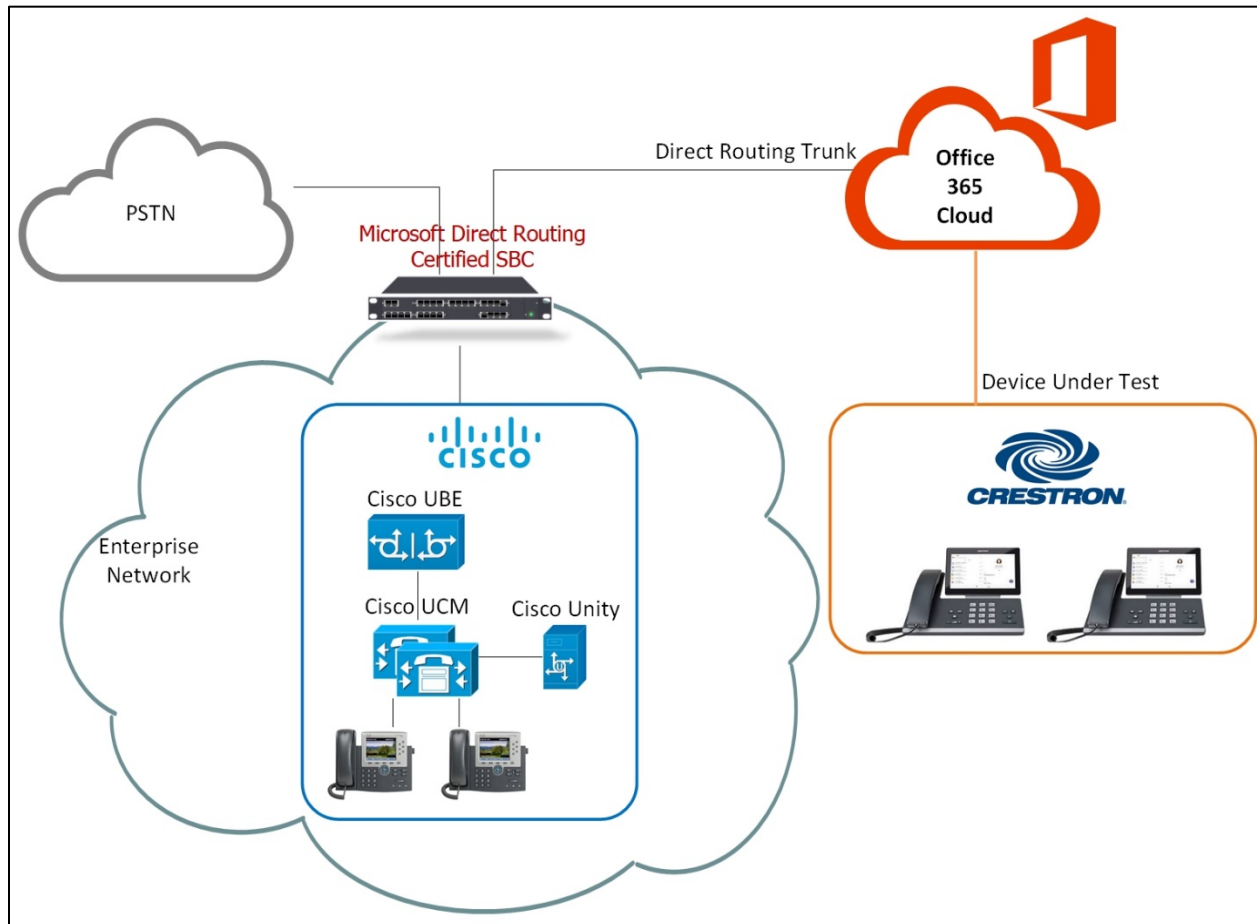


Figure 1 Network Topology

Numbering Plan

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

Dialing Plan

- Teams users and Cisco users call PSTN either doing 10 digits 11 digits dialing or E164 dialing
- Teams users call Cisco users by dialing 65XX

- Cisco users call Teams users by dialing 8XXX and AudioCodes will include the prefix +1972XXX and will send to Teams.

2.1 Hardware Components

- Microsoft Office 365 tenant with E5 without Audio Conferencing assigned to Teams users
- AudioCodes Mediant VE SBC for Teams Direct Routing serves as the demarcation point between customer's network and O365 WAN network
- Crestron UC-PHONE-PLUS and Crestron UC-PHONE phones
- Cisco UCM running on ESXi
- Cisco Unity Connection running on ESXi
- Cisco UBE v CISCO2921/K9
- PSTN Gateway

2.2 Software Requirements

- AudioCodes Mediant VE SBC v7.20A.250.003
- Cisco UCM v11.5.1.16900-16
- Cisco Unity Connection v11.5.1.12900-21
- Cisco UBE v 11.5.2
- Crestron UC-PHONE-PLUS v58.15.91.15

3 Features

3.1 Features Supported

- Basic Inbound and Basic Outbound
- Call hold and resume
- Call transfer (semi-attended and consultative)
- Conference
- Call forward (all, no answer)
- Busy On Busy
- Simultaneous ring
- Calling line identification restriction
- DTMF relay both directions (RFC2833)
- Call Failover

3.2 Caveats and Limitations

- Direct Routing supports call escalation to an adhoc conference without Audioconferencing license. However the UC-PHONE-PLUS and UC-PHONE desk phones could not add a user into conference without Audio Conferencing license.
- The UC-PHONE-PLUS desk phone is unable to resume a held call using soft-key, if the call has been answered by the phone using receiver or speaker button.

4 Configuration

4.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure **Microsoft Teams, Cisco UBE, Cisco UCM and AudioCodes** for SIP Trunking with **Microsoft Teams Direct Routing**.

Table 1 – PBX Configuration Steps

Steps	Description	Reference
Step 1	Microsoft Teams Configuration	Section 4.3
Step 2	AudioCodes VE SBC Configuration	Section 4.4
Step 3	Cisco UBE Configuration	Section 4.5
Step 4	Cisco UCM Configuration	Section 4.6

4.2 IP Address Worksheet

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

Table 2 – IP Addresses

Component	Lab Value
AudioCodes	
LAN IP Address	10.64.3.10
LAN Subnet Mask	255.255.255.0
WAN IP Address	192.XX.XX.XX
WAN Subnet Mask	255.255.255.128
Cisco UCM	
IP Address	172.16.29.81
Subnet Mask	255.255.255.0
Cisco UBE	
LAN IP Address	10.64.4.182
LAN Subnet Mask	255.255.255.0
WAN IP Address	10.70.69.70
WAN Subnet Mask	255.255.255.0
Cisco Unity	
LAN IP Address	10.80.18.5

LAN Subnet Mask	255.255.255.0
-----------------	---------------

4.3 Microsoft Teams Configuration

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

4.3.1 Teams User Configuration

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

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

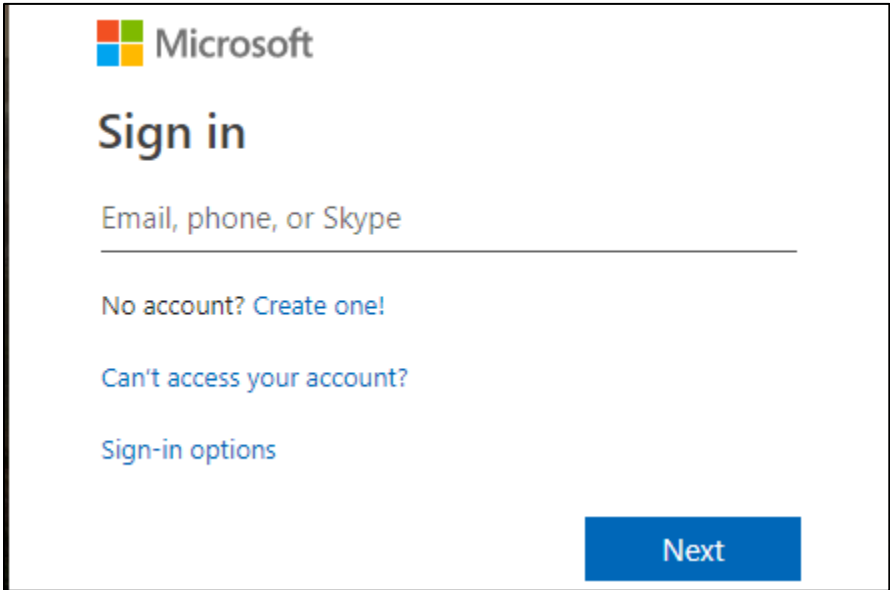


Figure 2: Office 365 Portal Login

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

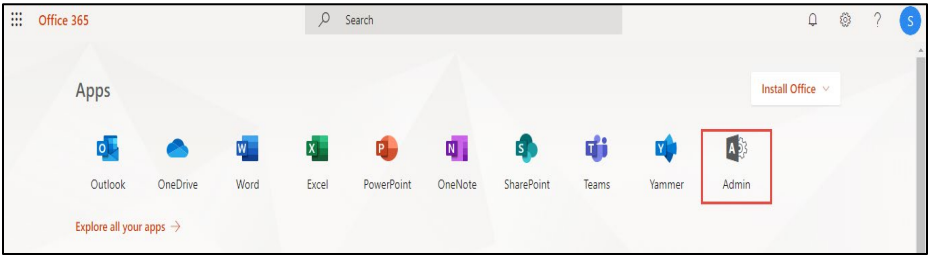


Figure 3: Office 365 Portal Login

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

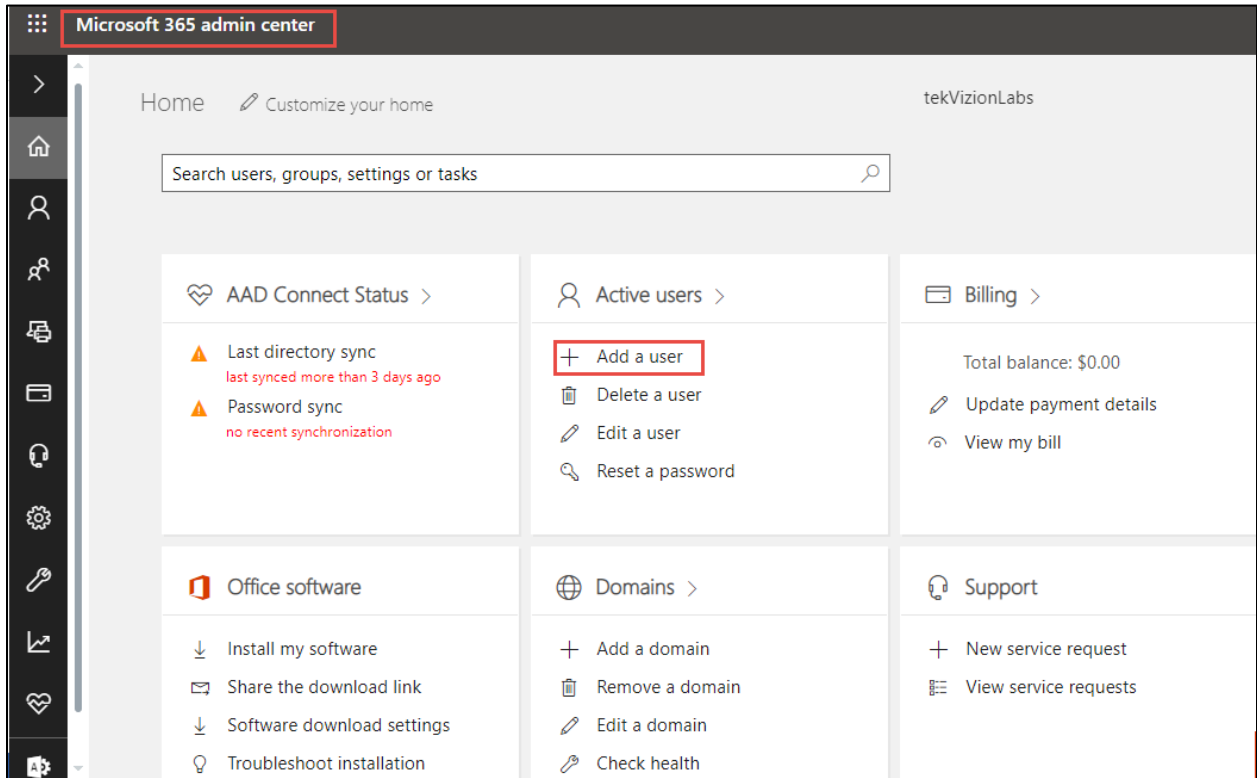


Figure 4: Teams User Creation

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

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.

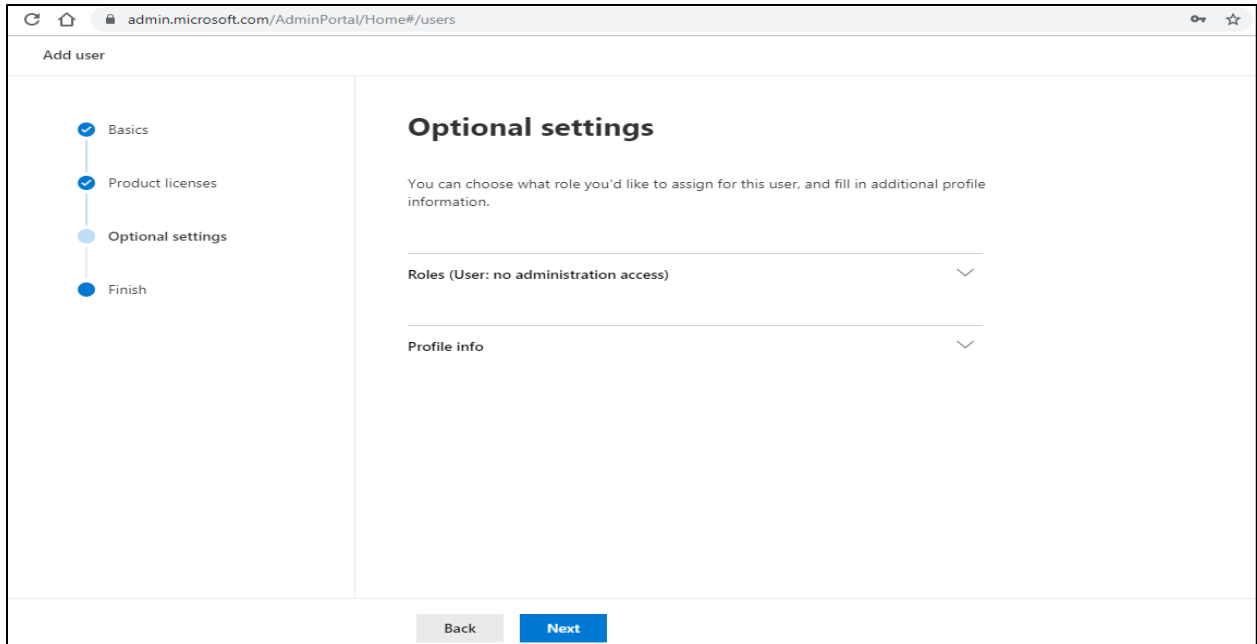


Figure 7: Teams User Creation – Contd.

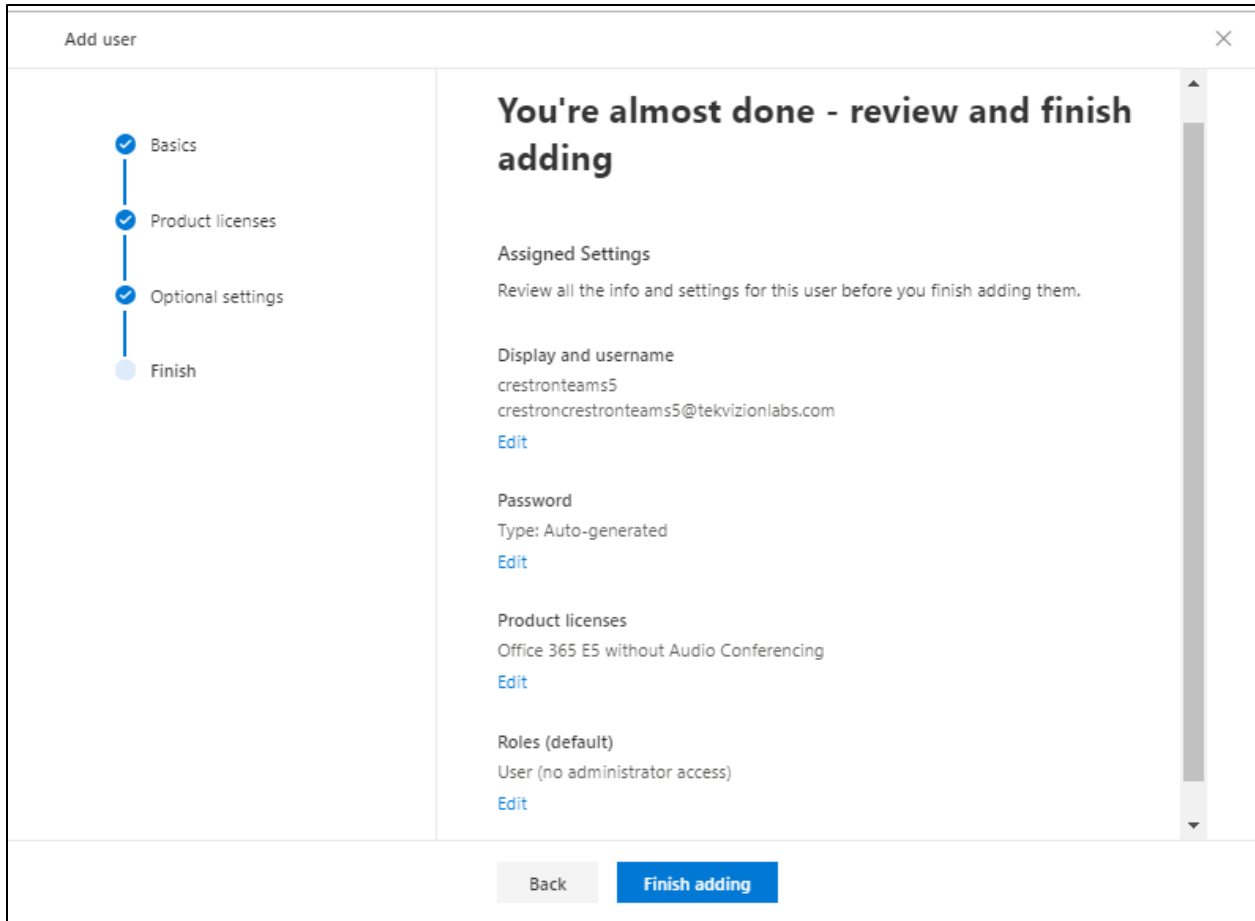


Figure 8: Teams User Creation – Contd.

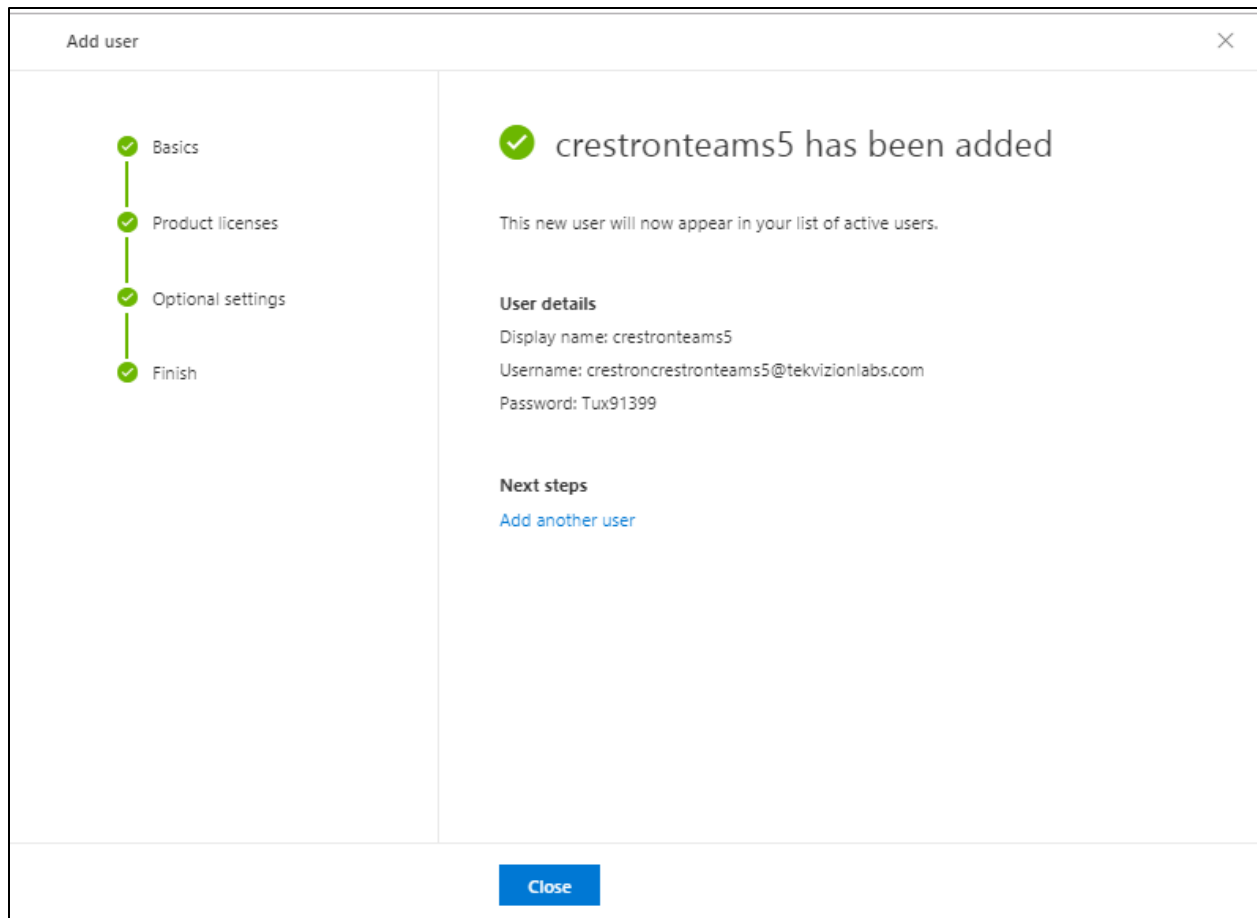


Figure 9: Teams User Creation – Contd.

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

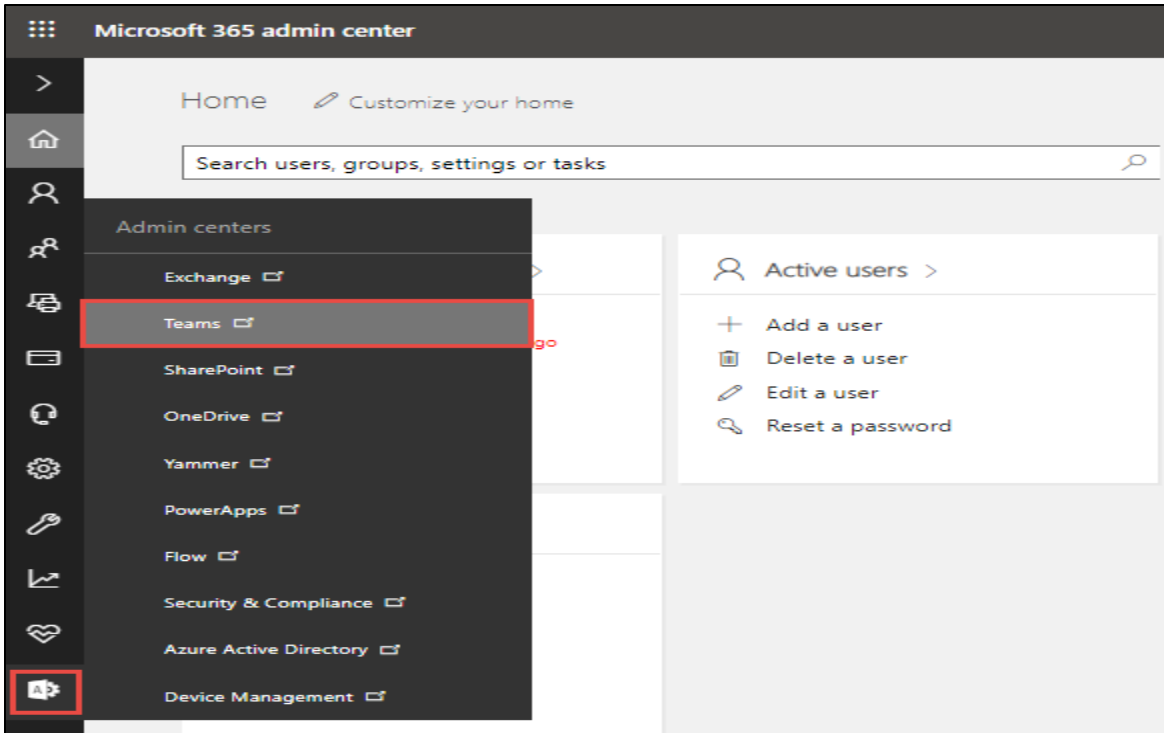


Figure 10: Microsoft O365 admin

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

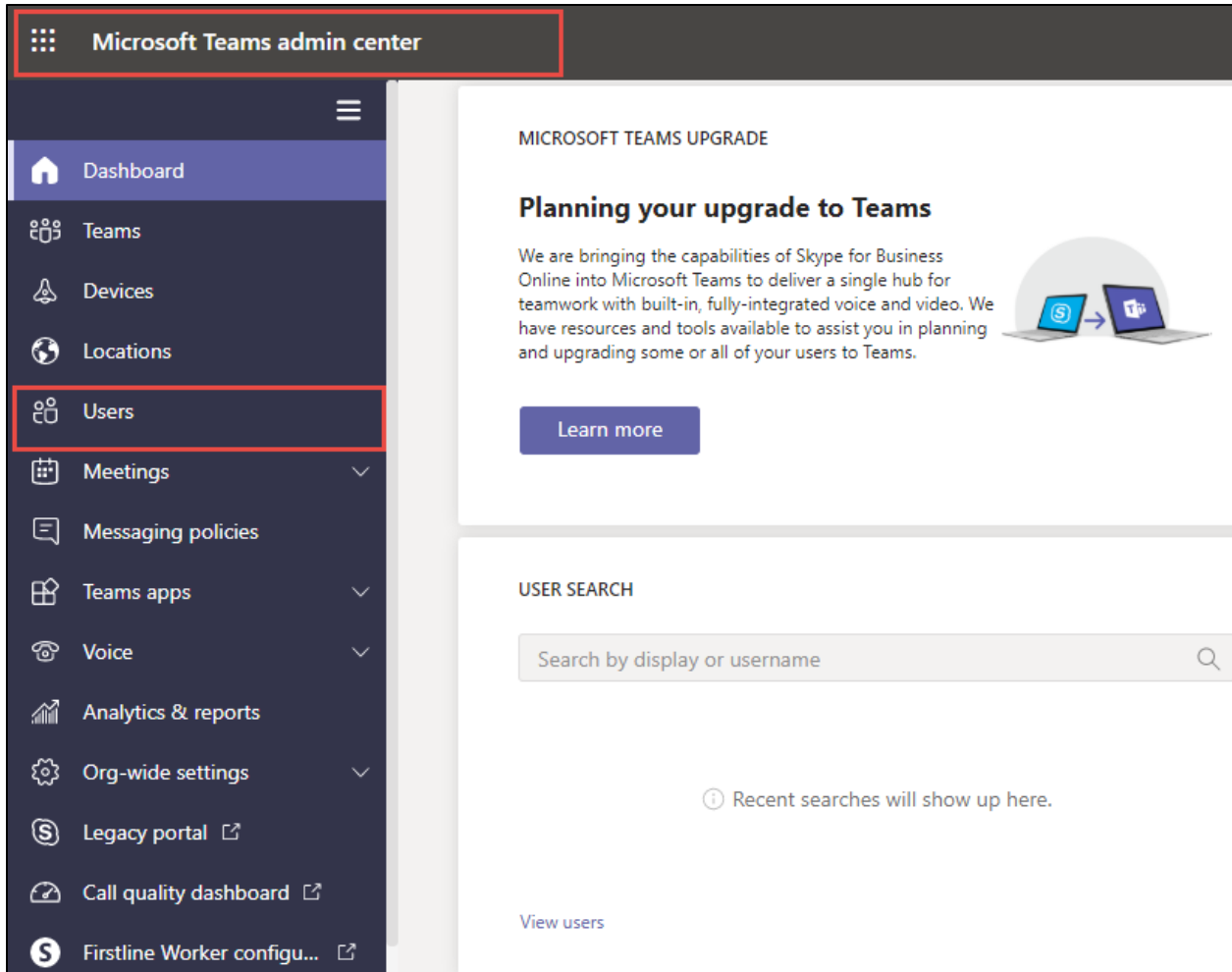


Figure 11: Microsoft O365 admin

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

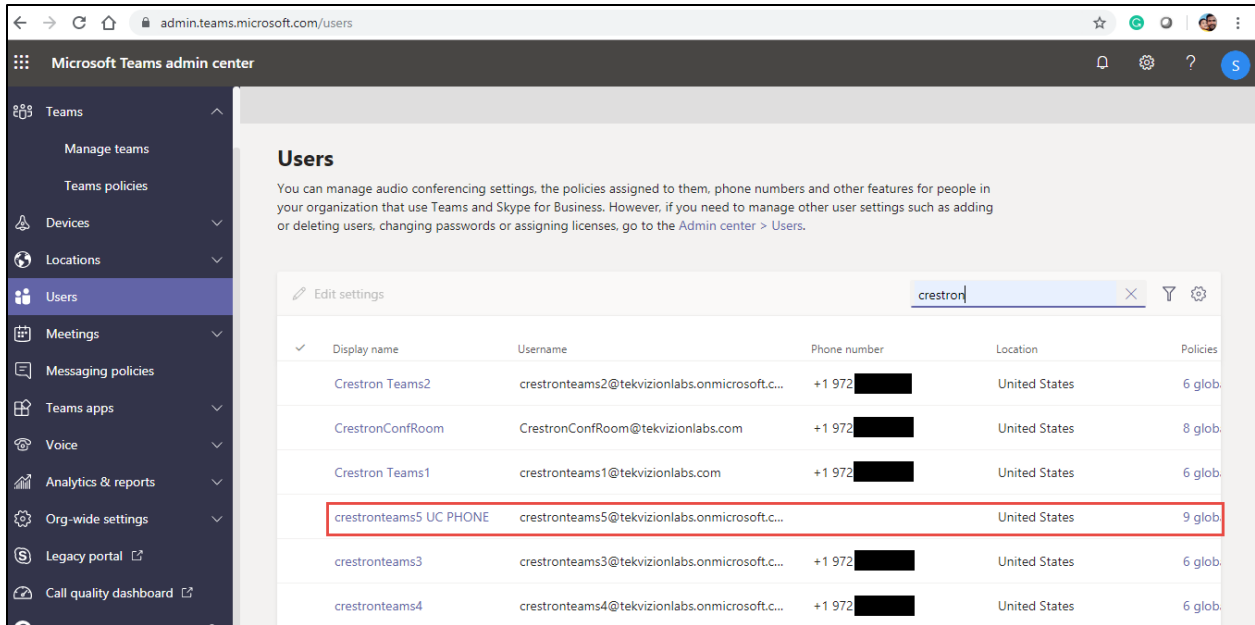


Figure 12: Microsoft O365 admin

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

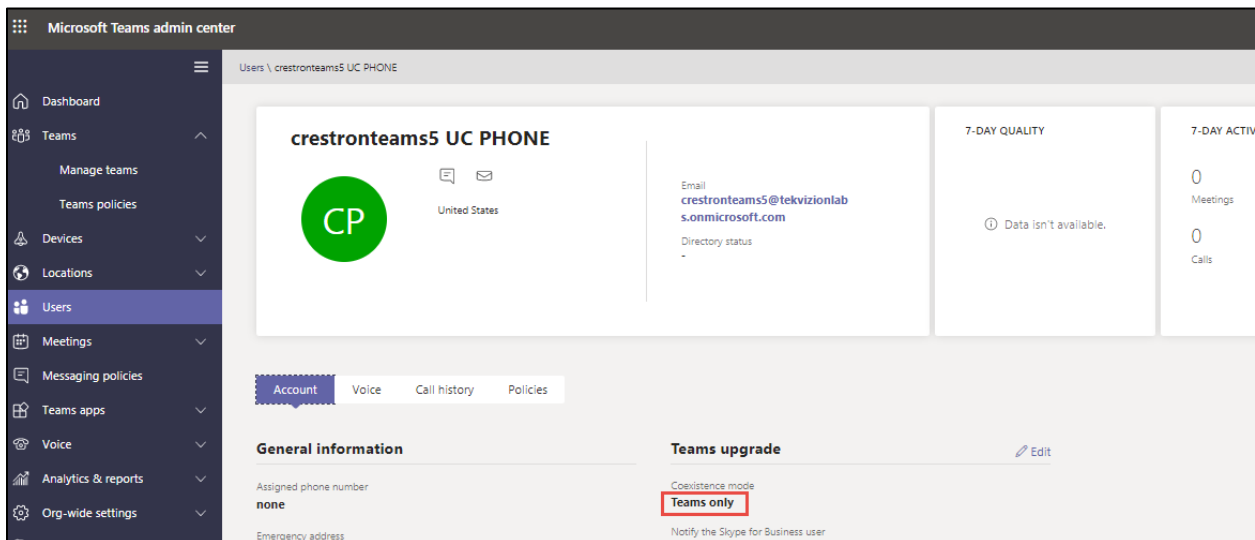


Figure 13: Teams User

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.

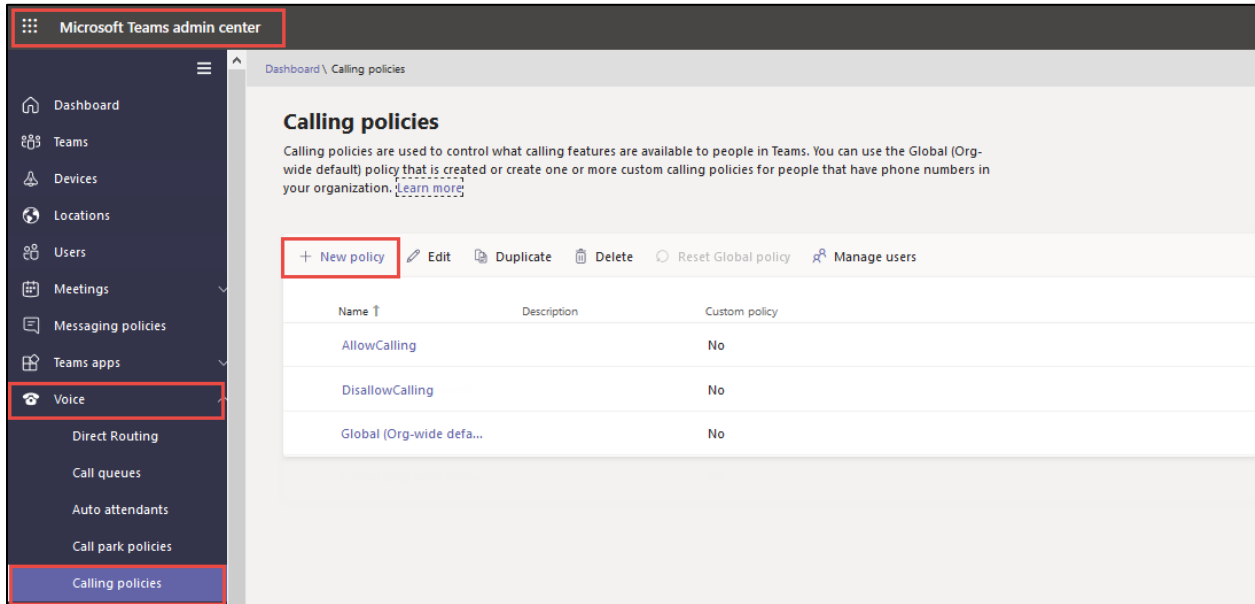


Figure 14 – Calling Policy

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 User controlled

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

Save Cancel

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" -EnterpriseVocieEnabled $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

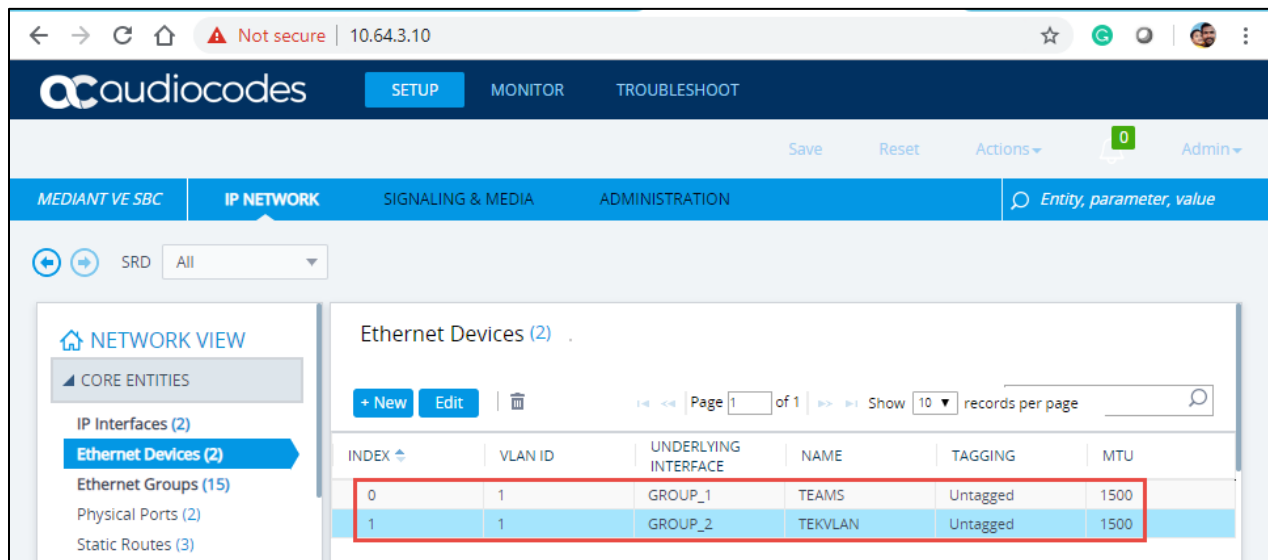
4.4.1 General

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. A table lists two Ethernet Devices:

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:

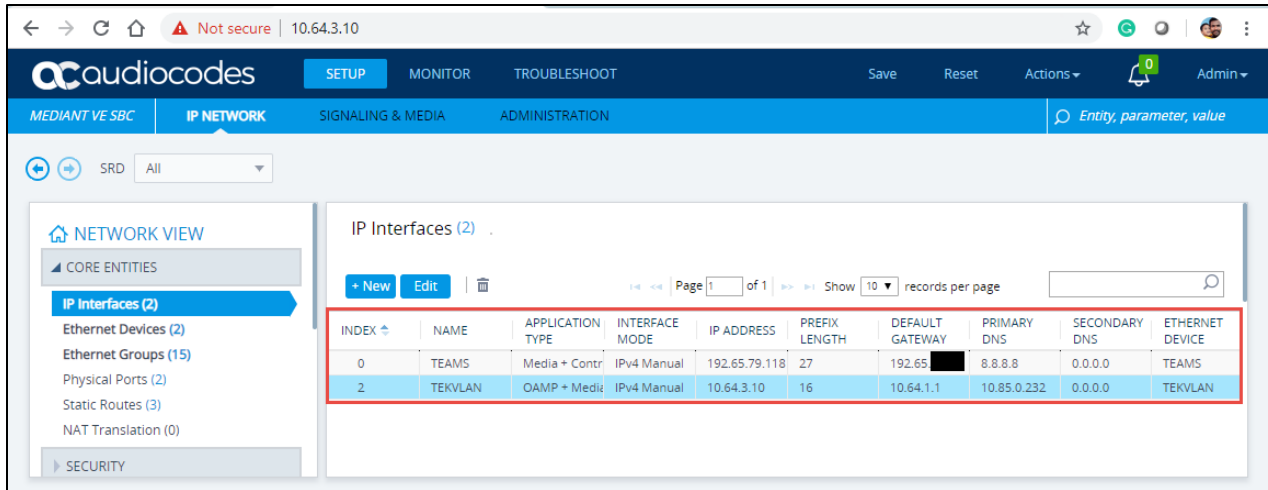


Figure 27 – IP interface Devices

IP interface TEAMS

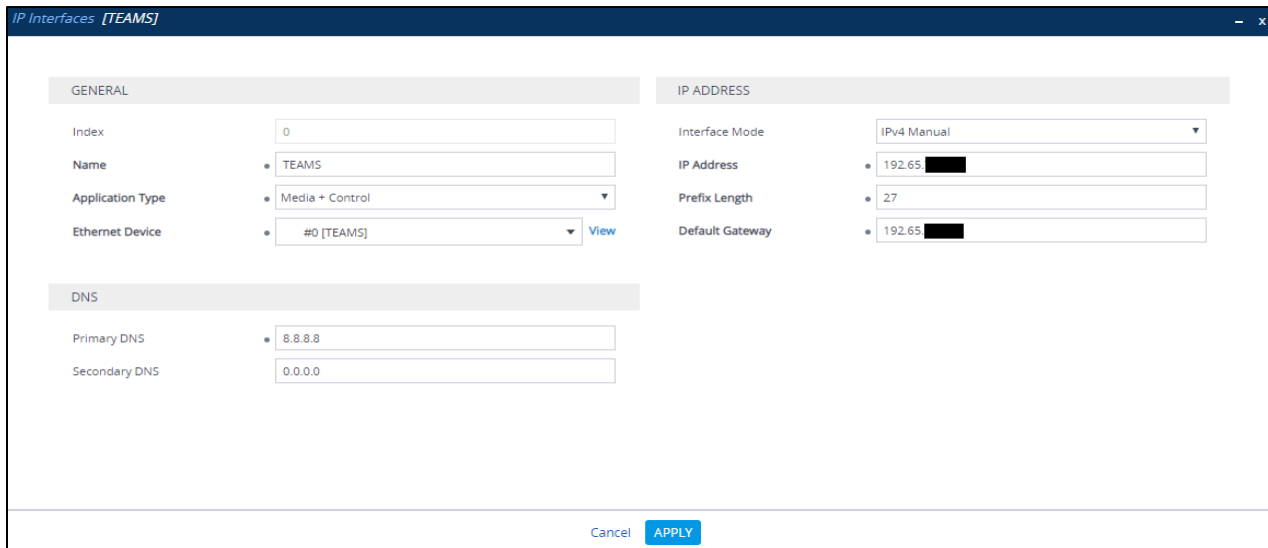


Figure 28 – IP interface Devices

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

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

Figure 30 – DNS SRV Records

4.4.5 Configure SRTP

By default, SRTP is disabled.

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

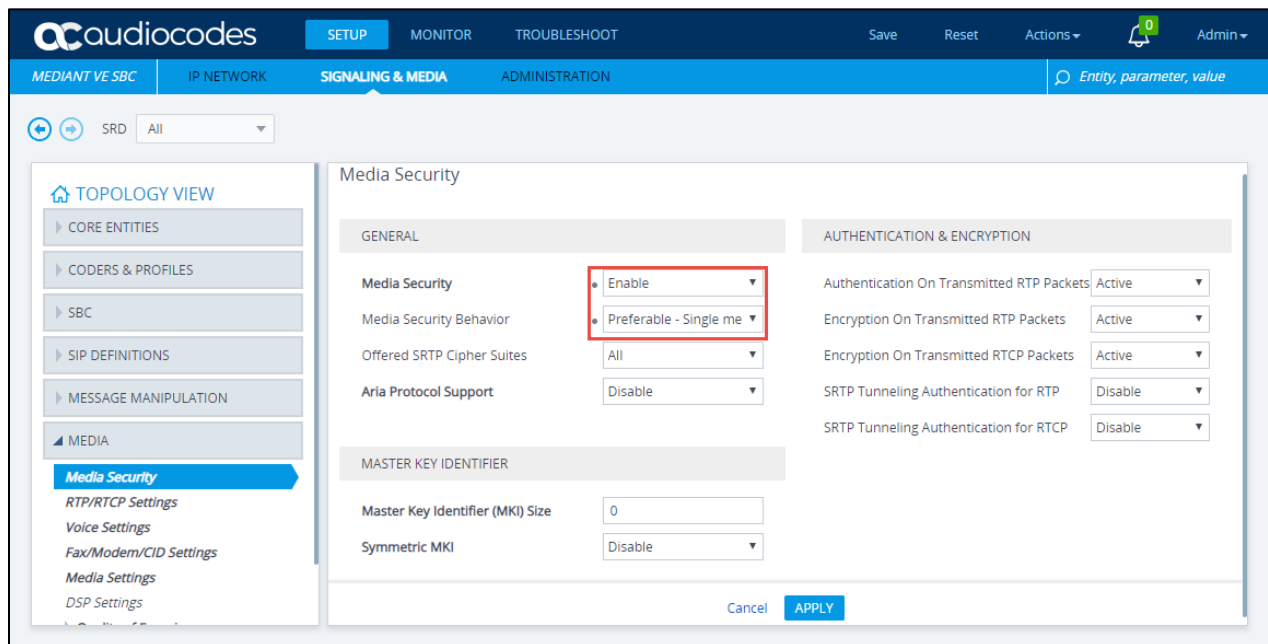


Figure 31 – Media Security

4.4.6 Configure TLS contexts

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

- AffirmTrust
- AddTrust External CA Root
- Baltimore CyberTrust Root
- Buypass
- Cybertrust
- Class 3 Public Primary Certification Authority
- Comodo Secure Root CA
- Deutsche Telekom

- DigiCert Global Root CA
- DigiCert High Assurance EV Root CA
- Entrust
- GlobalSign
- Go Daddy
- GeoTrust
- Verisign, Inc.
- Starfield
- Symantec Enterprise Mobile Root for Microsoft
- SwissSign
- Thawte Timestamping CA
- Trustwave
- TeliaSonera
- T-Systems International GmbH (Deutsche Telekom)
- QuoVadis

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

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

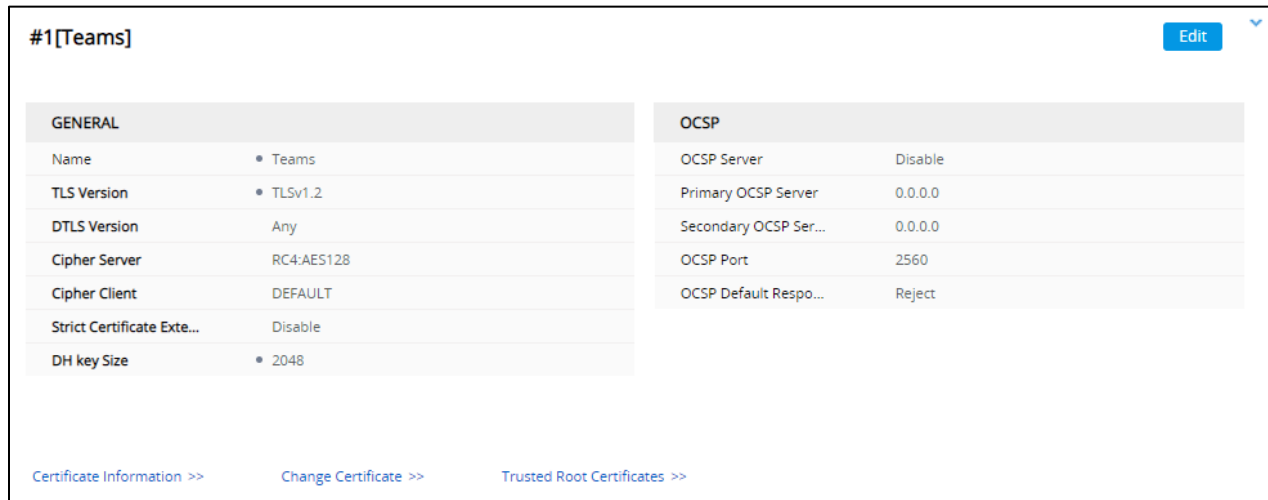


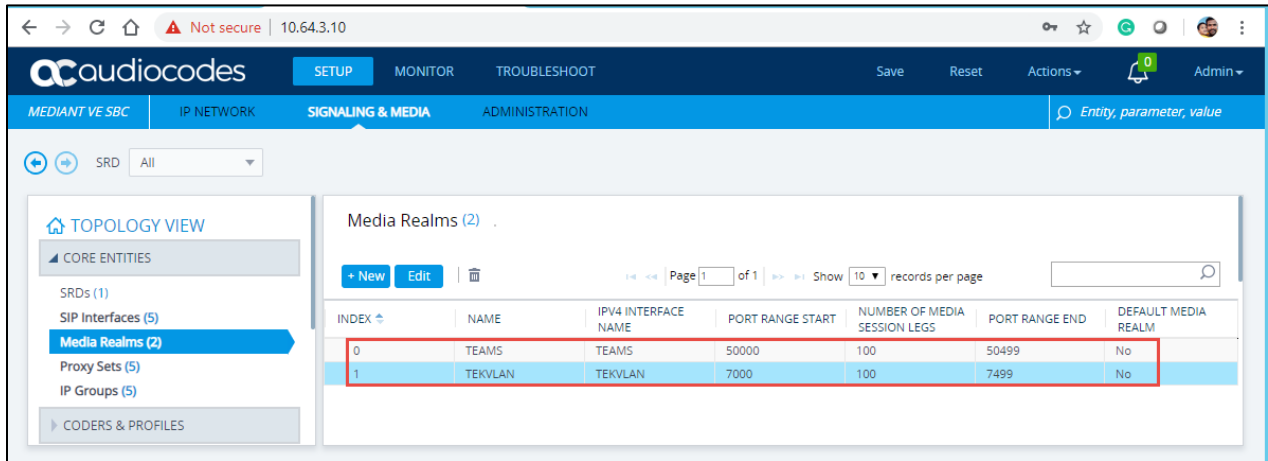
Figure 32 – Teams TLS

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

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

4.4.7 Configure Media Realms

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

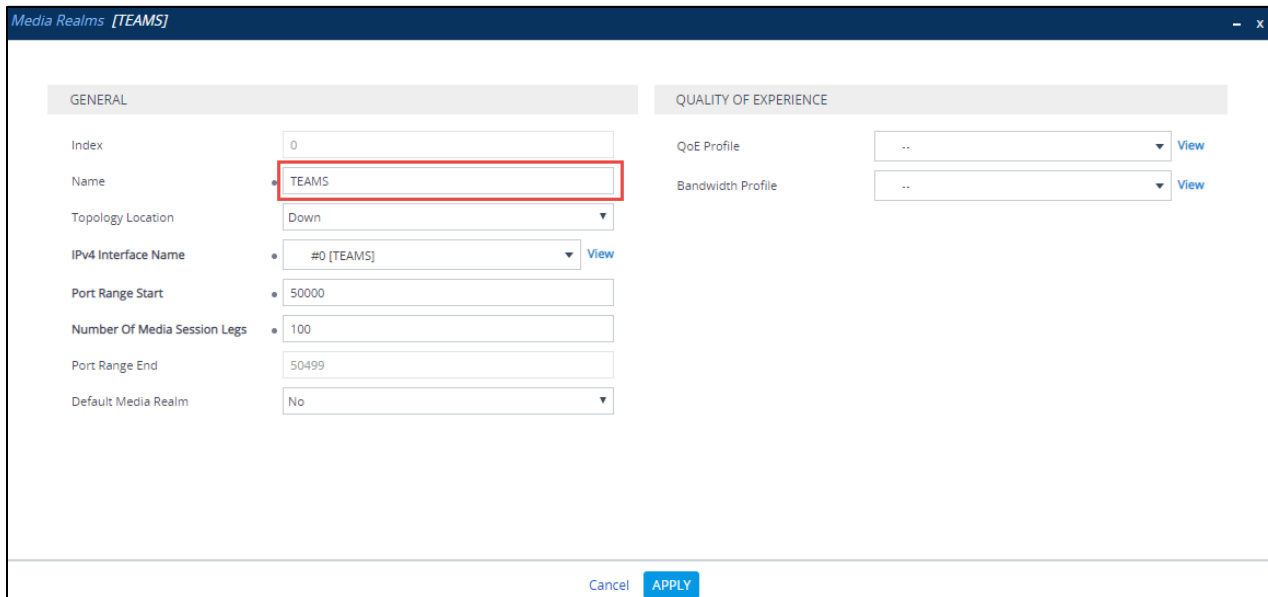


The screenshot shows the Audiocodes management console interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The main navigation tabs are 'MEDIANT VE SBC', 'IP NETWORK', 'SIGNALING & MEDIA', and 'ADMINISTRATION'. The 'SIGNALING & MEDIA' tab is active, and the 'CORE ENTITIES' 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 'Name' field is highlighted with a red box. The configuration is as follows:

GENERAL	QUALITY OF EXPERIENCE
Index: 0	QoE Profile: .. View
Name: TEAMS	Bandwidth Profile: .. View
Topology Location: Down	
IPv4 Interface Name: #0 [TEAMS] View	
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] View		
Used By Routin...	Not Used		
Dial Plan	# [-] View		
CAC Profile	# [-] View		

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	MEDIA
Index: 1	Media Realm: #1 [TEKVLAN] View
Name: PSTNGW	Direct Media: Disable
Topology Location: Up	
Network Interface: #2 [TEKVLAN] View	
Application Type: SBC	
UDP Port: 5060	
TCP Port: 0	
TLS Port: 0	
Additional UDP Ports:	
Additional UDP Ports Mode: Always Open	
Encapsulating Protocol: No encapsulation	

SECURITY
TLS Context Name: -- View
TLS Mutual Authentication: --
Message Policy: -- View
User Security Mode: Not Configured
Enable Un-Authenticated Registrations: Not configured
Max. Number of Registered Users: -1

Figure 39 – PSTN

Enable TCP Keepalive: Disable

Used By Routing Server: Not Used

Pre-Parsing Manipulation Set: -- [View](#)

CAC Profile: -- [View](#)

CLASSIFICATION

Classification Failure Response Type: 500

Pre-classification Manipulation Set ID: -1

Call Setup Rules Set ID: -1

Cancel [APPLY](#)

Figure 40 – PSTN

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

SIP Interfaces [CISCO]

SRD #0 [DefaultSRD]

GENERAL	MEDIA	SECURITY
Index: 2	Media Realm: #1 [TEKVLAN] View	TLS Context Name: #0 [default] View
Name: CISCO	Direct Media: Disable	TLS Mutual Authentication: --
Topology Location: Down		Message Policy: -- View
Network Interface: #2 [TEKVLAN] View		User Security Mode: Not Configured
Application Type: SBC		Enable Un-Authenticated Registrations: Not configured
UDP Port: 5062		Max. Number of Registered Users: -1
TCP Port: 0		
TLS Port: 0		
Additional UDP Ports: --		
Additional UDP Ports Mode: Always Open		
Encapsulating Protocol: No encapsulation		

Figure 41 – Cisco

Enable TCP Keepalive	Disable
Used By Routing Server	Not Used
Pre-Parsing Manipulation Set	-- View
CAC Profile	-- View

CLASSIFICATION

Classification Failure Response Type	500
Pre-classification Manipulation Set ID	-1
Call Setup Rules Set ID	-1

Cancel [APPLY](#)

Figure 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.

Proxy Sets [TEAMS]

SRD #0 [DefaultSRD]

GENERAL

Index: 0

Name: TEAMS

SBC IPv4 SIP Interface: #0 [TEAMS] [View](#)

TLS Context Name: #1 [Teams] [View](#)

KEEP ALIVE

Proxy Keep-Alive: Using OPTIONS

Proxy Keep-Alive Time [sec]: 60

Keep-Alive Failure Responses:

Success Detection Retries: 1

Success Detection Interval: 10

REDUNDANCY

Redundancy Mode: Homing

Proxy Hot Swap: Enable

Proxy Load Balancing Method: Random Weights

Min. Active Servers for Load Balancing: 1

ADVANCED

Classification Input: IP Address only

DNS Resolve Method: SRV

Cancel **APPLY**

Figure 43 – Teams

TOPOLOGY VIEW

- CORE ENTITIES
 - SRDs (1)
 - SIP Interfaces (5)
 - Media Realms (2)
 - Proxy Sets (5)**
 - IP Groups (5)
- CODERS & PROFILES
- SBC
 - Classification (2)
 - Routing
 - Routing Policies (1)
 - IP-to-IP Routing (9)
 - Alternative Routing Reasons (0)
 - IP Group Set (0)
 - Manipulation
 - SBC General Settings
 - Call Admission Control Profile (0)

Proxy Sets [#0] > Proxy Address (1)

[New](#) [Edit](#) [Delete](#) | Page 1 of 1 | Show 10 records per page

INDEX	PROXY ADDRESS	TRANSPORT TYPE
0	teams.local	TLS

#0 [Edit](#)

GENERAL

Proxy Address: teams.local

Transport Type: TLS

Proxy Priority: 0

Proxy Random Weight: 0

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] View	Proxy Load Balancing Method	Disable
TLS Context Name	-- View	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
Cancel APPLY	

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 (sbc4.tekvizionlabs.com), Created By Routing Server (No), and Used By Routing Server (Not Used). The 'MESSAGE MANIPULATION' section includes Inbound Message Manipulation Set (1), Outbound Message Manipulation Set (2), Message Manipulation User-Defined String 1 and 2, and Proxy Keep-Alive using IP Group settings (Enable).

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 (sbc4.tekvizionlabs.com), UII Format (Disable), and Always Use Src Address (No). The right side of the page 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 STATUS' section includes GW Group Registered IP Address and GW Group Registered Status (Not Registered).

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] View
Keep Original Call-ID	No
Dial Plan	-- View
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: .. View
Name: PSTNGW	Bandwidth Profile: .. View
Topology Location: Up	
Type: Server	
Proxy Set: #1 [PSTNGW] View	
IP Profile: #2 [PSTNGW_Profile] View	
Media Realm: #1 [TEKVLAN] View	
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

SBC GENERAL

Classify By Proxy Set: Enable

SBC Operation Mode: Not Configured

SBC Client Forking Mode: Sequential

CAC Profile: .. [View](#)

ADVANCED

Local Host Name:

UUI Format: Disable

Always Use Src Address: No

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:

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 SRD is 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 section. The configuration is divided into several sections:

- 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.

GW Group Registered Status: Not Registered

SBC ADVANCED

Source URI Input: [Dropdown]

Destination URI Input: [Dropdown]

SIP Connect: No [Dropdown]

SBC PSAP Mode: Disable [Dropdown]

Route Using Request URI Port: Disable [Dropdown]

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

Keep Original Call-ID: No [Dropdown]

Dial Plan: .. [Dropdown] [View](#)

Call Setup Rules Set ID: -1 [Text]

Tags: [Text]

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 [Text]

Name: **TEAMS_Profile** [Text]

Created by Routing Server: No [Text]

MEDIA SECURITY

SBC Media Security Mode: **SRTP** [Dropdown]

Symmetric MKI: Disable [Dropdown]

MKI Size: 1 [Text]

SBC Enforce MKI Size: Don't enforce [Dropdown]

SBC Media Security Method: SDES [Dropdown]

Reset SRTP Upon Re-key: Disable [Dropdown]

Generate SRTP Keys Mode: **Always** [Dropdown]

SBC SIGNALING

PRACK Mode: **Optional** [Dropdown]

P-Asserted-Identity Header Mode: As Is [Dropdown]

Diversion Header Mode: As Is [Dropdown]

History-Info Header Mode: As Is [Dropdown]

Session Expires Mode: Transparent [Dropdown]

Remote Update Support: Not Supported [Dropdown]

Remote re-INVITE: Not Supported [Dropdown]

Remote Delayed Offer Support: Not Supported [Dropdown]

Remote Representation Mode: According to Operation Mode [Dropdown]

Keep Incoming Via Headers: According to Operation Mode [Dropdown]

Keep Incoming Routing Headers: According to Operation Mode [Dropdown]

Keep User-Agent Header: According to Operation Mode [Dropdown]

Figure 58 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the TEAMS_Profile IP profile. The SBC EARLY MEDIA section includes settings for 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), and Generate RTP (None). The SBC REGISTRATION section includes User Registration Time (0), NAT UDP Registration Time (-1), and NAT TCP Registration Time (-1). Other settings include Handle X-Detect (No), ISUP Body Handling (Transparent), ISUP Variant (itu92), and Max Call Duration (0).

Figure 59 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the TEAMS_Profile IP profile. The SBC MEDIA section includes settings for 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), and SDP Ptime Answer (Preferred Value). The SBC HOLD section includes Remote 3xx Mode (Handle Locally), Remote Hold Format (Inactive), Reliable Held Tone Source (Yes), and Play Held Tone (No). The SBC FAX section includes Fax Coders Group (..), Fax Mode (As Is), Fax Offer Mode (All coders), Fax Answer Mode (Single coder), Remote Renegotiate on Fax Detection (Transparent), and 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	▼

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

Figure 61 – IP Profile – Teams – Contd.

IP Profiles [TEAMS_Profile]

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

Cancel **APPLY**

Figure 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.

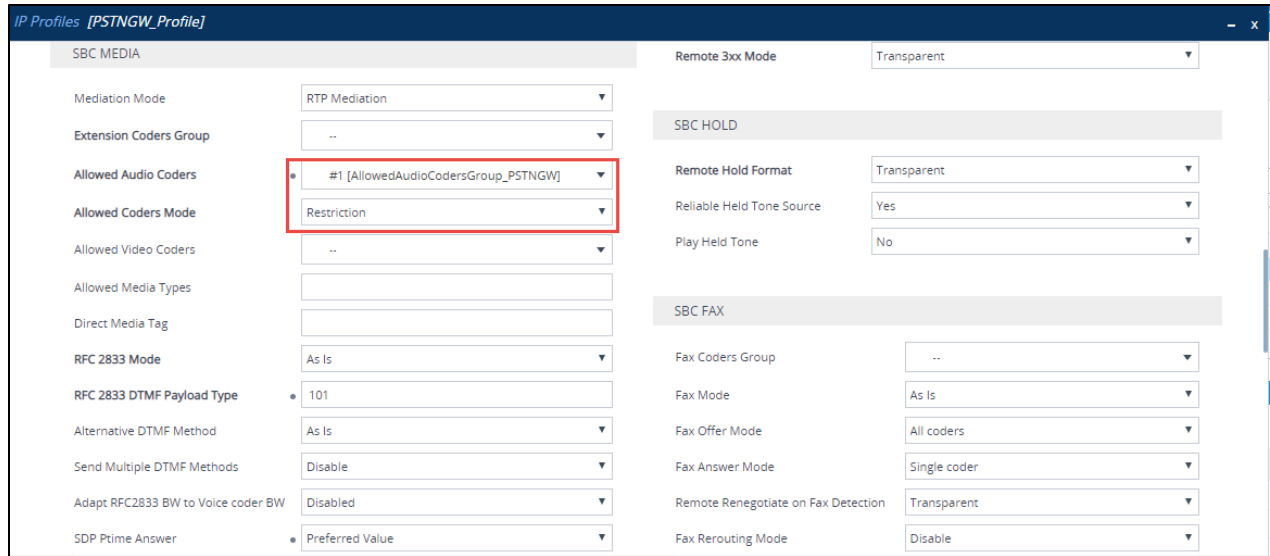


Figure 65 – IP Profile – PSTN Gateway – Contd.

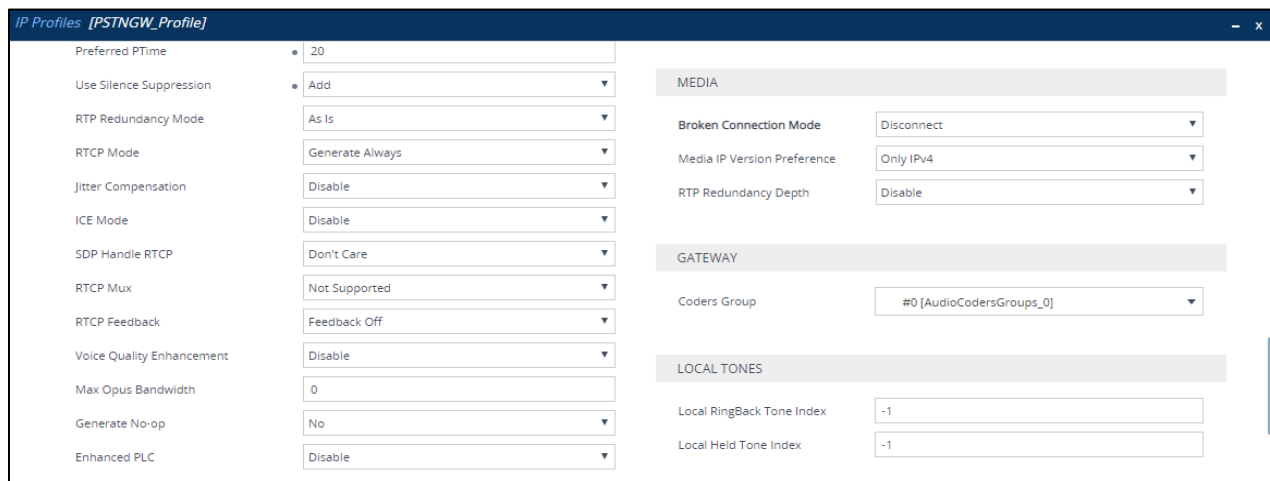


Figure 66 – IP Profile – PSTN Gateway – Contd.

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
SBC EARLY MEDIA		ISUP Variant	Itu92
Remote Early Media	Supported	Max Call Duration [min]	0
Remote Multiple 18x	Supported	SBC REGISTRATION	
Remote Early Media Response Type	Transparent	User Registration Time	0
Remote Multiple Early Dialogs	According to Operation Mode	NAT UDP Registration Time	-1
Remote Multiple Answers Mode	Disable	NAT TCP Registration Time	-1
Remote Early Media RTP Detection Mode	By Signaling	SBC FORWARD AND TRANSFER	
Remote RFC 3960 Support	Not Supported	Remote REFER Mode	Handle Locally
Remote Can Play Ringback	Yes	Remote Replaces Mode	Handle Locally
Generate RTP	None	Play RBT To Transferee	Yes

Figure 69 – IP Profile – Cisco – Contd.

IP Profiles [CISCO_Profile]

SBC MEDIA		Remote 3xx Mode	Transparent
Mediation Mode	RTP Mediation	SBC HOLD	
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	..	SBC FAX	
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:

- General Settings:**
 - 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:**
 - Broken Connection Mode: Disconnect
 - Media IP Version Preference: Only IPv4
 - RTP Redundancy Depth: Disable
- GATEWAY:**
 - Coders Group: #0 [AudioCodersGroups_0]
- LOCAL TONES:**
 - Local RingBack Tone Index: -1
 - Local Held Tone Index: -1

Figure 71 – IP Profile – Cisco – Contd.

The screenshot shows the 'IP Profiles [CISCO_Profile]' configuration window, continuing from the previous one. It includes the following sections:

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

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

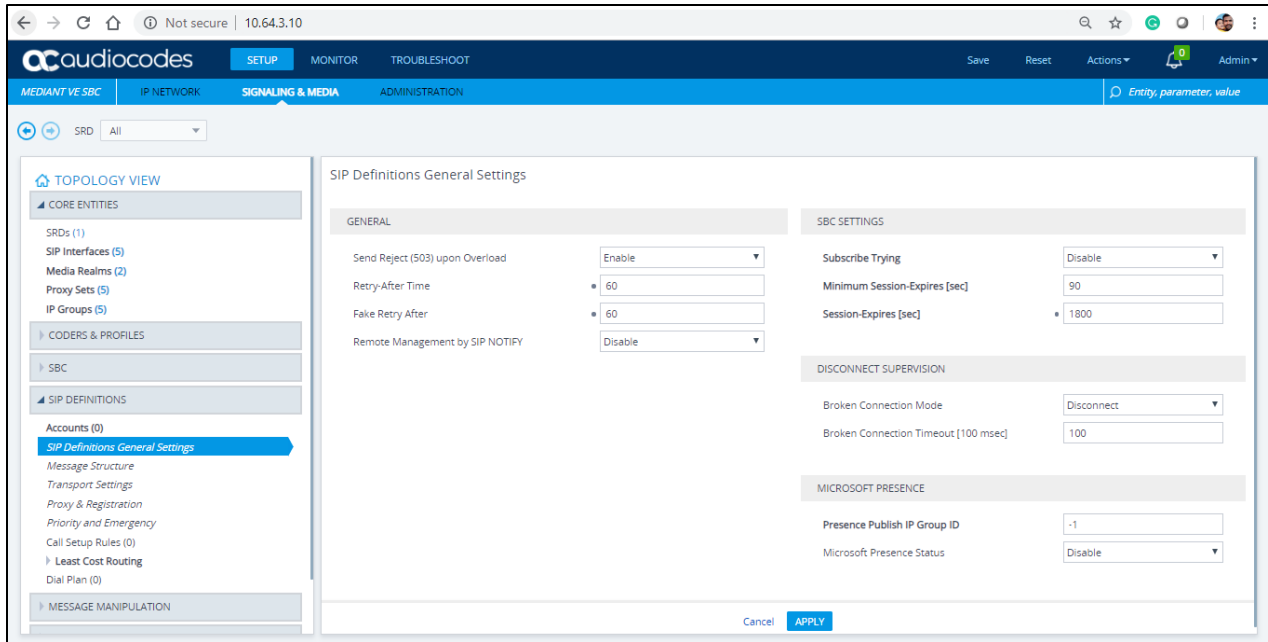


Figure 73 – SIP Definition

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.

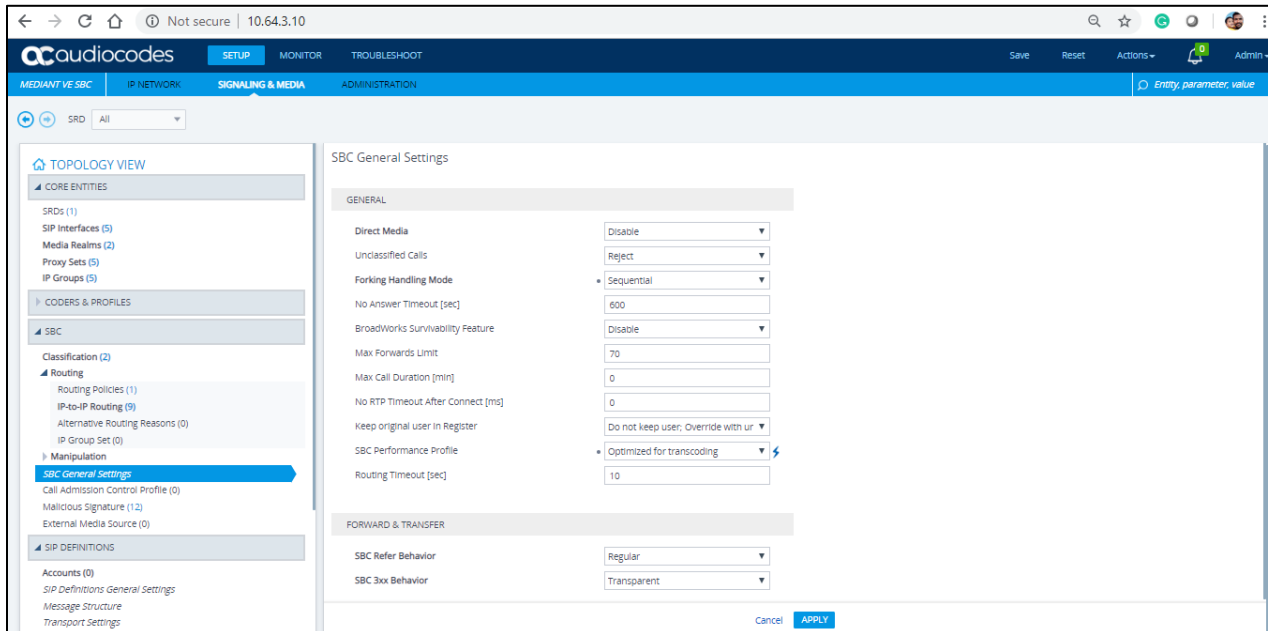


Figure 74 – SBC General Setting – Contd.

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 APPLY			

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]
MATCH	Destination Address: <input type="text"/>
Source IP Group: #1 [PSTNGW]	Destination Port: 0
Request Type: All	Destination Transport Type: <input type="text"/>
Source Username Pattern: *	IP Group Set: ..
Source Host: *	Call Setup Rules Set ID: .1
Source Tag: <input type="text"/>	Group Policy: Sequential
Destination Username Pattern: *	Cost Group: ..
	Routing Tag Name: default

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 APPLY			

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: TEAMS to CISCO	Destination IP Group: #2 [CISCO]
Alternative Route Options: Route Row	Destination SIP Interface: #2 [CISCO]
MATCH	
Source IP Group: #0 [TEAMS]	Destination Address:
Request Type: All	Destination Port: 5060
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: ..
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: ..

Figure 79 – Teams to Cisco – Contd.

Destination Username Pattern: 6	Routing Tag Name: default
Destination Host: *	Internal Action: Editor
Destination Tag:	
Message Condition: .. View	
Call Trigger: Any	
ReRoute IP Group: Any View	
Cancel APPLY	

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: Cisco -> Teams Extn dialing	Destination IP Group: #0 [TEAMS]
Alternative Route Options: Route Row	Destination SIP Interface: #0 [TEAMS]
MATCH	
Source IP Group: #2 [CISCO]	Destination Address:
Request Type: All	Destination Port: 0
Source Username Pattern: *	Destination Transport Type:
Source Host: *	IP Group Set: ..
Source Tag:	Call Setup Rules Set ID: -1
	Group Policy: Sequential
	Cost Group: ..

Figure 81 – Cisco to Teams – Contd.

Destination Username Pattern: *	Routing Tag Name: default
Destination Host: *	Internal Action: Editor
Destination Tag:	
Message Condition: .. View	
Call Trigger: Any	
ReRoute IP Group: Any View	
Cancel APPLY	

Figure 82 – Cisco to Teams

4.4.16 IP Group

IP Group – Teams

IP Groups [TEAMS]

SRD #0 [DefaultSRD]

GENERAL	QUALITY OF EXPERIENCE
Index: 0	QoE Profile: -- View
Name: TEAMS	Bandwidth Profile: -- View
Topology Location: Down	
Type: Server	
Proxy Set: #0 [TEAMS] View	
IP Profile: #1 [TEAMS_Profile] View	
Media Realm: #0 [TEAMS] View	
Contact User:	
SIP Group Name: sdc4.tekvizionlabs.com	
Created By Routing Server: No	

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

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.tekvizionlabs.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="--"/> View
Name	<input type="text" value="CISCO"/>	Bandwidth Profile	<input type="text" value="--"/> View
Topology Location	<input type="text" value="Down"/>	MESSAGE MANIPULATION	
Type	<input type="text" value="Server"/>	Inbound Message Manipulation Set	<input type="text" value="4"/>
Proxy Set	<input type="text" value="#2 [CISCO]"/> View	Outbound Message Manipulation Set	<input type="text" value="5"/>
IP Profile	<input type="text" value="#3 [CISCO_Profile]"/> View	Message Manipulation User-Defined String 1	<input type="text"/>
Media Realm	<input type="text" value="#1 [TEKVLAN]"/> View	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"/>	SBC REGISTRATION AND AUTHENTICATION	
Proxy Set Connectivity	<input type="text" value="NA"/>	Max. Number of Registered Users	<input type="text" value="-1"/>
SBC GENERAL		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="--"/> View	Authentication Method List	<input type="text"/>
ADVANCED		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="--"/> View
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="....."/>
		GW GROUP STATUS	
		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]"/> View
Keep Original Call-ID	<input type="text" value="No"/>
Dial Plan	<input type="text" value="--"/> View
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 is on the right, showing 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 Action Subject, Type, and Value fields.

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 is on the right, showing 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 Action Subject, Type, and Value fields.

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

Figure 104 – SIP Message Manipulation – From

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

Figure 105 – SIP Message Manipulation – Referred By

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

Message Manipulations [From header Teams -> Cisco]

GENERAL

Index: 13

Name: From header Teams -> Cisco

Manipulation Set ID: 5

Row Role: Use Current Condition

MATCH

Message Type: Options

Condition: Param.Message.Address.dst.SIPInterface=='2'

ACTION

Action Subject: Header.From.URL.host

Action Type: Modify

Action Value: 10.64.3.10

Buttons: Cancel, APPLY

Figure 106 – SIP Message Manipulation – From

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

Message Manipulations [To header Teams -> Cisco]

GENERAL

Index: 14

Name: To header Teams -> Cisco

Manipulation Set ID: 5

Row Role: Use Current Condition

MATCH

Message Type: Options

Condition: Param.Message.Address.dst.SIPInterface=='2'

ACTION

Action Subject: header.to.uri.host

Action Type: Modify

Action Value: 10.64.3.10

Buttons: Cancel, APPLY

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
```

¹ Hide signaling and media peer addresses from endpoints other than gateway.

² If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the Cisco

³ This command enables Cisco UBE basic IP-to-IP voice communication feature.

⁴ This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).

```
midcall-signaling 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
```

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


```
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  
!
```

⁸ 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
```

⁹ Outbound Dial-peer towards AudioCodes

¹⁰ 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
```

¹¹ Outbound Dial peer towards Cisco UCM

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

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

Crestron_Teams#

4.6 Cisco UCM Configuration

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

4.6.1 Version

Cisco UCM version



Figure 108 – Cisco UCM Version

4.6.2 Cisco UCM Audio Codec Preference List

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration **Go**
administrator | Search Documentation | About | Logout

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

Audio Codec Preference List Configuration Related Links: [Back To Find/List](#) **Go**

Save Delete Copy Add New

Audio Codec Preference List Information

Name*

Description*

Codecs in List*

- G.711 U-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 64k
- G.711 A-Law 56k
- G.729a 8k
- G.729b 8k
- G.729ab 8k
- G.729 8k
- OPUS (6k-510k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- ISAC 32k
- AMR-WB (7k-24k)
- MP4A-LATM 32k
- G.722 64k
- G.722.1 32k
- G.722 56k
- G.722.1 24k
- G.722 48k
- MP4A-LATM 24k
- ILBC 16k
- G.728 16k
- AMR (5k-13k)
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- GSM Half Rate 6k

Save Delete Copy Add New

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: g711_region

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	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
g711_region	Factory Default lowloss crestron	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
Default g711_region g729_region	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> [] kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> [] kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> [] kbps

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 device pool name is "Crestron_DevicePool (13 members**)". The configuration is organized into several sections:

- Device Pool Information:** Shows the device pool name "Crestron_DevicePool".
- Device Pool Settings:** Includes fields for "Device Pool Name" (Crestron_DevicePool), "Cisco Unified Communications Manager Group" (Default), "Calling Search Space for Auto-registration" (< None >), "Adjunct CSS" (< None >), "Reverted Call Focus Priority" (Default), and "Intercompany Media Services Enrolled Group" (< None >).
- Roaming Sensitive Settings:** Includes "Date/Time Group" (CMLocal), "Region" (g711_region), "Media Resource Group List" (MRGL), "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 >), and "Wireless LAN Profile Group" (< None >).
- Local Route Group Settings:** Shows "Standard Local Route Group" as (< None >).
- Device Mobility Related Information:** Includes "Device Mobility Calling Search Space" (< None >), "AAR Calling Search Space" (< None >), "AAR Group" (< None >), "Calling Party Transformation CSS" (< None >), and "Called Party Transformation CSS" (< None >).
- Geolocation Configuration:** (Section header visible at the bottom).

Red boxes highlight the "Device Pool Name" field and the "Region" and "Media Resource Group List" dropdowns.

Figure 111 – Cisco UCM Device Pool – Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go
administrator 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_tekcucm5-cucmpub. This is used for this example

Set Device Pool* = Crestron_DevicePool

The screenshot displays the Cisco Unified CM Administration interface for configuring an Annunciator. The page title is "Annunciator Configuration". The status is "Ready". The "Annunciator Information" section contains the following fields:

Registration:	Registered with Cisco Unified Communications Manager 172.16.29.81
IPv4 Address:	172.16.29.81
<input checked="" type="checkbox"/> Device is trusted	
Server*	172.16.29.81
Name*	ANN_2
Description	ANN_tekcucm5-cucmpub
Device Pool*	Crestron_DevicePool
Location*	Hub_None
Use Trusted Relay Point*	Off

Buttons: Save, Reset, Apply Config

*- indicates required item.

Figure 113 – Cisco UCM Annunciator

4.6.6 Cisco UCM Conference Bridge

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

Set Name* = CFB_2

Set Description = ANN_tekcucm5-cucmpub. This is used for this example

Set Device Pool* = Crestron_DevicePool

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Conference Bridge. The page title is "Conference Bridge Configuration" and it includes navigation tabs for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

Status: Ready

Conference Bridge Information:
Conference Bridge : CFB_2 (CFB_tekcucm5-cucmpub)
Registration: Registered with Cisco Unified Communications Manager 172.16.29.81
IPv4 Address: 172.16.29.81

Software Conference Bridge Info:
Conference Bridge Type* Cisco Conference Bridge Software
Host Server 172.16.29.81
⚠ Device is not trusted
Conference Bridge Name* CFB_2
Description CFB_tekcucm5-cucmpub
Device Pool* Crestron_DevicePool
Common Device Configuration < None >
Location* Hub_None
Use Trusted Relay Point* Default

Buttons: Save, Reset, Apply Config

ⓘ *- indicates required item.

Figure 114 – Cisco UCM Conference Bridge

4.6.7 Cisco UCM MTP

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

Set Name* = MTP_2

Set Description = MTP_tekcucm5-cucmpub. This is used for this example

Set Device Pool* = Crestron_DevicePool

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu shows "Media Resources" selected, leading to "Media Termination Point Configuration". The page title is "Media Termination Point Configuration" with a "Related Links: Back To Find/List" dropdown. Below the title are "Save", "Reset", and "Apply Config" buttons. A status message indicates "Update successful". The "Media Termination Point Information" section shows the following details: Registration: Registered with Cisco Unified Communications Manager 172.16.29.81; IPv4 Address: 172.16.29.81; Media Termination Point Type*: Cisco Media Termination Point Software; Host Server*: 172.16.29.81; Media Termination Point Name*: MTP_2; Description: MTP_tekcucm5-cucmpub; Device Pool*: Crestron_DevicePool. A red box highlights the Name, Description, and Device Pool fields. At the bottom, there are "Save", "Reset", and "Apply Config" buttons, and a note: "i *- indicates required item."

Figure 115 – Cisco UCM MTP

4.6.8 Cisco Media Resource Group

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

Set Name* = MRG

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

The screenshot shows the Cisco Unified CM Administration interface for configuring a Media Resource Group (MRG). The page title is "Media Resource Group Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Media Resource Group Configuration" page has a "Related Links" section with "Back To Find/List" and a "Go" button. Below the navigation is a toolbar with "Save", "Delete", "Copy", and "Add New" buttons. The main content area is divided into several sections: "Status" (Ready), "Media Resource Group Status" (MRG, used by 15 devices), "Media Resource Group Information" (Name: MRG, Description: empty), and "Devices for this Group". The "Devices for this Group" section has two lists: "Available Media Resources" (ANN_3, CFB_3, IVR_3, MOH_3, MTP_2) and "Selected Media Resources" (ANN_2 (ANN), CFB_2 (CFB), IVR_2 (IVR), MOH_2 (MOH)). The "Selected Media Resources" list is highlighted with a red box. There is also a checkbox for "Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)". At the bottom of the page are "Save", "Delete", "Copy", and "Add New" buttons.

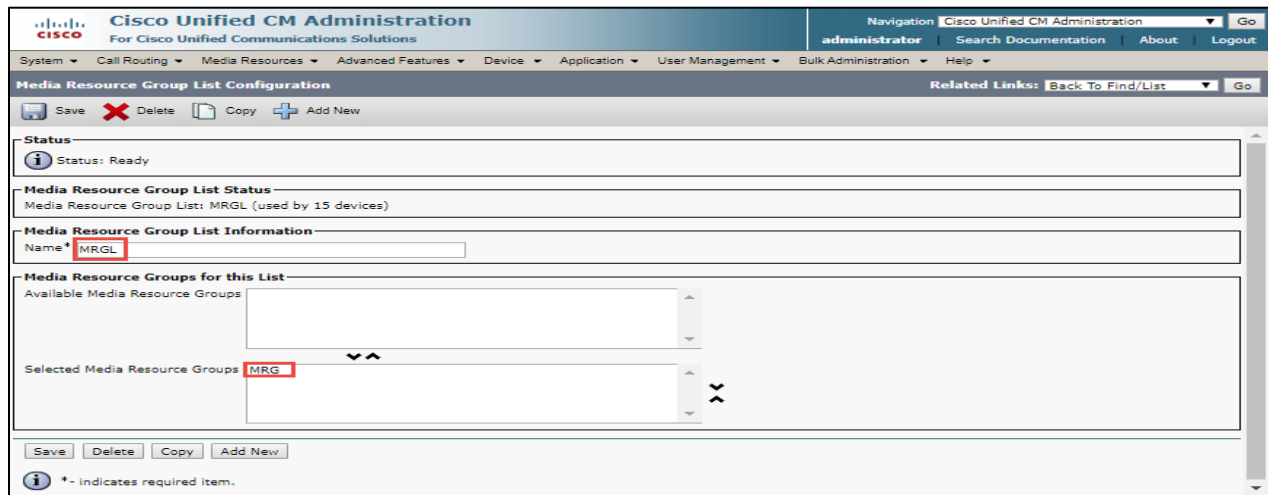
Figure 116 – Cisco UCM MTP

4.6.9 Cisco Media Resource Group List

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

Set Name* = MRGL

Selected Media Resources Groups = MRG



The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group List (MRGL). 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 user is logged in as "administrator".

At the top, there are action buttons: Save, Delete, Copy, and Add New. A "Related Links" section contains "Back To Find/List" and "Go".

The configuration area is divided into several sections:

- Status:** Shows "Status: Ready".
- Media Resource Group List Status:** Displays "Media Resource Group List: MRGL (used by 15 devices)".
- Media Resource Group List Information:** Contains a "Name*" field with the value "MRGL".
- Media Resource Groups for this List:** Features two list boxes. The "Available Media Resource Groups" list is empty. The "Selected Media Resource Groups" list contains "MRG".

At the bottom, there are "Save", "Delete", "Copy", and "Add New" buttons, along with an information icon and the text "* - Indicates required item."

Figure 117 – Cisco UCM MRGL

4.6.10 Cisco UCM SIP Trunk towards Cisco UBE

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

Set **Device Name*** = Cube_Crestron_Teams. This is used for this example

Set **Description** = Cube_Crestron_Teams. This is used for this example

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

Set **Media Resource Group List** = MRGL

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 user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "Trunk Configuration" with a "Related Links" section containing "Back To Find/List".

The configuration form is titled "SIP Trunk Status" and shows the following details:

- Service Status:** Full Service
- Duration:** Time In Full Service: 0 day 0 hour 43 minutes

The "Device Information" section contains the following fields and values:

- Product: SIP Trunk
- Device Protocol: SIP
- Trunk Service Type: None(Default)
- Device Name*: Cube_Crestron_Teams
- Description: Cube_Crestron_Teams
- Device Pool*: Crestron_DevicePool
- 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

Additional configuration options include:

- 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 | Go
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 | Go
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*** = Crestron Profile. This is used in this example.
 Set **DTMF Signaling Method*** = No Preference. This is used in this example.

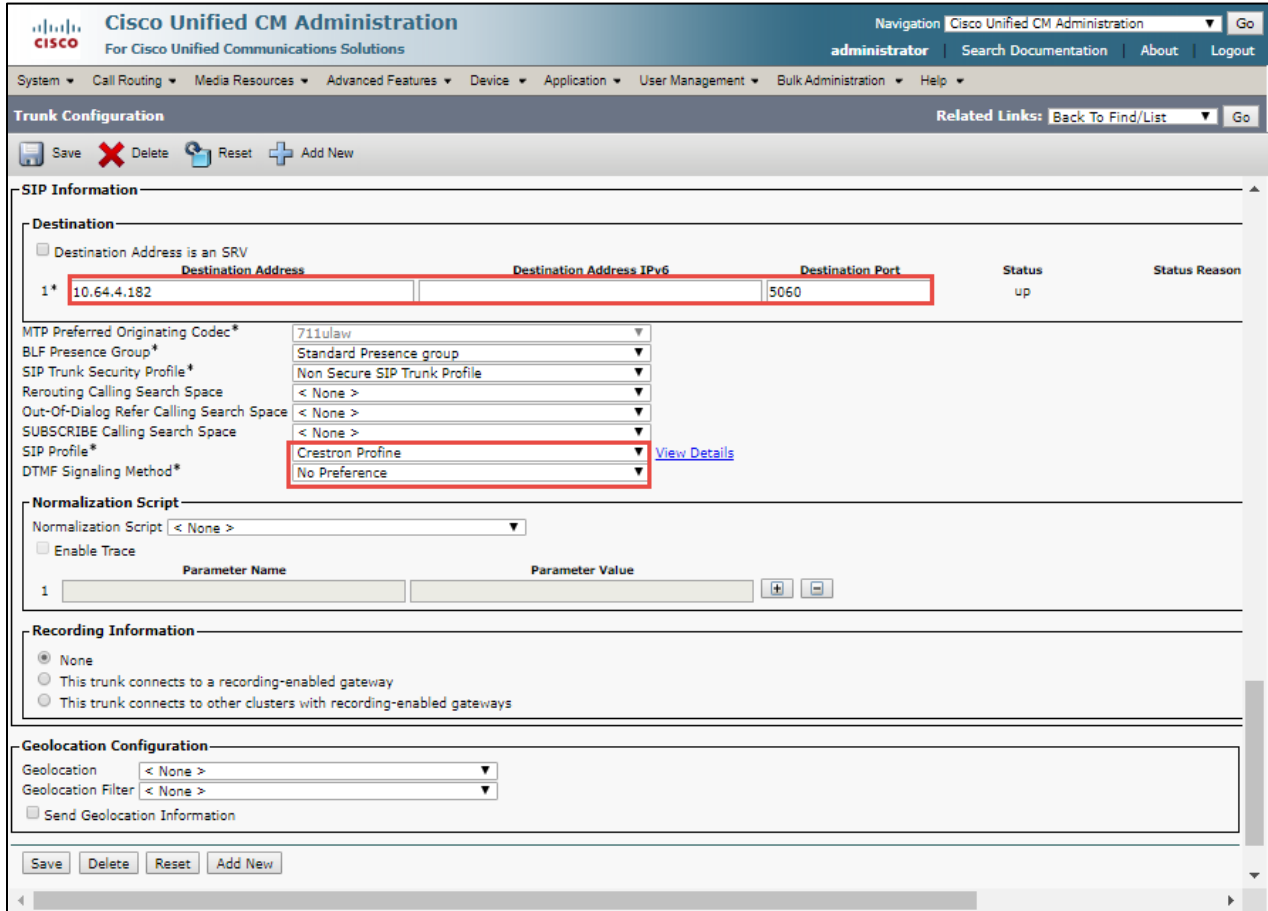


Figure 121 – SIP Trunk – Cisco UBE

4.6.11 Cisco UCM SIP Trunk towards Cisco Unity

Set **Device Name*** = Unity. This is used for this example

Set **Description** = Unity Connection. This is used for this example

Set **Media Resource Group List** = MRGL

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Cisco Unified CM Administration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main heading is "Trunk Configuration" with a "Related Links: Back To Find/List" button. Below the heading are buttons for Save, Delete, Reset, and Add New. The configuration is organized into several sections:

- Status:** Shows "Status: Ready".
- SIP Trunk Status:** Shows "Service Status: Full Service" and "Duration: Time In Full Service: 7 days 1 hour 43 minutes".
- Device Information:** A table of configuration parameters for the SIP Trunk.

Parameter	Value
Product	SIP Trunk
Device Protocol	SIP
Trunk Service Type	None(Default)
Device Name*	Unity
Description	Unity_Connection
Device Pool*	Default
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
Media Termination Point Required	<input type="checkbox"/>
Retry Video Call as Audio	<input checked="" 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"/>
S RTP 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.	<input type="checkbox"/>
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default

Figure 122 – SIP Trunk – Cisco Unity – Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
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.

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

SIP Information

Figure 124 – SIP Trunk – Cisco Unity – Contd.

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

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

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason
1*	10.80.18.5		5060	up	

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* Non Secure SIP Trunk Profile Unity

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile - OPTIONS [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 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, there are several sections with checkboxes: 'Message Waiting Indicators' (Send Message Counts, Use Same Port for Enabling and Disabling MWIs, Force All MWIs Off for this Phone System), 'Call Loop Detection by Using DTMF' (Enable for Supervised Transfers, Enable for Forwarded Message Notification Calls (by Using DTMF)), and 'Call Loop Detection by Using Extension' (Enable for Forwarded Message Notification Calls (by Using Extension)). The 'DTMF Tone To Use' is set to 'A' and 'Guard Time' is 2500 milliseconds. The 'Phone View Settings' section has 'Enable Phone View' unchecked. 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. The left sidebar is expanded to 'Tools' > 'Migration Utilities'. The main content area has 'Enable outgoing calls' selected with a radio button. Other options are 'Disable all outgoing calls immediately' and 'Disable all outgoing calls between'. The 'Beginning Time' is set to 12:00 AM and the 'Ending Time' is set to 12:00 AM. At the bottom, there are 'Save', 'Delete', 'Previous', and 'Next' buttons. A note at the bottom states: '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 (CiscoUCM11.5_1)'. It includes a navigation bar with 'Save', 'Delete', 'Previous', and 'Next' buttons. The 'Port Group' section contains a 'Display Name*' field with the value 'CiscoUCM11.5_1', an 'Integration Method' dropdown set to 'SIP', and a 'Reset Status' section with 'Reset Not Required' and a 'Reset' button. The 'Session Initiation Protocol (SIP) Settings' section has checkboxes for 'Register with SIP Server' and 'Authenticate with SIP Server', both of which are unchecked. Below these are input fields for 'Authentication Username', 'Authentication Password', and 'Contact Line Name'. There are also dropdown menus for 'SIP Security Profile' (set to '5060') and 'SIP Transport Protocol' (set to 'TCP'). The 'Advertised Codec Settings' section features a 'Change Advertising' button and a table with two 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'. Both rows are highlighted with a red border.

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

Figure 128 –Phone Group – Contd.

The screenshot shows the 'Message Waiting Indicator Settings' section of the Cisco Unity Connection Administration interface. The left sidebar is expanded to 'Telephony Integrations' > 'Port Group'. The main content area is titled 'Message Waiting Indicator Settings'. It includes a 'Change Advertising' button and a checkbox for 'Enable Message Waiting Indicators', which is checked. Below this are input 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, there are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (*) are required.'

Figure 129 –Phone Group

4.7.3 Port

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

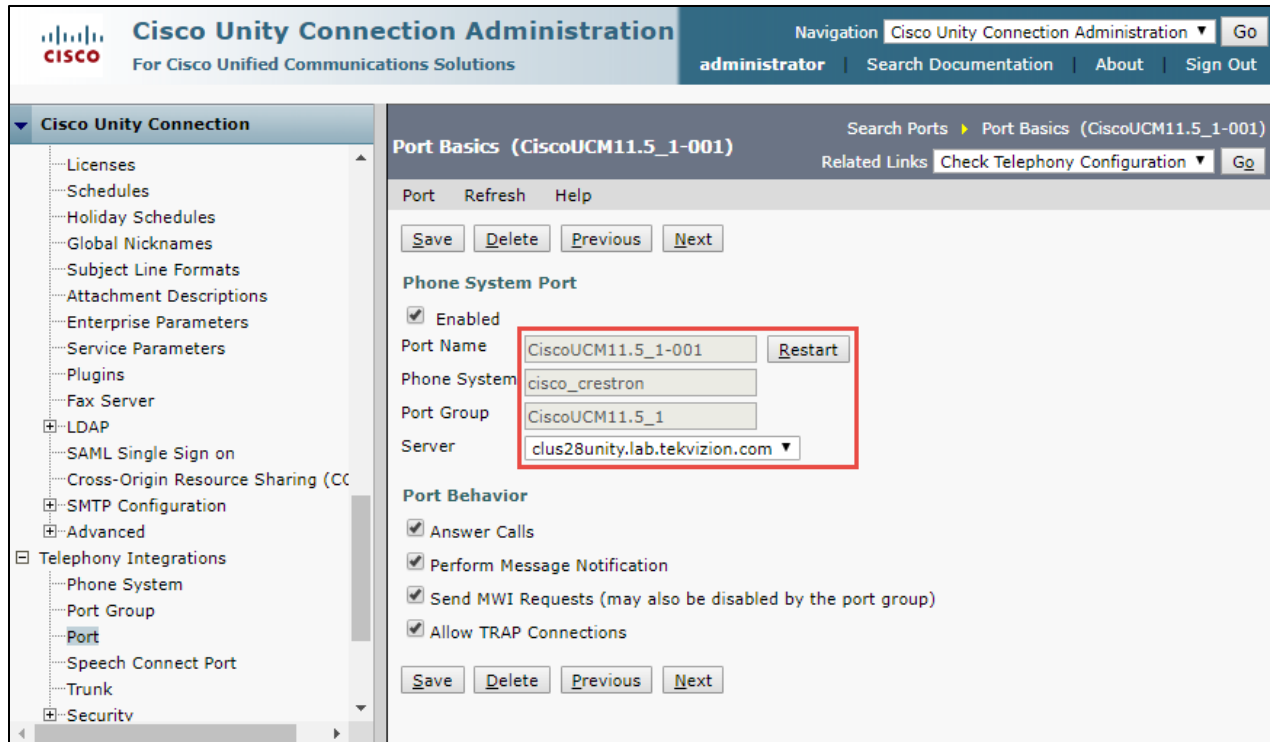


Figure 130 – Port

4.7.4 User

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

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

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

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

All other values are default.

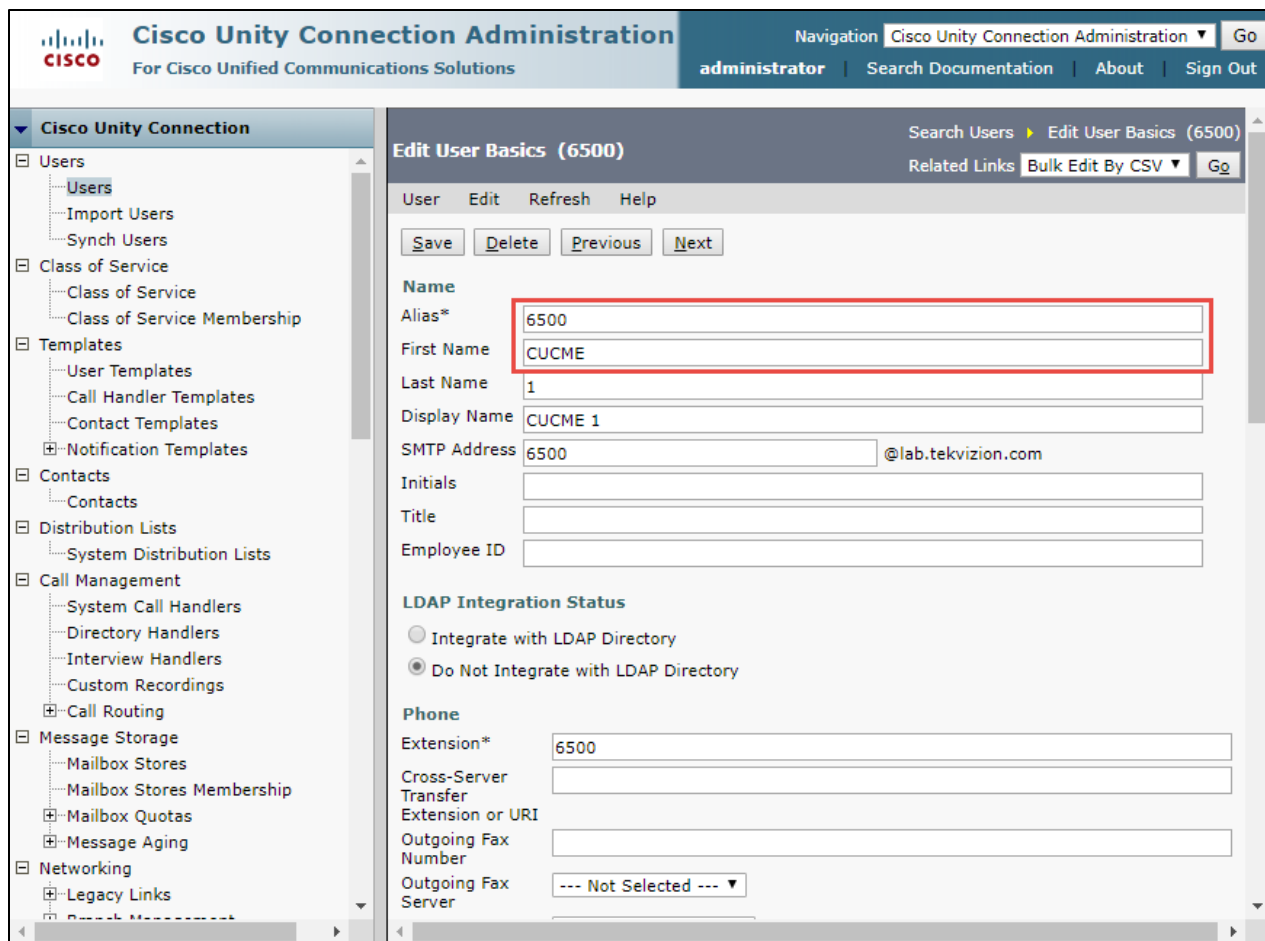


Figure 131 – User

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

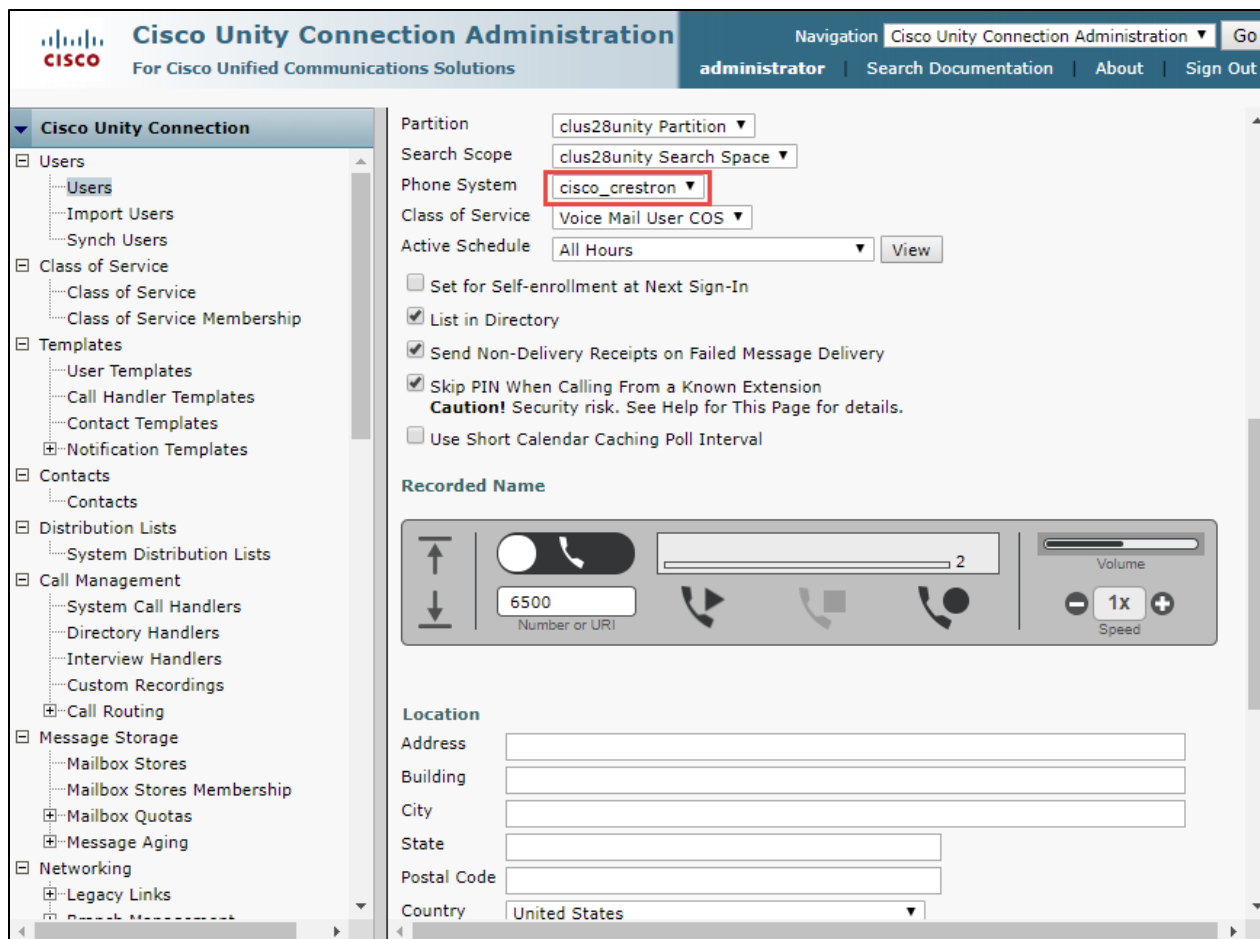


Figure 132 – User

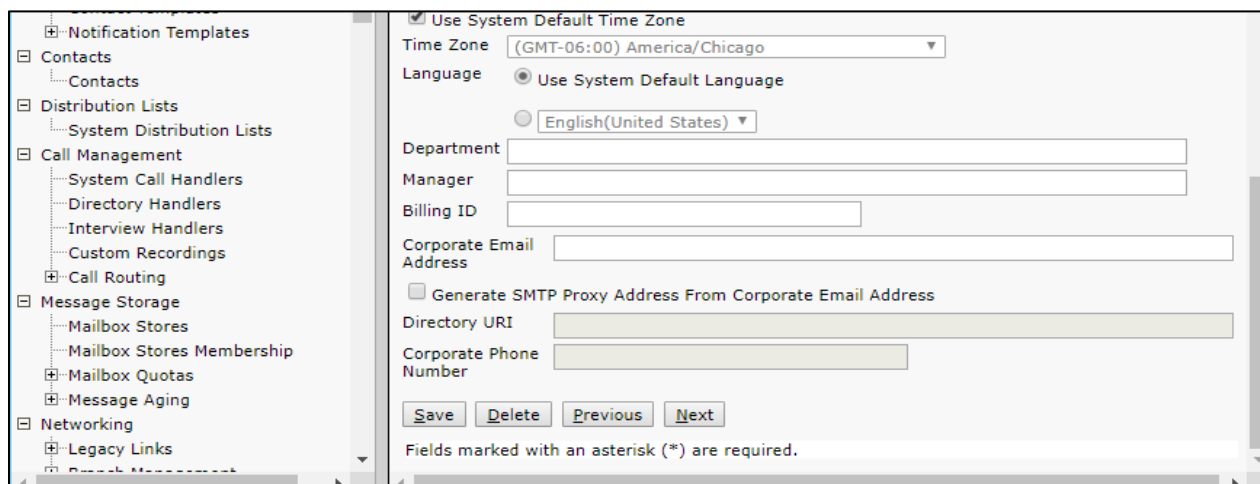


Figure 133 – User

5 Acronyms

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

6 Summary of Tests and Results

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
57	Teams user user Calls PBX A user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to PBX A user 2. Teams user hears Ring back Tone 3. PBX A user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
59	Teams user user Calls PBX B user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to PBX B user 1 2. Teams user hears Ring back Tone 3. PBX B user 1 answers the call 4. Verify two way audio 5. Teams user adds PBX B user 2 to the ongoing call 6. PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams
60	Teams user user Calls PBX B user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
61	Teams user user Calls PBX B user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to PBX B user 2. Teams user hears Ring back Tone 3. PBX B user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams
62	Teams user user Calls PSTN user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to PSTN user 2. Teams user hears Ring back Tone 3. PSTN user answers the call 4. Verify two way audio 5. Teams user adds PBX B user to the ongoing call 6. PBX B user starts ringing 7. PBX B user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams

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

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

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

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

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
68	PBX B user Calls Teams user, Teams user adds PBX B user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from PBX B user 1 to Teams user 2. PBX B user 1 hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX B user 2 to the ongoing call 6. PBX B user 2 starts ringing 7. PBX B user 2 answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams

External ID	Title	Procedure	Expected Results	Status	Comments
69	PBX B user Calls Teams user, Teams user adds PBX A user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PBX A user to the ongoing call 6. PBX A user starts ringing 7. PBX A user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams
70	PBX B user Calls Teams user, Teams user adds PSTN user to the ongoing call	<ol style="list-style-type: none"> 1. Make a voice call from PBX B user to Teams user 2. PBX B user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Teams user adds PSTN user to the ongoing call 6. PSTN user starts ringing 7. PSTN user answers the call 9. Verify all three users are able to hear each other 10. Teams user hangs up the call 11. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Third user is added to the call successfully 2. All three users are able to hear each other 	NOT APPLICABLE	This testing is for only one PBX with Teams

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

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

External ID	Title	Procedure	Expected Results	Status	Comments
		Skype for Business user is internal and external			
111	Skype for Business user calls Teams user	<ol style="list-style-type: none"> 1. Make a voice call from Skype for Business user to Teams user 2. Skype for Business user hears Ring back Tone 3. Teams user answers the call 4. Verify two way audio 5. Skype for Business user hangs up the call 6. Verify call is cleared successfully 7. Verify the same scenario where Skype for Business user is internal and external 		NOT APPLICABLE	Not applicable for this topology
112	Teams user calls Skype for Business External Mobile user	<ol style="list-style-type: none"> 1. Skype for business user is an External Mobile user 2. Make a voice call from Teams user to Skype for Business user 3. Teams user hears Ring back Tone 4. Skype for Business user answers the call 5. Verify two way audio 6. Teams user hangs up the call 7. Verify call is cleared successfully 		NOT APPLICABLE	Not applicable for this topology

External ID	Title	Procedure	Expected Results	Status	Comments
113	Skype for Business External Mobile user calls Teams user	<ol style="list-style-type: none"> 1. Skype for business user is an External Mobile user 2. Make a voice call from Skype for Business user to Teams user 3. Skype for Business user hears Ring back Tone 4. Teams user answers the call 5. Verify two way audio 6. Skype for Business user hangs up the call 7. Verify call is cleared successfully 		NOT APPLICABLE	Not applicable for this topology
114	Teams user call other tenant users	<ol style="list-style-type: none"> 1. Make a voice call from Teams user to another tenant users (Teams desktop client user, Teams mobile user, Skype for Business Online user) 2. Verify call is successful 3. Make one call to each different user one by one 		NOT APPLICABLE	Not applicable for this topology
115	Teams users joins a meeting scheduled by Skype for business On-premises user	<ol style="list-style-type: none"> 1. Skype for business user schedules a meeting and invites Teams user 1 and Teams user 2 2. Teams user 1 joins the meeting using the Join button 3. Teams user 2 joins the meeting using the dial-in conferencing number 4. Verify Teams users are able to join the meeting successfully 5. Verify all three users are able to hear 		NOT APPLICABLE	Not applicable for this topology

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