



Crestron UC-PHONE and UC-PHONE-PLUS

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

September 2019

Document History

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Table of Contents

1	Audience	6
1.1	tekVizion Labs.....	6
2	SIP Trunking Network Components.....	7
2.1	Hardware Components.....	8
2.2	Software Requirements.....	8
3	Features	8
3.1	Features Supported.....	8
3.2	Caveats and Limitations	9
4	Configuration.....	9
4.1	Configuration Checklist	9
4.2	IP Address Worksheet	9
4.3	Microsoft Skype for Business Hybrid Configuration.....	10
4.3.1	Create tenant account for Office 365.....	10
4.3.2	Add on-prem domain to O365	11
4.3.3	Setup AD synchronization	11
4.3.4	Configure skype for Business hybrid	16
4.4	Microsoft Teams Configuration.....	25
4.4.1	Teams User Configuration	25
4.4.2	Configure Calling policy to Users	34
4.4.3	Configure user parameters.....	35
4.4.4	Create Online PSTN Gateway	35
4.4.5	Configure Online PSTN Usage.....	36
4.4.6	Configure Online Voice Route	36
4.4.7	Configure Online Voice Route Policy.....	37
4.4.8	Configure Online Voice Route Policy to user	37

4.4.9	Configure Tenant Dial Plan.....	37
4.4.10	Create Normalization Rule	38
4.4.11	Associate Normalization rule to tenant dial plan	38
4.4.12	Associate tenant Dial plan to user	38
4.4.13	Calling Line Identity Policy.....	38
4.5	AudioCodes VE SBC Configuration.....	40
4.5.1	General	40
4.5.2	Configure VLANs.....	40
4.5.3	Configure IP Network Interfaces.....	41
4.5.4	Configure DNS SRV Records.....	42
4.5.5	Configure SRTP	43
4.5.6	Configure TLS contexts.....	43
4.5.7	Configure Media Realms	45
4.5.8	Configure the SRD	46
4.5.9	Configure SIP Signaling Interface	47
4.5.10	Configure Proxy Sets	49
4.5.11	Configure IP Groups	52
4.5.12	Configure IP Profile.....	56
4.5.13	Configure SIP Definition and General Setting.....	64
4.5.14	Configure SBC General Settings.....	65
4.5.15	Configure IP-to-IP Routing Rules.....	66
4.5.16	IP Group.....	69
4.5.17	Message Manipulation	73
4.6	Cisco UBE Configuration.....	83
4.7	Cisco UCM Configuration	92
4.7.1	Version.....	92
4.7.2	Cisco UCM Audio Codec Preference List.....	92
4.7.3	Cisco UCM Region Configuration	94
4.7.4	Cisco UCM Device Pool.....	94
4.7.5	Cisco UCM Annunciator Configuration	96

4.7.6	Cisco UCM Conference Bridge	97
4.7.7	Cisco UCM MTP	98
4.7.8	Cisco Media Resource Group	98
4.7.9	Cisco Media Resource Group List.....	99
4.7.10	Cisco UCM SIP Trunk towards Cisco UBE.....	100
4.7.11	Cisco UCM SIP Trunk towards Cisco Unity	102
4.8	Cisco Unity Connection (CUC)	107
4.8.1	Telephony Integration – Phone System	107
4.8.2	Phone Group	108
4.8.3	Port	109
4.8.4	User.....	109
5	Acronyms	112
6	Summary of Tests and Results	113

1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring **Crestron UC-PHONE and UC-PHONE-PLUS with Microsoft Teams Direct Routing using AudioCodes Mediant VE SBC, Cisco UCM 11.5 as Customer PBX and Skype for Business 2015 (Hybrid)**.

1.1 tekVizion Labs

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

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

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

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

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

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representation of Crestron UC-PHONE and UC-PHONE-PLUS connected O365 Cloud with Microsoft Teams Direct Routing to Cisco UCM 11.5 environment and Skype for Business 2015(Hybrid) using AudioCodes Mediant VE SBC and PSTN Gateway for PSTN connectivity. Media bypass enables Configured teams side used in this topology.

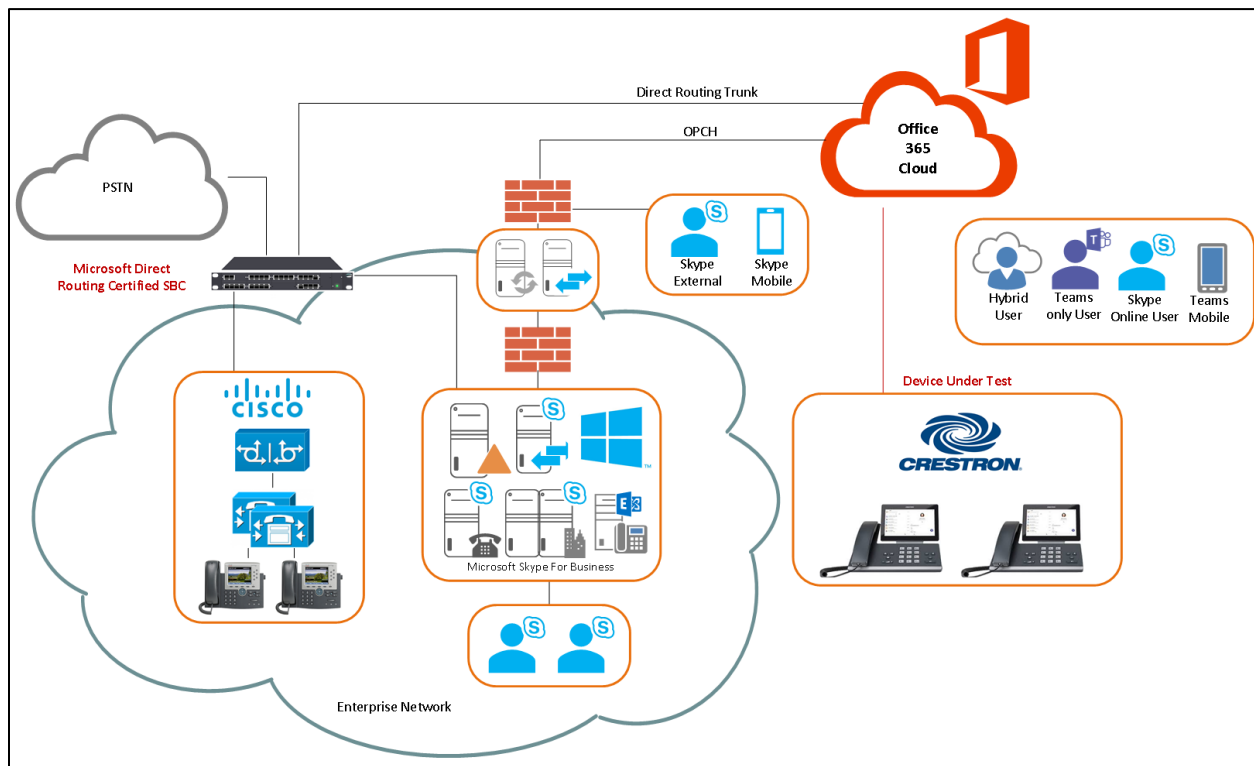


Figure 1 Network Topology

Numbering Plan

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

Dialing Plan

- Teams users and Cisco users call PSTN either doing 10 digits 11 digits dialing or E164 dialing
- Teams users call Cisco users by dialing 65XX
- Cisco users call Teams users by dialing 8XXX and AudioCodes will include the prefix +1972XXX and will send to Teams.
- Teams user calls Pure on-prem user via SIP URI dialing
- On-Prem user calls Teams user via SIP URI dialing

2.1 Hardware Components

- Microsoft Office 365 tenant with E5 without Audio Conferencing assigned to Teams users
- AudioCodes Mediant VE SBC for Teams Direct Routing serves as the demarcation point between customer's network and O365 WAN network
- Crestron UC-PHONE-PLUS and Crestron UC-PHONE phones
- Skype For Business 2015 on System running Windows 2012 R2
- Cisco UCM running on ESXi
- Cisco Unity Connection running on ESXi
- Cisco UBE v CISCO2921/K9
- PSTN Gateway

2.2 Software Requirements

- AudioCodes Mediant VE SBC v7.20A.250.003
- Skype For Business 2015 Version (6.0.9319)
- Cisco UCM v11.5.1.16900-16
- Cisco Unity Connection v11.5.1.12900-21
- Cisco UBE v 11.5.2
- Crestron UC-PHONE-PLUS v58.15.91.15

3 Features

3.1 Features Supported

- Basic Inbound and Basic Outbound
- Call hold and resume
- Call transfer (semi-attended and consultative)
- Conference
- Call forward (all, no answer)
- Busy On Busy
- Simultaneous ring

- Calling line identification restriction
- DTMF relay both directions (RFC2833)

3.2 Caveats and Limitations

- Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license.
- The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.

4 Configuration

4.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure **Microsoft Teams, Cisco UBE, Cisco UCM, Skype for Business 2015 and AudioCodes** for SIP Trunking with **Microsoft Teams Direct Routing**.

Table 1 – PBX Configuration Steps

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

4.2 IP Address Worksheet

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

Component	Lab Value
AudioCodes	
LAN IP Address	10.64.3.10

LAN Subnet Mask	255.255.255.0
WAN IP Address	192.XX.XX.XX
WAN Subnet Mask	255.255.255.128
Cisco UCM	
IP Address	172.16.29.81
Subnet Mask	255.255.255.0
Cisco UBE	
LAN IP Address	10.64.4.182
LAN Subnet Mask	255.255.255.0
WAN IP Address	10.70.69.70
WAN Subnet Mask	255.255.255.0
Cisco Unity	
LAN IP Address	10.80.18.5
LAN Subnet Mask	255.255.255.0
Skype for Business 2015	
Skype for Business - FQDN	Fe0101.tekvizionlabs.com
Edge Server - FQDN	accessedge02.tekvizionlabs.com
Exchange - FQDN	exum.tekvizionlabs.com

Table 2 – IP Addresses

4.3 Microsoft Skype for Business Hybrid Configuration

Configure hybrid connectivity between Skype for Business Server and Teams or Skype for Business Online. Hybrid connectivity enables you to move your on-premises users to Teams or Skype for Business Online, and enables your users signed in Teams using Crestron UC-PHONE to take advantage of cloud services. The scenario assumes with this guide that an Edge Server is already in production and operational.

A SIP address (Session Initiation Protocol) is an identifier that must be unique for each user. In Hybrid mode, it is necessary to configure the Office365 tenant in a shared mode for the SIP domain used with Skype for Business 2015 on-premises.

4.3.1 Create tenant account for Office 365

Follow these steps to set up an Office 365 Enterprise tenant if customer does not have one set up already.

1. Navigate to <https://www.microsoft.com/en-us/microsoft-365/>
2. Select the O365 Plan - O365 tend to fall into 5 main categories: Small Business, Midsize Business, Enterprise, Education and Government. Most of these categories

have trial accounts, and all of these can be converted to regular licensed accounts if required.

3. Enter the Correct information - **Once set up, the Tenant account name cannot be changed.** When administrator first creates the Tenant account, it will be in the form of .onmicrosoft.com, but administrator can add in and use your own registered domain name once the Tenant account is created.
4. Complete the sign-in process by validating the text message or phone call.

4.3.2 Add on-prem domain to O365

To add, modify or remove domains the engineer must be a Global Administrator of a business or enterprise plan. These changes affect the whole tenant, customized administrators or regular users won't be able to make these changes.

1. Go to the admin center at **https://admin.microsoft.com**.
2. Go to the **Setup → Domains** page.
3. Select Add domain and enter the domain name to be added. **tekvizionlabs.com** is of the domain used for this setup
4. Select **Next** and **Finish**
5. Wait at least five minutes for replication to complete.

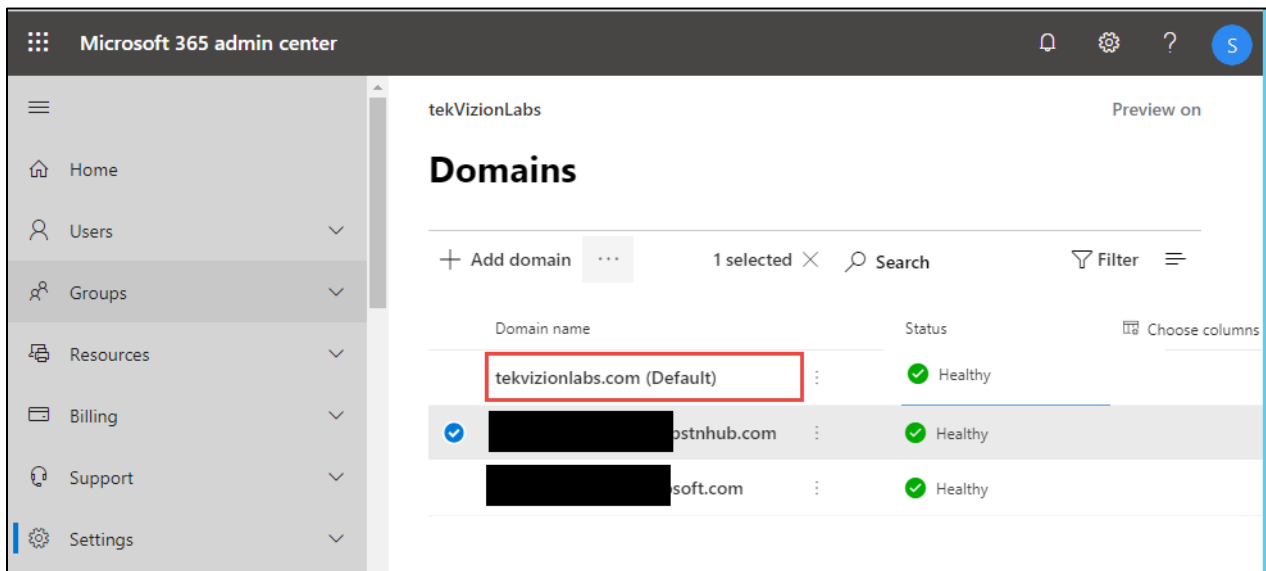


Figure 2 – Add Domain

4.3.3 Setup AD synchronization

Active Directory synchronization keeps on-premises Active Directory continuously synchronized with Office 365. This lets you to create a synchronized version of each user account and group.

The configuration steps below guides the administrator to setup the Azure AD Connect tool downloaded from Microsoft Site and provision in the on-prem server. On the Connect to Azure AD screen, use a global admin account and password. Recommendation is to use an account in the default **onmicrosoft.com** domain, which comes with Azure AD tenant.

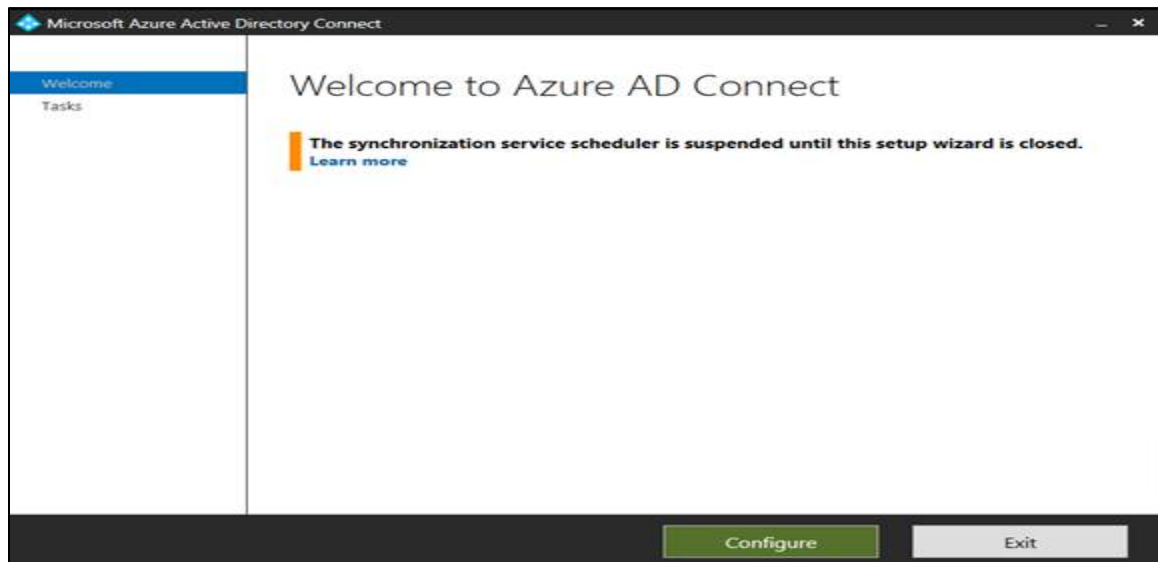


Figure 3 – Azure AD – Sync Process

1. On the Connect to Azure AD screen, use a global admin account and password. Recommendation is to use an account in the default **onmicrosoft.com** domain, which comes with Azure AD tenant.

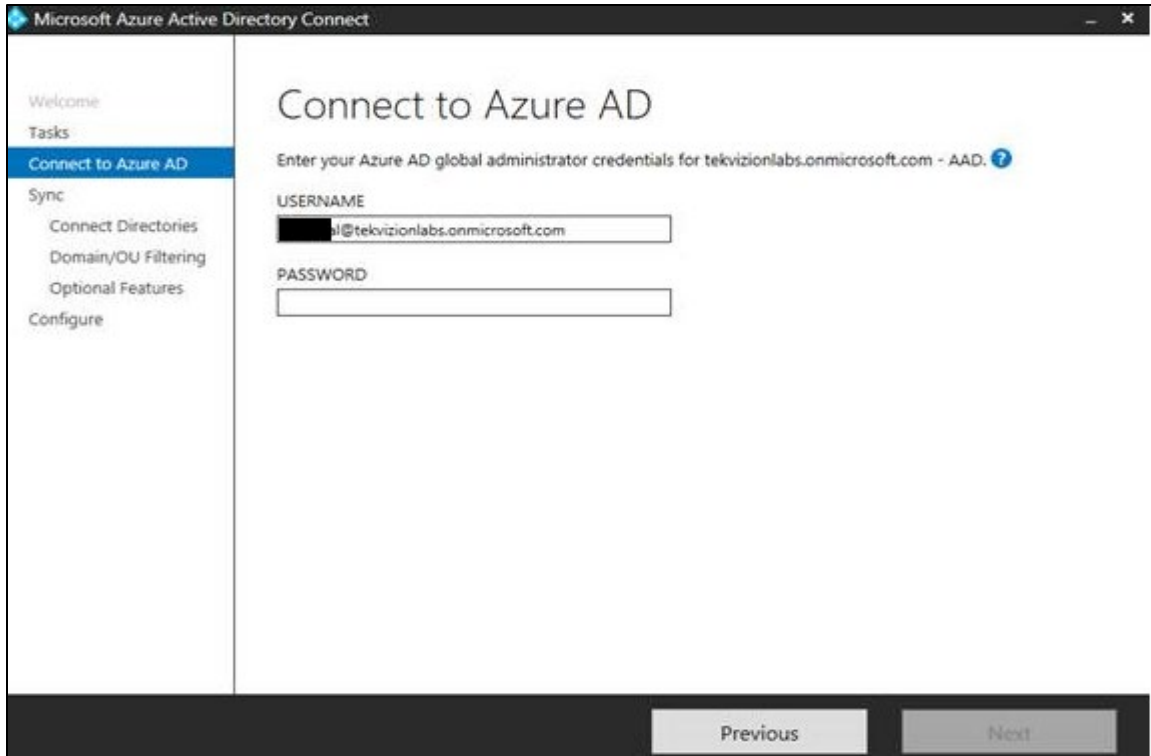


Figure 4 – Azure AD – Sync Process

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

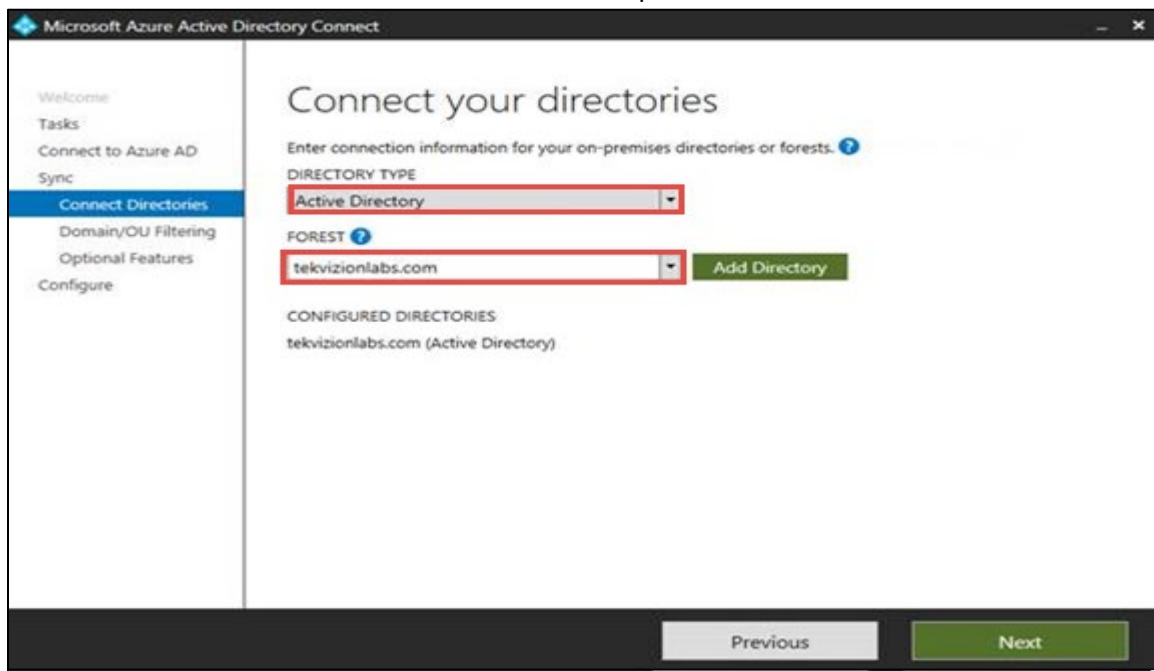


Figure 5 – Azure AD – Sync Process

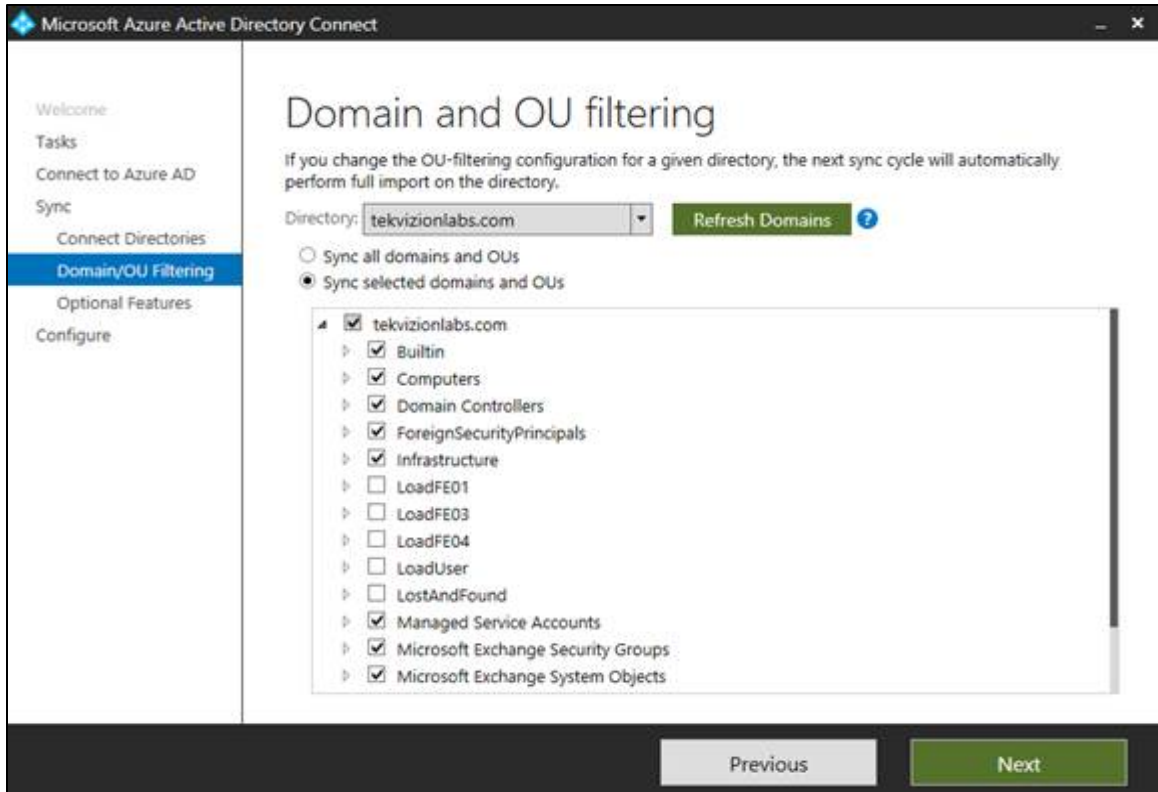


Figure 6 – Azure AD – Sync Process

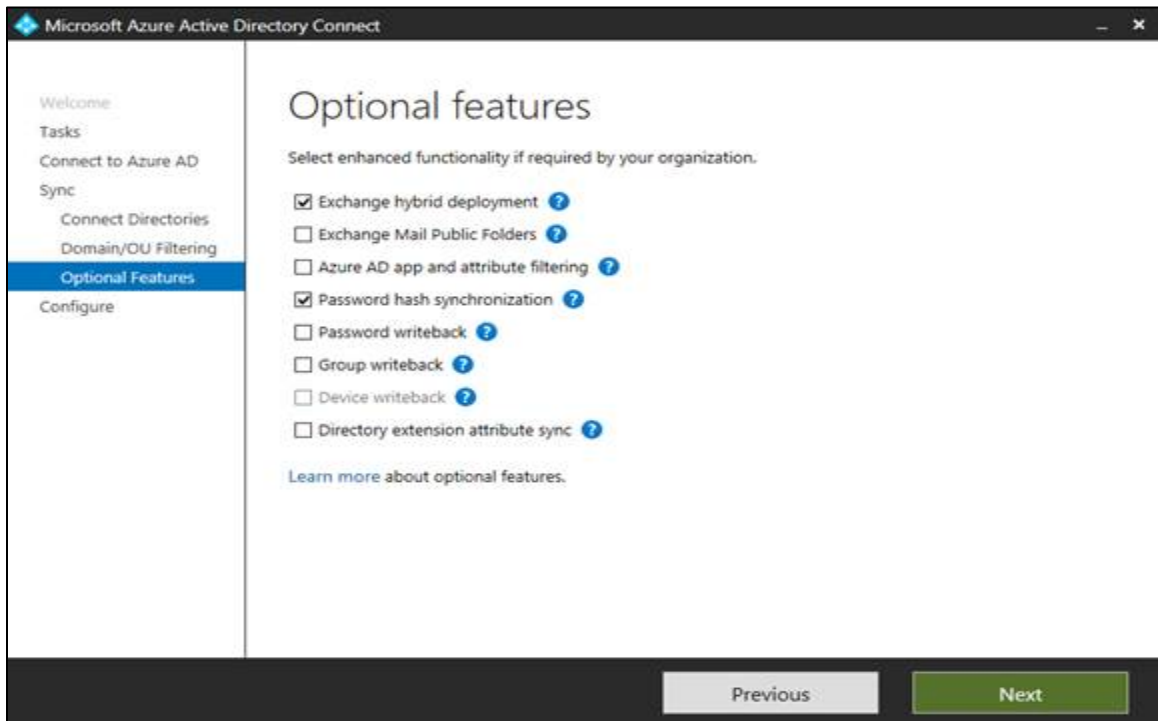


Figure 7 – Azure AD – Sync Process

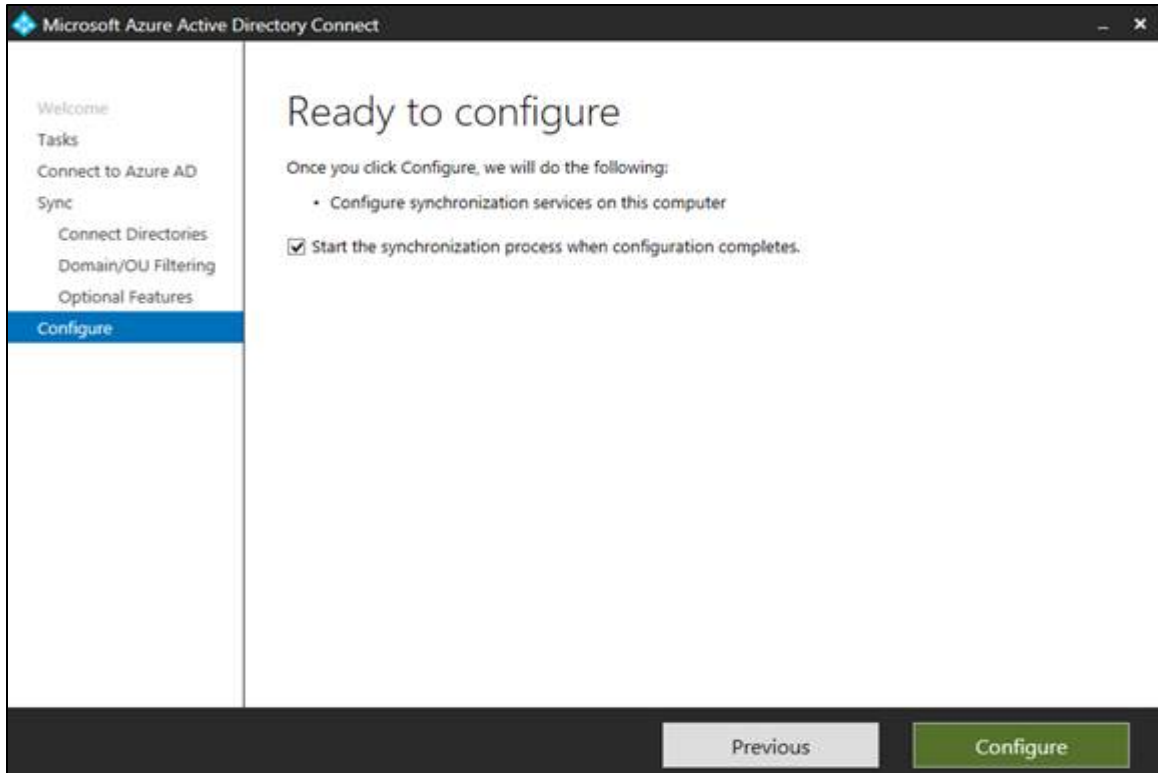


Figure 8 – Azure AD – Sync Process

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

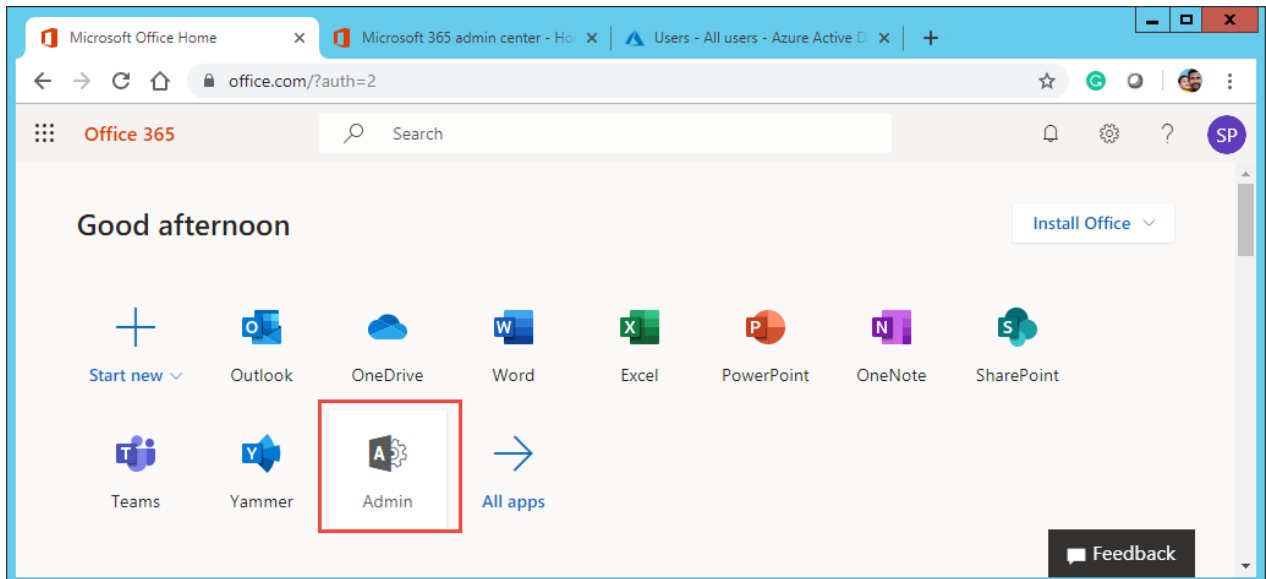


Figure 9 – Azure AD – Sync Process

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

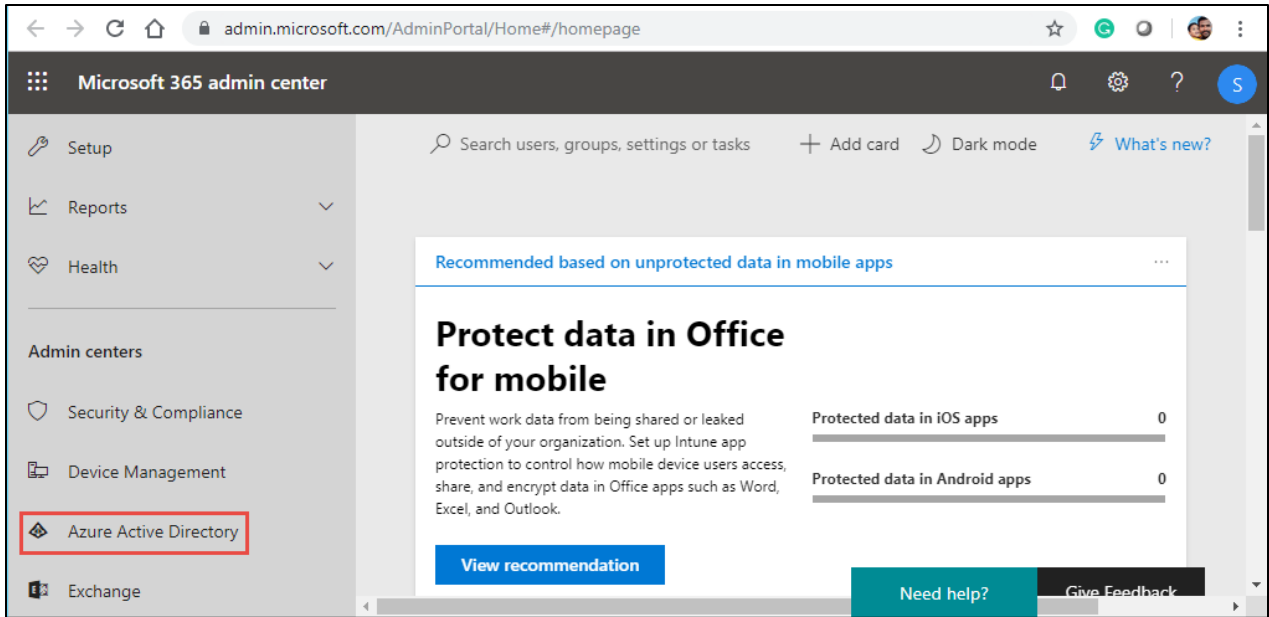


Figure 10 – Azure AD – Sync Process

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

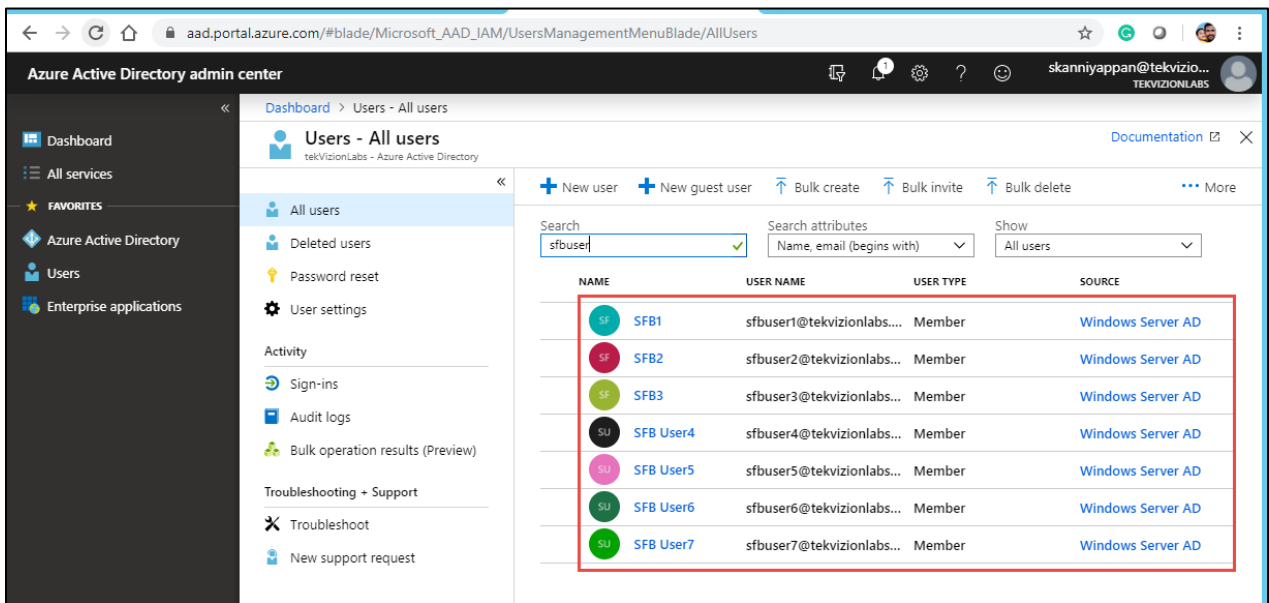


Figure 11 – Azure AD – Sync Process

4.3.4 Configure skype for Business hybrid

The below steps shows how to configure hybrid connectivity between Skype for Business Server and Teams or Skype for Business Online. Hybrid connectivity enables the ability to

move on-premises users to Teams or Skype for Business Online, and enable users to take advantage of cloud services.

4.3.4.1 Configure your on-premises Edge service to federate with Office 365

Federation allows users in on-premises deployment to communicate with Office 365 users in organization. To configure federation, run the following cmdlet in the Skype for Business Server Management Shell:

Set-CsAccessEdgeConfiguration -AllowOutsideUsers 1 -AllowFederatedUsers 1 - EnablePartnerDiscovery 1 -UseDnsSrvRouting

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

Identity                : Global
AllowAnonymousUsers    : True
AllowFederatedUsers    : True
AllowOutsideUsers      : True
BeClearingHouse        : False
EnablePartnerDiscovery : True
DiscoveredPartnerVerificationLevel : UseSourceVerification
EnableArchivingDisclaimer : True
EnableUserReplicator   : False
KeepCr1sUpToDateForPeers : True
MarkSourceVerifiableOnOutgoingMessages : True
OutgoingTlsCountForFederatedPartners : 4
DnsSrvCacheRecordCount : 131072
DiscoveredPartnerStandardRate : 20
EnabledDiscoveredPartnerContactsLimit : True
MaxContactsPerDiscoveredPartner : 1000
DiscoveredPartnerReportPeriodMinutes : 60
MaxAcceptedCertificatesStored : 1000
MaxRejectedCertificatesStored : 500
CertificatesDeletedPercentage : 20
SkypeSearchUrl         : https://skypegraph.skype.com/search/v1.0
RoutingMethod          : UseDnsSrvRouting

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

Figure 12 – Edge Federation

4.3.4.2 Configure your on-premises environment to enable shared SIP address space with Office 365

Configure your on-premises environment to trust Office 365 and enable shared SIP address space with Office 365. Office 365 can potentially host user accounts for the same set of SIP domains as your on-premises environment, and messages can be routed between users hosted on premises and online. This is achieved by configuring a hosting provider with **ProxyFqdn=sipfed.online.lync.com** as described below.

Create a new hosting provider using the **New-CsHostingProvider** cmdlet as follows:

New-CsHostingProvider -Identity Office365 -ProxyFqdn "sipfed.online.lync.com" - Enabled \$true -EnabledSharedAddressSpace \$true -HostsOCSUsers \$true -

**VerificationLevel UseSourceVerification -IsLocal \$false -AutodiscoverUrl
https://webdir.online.lync.com/Autodiscover/AutodiscoverService.svc/root**

The below is the output taken from on-prem Skype for Business Server

```
PS C:\Users\administrator.TEKVIZIONLABS> Get-CsHostingProvider
Identity           : LyncOnline
Name               : LyncOnline
ProxyFqdn          : sipfed.online.lync.com
VerificationLevel  : AlwaysVerifiable
Enabled            : True
EnabledSharedAddressSpace : True
HostsOCSUsers      : True
IsLocal            : False
AutodiscoverUrl    : https://webdir1a.online.lync.com/Autodiscover/AutodiscoverService.svc/root

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

Figure 13 – Hosting Provider

4.3.4.3 Enable shared SIP address space in your Office 365 tenant

In addition to the change made in on-premises deployment, make the corresponding change in Office 365 tenant to enable shared SIP address space with on-premises deployment. A SIP address (Session Initiation Protocol) is an identifier that must be unique for each user. In Hybrid mode, it is necessary to configure the Office365 tenant in a shared mode for the SIP domain used with Skype for Business 2015 on-premises.

Set-CsTenantFederationConfiguration -SharedSipAddressSpace \$true

The SharedSipAddressSpace attribute needs to remain "True" until migration to online is completed, and no users remain on-premises.

4.3.4.4 Skype for Business 2015 Hybrid Mode Configuration

Skype for Business control panel provides an option to configure the Hybrid Mode .

- 1) In Control Panel home page, select the link « Set up hybrid with Teams and Skype for Business Online

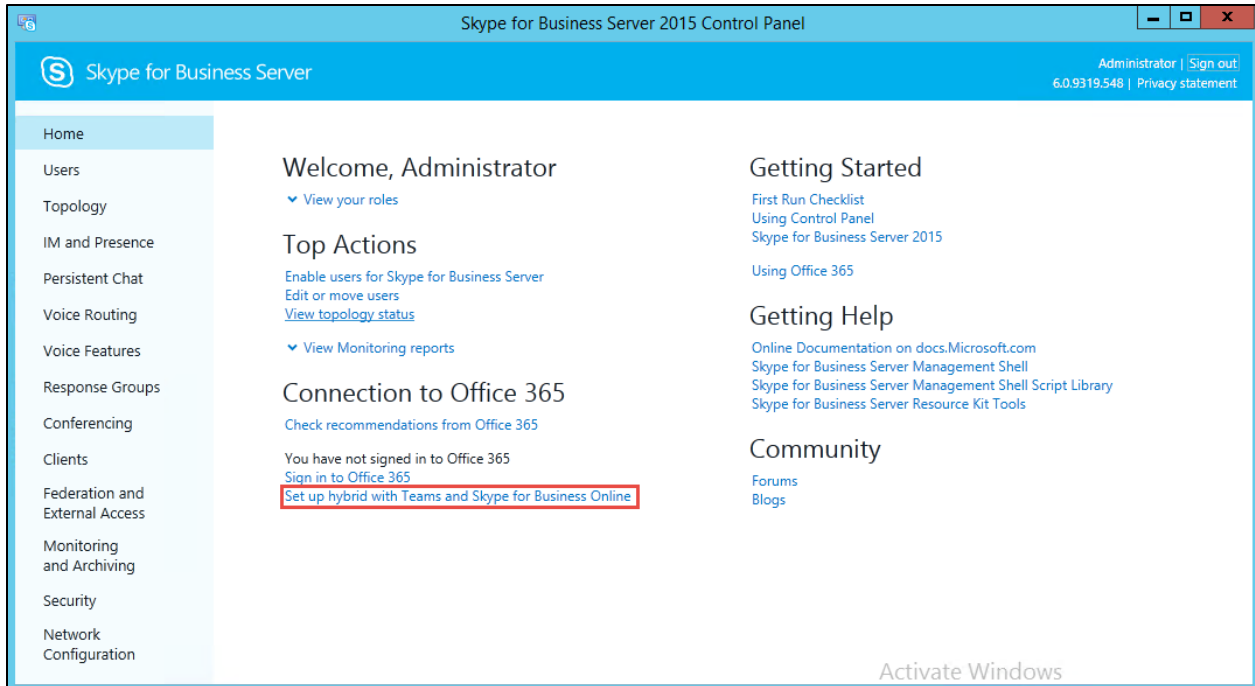


Figure 14 – Hybrid with Teams and SFB

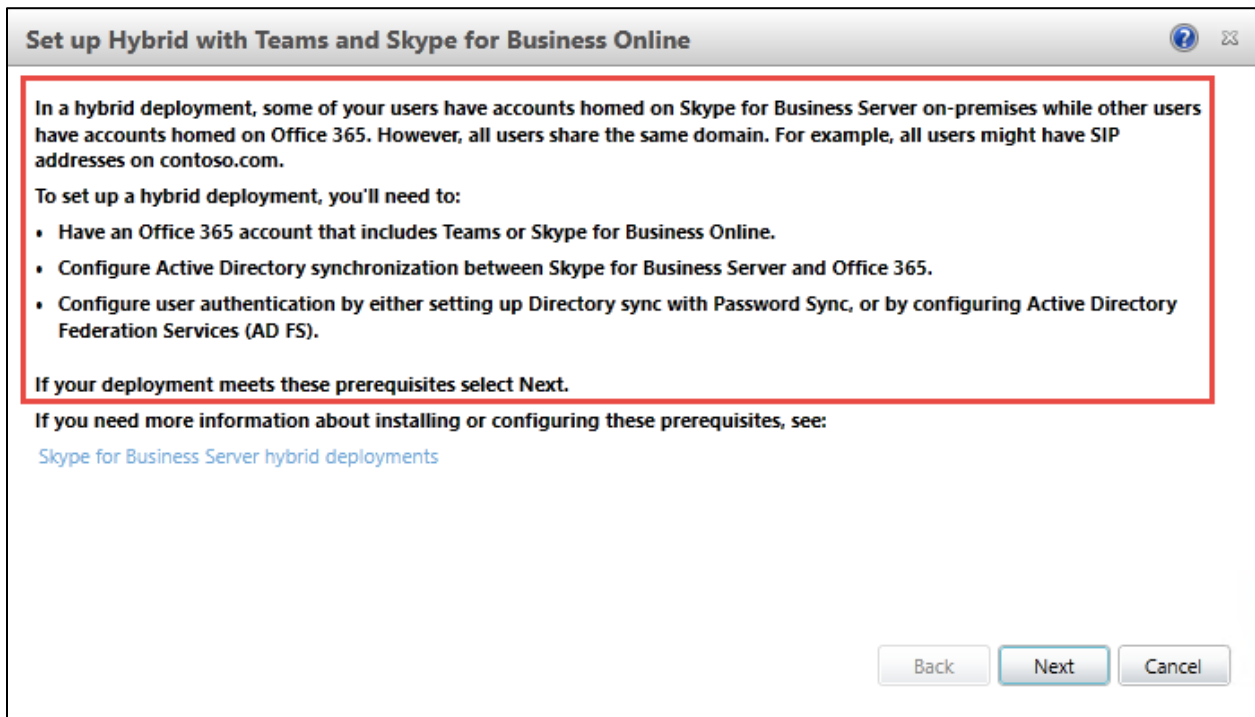


Figure 15 – Hybrid with Teams and SFB Contd.

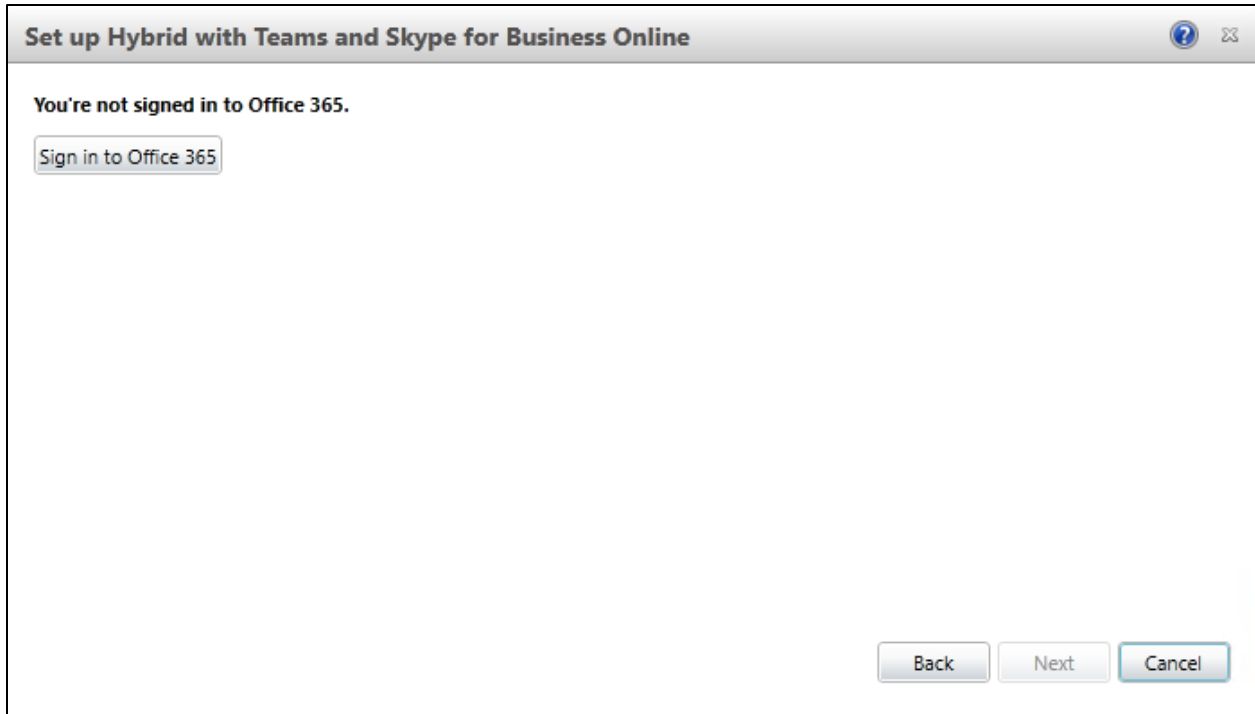


Figure 16 – Hybrid with Teams and SFB Contd.

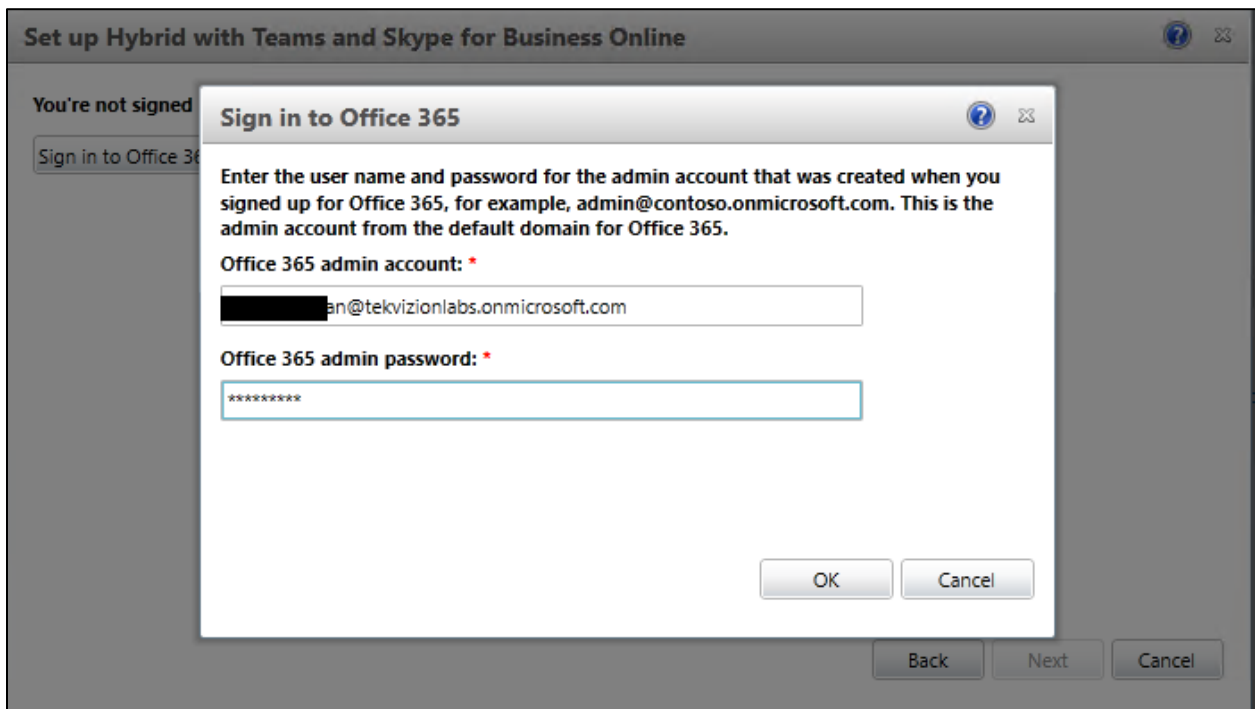


Figure 17 – Hybrid with Teams and SFB Contd.

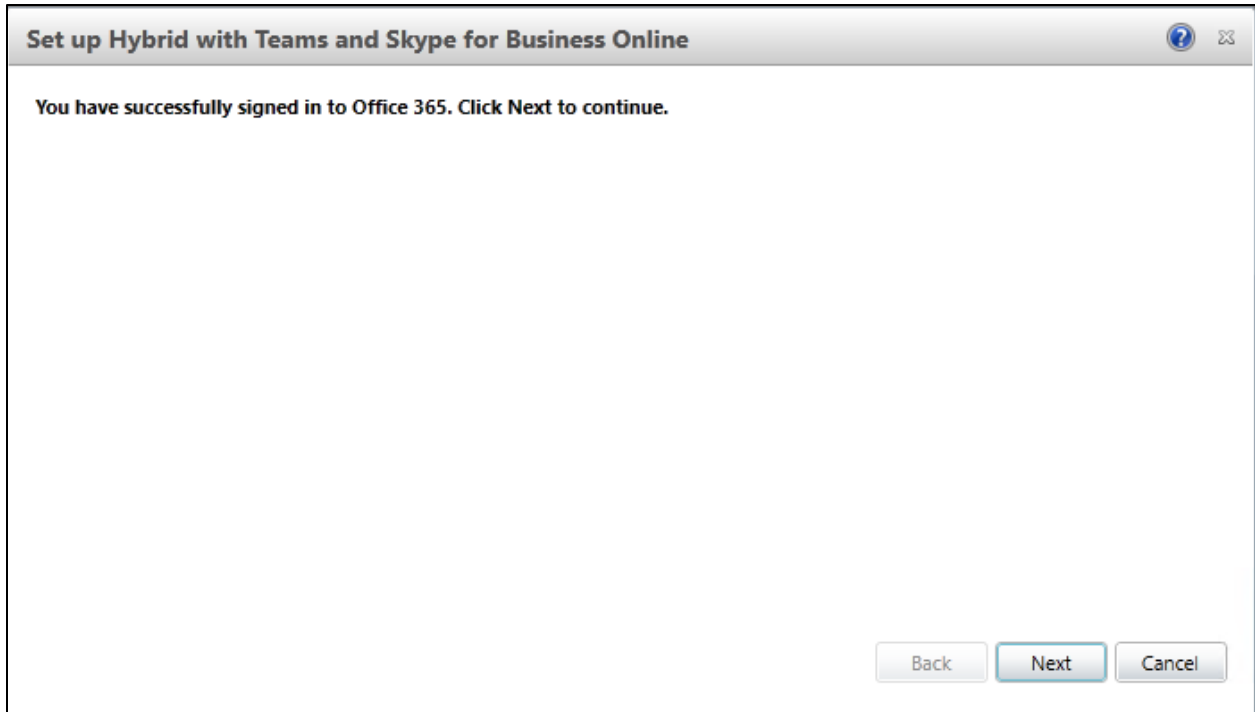


Figure 18 – Hybrid with Teams and SFB Contd.

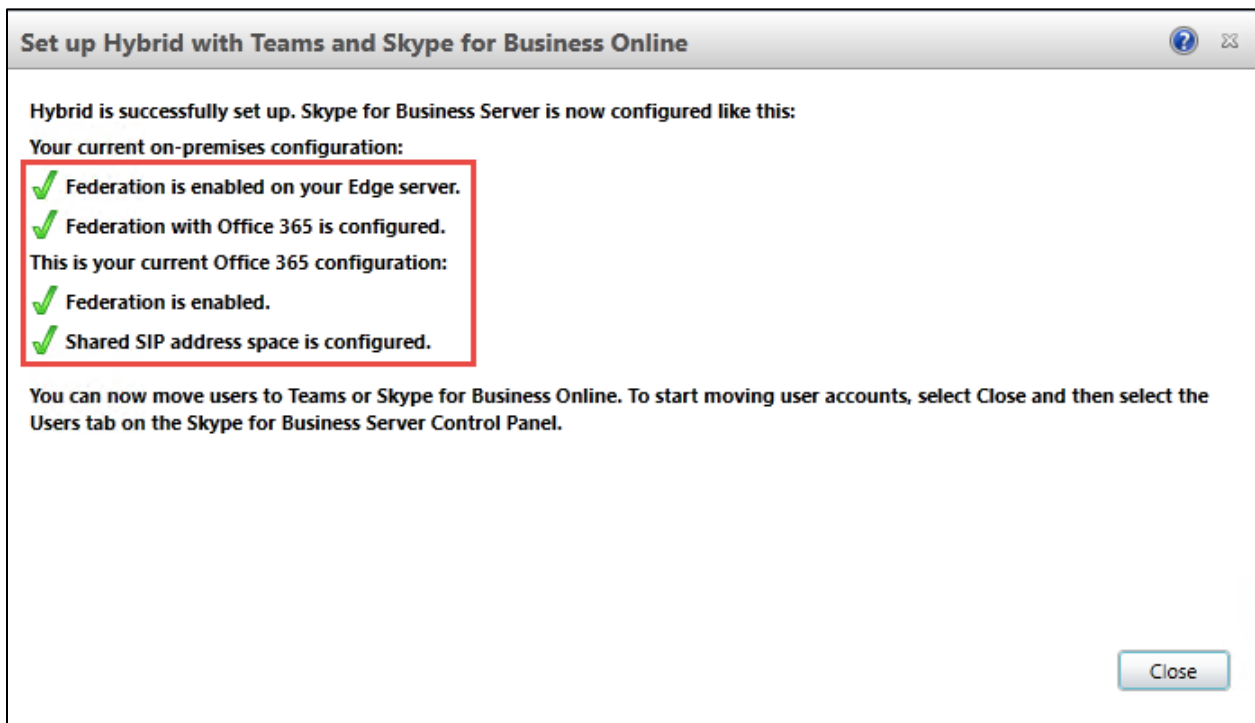


Figure 19 – Hybrid with Teams and SFB Contd.

4.3.4.5 Move to Teams using Skype for Business Server Control Panel

In an on-premises deployment of Skype for Business Server that is enabled for hybrid, users can be moved between on-premises environment and cloud (either to Microsoft Teams or to Skype for Business Online).

When a user is moved online, the user is allowed to use Skype for Business Online or Teams Only or both (Islands mode). Microsoft strongly recommends that users moved online to be configured in Teams only mode, which will ensure that routing of all incoming chats and calls lands in their Teams client. This is configured in this setup.

1. Open the Skype for Business Server Control Panel app.
2. In the left navigation, choose **Users**.
3. Use **Find** to locate the user(s) you would like to move to Teams.
4. Select the user(s), and then, from the **Action** dropdown above the list, choose **Move selected users to Teams**.
5. In the wizard, click **Next**.
6. If prompted, sign in to Office 365, with an account that ends in .onmicrosoft.com and has sufficient permissions (tenant user with Global Admin role).
7. Click **Next**, and then **Next** one more time to move the user.
8. Note that status messages regarding success or failure are provided at the top of the main Control Panel app, not in the wizard.

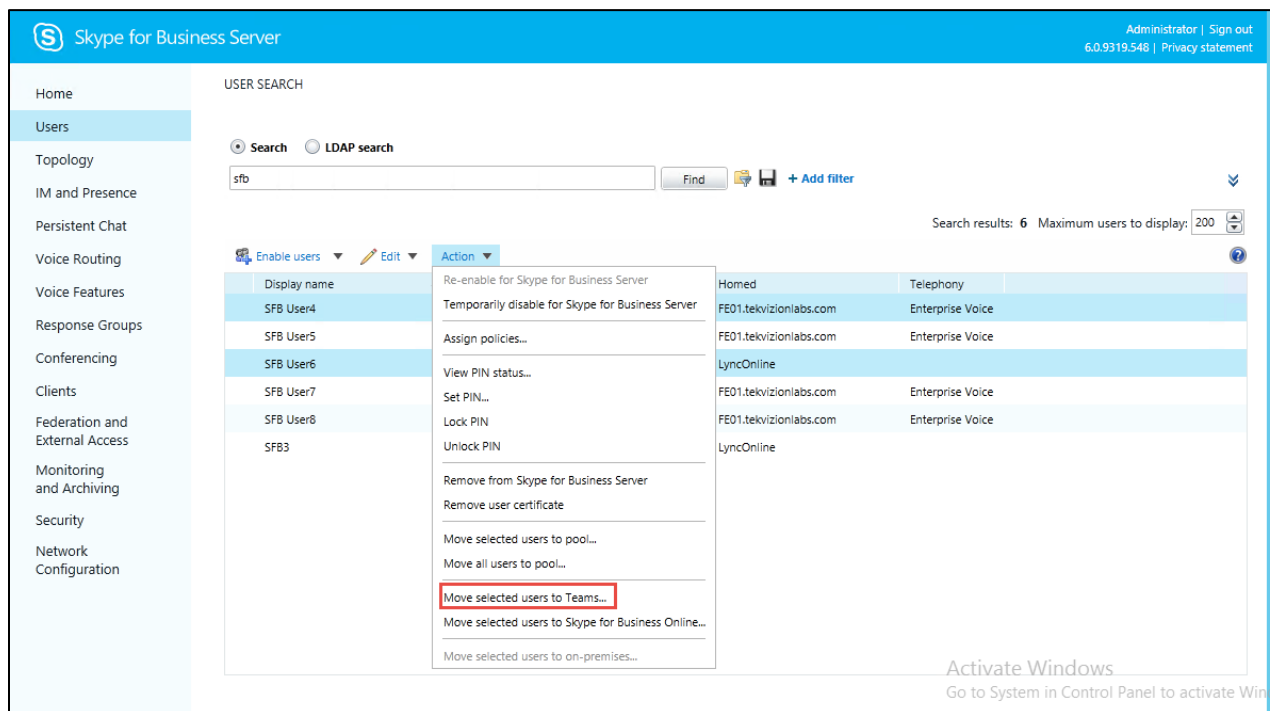


Figure 20 – Move Users to Teams

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

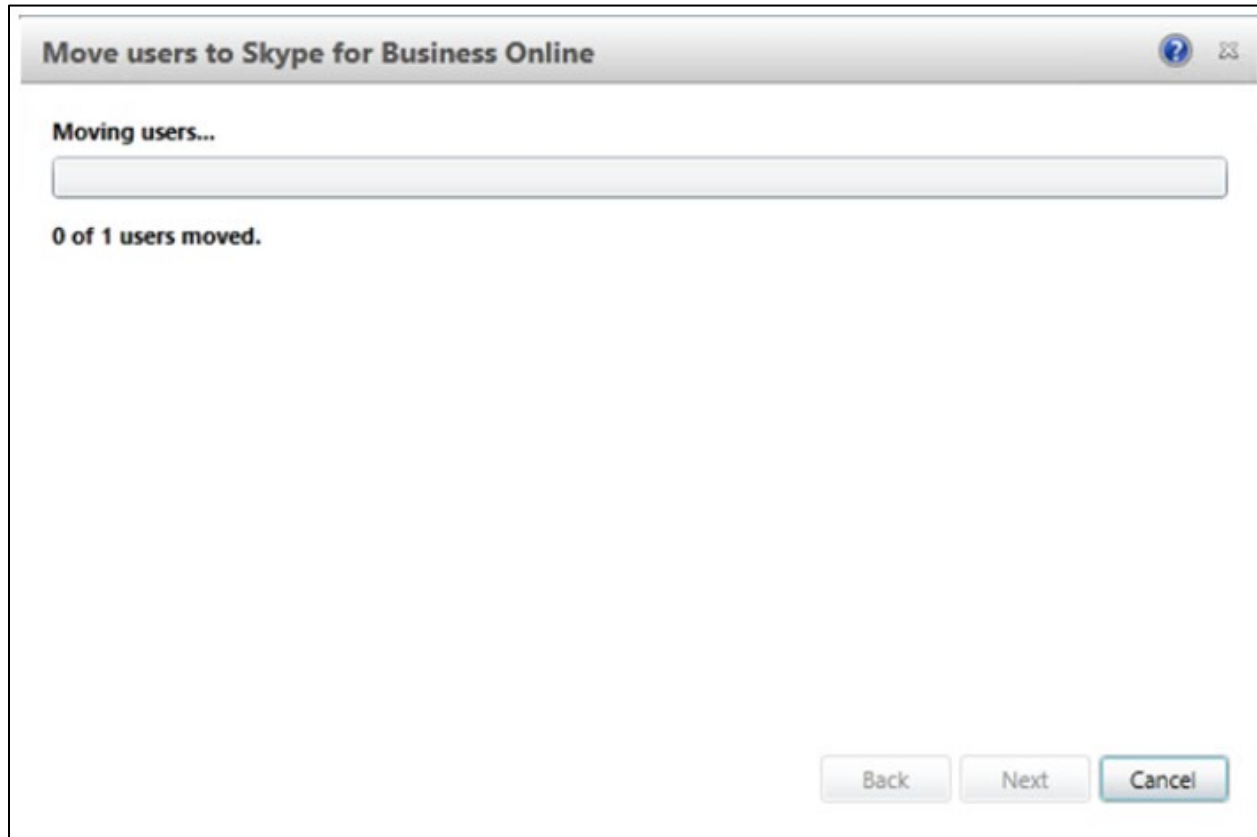


Figure 21 - Move Users to Teams Contd.

The migration of the selected user is complete.

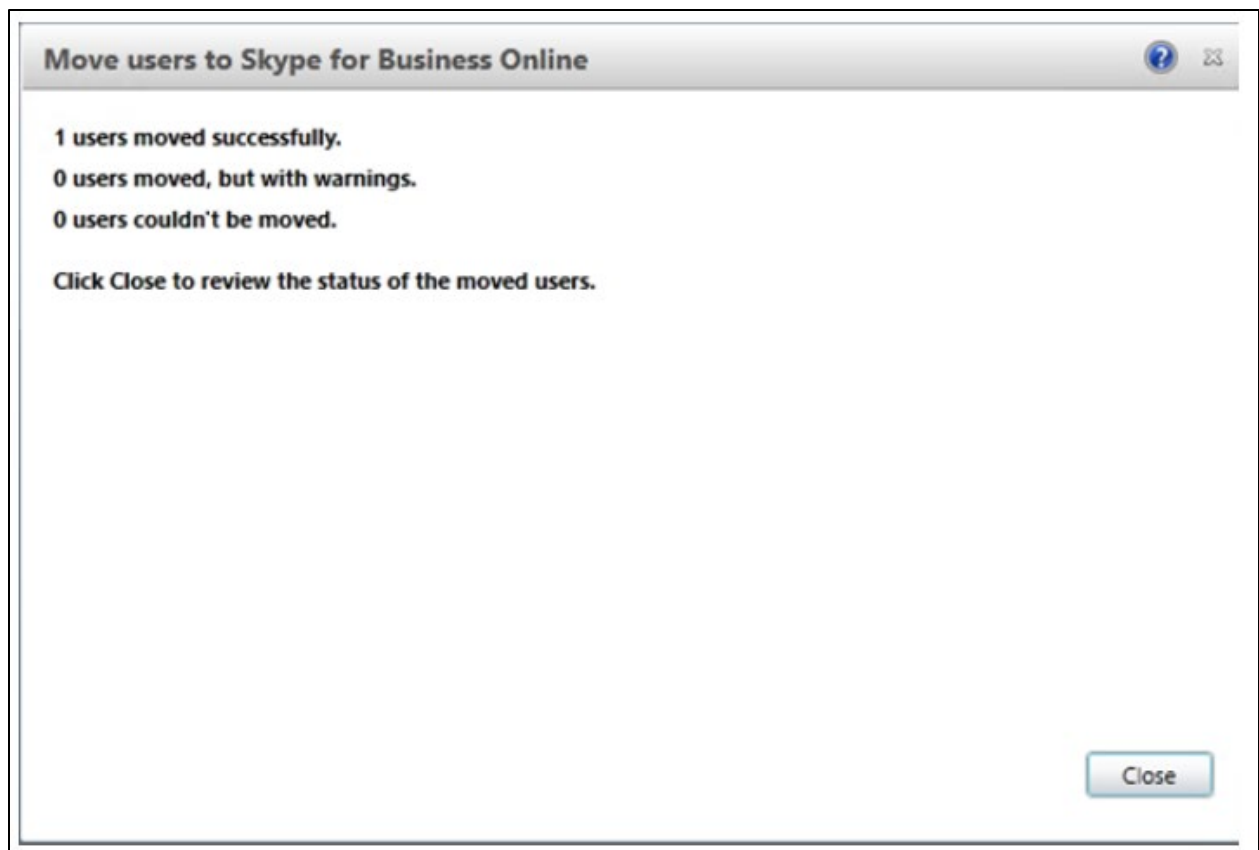


Figure 22 – Move Users to Teams Contd.

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

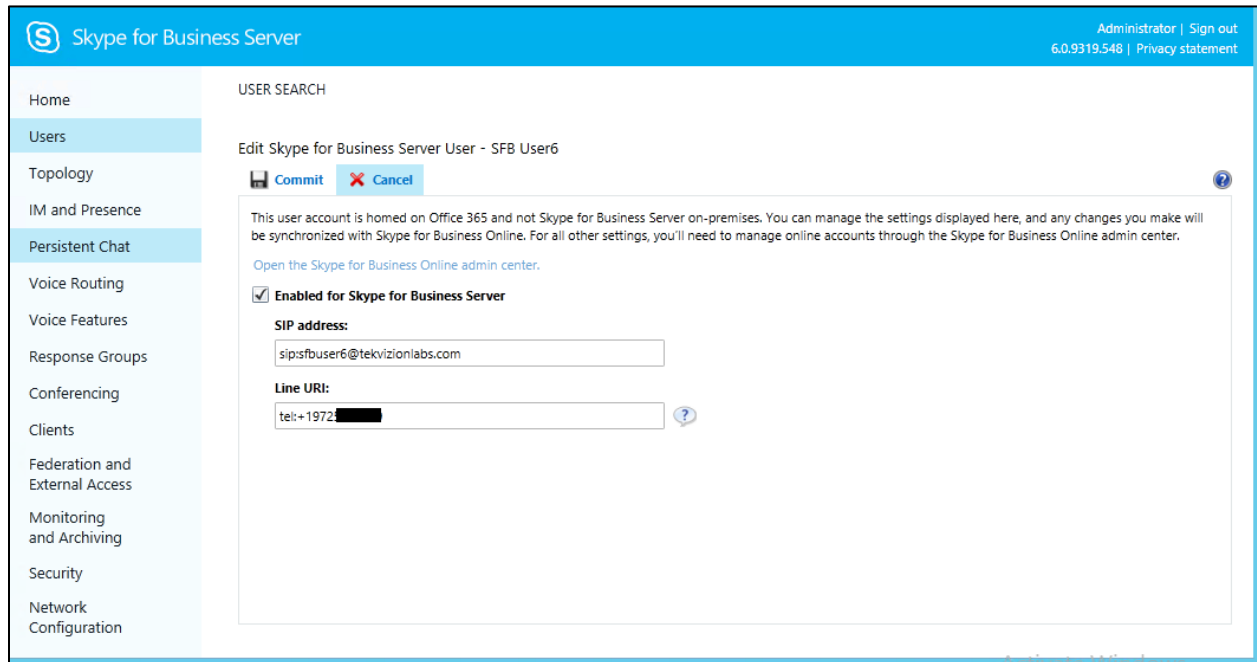


Figure 23 – Move Users to Teams

4.4 Microsoft Teams Configuration

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

4.4.1 Teams User Configuration

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

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

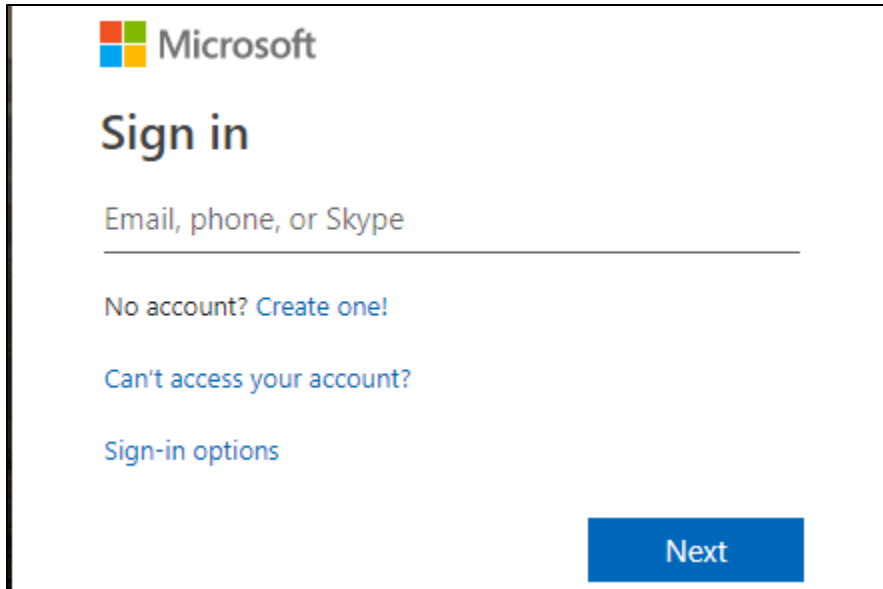


Figure 24: Office 365 Portal Login

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

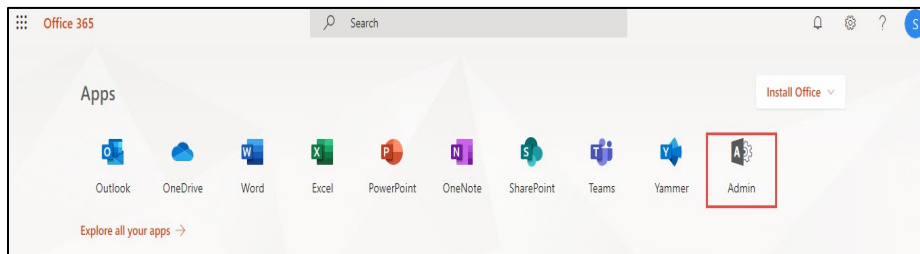


Figure 25: Office 365 Portal Login

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

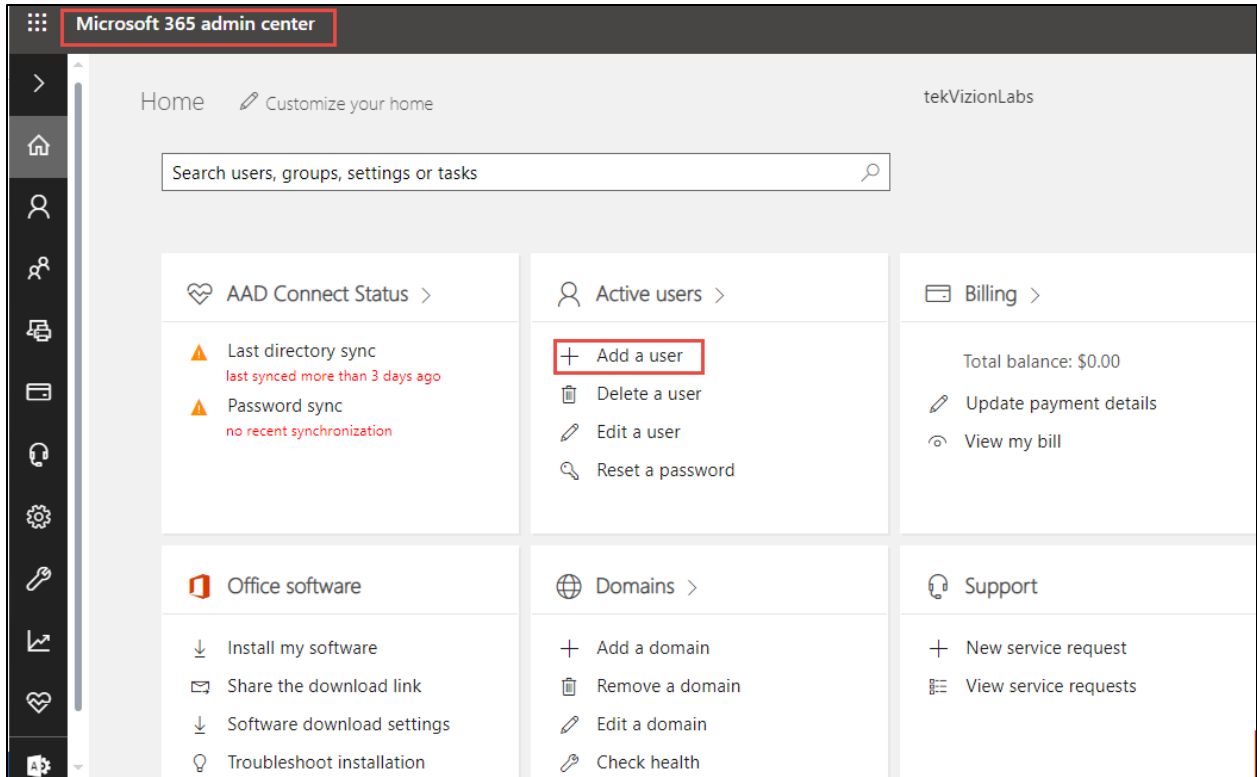


Figure 26: Teams User Creation

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

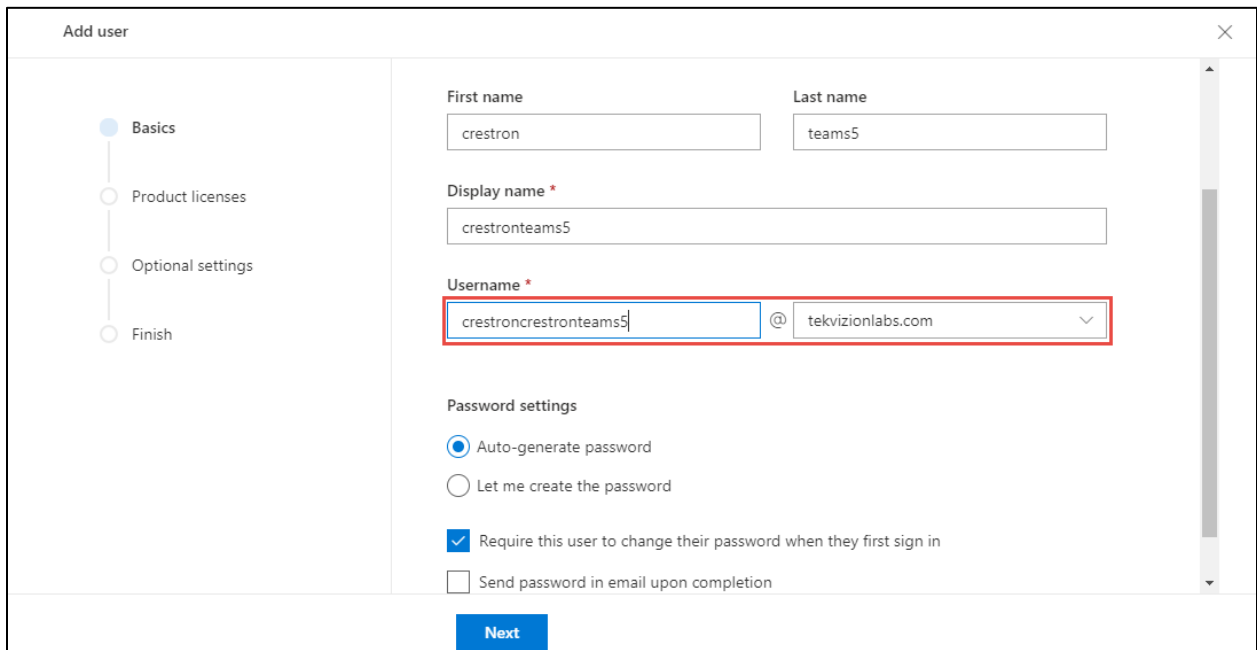


Figure 27: Teams User Creation – Contd.

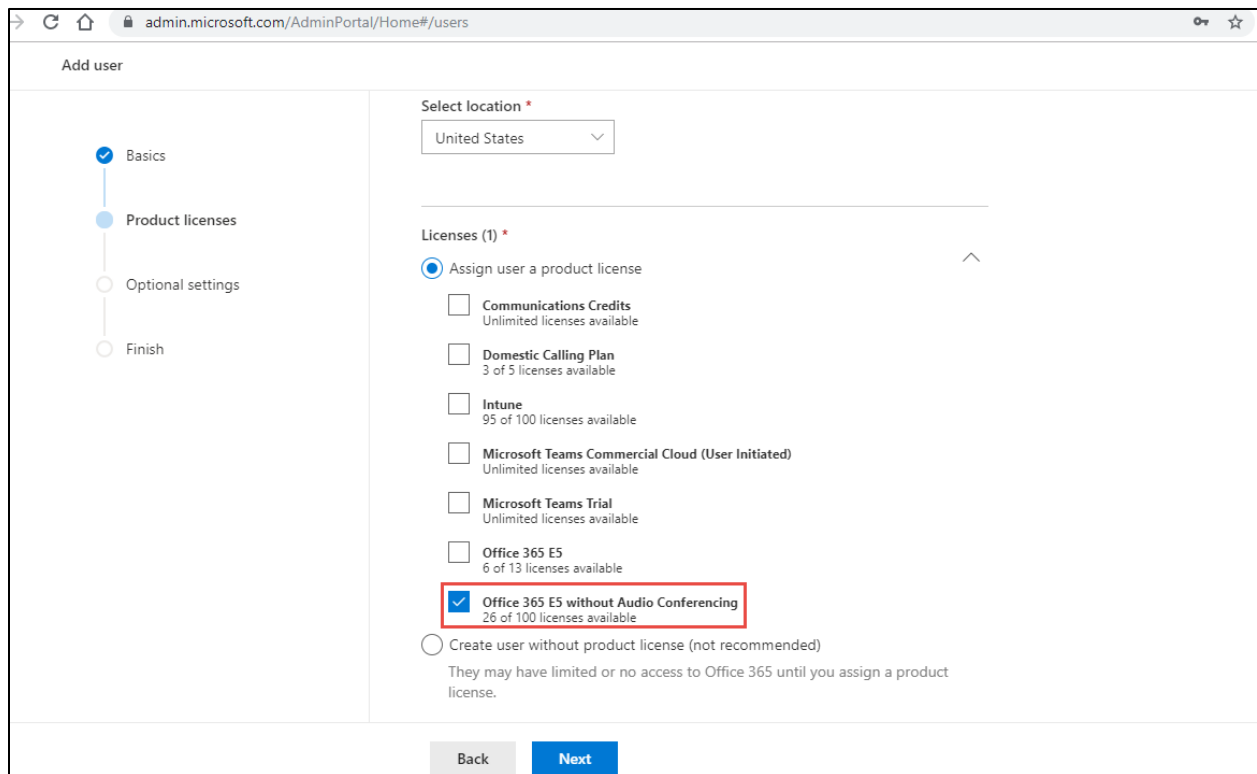


Figure 28: Teams User Creation – Contd.

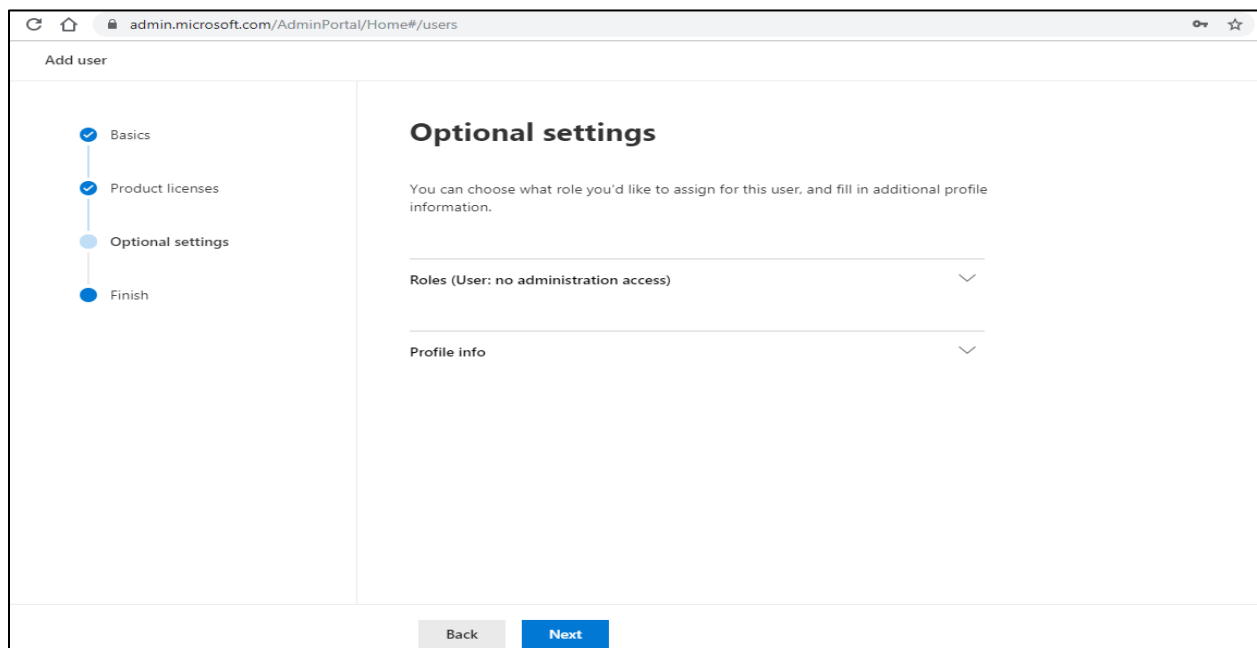


Figure 29: Teams User Creation – Contd.

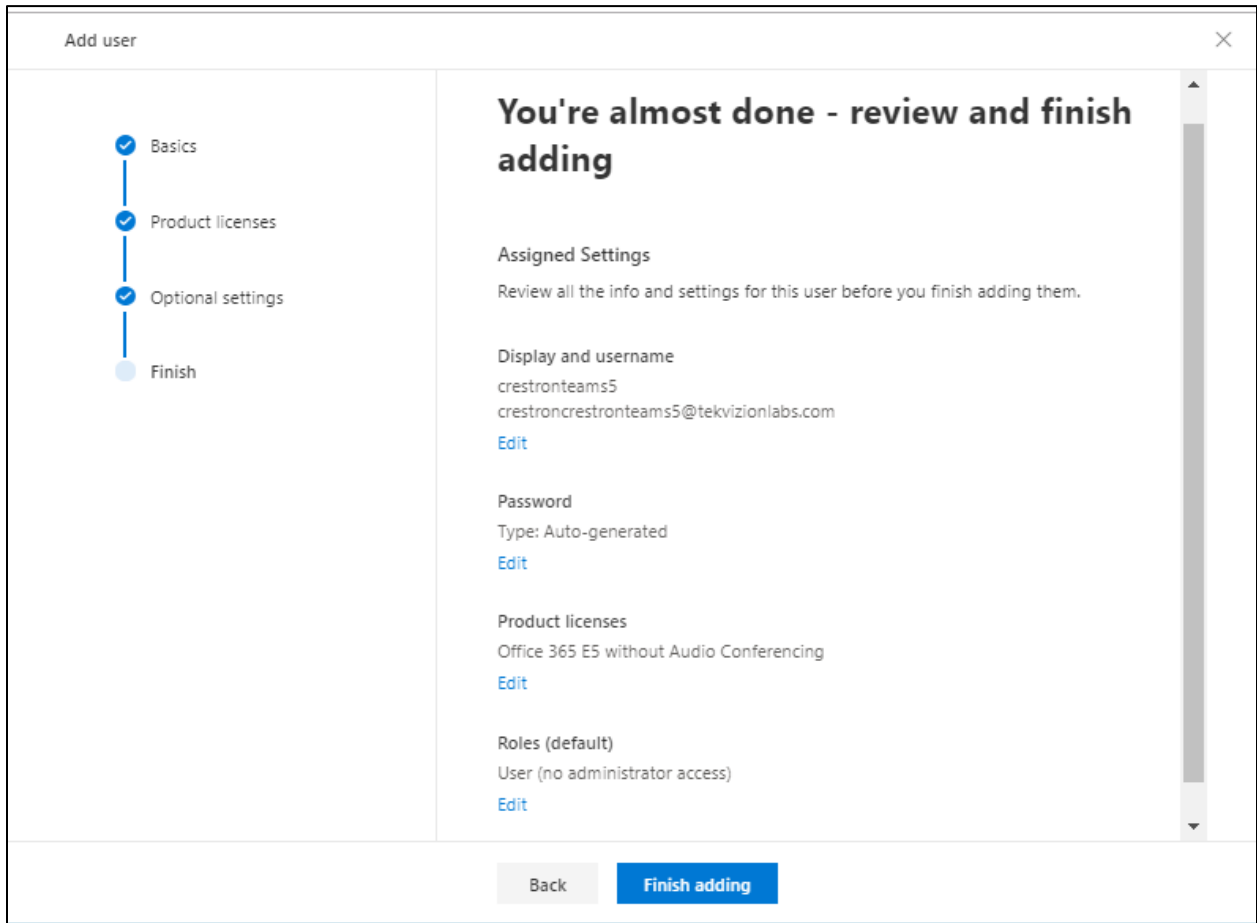


Figure 30: Teams User Creation – Contd.

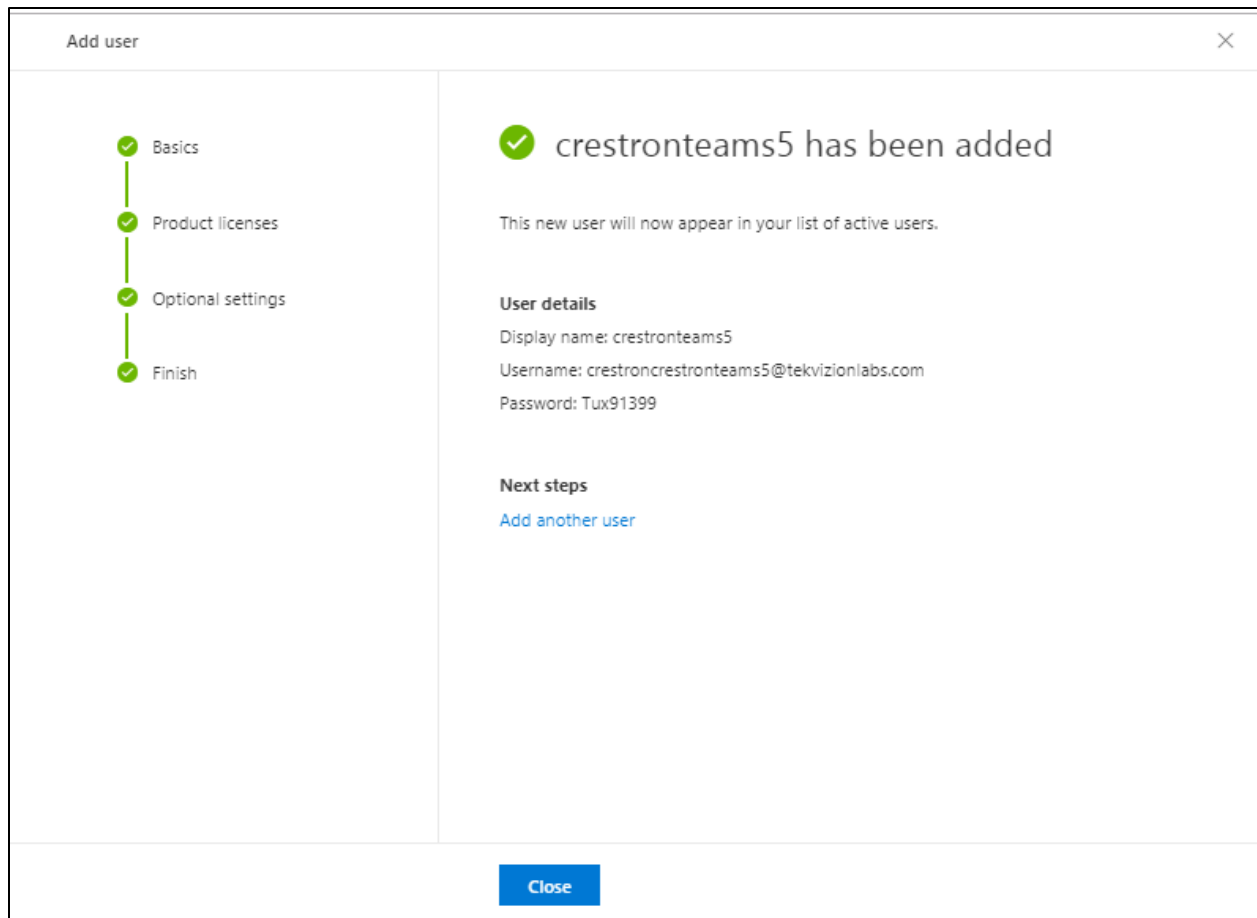


Figure 31: Teams User Creation – Contd.

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

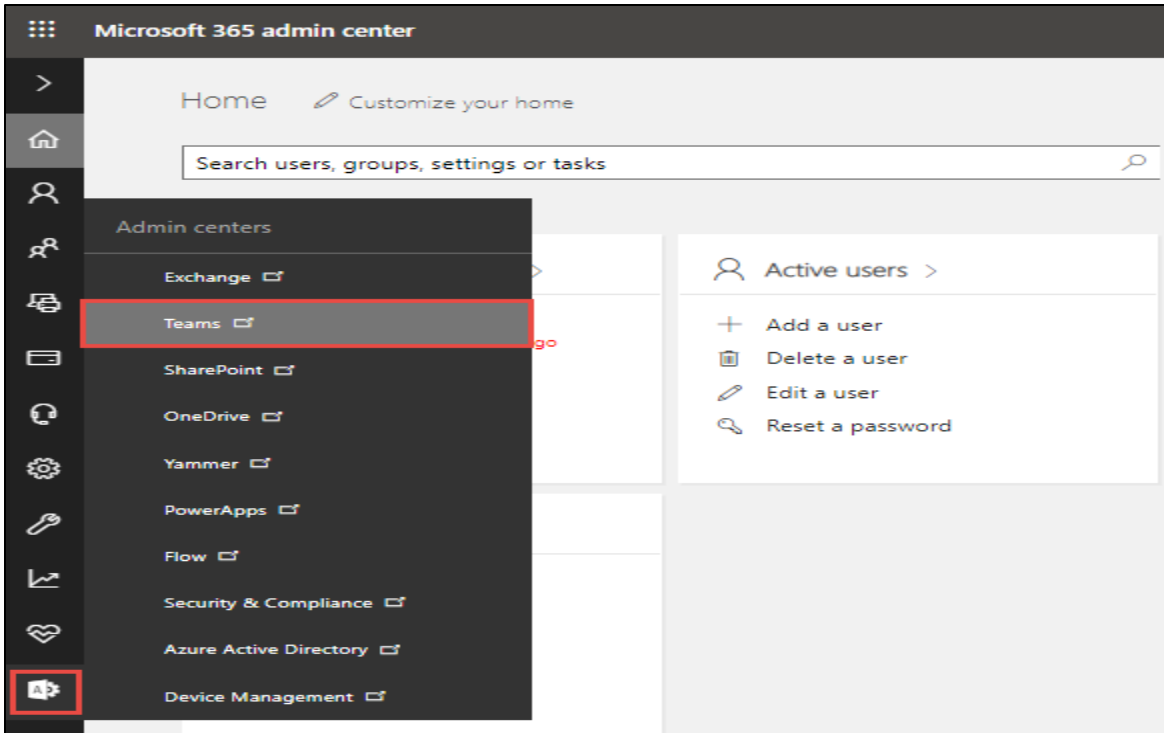


Figure 32: Microsoft O365 admin

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

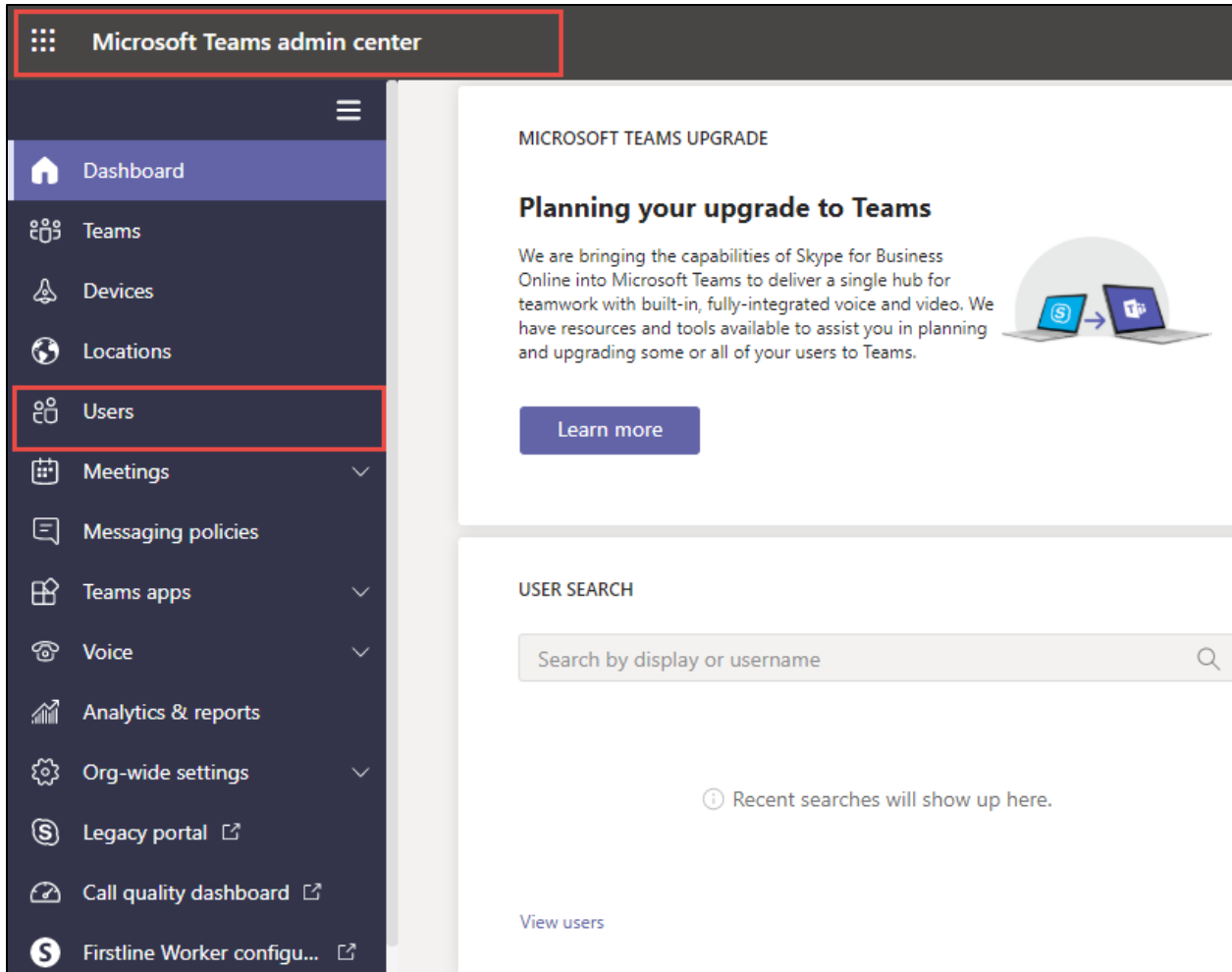


Figure 33: Microsoft O365 admin

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

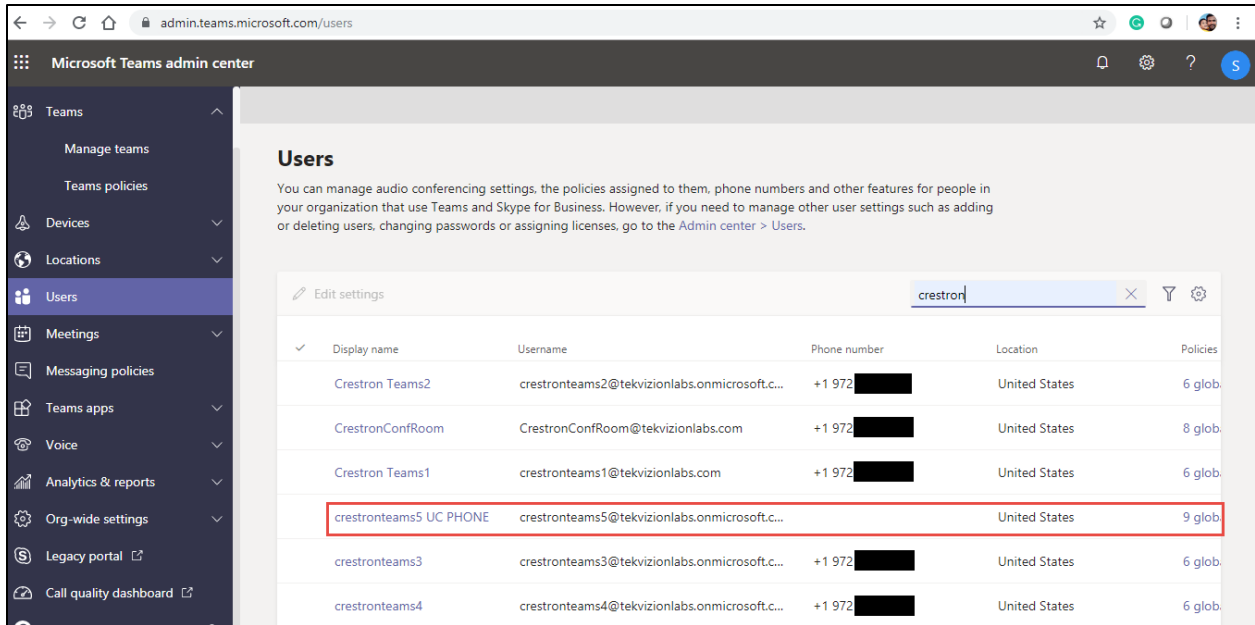


Figure 34: Microsoft O365 admin

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

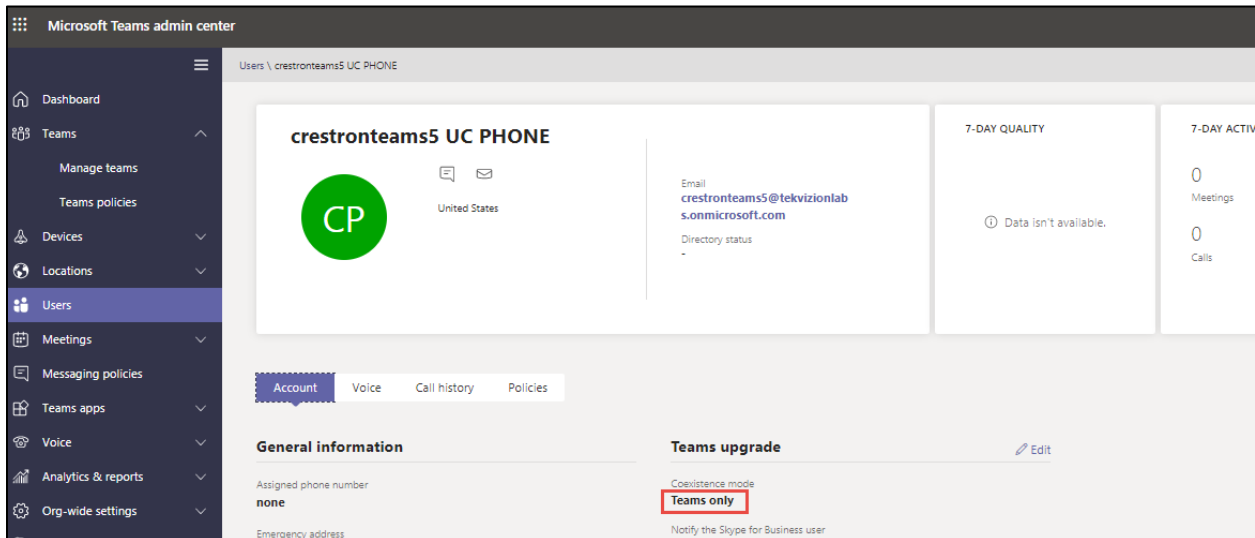


Figure 35: Teams User

4.4.2 Configure Calling policy to Users

- 1) Under user properties, navigate to Policies and set the Calling Policy as shown below. Here in the below example custom policy “Busy on Busy enabled” is assigned to user. Procedure to create custom policy is shown in the next section.

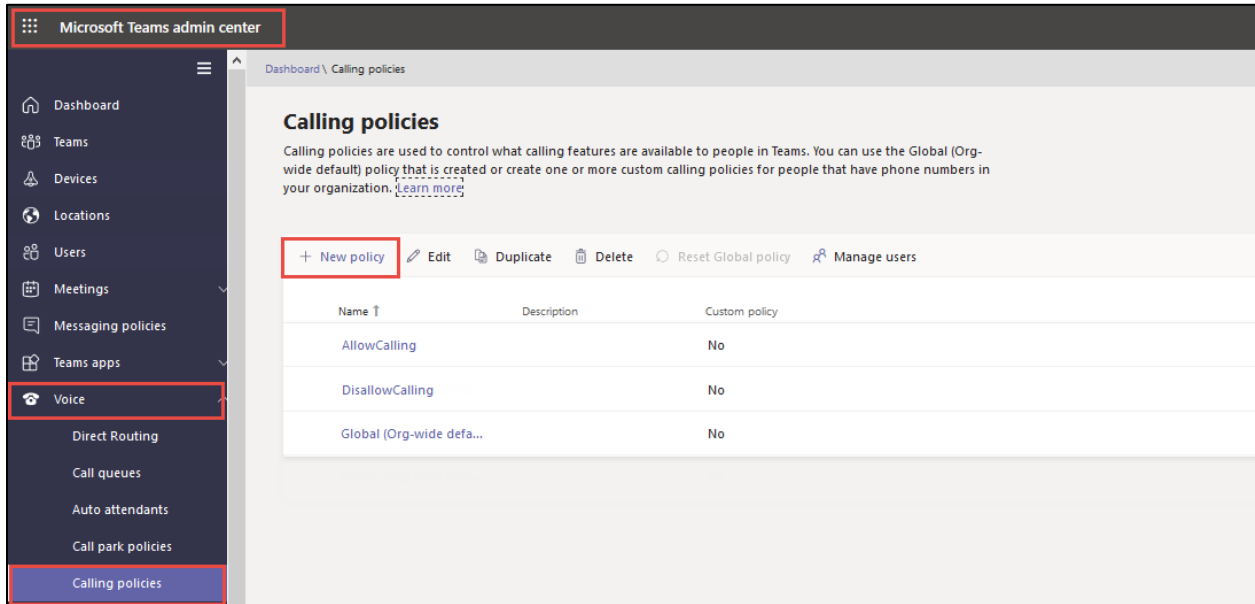


Figure 36 – Calling Policy

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

Dashboard \ Calling policies \ New policy

Busy on Busy Enabled

Description

Make private calls On

Call forwarding and simultaneous ringing to people in your organization On

Call forwarding and simultaneous ringing to external phone numbers On

Voicemail is available for routing inbound calls

Inbound calls can be routed to call groups On

Allow delegation for inbound and outbound calls On

Prevent toll bypass and send calls through the PSTN Off

Busy on busy is available when in a call On

Figure 37 – Calling Policy

4.4.3 Configure user parameters.

Using the Remote PowerShell connect to Microsoft office 365 Tenant. Use the below commands to set DID and enable Enterprise Voice, Hosted Voicemail for Teams users.

```
Set-CsUser -identity "crestronteam5@tekvisionlabs.com" -EnterpriseVoiceEnabled $true -HostedVoicemail $true
```

```
Set-CsUser -identity "crestronteam5@tekvisionlabs.com" -OnPremLineURI tel:+197259800xx
```

4.4.4 Create Online PSTN Gateway

Use the below command to pair the SBC to the tenant.

```
New-CsOnlinePSTNGateway -Fqdn <SBC FQDN> -SipSignallingPort <SBC SIP Port>
```

```
-ForwardCallHistory $true -ForwardPai $true -MaxConcurrentSessions <Max Concurrent Sessions the SBC can handle> -Enabled $true -MediaBypass $true
```

```

PS C:\Users\spandian> Get-CsOnlinePSTNGateway -Identity sbc4.tekvizionlabs.com

Identity           : sbc4.tekvizionlabs.com
Fqdn               : sbc4.tekvizionlabs.com
SipSignallingPort  : 5061
FailoverTimeSeconds : 10
ForwardCallHistory : True
ForwardPai        : True
SendsipOptions    : True
MaxConcurrentSessions : 100
Enabled           : True
MediaBypass       : True
GatewaySiteId     :
GatewaySiteLbrEnabled : False
FailoverResponseCodes : 408, 503, 504
GenerateRingingWhileLocatingUser : True
PidfloSupported   : True
MediaRelayRoutingLocationOverride :
ProxySbc          :
BypassMode        : None

```

Figure 38 - Online PSTN Gateway

4.4.5 Configure Online PSTN Usage

Use the below command to add a new PSTN usage.

Set-CsOnlinePstnUsage -identity Global -Usage @{Add="<usage name>"}

After creating Online PSTN usage use the command "**(Get-CsOnlinePstnUsage).usage**" to view the online pstn usage created. Example is shown below.

```

PS C:\WINDOWS\system32> (Get-CsOnlinePstnUsage).usage
US and Canada
Test
CCE
Non E.164
ThinkTel
sbc3
sbc4

```

Figure 39 - Microsoft Teams - Online PSTN usage reference

4.4.6 Configure Online Voice Route

Use the below command to add a new online Voice Route.

```

New-CsOnlineVoiceRoute -Identity "<Route name>" -NumberPattern ".*"
-OnlinePstnGatewayList "<SBCFQDN>" -Priority 1 -OnlinePstnUsages "<PSTN usage
name>"}

```

```
PS C:\WINDOWS\system32> Get-CsOnlineVoiceRoute -Identity sbc4

Identity           : sbc4
Priority            : 5
Description        :
NumberPattern      : .*
OnlinePstnUsages   : {sbc4}
OnlinePstnGatewayList : {sbc4.tekvizionlabs.com}
Name               : sbc4
```

Figure 40 - Microsoft Teams - Online PSTN Voice Route reference

4.4.7 Configure Online Voice Route Policy

Create a new online Voice Routing Policy using the below command.

```
New-CsOnlineVoiceRoutingPolicy "<policy name>" -OnlinePstnUsages "<pstn usage name>"
```

```
PS C:\WINDOWS\system32> Get-CsOnlineVoiceRoutingPolicy

Identity           : Tag:sbc4
OnlinePstnUsages   : {sbc4}
Description        :
RouteType          : BYOT
```

Figure 41 - Microsoft Teams - Online Voice Route Policy

4.4.8 Configure Online Voice Route Policy to user

Assign a online Voice Routing Policy to user using the below command.

```
Grant-CsOnlineVoiceRoutingPolicy -Identity "<Teams User>" -PolicyName "<PSTN Usage>"
```

```
> Grant-CsOnlineVoiceRoutingPolicy -Identity "crestronteam5" -PolicyName "sbc4"
```

Figure 42 - Microsoft Teams - Online Voice Route Policy to User

4.4.9 Configure Tenant Dial Plan

Tenant dial plan added to provision custom dial plan to user. Example is shown below

```
New-CsTenantDialPlan -Identity <dial plan name> -Description "For Extension Calling"
```

```
> Get-CsTenantDialPlan -Identity crestron

Identity           : Tag:crestron
Description        : For Extention Dialing
```

```
NormalizationRules :  
{Description=crestron;Pattern=^(.*)$;Translation=$1;Name=crestron;IsInternalExtension=False}  
ExternalAccessPrefix :  
SimpleName : crestron  
OptimizeDeviceDialing : False
```

Figure 43 - Microsoft Teams – Configure Tenant Dial Plan

4.4.10 Create Normalization Rule

Create a new Voice Normalization Rule using the below command.

```
$rule1 = New-CsVoiceNormalizationRule -Parent Global -Description "description" -  
Pattern '^(.*)$' -Translation '$1' -Name <dial plan name> -IsInternalExtension $false  
-InMemory
```

```
> $rule1 = New-CsVoiceNormalizationRule -Parent Global -Description "crestron" -Pattern '^(.*)$' -Translation '$1' -Name crestron -IsInternalExtension $false -InMemory
```

Figure 44 - Microsoft Teams – Normalization Rule

4.4.11 Associate Normalization rule to tenant dial plan

Associate the Voice Normalization Rule to tenant dial plan created earlier using the below command.

```
Set-CsTenantDialPlan -Identity <dial plan name> -NormalizationRules  
@{add=$rule1}
```

```
> Set-CsTenantDialPlan -Identity crestron -NormalizationRules @{add=$rule1}
```

Figure 45 - Microsoft Teams – Normalization Rule to tenant dial plan

4.4.12 Associate tenant Dial plan to user

Assign the Tenant dial plan to the user using below command.

```
Grant-CsTenantDialPlan -identity <username> -PolicyName <dial plan name>
```

```
> Grant-CsTenantDialPlan -identity crestronteam5 -PolicyName crestron
```

Figure 46 - Microsoft Teams – tenant dial plan to user

4.4.13 Calling Line Identity Policy

Calling Line Identity Policy is used to present/restrict users Caller ID.

```
New-CsCallingLineIdentity -Identity anonymous_policy -Description "clid restricted" -CallingIDSubstitute Anonymous -EnableUserOverride $true
```

Use the command **Get-CsCallingLineIdentity** to view the Calling Line Identity policy created.

```
PS C:\WINDOWS\system32> Get-CsCallingLineIdentity -Identity anonymous_policy

Identity           : Tag:Anonymous_policy
Description        : clid restricted
EnableUserOverride : True
ServiceNumber     :
CallingIDSubstitute : Anonymous
BlockIncomingPstnCallerID : False
```

Figure 47 – Privacy Policy

Associate the policy created above to the users using the below command.

```
Grant-CsCallingLineIdentity -Identity "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.

4.5 AudioCodes VE SBC Configuration

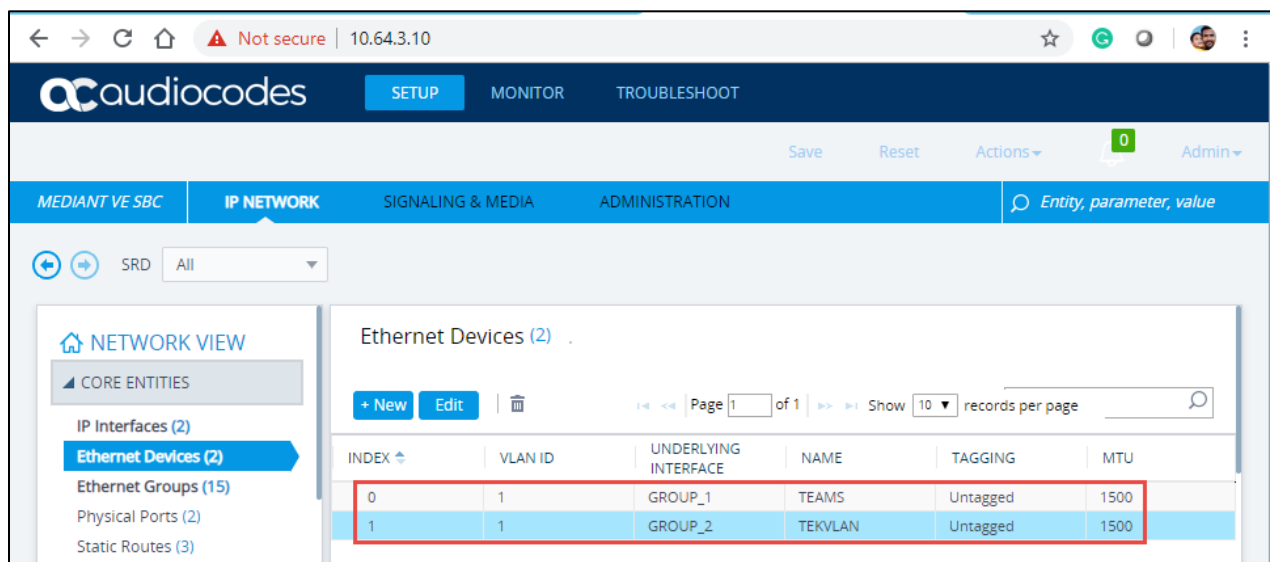
4.5.1 General

AudioCodes Mediant 1000 SBC was used as it can meet the requirements and support the enhancements for Microsoft Teams Direct Routing. PSTN Gateway SIP Trunk is a non-registering trunk that connects to E-SBC using UDP. Cisco UBE SIP Trunk that connects to E-SBC using UDP. The SBC must be configured to perform back to back User Agent (B2BUA) functionality. For the B2BUA configuration, it is recommended that Physical interfaces are connected with two different customer WAN networks.

4.5.2 Configure VLANs

To configure VLANs, navigate to **IP Network tab** → **Core Entities menu** → **Ethernet Devices**

Add an entry with VLAN ID for underlying Teams and CenturyLink Voice Complete® interface Groups configured.



The screenshot shows the AudioCodes Mediant VE SBC configuration interface. The browser address bar displays "10.64.3.10". The interface includes a navigation menu with "IP NETWORK" selected. The "Ethernet Devices (2)" section is active, showing a table with two entries. The table has columns for INDEX, VLAN ID, UNDERLYING INTERFACE, NAME, TAGGING, and MTU. The first entry (INDEX 0) is for VLAN ID 1, UNDERLYING INTERFACE GROUP_1, NAME TEAMS, TAGGING Untagged, and MTU 1500. The second entry (INDEX 1) is for VLAN ID 1, UNDERLYING INTERFACE GROUP_2, NAME TEKVLAN, TAGGING Untagged, and MTU 1500. A red box highlights the second entry.

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

Figure 48 – Ethernet Devices

4.5.3 Configure IP Network Interfaces

To configure IP Network interfaces, navigate to the **IP Network tab->Core Entities menu ->Interfaces Table**.

Configure the WAN and LAN interface (interface towards Teams and LAN) as shown below:

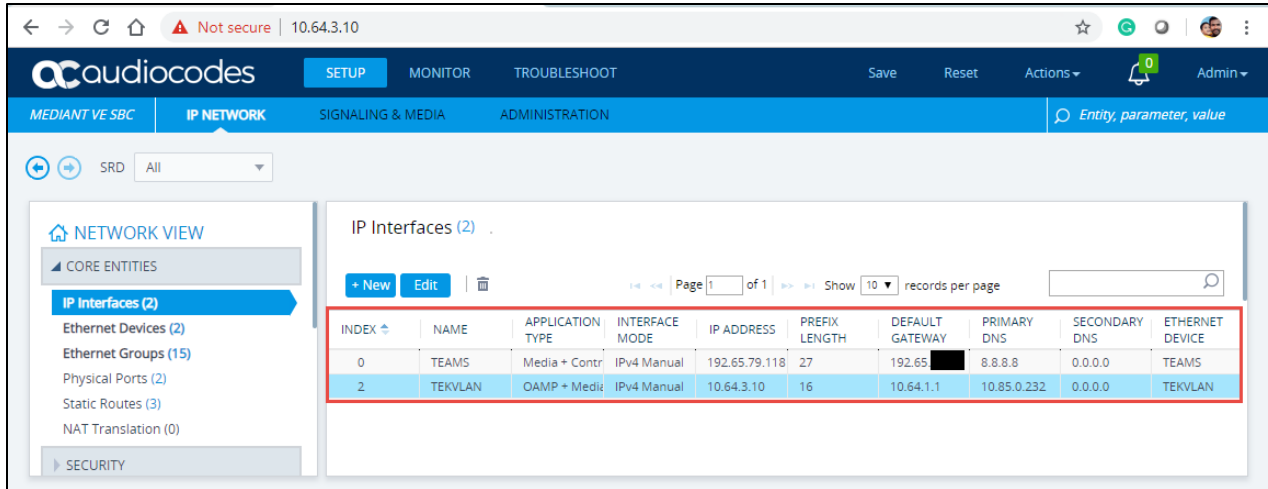


Figure 49 – IP interface Devices

IP interface TEAMS

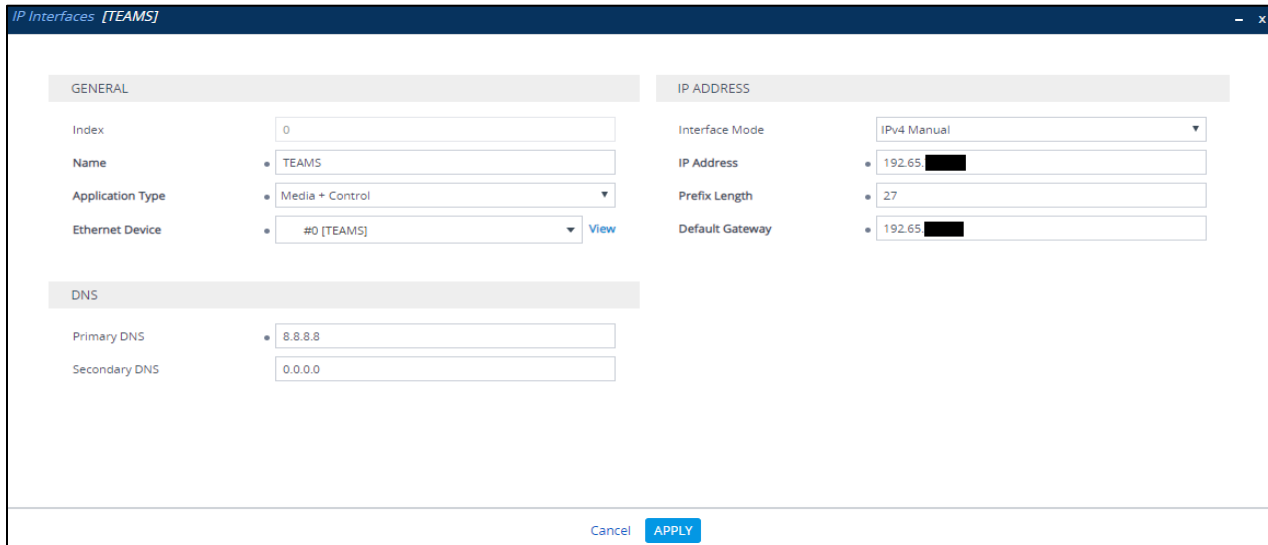


Figure 50 – IP interface Devices

IP Interfaces – TEKVLAN

The screenshot shows the configuration interface for an IP interface named 'TEKVLAN'. The 'GENERAL' tab is active, showing the following settings:

- Index: 2
- Name: TEKVLAN
- Application Type: OAMP + Media + Control
- Ethernet Device: #1 [TEKVLAN]

The 'IP ADDRESS' section is also visible, showing:

- Interface Mode: IPv4 Manual
- IP Address: 10.64.3.10
- Prefix Length: 16
- Default Gateway: 10.64.1.1

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

Figure 51 – IP interface Devices

4.5.4 Configure DNS SRV Records

Microsoft Teams Direct Routing uses primary, secondary and tertiary datacenters for call routing.

AudioCodes Mediant 1000 SBC uses internal SRV records to resolve the FQDN of these datacenters.

To configure DNS SRV records, navigate to the **IP Network tab** → **DNS menu** → **Internal SRV Table**.

Configure a DNS SRV records as shown below and associate it under proxy set towards Teams

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

Figure 52 – DNS SRV Records

4.5.5 Configure SRTP

By default, SRTP is disabled.

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

The screenshot shows the Audiocodes web interface for configuring Media Security. The left sidebar shows a navigation menu with 'Media Security' selected. The main content area is titled 'Media Security' and is divided into 'GENERAL' and 'AUTHENTICATION & ENCRYPTION' sections. In the 'GENERAL' section, the 'Media Security' dropdown is set to 'Enable' (highlighted with a red box), and 'Media Security Behavior' is set to 'Preferable - Single me' (also highlighted with a red box). Other settings include 'Offered SRTP Cipher Suites' set to 'All', 'Aria Protocol Support' set to 'Disable', and 'MASTER KEY IDENTIFIER' settings for 'Master Key Identifier (MKI) Size' (0) and 'Symmetric MKI' (Disable). The 'AUTHENTICATION & ENCRYPTION' section has several dropdowns: 'Authentication On Transmitted RTP Packets' (Active), 'Encryption On Transmitted RTP Packets' (Active), 'Encryption On Transmitted RTCP Packets' (Active), 'SRTP Tunneling Authentication for RTP' (Disable), and 'SRTP Tunneling Authentication for RTCP' (Disable). At the bottom right, there are 'Cancel' and 'APPLY' buttons.

Figure 53 – Media Security

4.5.6 Configure TLS contexts

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

- AffirmTrust
- AddTrust External CA Root
- Baltimore CyberTrust Root
- Buypass
- Cybertrust
- Class 3 Public Primary Certification Authority
- Comodo Secure Root CA
- Deutsche Telekom
- DigiCert Global Root CA
- DigiCert High Assurance EV Root CA
- Entrust
- GlobalSign
- Go Daddy
- GeoTrust
- Verisign, Inc.
- Starfield
- Symantec Enterprise Mobile Root for Microsoft
- SwissSign
- Thawte Timestamping CA
- Trustwave
- TeliaSonera
- T-Systems International GmbH (Deutsche Telekom)
- QuoVadis

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

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

#1[Teams] Edit

GENERAL		OCSP	
Name	• Teams	OCSP Server	Disable
TLS Version	• TLSv1.2	Primary OCSP Server	0.0.0.0
DTLS Version	Any	Secondary OCSP Ser...	0.0.0.0
Cipher Server	RC4:AE5128	OCSP Port	2560
Cipher Client	DEFAULT	OCSP Default Respo...	Reject
Strict Certificate Exte...	Disable		
DH key Size	• 2048		

Certificate Information >> Change Certificate >> Trusted Root Certificates >>

Figure 54 – Teams TLS

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

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

4.5.7 Configure Media Realms

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

Media Realms (2)

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 55 – Media Realms

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

The screenshot shows the 'Media Realms [TEAMS]' configuration window. It is divided into two main sections: 'GENERAL' and 'QUALITY OF EXPERIENCE'. In the 'GENERAL' section, the 'Name' field is highlighted with a red box and contains the text 'TEAMS'. Other fields include 'Index' (0), 'Topology Location' (Down), 'IPv4 Interface Name' (#0 [TEAMS]), 'Port Range Start' (50000), 'Number Of Media Session Legs' (100), 'Port Range End' (50499), and 'Default Media Realm' (No). The 'QUALITY OF EXPERIENCE' section has 'QoE Profile' and 'Bandwidth Profile' both set to '..'. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 56 – Teams

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

The screenshot shows the 'Media Realms [TEKVLAN]' configuration window. It is divided into two main sections: 'GENERAL' and 'QUALITY OF EXPERIENCE'. In the 'GENERAL' section, the 'Name' field is highlighted with a red box and contains the text 'TEKVLAN'. Other fields include 'Index' (1), 'Topology Location' (Up), 'IPv4 Interface Name' (#2 [TEKVLAN]), 'Port Range Start' (7000), 'Number Of Media Session Legs' (100), 'Port Range End' (7499), and 'Default Media Realm' (No). The 'QUALITY OF EXPERIENCE' section has 'QoE Profile' and 'Bandwidth Profile' both set to '..'. At the bottom, there are 'Cancel' and 'APPLY' buttons.

Figure 57 – LAN LAB

4.5.8 Configure the SRD

To configure Signaling Routing Domains (SRD), navigate to **Signaling & Media tab** → **Core Entities menu** → **SRD Table**

Here the default SRD is used as shown below.

The screenshot shows the configuration page for the default SRD, titled "#0[DefaultSRD]". It features two main sections: GENERAL and REGISTRATION. The GENERAL section includes fields for Name (DefaultSRD), Sharing Policy (Shared), SBC Operation (B2BUA), SBC Routing Policy (# [Default_SBCRoutingPolicy] with a View link), Used By Routine (Not Used), Dial Plan (# [-] with a View link), and CAC Profile (# [-] with a View link). The REGISTRATION section includes Max. Number of... (-1), User Security M... (Accept All), and Enable Un-Auth... (Enable). An Edit button is located in the top right corner.

GENERAL	
Name	• DefaultSRD
Sharing Policy	Shared
SBC Operation ...	B2BUA
SBC Routing Pol...	• # [Default_SBCRoutingPolicy] View
Used By Routin...	Not Used
Dial Plan	# [-] View
CAC Profile	# [-] View

REGISTRATION	
Max. Number o...	-1
User Security M...	Accept All
Enable Un-Auth...	Enable

Figure 58 – Default SRD

4.5.9 Configure SIP Signaling Interface

For this test, three external SIP interfaces were configured on the SBC. To configure SIP interfaces, navigate to **Signaling & Media** tab → **Core Entities** menu → **SIP Interface Table**.

Configure a SIP interface for the WAN (towards Teams) as shown below.

The screenshot shows the configuration page for a SIP interface titled "SIP Interfaces [TEAMS]". The SRD is set to "#0 [DefaultSRD]". The configuration is divided into GENERAL, MEDIA, and SECURITY sections. Several fields are highlighted with red boxes: Name (TEAMS), Network Interface (#0 [TEAMS]), Media Realm (#0 [TEAMS]), Network Interface (#0 [TEAMS]), Application Type (SBC), UDP Port (5060), TLS Context Name (#1 [Teams]), and TLS Mutual Authentication (Enable).

GENERAL	
Index	0
Name	• TEAMS
Topology Location	Down
Network Interface	• #0 [TEAMS] View
Application Type	SBC
UDP Port	5060
TCP Port	• 0
TLS Port	5061
Additional UDP Ports	
Additional UDP Ports Mode	Always Open
Encapsulating Protocol	No encapsulation

MEDIA	
Media Realm	• #0 [TEAMS] View
Direct Media	Disable

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

Figure 59 – Teams

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

CLASSIFICATION

Classification Failure Response Type: 0
Pre-classification Manipulation Set ID: -1
Call Setup Rules Set ID: -1

Cancel APPLY

Figure 60 – Teams

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

SIP Interfaces [PSTNGW] SRD: #0 [DefaultSRD]

GENERAL

Index: 1
Name: PSTNGW
Topology Location: Up
Network Interface: #2 [TEKVLAN] 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

MEDIA

Media Realm: #1 [TEKVLAN] View
Direct Media: Disable

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 61 – 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 62 – 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 63 – 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 64 – Cisco

4.5.10 Configure Proxy Sets

The Proxy Set defines the destination address (IP address or FQDN) of the SIP entity server.

For the test, three Proxy Sets were configured: one for the Microsoft Teams, PSTN Gateway and another one towards Cisco UBE. These proxy sets were later associated with IP Groups.

To configure Proxy Sets, navigate to **Signaling & Media** tab → **Core Entities** menu → **Proxy Sets Table**

Configure a Proxy Set for the Teams as shown below.

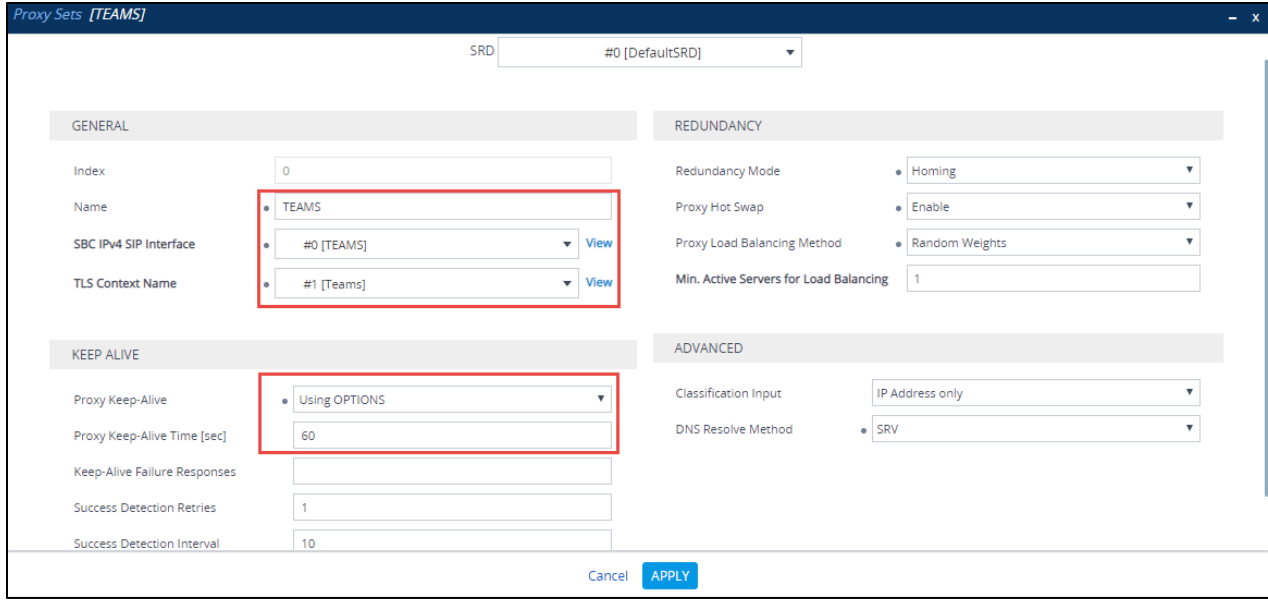


Figure 65 – Teams

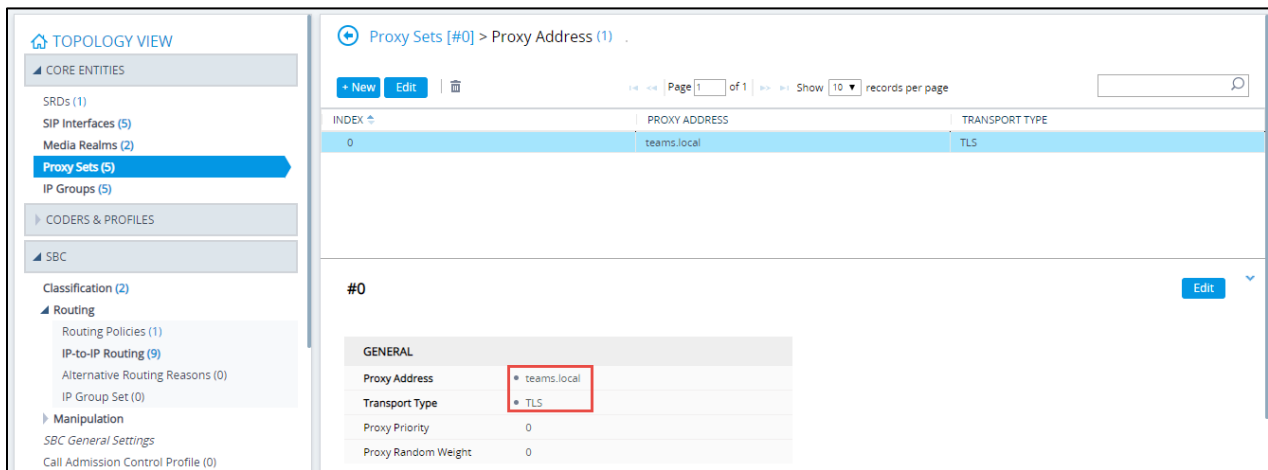


Figure 66 – Teams

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

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 67 – PSTN Gateway

Keep-Alive Failure Responses	
Success Detection Retries	1
Success Detection Interval	10
Cancel APPLY	

Figure 68 – PSTN Gateway

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

Proxy Sets [CISCO]

SRD #0 [DefaultSRD]

GENERAL		REDUNDANCY	
Index	2	Redundancy Mode	
Name	CISCO	Proxy Hot Swap	Disable
SBC IPv4 SIP Interface	#2 [CISCO] View	Proxy Load Balancing Method	Disable
TLS Context Name	.. View	Min. Active Servers for Load Balancing	1
KEEP ALIVE		ADVANCED	
Proxy Keep-Alive	Disable	Classification Input	IP Address only
Proxy Keep-Alive Time [sec]	60	DNS Resolve Method	
Keep-Alive Failure Responses			
Success Detection Retries	1		

Figure 69 – Cisco UCM

Success Detection Interval	<input type="text" value="10"/>
Failure Detection Retransmissions	<input type="text" value="-1"/>
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>	

Figure 70 – Cisco UCM

4.5.11 Configure IP Groups

The IP Group represents an IP entity on the network with which the SBC communicates. For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting the source and destination of the call.

For the test, IP Groups were configured for the following IP entities:

- Microsoft Teams
- PSTN Gateway – SIP Trunk
- Cisco – SIP Trunk

To configure IP groups, navigate to **Signaling & Media** tab → **Core Entities** menu → **IP Group Table**

Configure an IP Group for Microsoft Teams as shown below

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) and Outbound Message Manipulation Set (2). The 'QUALITY OF EXPERIENCE' section includes QoS Profile and Bandwidth Profile. The 'SBC REGISTRATION AND AUTHENTICATION' section is partially visible at the bottom.

Figure 71 – IP Group – Teams – Contd.

IP Groups [TEAMS]

Proxy Set Connectivity: Connected

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:

SBC GENERAL

Classify By Proxy Set: Disable

SBC Operation Mode: Not Configured

SBC Client Forking Mode: Sequential

CAC Profile: .. View

ADVANCED

Local Host Name: sbc4.tekvizionlabs.com

UII Format: Disable

Always Use Src Address: No

GW GROUP STATUS

GW Group Registered IP Address:

GW Group Registered Status: Not Registered

Figure 72 – IP Group – 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 73 – 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	MESSAGE MANIPULATION	
Type	Server	Inbound Message Manipulation Set	0
Proxy Set	#1 [PSTNGW] View	Outbound Message Manipulation Set	3
IP Profile	#2 [PSTNGW_Profile] View	Message Manipulation User-Defined String 1	
Media Realm	#1 [TEKVLAN] View	Message Manipulation User-Defined String 2	
Contact User		Proxy Keep-Alive using IP Group settings	Enable
SIP Group Name	10.64.1.72	SBC REGISTRATION AND AUTHENTICATION	
Created By Routing Server	No		
Used By Routing Server	Not Used		

Figure 74 – IP Group – PSTN – Contd.

IP Groups [PSTNGW] SRD #0 [DefaultSRD]

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

Figure 75 – IP Group – PSTN – 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	#0 [default] View
Keep Original Call-ID	No
Dial Plan	-- View
Call Setup Rules Set ID	-1
Tags	<input type="text"/>

Cancel [APPLY](#)

Figure 76 – IP Group

Configure an IP Group for Cisco UCM as shown below

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"/>	
Type: <input type="text" value="Server"/>	
Proxy Set: <input type="text" value="#2 [CISCO]"/> View	MESSAGE MANIPULATION
IP Profile: <input type="text" value="#3 [CISCO_Profile]"/> View	Inbound Message Manipulation Set: <input type="text" value="4"/>
Media Realm: <input type="text" value="#1 [TEKVLAN]"/> View	Outbound Message Manipulation Set: <input type="text" value="5"/>
Contact User: <input type="text"/>	Message Manipulation User-Defined String 1: <input type="text"/>
SIP Group Name: <input type="text" value="10.70.69.70"/>	Message Manipulation User-Defined String 2: <input type="text"/>
Created By Routing Server: <input type="text" value="No"/>	Proxy Keep-Alive using IP Group settings: <input type="text" value="Disable"/>

Figure 77 – IP Group – Cisco – Contd.

The screenshot shows the configuration interface for IP Groups in Cisco. It is divided into several sections:

- General Settings:**
 - 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: .. (with a View link)
- ADVANCED:**
 - Local Host Name: (empty text field)
 - 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: (empty text field)
 - SBC Server Authentication Type: According to Global Parameter
 - OAuth HTTP Service: .. (with a View link)
 - Username: Admin
 - Password:
- GW GROUP STATUS:**
 - GW Group Registered IP Address: (empty text field)

Figure 78 – IP Group – Cisco – Contd.

The screenshot shows the SBC ADVANCED configuration page. It includes the following settings:

- SBC ADVANCED:**
 - Source URI Input: (empty text field)
 - Destination URI Input: (empty text field)
 - SIP Connect: No
 - SBC PSAP Mode: Disable
 - Route Using Request URI Port: Disable
 - DTLS Context: #0 [default] (with a View link)
 - Keep Original Call-ID: No
 - Dial Plan: .. (with a View link)
 - Call Setup Rules Set ID: -1
 - Tags: (empty text field)
- GW Group Registered Status:** Not Registered

At the bottom of the page, there are buttons for "Cancel" and "APPLY".

Figure 79 – IP Group

4.5.12 Configure IP Profile

The IP Profile defines a set of call capabilities relating to signaling.

For this test, IP Profiles were configured for the following IP entities:

- Microsoft Teams
- PSTN Gateway – SIP Trunk
- Cisco – SIP Trunk

To configure IP profiles, navigate to **Signaling & Media** tab → **Coders and Profiles** → **IP Profile Settings**.
Click **Add**.

Configure the IP Profile for the Microsoft Teams as shown below.

The screenshot shows the configuration for the IP Profile 'TEAMS_Profile'. The 'GENERAL' tab is active, showing the profile name and index. The 'MEDIA SECURITY' tab shows SBC Media Security Mode set to SRTP, Symmetric MKI set to Disable, MKI Size set to 1, SBC Enforce MKI Size set to Don't enforce, SBC Media Security Method set to SDES, Reset SRTP Upon Re-key set to Disable, and Generate SRTP Keys Mode set to Always. The 'SBC SIGNALING' tab shows PRACK Mode set to Optional, P-Asserted-Identity Header Mode, Diversion Header Mode, and History-Info Header Mode all set to As Is, Session Expires Mode set to Transparent, Remote Update Support, Remote re-INVITE, and Remote Delayed Offer Support all set to Not Supported, Remote Representation Mode set to According to Operation Mode, Keep Incoming Via Headers, Keep Incoming Routing Headers, and Keep User-Agent Header all set to According to Operation Mode.

Figure 80 – IP Profile – Teams – Contd.

The screenshot shows the configuration for the IP Profile 'TEAMS_Profile' in the SBC EARLY MEDIA, SBC REGISTRATION, and SBC FORWARD AND TRANSFER tabs. The 'SBC EARLY MEDIA' tab shows Remote Early Media, Remote Multiple 18x, and Remote Early Media RTP Detection Mode all set to Supported, Remote Early Media Response Type set to Transparent, Remote Multiple Early Dialogs set to According to Operation Mode, Remote Multiple Answers Mode set to Disable, Remote RFC 3960 Support set to Not Supported, Remote Can Play Ringback set to No, and Generate RTP set to None. The 'SBC REGISTRATION' tab shows Handle X-Detect set to No, ISUP Body Handling set to Transparent, ISUP Variant set to Itu92, Max Call Duration [min] set to 0, User Registration Time set to 0, NAT UDP Registration Time set to -1, and NAT TCP Registration Time set to -1. The 'SBC FORWARD AND TRANSFER' tab shows Remote REFER Mode set to Regular, Remote Replaces Mode set to Standard, and Play RBT To Transferee set to Yes.

Figure 81 – IP Profile – Teams – Contd.

IP Profiles [TEAMS_Profile]

SBC MEDIA

Mediation Mode: RTP Mediation

Extension Coders Group: #0 [AudioCodersGroups_0]

Allowed Audio Coders: #0 [AllowedAudioCodersGroup_TEAMS]

Allowed Coders Mode: Preference

Allowed Video Coders: ..

Allowed Media Types:

Direct Media Tag:

RFC 2833 Mode: As Is

RFC 2833 DTMF Payload Type: 101

Alternative DTMF Method: As Is

Send Multiple DTMF Methods: Disable

Adapt RFC2833 BW to Voice coder BW: Disabled

SDP Ptime Answer: Preferred Value

Remote 3xx Mode: Handle Locally

SBC HOLD

Remote Hold Format: Inactive

Reliable Held Tone Source: Yes

Play Held Tone: No

SBC FAX

Fax Coders Group: ..

Fax Mode: As Is

Fax Offer Mode: All coders

Fax Answer Mode: Single coder

Remote Renegotiate on Fax Detection: Transparent

Fax Rerouting Mode: Disable

Figure 82 – 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 83 – 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 84 – IP Profile – Teams – Contd.

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

IP Profiles [PSTNGW_Profile]

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: 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 85 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW_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 86 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW_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	101	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	Preferred Value		

Figure 87 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW_Profile]

Preferred PTime	20	
Use Silence Suppression	Add	
RTP Redundancy Mode	As Is	
RTCP Mode	Generate Always	
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 88 – IP Profile – PSTN Gateway – Contd.

IP Profiles [PSTNGW_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 89 – IP Profile – PSTN Gateway

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

IP Profiles [CISCO_Profile]

GENERAL		SBC SIGNALING	
Index	3	PRACK Mode	Transparent
Name	CISCO_Profile	P-Asserted-Identity Header Mode	As Is
Created by Routing Server	No	Diversion Header Mode	As Is
MEDIA SECURITY		History-Info Header Mode	As Is
SBC Media Security Mode	RTP	Session Expires Mode	Supported
Symmetric MKI	Disable	Remote Update Support	Supported Only After Connect
MKI Size	0	Remote re-INVITE	Supported only with SDP
SBC Enforce MKI Size	Don't enforce	Remote Delayed Offer Support	Not Supported
SBC Media Security Method	SDES	Remote Representation Mode	According to Operation Mode
Reset SRTP Upon Re-key	Disable	Keep Incoming Via Headers	According to Operation Mode
Generate SRTP Keys Mode	Only If Required	Keep Incoming Routing Headers	According to Operation Mode
		Keep User-Agent Header	According to Operation Mode

Figure 90 – 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 91 – IP Profile – Cisco – Contd.

IP Profiles [CISCO_Profile] - x

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

Figure 92 – IP Profile – Cisco – Contd.

IP Profiles [CISCO_Profile] - x

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 93 – IP Profile – Cisco – Contd.

IP Profiles [CISCO_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 94 – IP Profile – Cisco

4.5.13 Configure SIP Definition and General Setting

The screenshot below captures the configuration of the **SIP Definitions General Settings** that were used during the test for the successful test execution

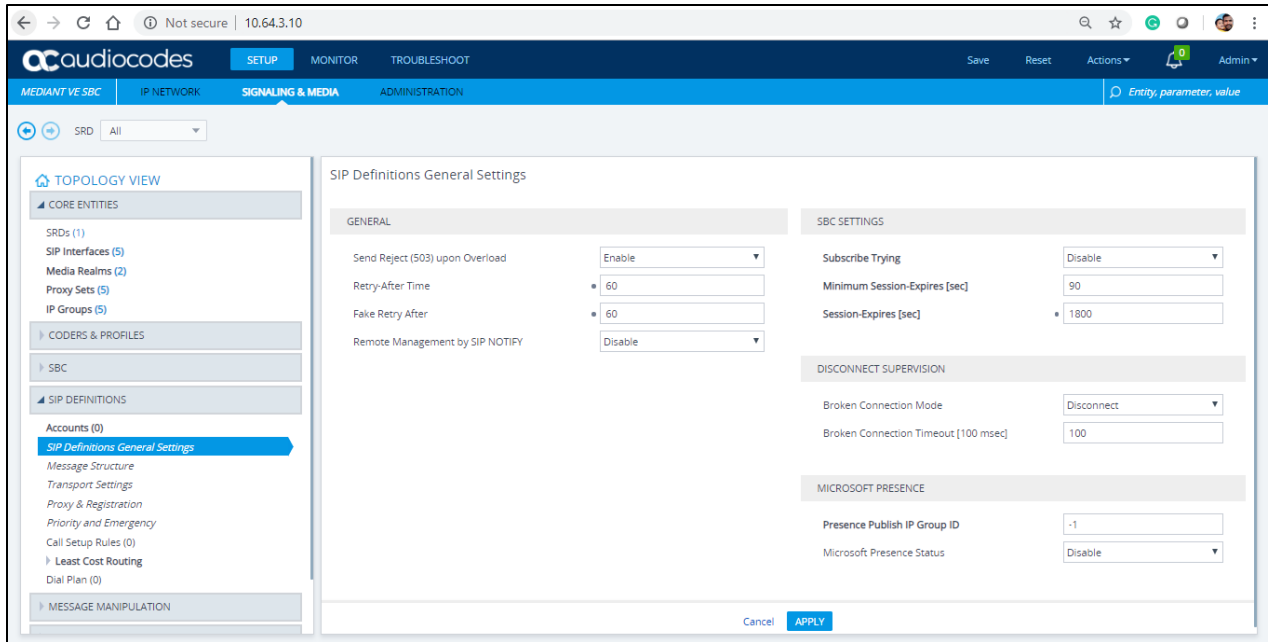


Figure 95 – SIP Definition

4.5.14 Configure SBC General Settings

The screenshot below captures the configuration of the **SBC General Parameters** that was used during the test for the successful test execution.

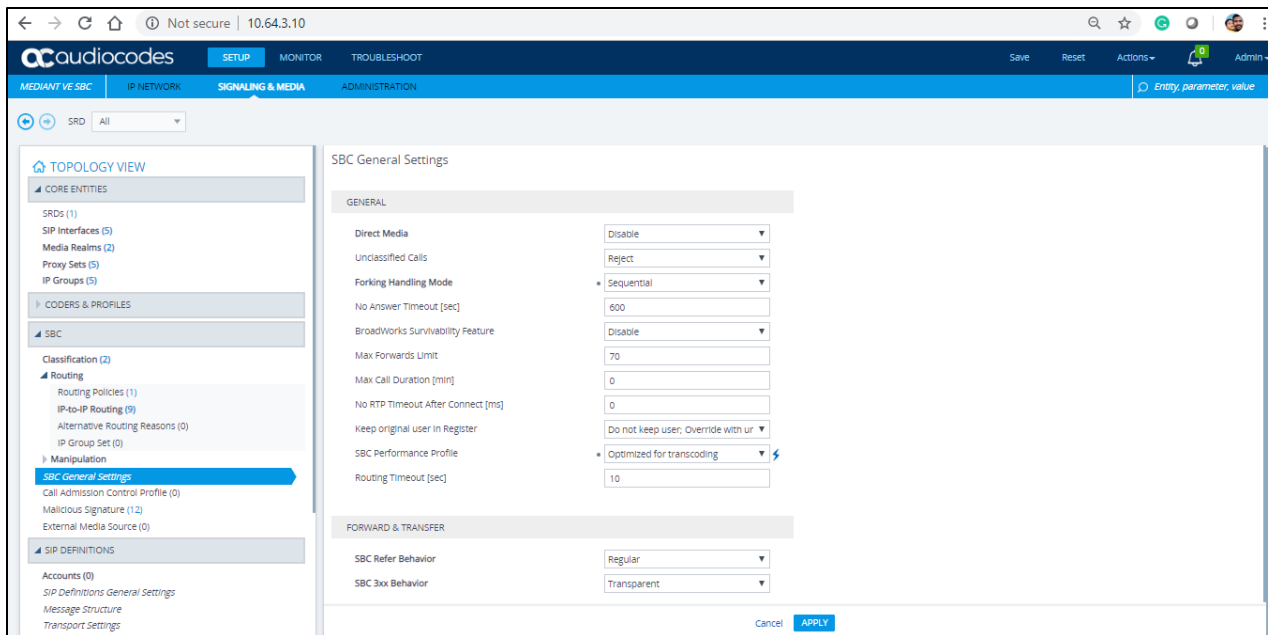


Figure 96 – SBC General Setting – Contd.

4.5.15 Configure IP-to-IP Routing Rules

This section describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected. The routing rules use the configured IP Groups to denote the source and destination of the call.

For the test, the following IP-To-IP Routing rules were configured to route calls between the Teams and CenturyLink

- Calls from Teams to PSTN Gateway
- Calls from PSTN Gateway to Teams
- Calls from Teams to Cisco
- Calls from Cisco to Teams

To configure IP-to-IP routing rules, navigate to **Signaling & Media** tab → **SBC** menu → **Routing** → **IP-to-IP Routing Table**. Click **Add**.

Calls from Teams to PSTN Gateway

The screenshot shows the configuration window for an IP-to-IP Routing rule titled "IP-to-IP Routing [TEAMS -> PSTN]". The window is divided into two main sections: GENERAL and ACTION.

GENERAL Section:

- Index:** 4
- Name:** TEAMS -> PSTN (highlighted with a red box)
- Alternative Route Options:** Route Row
- MATCH Section:**
 - Source IP Group:** #0 [TEAMS] (highlighted with a red box)
 - Request Type:** All
 - Source Username Pattern:** *
 - Source Host:** *
 - Source Tag:** (empty)

ACTION Section:

- Destination Type:** IP Group
- Destination IP Group:** #1 [PSTNGW] (highlighted with a red box)
- Destination SIP Interface:** #1 [PSTNGW] (highlighted with a red box)
- Destination Address:** (empty)
- Destination Port:** 0
- Destination Transport Type:** (empty)
- IP Group Set:** ..
- Call Setup Rules Set ID:** -1
- Group Policy:** Sequential
- Cost Group:** ..

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

Figure 97 – 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 98 – Teams to PSTN

Calls from PSTN Gateway to Teams

IP-to-IP Routing [PSTNGW_to_TEAMS] - x

Routing Policy #0 [Default_SBCRoutingPolicy]

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

Figure 99 – 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 100 – 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 101 –Teams to Cisco – Contd.

Destination Username Pattern: 6	Routing Tag Name: default
Destination Host: *	Internal Action: Editor
Destination Tag:	
Message Condition: ..	
Call Trigger: Any	
ReRoute IP Group: Any	
<input type="button" value="Cancel"/> <input type="button" value="APPLY"/>	

Figure 102 –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 103 –Cisco to Teams – Contd.

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

Figure 104 –Cisco to Teams

4.5.16 IP Group

IP Group – Teams

IP Groups [TEAMS] - x

SRD

GENERAL		QUALITY OF EXPERIENCE	
Index	<input type="text" value="0"/>	QoE Profile	<input type="text" value="--"/> View
Name	<input type="text" value="TEAMS"/>	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="1"/>
Proxy Set	<input type="text" value="#0 [TEAMS]"/> View	Outbound Message Manipulation Set	<input type="text" value="2"/>
IP Profile	<input type="text" value="#1 [TEAMS_Profile]"/> View	Message Manipulation User-Defined String 1	<input type="text"/>
Media Realm	<input type="text" value="#0 [TEAMS]"/> 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="Enable"/>
SIP Group Name	<input type="text" value="sbc4.tekvizionlabs.com"/>		
Created By Routing Server	<input type="text" value="No"/>		

Figure 105 – IP Groups Teams – Contd.

IP Groups [TEAMS] - x

Used By Routing Server	<input type="text" value="Not Used"/>	SBC REGISTRATION AND AUTHENTICATION	
Proxy Set Connectivity	<input type="text" value="Connected"/>	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="Disable"/>	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" value="sbc4.tekvizionlabs.com"/>	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 106 – IP Groups Teams – 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="#1 [Teams]"/> 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 107 – IP Groups Teams

IP Group – PSTN Gateway

IP Groups [PSTNGW] - x

SRD

GENERAL		QUALITY OF EXPERIENCE	
Index	<input type="text" value="1"/>	QoE Profile	<input type="text" value="--"/> View
Name	<input type="text" value="PSTNGW"/>	Bandwidth Profile	<input type="text" value="--"/> View
Topology Location	<input type="text" value="Up"/>		
Type	<input type="text" value="Server"/>		
Proxy Set	<input type="text" value="#1 [PSTNGW]"/> View		
IP Profile	<input type="text" value="#2 [PSTNGW_Profile]"/> View		
Media Realm	<input type="text" value="#1 [TEKVLAN]"/> View		
Contact User			
SIP Group Name	<input type="text" value="10.64.1.72"/>		
Created By Routing Server	<input type="text" value="No"/>		

MESSAGE MANIPULATION	
Inbound Message Manipulation Set	<input type="text" value="0"/>
Outbound Message Manipulation Set	<input type="text" value="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	<input type="text" value="Enable"/>

Figure 108 – IP Groups PSTN – Contd.

IP Groups [PSTNGW] - x

Used By Routing Server	<input type="text" value="Not Used"/>	SBC REGISTRATION AND AUTHENTICATION	
Proxy Set Connectivity	<input type="text" value="Connected"/>	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 109 – IP Groups PSTN – 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 110 – IP Groups PSTN

IP Group – Cisco

IP Groups [CISCO]

SRD: #0 [DefaultSRD]

GENERAL	QUALITY OF EXPERIENCE
Index: 2	QoE Profile: -- View
Name: CISCO	Bandwidth Profile: -- View
Topology Location: Down	
Type: Server	
Proxy Set: #2 [CISCO] View	
IP Profile: #3 [CISCO_Profile] View	
Media Realm: #1 [TEKVLAN] View	
Contact User:	
SIP Group Name: 10.70.69.70	
Created By Routing Server: No	

MESSAGE MANIPULATION
Inbound Message Manipulation Set: 4
Outbound Message Manipulation Set: 5
Message Manipulation User-Defined String 1:
Message Manipulation User-Defined String 2:
Proxy Keep-Alive using IP Group settings: Disable

Figure 111 – IP Groups Cisco – Contd.

IP Groups [CISCO]

Used By Routing Server: Not Used

Proxy Set Connectivity: NA

SBC GENERAL	SBC REGISTRATION AND AUTHENTICATION
Classify By Proxy Set: Enable	Max. Number of Registered Users: -1
SBC Operation Mode: Not Configured	Registration Mode: User Initiates Registration
SBC Client Forking Mode: Sequential	User Stickiness: Disable
CAC Profile: -- View	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:

ADVANCED	GW GROUP STATUS
Local Host Name:	GW Group Registered IP Address:
UI Format: Disable	
Always Use Src Address: No	

Figure 112 – IP Groups Cisco – Contd.

SBC ADVANCED

Source URI Input:	
Destination URI Input:	
SIP Connect:	No
SBC PSAP Mode:	Disable
Route Using Request URI Port:	Disable
DTLS Context:	#0 [default] View
Keep Original Call-ID:	No
Dial Plan:	-- View
Call Setup Rules Set ID:	-1
Tags:	

Cancel APPLY

Figure 113 – IP Groups Cisco

4.5.17 Message Manipulation

A Message Manipulation rule defines a manipulation sequence for SIP messages. SIP message manipulation enables the normalization of SIP messaging fields between communicating network segments. SIP message manipulations can also be implemented to resolve incompatibilities between SIP devices inside the enterprise network.

Each Message Manipulation rule is configured with a Manipulation Set ID. Groups (sets) of Message Manipulation rules can be created by assigning each of the relevant Message Manipulation rules to the same Manipulation Set ID.

The SIP message manipulation feature supports the following:

- Manipulation on SIP message type (Method, Request/Response, and Response type)
- Addition of new SIP headers
- Removal of SIP headers
- Modification of SIP header components such as values, header values (e.g., URI value of the P-Asserted-Identity header can be copied to the From header), call's parameter values
- Deletion of SIP body (e.g., if a message body is not supported at the destination network this body is removed)
- Translating one SIP response code to another
- Topology hiding (generally present in SIP headers such as Via, Record Route, Route and Service-Route).
- Configurable identity hiding (information related to identity of subscribers, for example P-Asserted-Identity, Referred-By, Identity and Identity-Info)

To configure Message Manipulation rules, navigate to **Signaling & Media** tab → **Message Manipulations** menu → **Message Manipulations**.

Click **Add** and populate the required fields in the screen that appears as below:

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

Figure 114 – SIP Message Manipulation

Then click **Add** again, once the parameters have been configured.

For this test, the following message manipulations were configured and assigned to one manipulation set ID.

Manipulation set ID = 1: Manipulation from Teams

Manipulation set ID = 2: Manipulation to Teams

Manipulation set ID = 3: Manipulation to PSTN

Manipulation set ID = 4: Manipulation from Cisco

Manipulation set ID = 5: Manipulation to Cisco

Manipulation from Teams

- To Remove "Privacy" header: To Remove Privacy Header from Teams

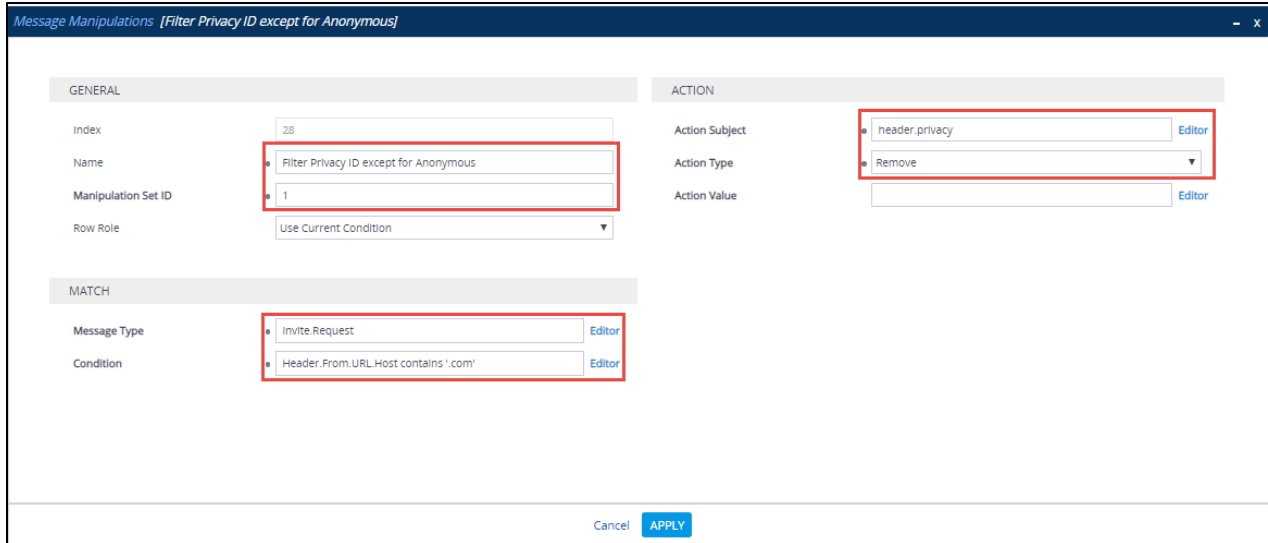


Figure 115 – SIP Message Manipulation - Privacy

Manipulation to Teams

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

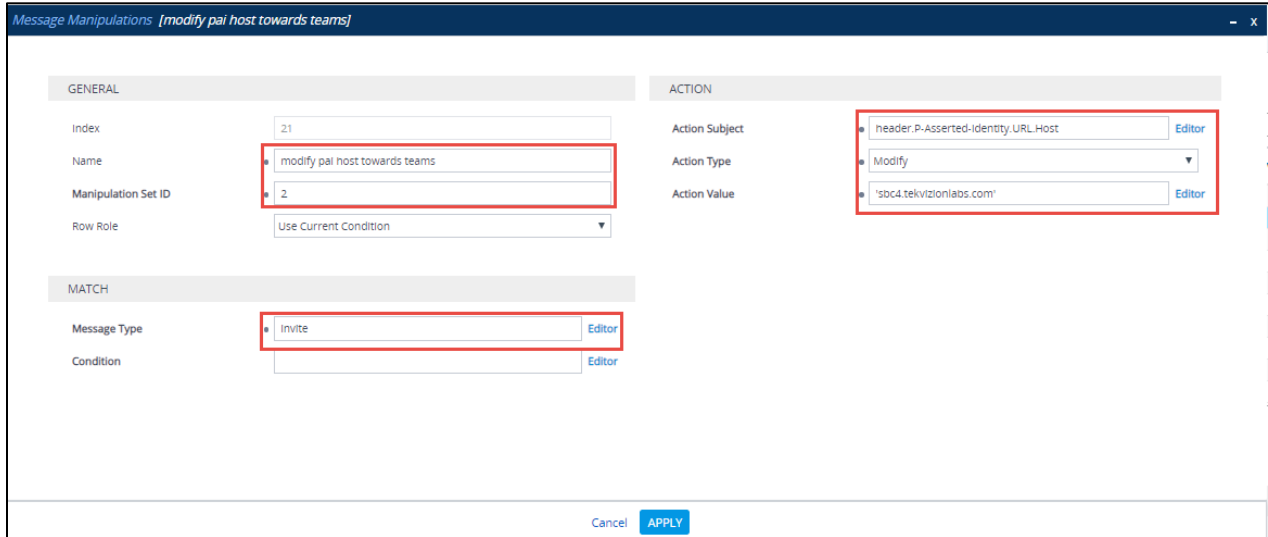


Figure 116 – SIP Message Manipulation - PAI

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

Message Manipulations [modify to towards teams]

GENERAL

Index: 19

Name: modify to towards: teams

Manipulation Set ID: 2

Row Role: Use Current Condition

MATCH

Message Type: Invite request

Condition:

ACTION

Action Subject: header.to.url.host

Action Type: Modify

Action Value: *sip.pstnhub.microsoft.com*

Buttons: Cancel, APPLY

Figure 117 – 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 118 – SIP Message Manipulation - From

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

Figure 119 – SIP Message Manipulation - Contact

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

Figure 120 – SIP Message Manipulation - From

Manipulation to PSTN

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

Message Manipulations [towards PSTNGW TO]

GENERAL		ACTION	
Index	3	Action Subject	header.to.url.host Editor
Name	towards PSTNGW TO	Action Type	Modify
Manipulation Set ID	3	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 121 – SIP Message Manipulation – To

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

Message Manipulations [Towards PSTNGW FROM]

GENERAL		ACTION	
Index	4	Action Subject	Header.From.URL.host Editor
Name	Towards PSTNGW FROM	Action Type	Modify
Manipulation Set ID	3	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 122 – SIP Message Manipulation – From

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

The screenshot shows the configuration for a SIP message manipulation. The window title is "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 123 – 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 124 – SIP Message Manipulation – From

Manipulation to Cisco

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

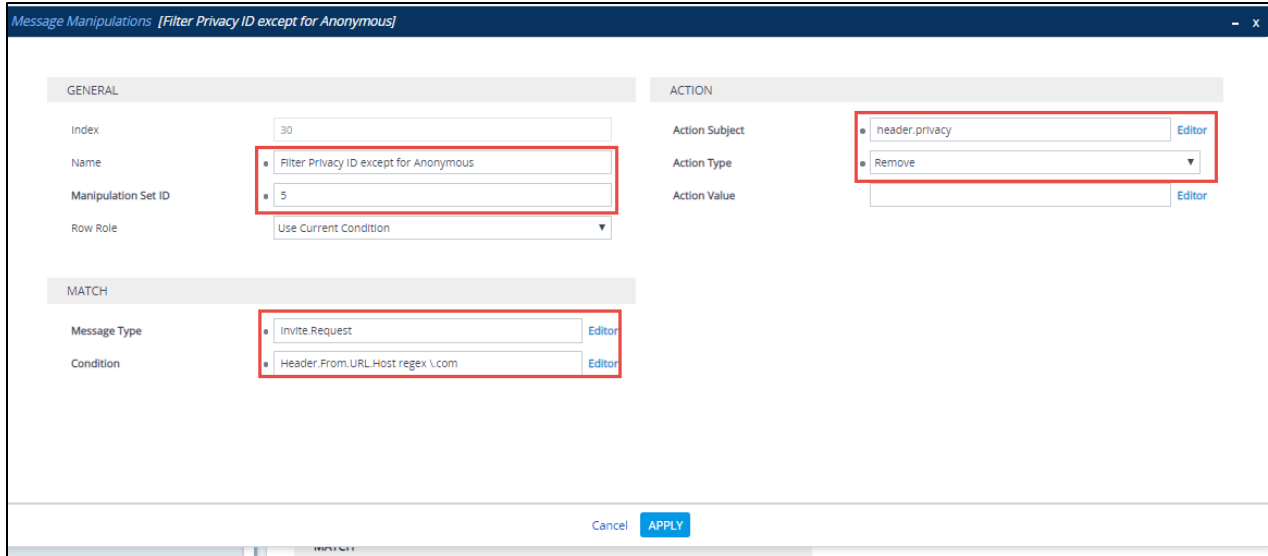


Figure 125 – SIP Message Manipulation – Privacy

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

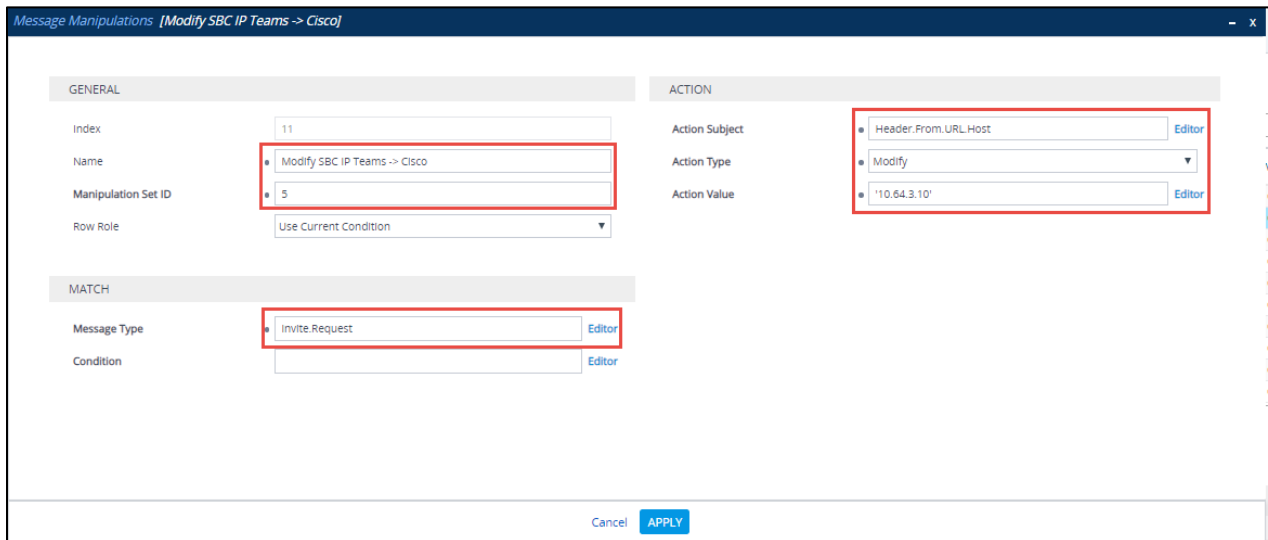


Figure 126 – SIP Message Manipulation – From

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

Message Manipulations [Referred-By Teams -> Cisco]

GENERAL

Index: 12

Name: Referred-By Teams -> Cisco

Manipulation Set ID: 5

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 127 – 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 128 – SIP Message Manipulation – From

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

Message Manipulations [To header Teams -> Cisco]

GENERAL		ACTION	
Index	14	Action Subject	header.to.url.host Editor
Name	To header Teams -> Cisco	Action Type	Modify
Manipulation Set ID	5	Action Value	'10.64.3.10' Editor
Row Role	Use Current Condition		
MATCH			
Message Type	Options Editor		
Condition	Param.Message.Address.dst.SIPInterface=='2' Editor		

Cancel APPLY

Figure 129 – SIP Message Manipulation – to

4.6 Cisco UBE Configuration

```
Crestron_Teams#sh run
Building configuration...
Current configuration : 6699 bytes
!
!
version 15.7
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname Crestron_Teams
!
boot-start-marker
boot system tftp c2900-universalk9-mz.SPA.157-3.M1.bin 255.255.255.255
boot-end-marker

!
enable secret 4 sKPgCY/XPea3wk8xoeSWo7UGFaNVwzXDEyXWhuDjeLk
enable password 7 071B244778580354471C
!
!
voice service voip
no ip address trusted authenticate
address-hiding1
mode border-element license capacity 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.

```
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
!  
interface Embedded-Service-Engine0/0  
no ip address  
shutdown  
!  
interface GigabitEthernet0/06  
description LAN to CUCM  
ip address 10.64.4.182 255.255.0.0  
duplex auto  
speed auto  
!  
interface GigabitEthernet0/17  
description WAN to AudioCodes  
ip address 10.70.69.70 255.255.255.0  
duplex auto
```

⁶ LAN interface pointing to CUCM

⁷ WAN interface pointing to AudioCodes

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

```
!  
!  
mgcp behavior rsip-range tgcp-only  
mgcp behavior comedia-role none  
mgcp behavior comedia-check-media-src disable  
mgcp behavior comedia-sdp-force disable  
!  
mgcp profile default  
!  
!  
!  
!  
dial-peer voice 10 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#

4.7 Cisco UCM Configuration

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

4.7.1 Version

Cisco UCM version

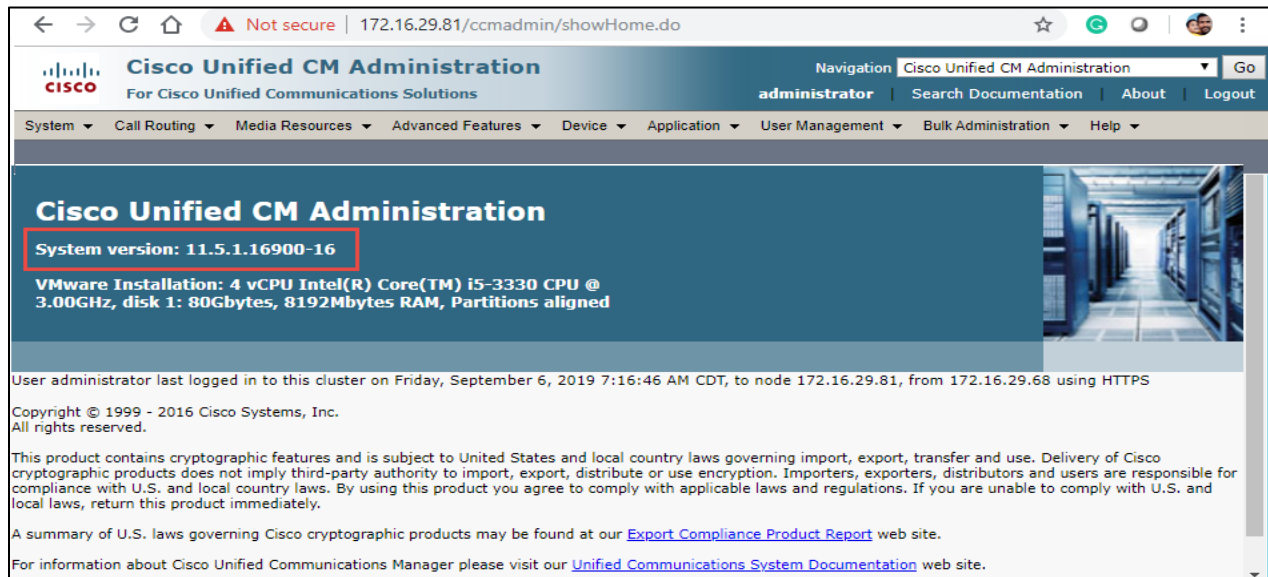


Figure 130 – Cisco UCM Version

4.7.2 Cisco UCM Audio Codec Preference List

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

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

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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* Factory Default lowloss crestron

Description* Lossy Codec List

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 131 – Audio Codec Preference List

4.7.3 Cisco UCM Region Configuration

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

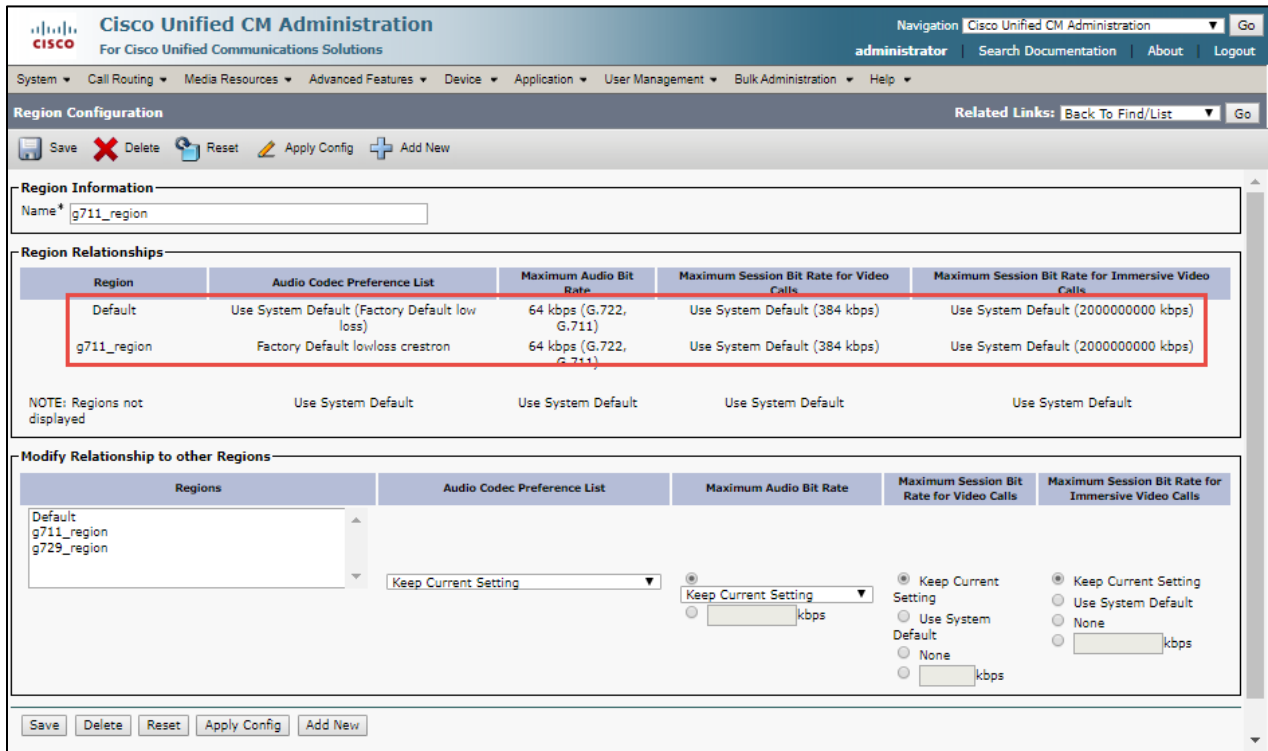


Figure 132 – Cisco UCM Region

4.7.4 Cisco UCM Device Pool

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

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

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

Save Delete Copy Reset Apply Config Add New

Device Pool Information
Device Pool: Crestron_DevicePool (13 members)**

Device Pool Settings

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

Intercompany Media Services Enrolled Group < None >

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* g711_region

Media Resource Group List MRGL

Location < None >

Network Locale < None >

SRST Reference* Disable

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Wireless LAN Profile Group < None > [View Details](#)

Local Route Group Settings

Standard Local Route Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Calling Party Transformation CSS < None >

Called Party Transformation CSS < None >

Geolocation Configuration

Figure 133 – Cisco UCM Device Pool – Contd.

Device Pool Configuration

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 134 – Cisco UCM Device Pool

4.7.5 Cisco UCM Annunciator Configuration

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

Set Name* = ANN_2.

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

Set Device Pool* = Crestron_DevicePool

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

Annunciator Configuration | Related Links: Back To Find/List | Go

Save | Reset | Apply Config

Status
Status: Ready

Annunciator Information
Registration: Registered with Cisco Unified Communications Manager 172.16.29.81
IPv4 Address: 172.16.29.81
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

Save | Reset | Apply Config

* - indicates required item.

Figure 135 – Cisco UCM Annunciator

4.7.6 Cisco UCM Conference Bridge

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

Set Name* = CFB_2

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

Set Device Pool* = Crestron_DevicePool

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

Conference Bridge Configuration | Related Links: Back To Find/List | Go

Save | Reset | Apply Config

Status
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

Save | Reset | Apply Config

* - indicates required item.

Figure 136 – Cisco UCM Conference Bridge

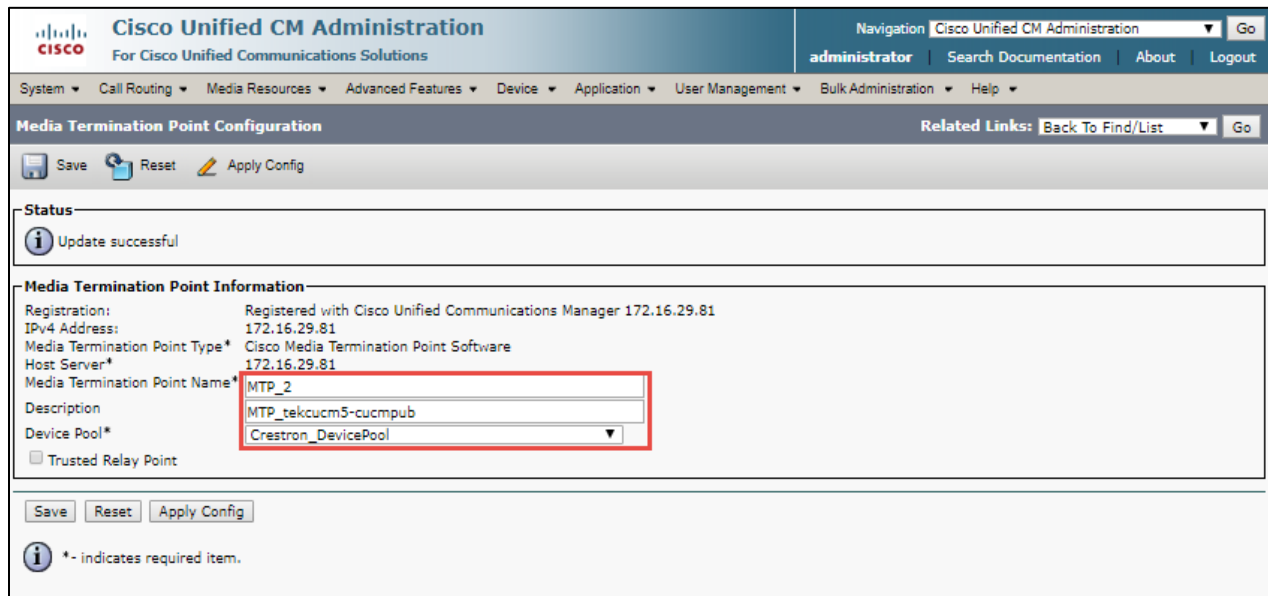
4.7.7 Cisco UCM MTP

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

Set Name* = MTP_2

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

Set Device Pool* = Crestron_DevicePool



The screenshot displays the Cisco Unified CM Administration web interface for configuring a Media Termination Point (MTP). The page title is "Media Termination Point Configuration". At the top, there is a navigation bar with "Navigation Cisco Unified CM Administration" and a "Go" button. Below this, there is a breadcrumb trail: "System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help". The main content area shows a "Status" section with an information icon and the text "Update successful". Below this is the "Media Termination Point Information" section, which contains the following fields:

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

There is a checkbox for "Trusted Relay Point" which is currently unchecked. At the bottom of the form, there are three buttons: "Save", "Reset", and "Apply Config". Below the buttons, there is an information icon and the text "*- indicates required item."

Figure 137 – Cisco UCM MTP

4.7.8 Cisco Media Resource Group

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

Set Name* = MRG

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

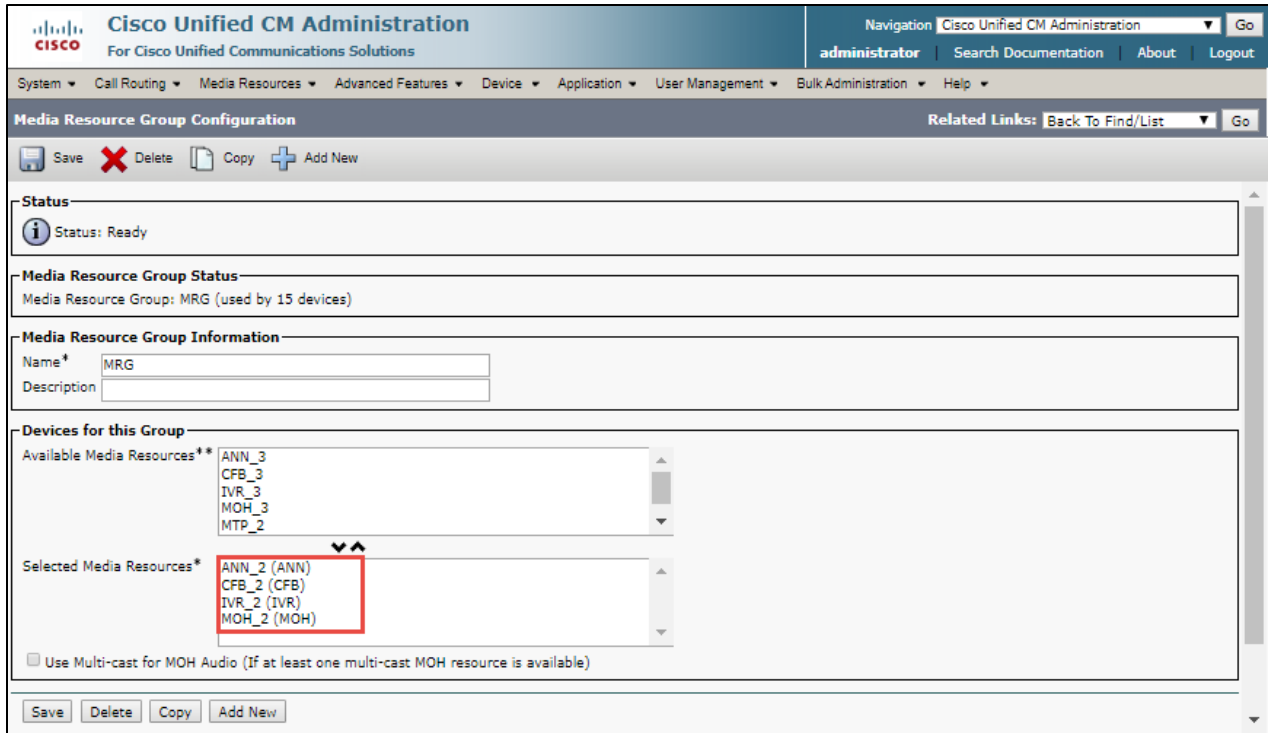


Figure 138 – Cisco UCM MTP

4.7.9 Cisco Media Resource Group List

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

Set Name* = MRGL

Selected Media Resources Groups = MRG

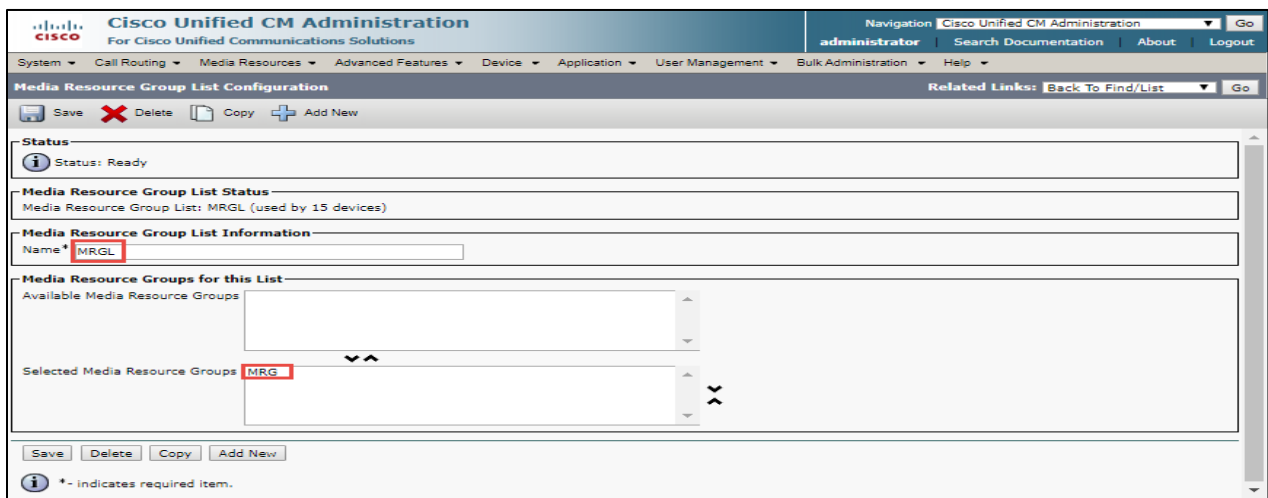


Figure 139 – Cisco UCM MRGL

4.7.10 Cisco UCM SIP Trunk towards Cisco UBE

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

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

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

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

Set **Media Resource Group List** = MRGL

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 current page is "Trunk Configuration" with a "Related Links" section containing "Back To Find/List".

The configuration is for a SIP Trunk with the following details:

- SIP Trunk Status:** Service Status: Full Service, Duration: Time In Full Service: 0 day 0 hour 43 minutes
- Device Information:**
 - 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
 - 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 140 – SIP Trunk – Cisco UBE – Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

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

Save | Delete | Reset | Add New

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

MLPP and Confidential Access Level Information
MLPP Domain: < None >
Confidential Access Mode: < None >
Confidential Access Level: < None >

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

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

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

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

Figure 141 – SIP Trunk – Cisco UBE – Contd.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

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

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

Save | Delete | Reset | Add New

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

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

Connected Party Settings
Connected Party Transformation CSS: < None >
 Use Device Pool Connected Party Transformation CSS

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

Caller Information
Caller ID DN:
Caller Name:
 Maintain Original Caller ID DN and Caller Name in Identity Headers

Figure 142 – SIP Trunk – Cisco UBE – Contd.

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

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

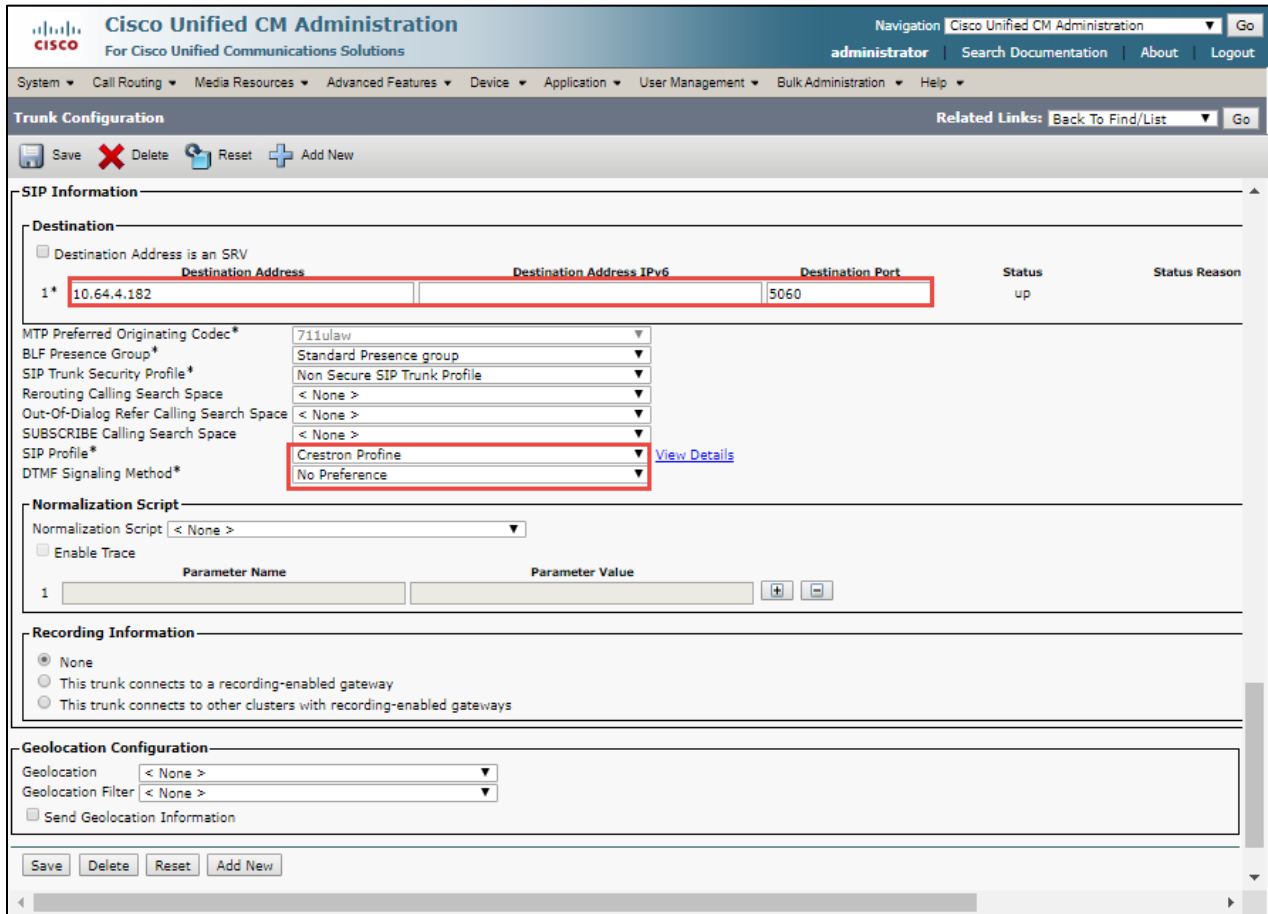


Figure 143 – SIP Trunk – Cisco UBE

4.7.11 Cisco UCM SIP Trunk towards Cisco Unity

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

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

Status
Status: Ready

SIP Trunk Status
Service Status: Full Service
Duration: Time In Full Service: 7 days 1 hour 43 minutes

Device Information

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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default

Figure 144 – 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](#)

Use Trusted Relay Point*

PSTN Access
 Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

MLPP and Confidential Access Level Information

MLPP Domain
 Confidential Access Mode
 Confidential Access Level

Call Routing Information

Remote-Party-Id
 Asserted-Identity
 Asserted-Type*
 SIP Privacy*

Inbound Calls

Significant Digits*
 Connected Line ID Presentation*
 Connected Name Presentation*
 Calling Search Space
 AAR Calling Search Space
 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.

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

Figure 145 – 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.

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 146 – SIP Trunk – Cisco Unity – Contd.

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

Set **SIP Trunk Security Profile*** = Non Secure Sip Trunk Profile.

Set **SIP Profile*** = Standard SIP Profile - OPTIONS. This is used in this example.

Set **DTMF Signaling Method*** = No Preference. This is used in this example.

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](#)

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*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

Figure 147 – SIP Trunk – Cisco Unity

4.8 Cisco Unity Connection (CUC)

4.8.1 Telephony Integration – Phone System

To configure CUC, **navigate to Telephony Integrations → Phone system**

Add New

Set Phone System Name* = Cisco_Crestron. This Name used for this test

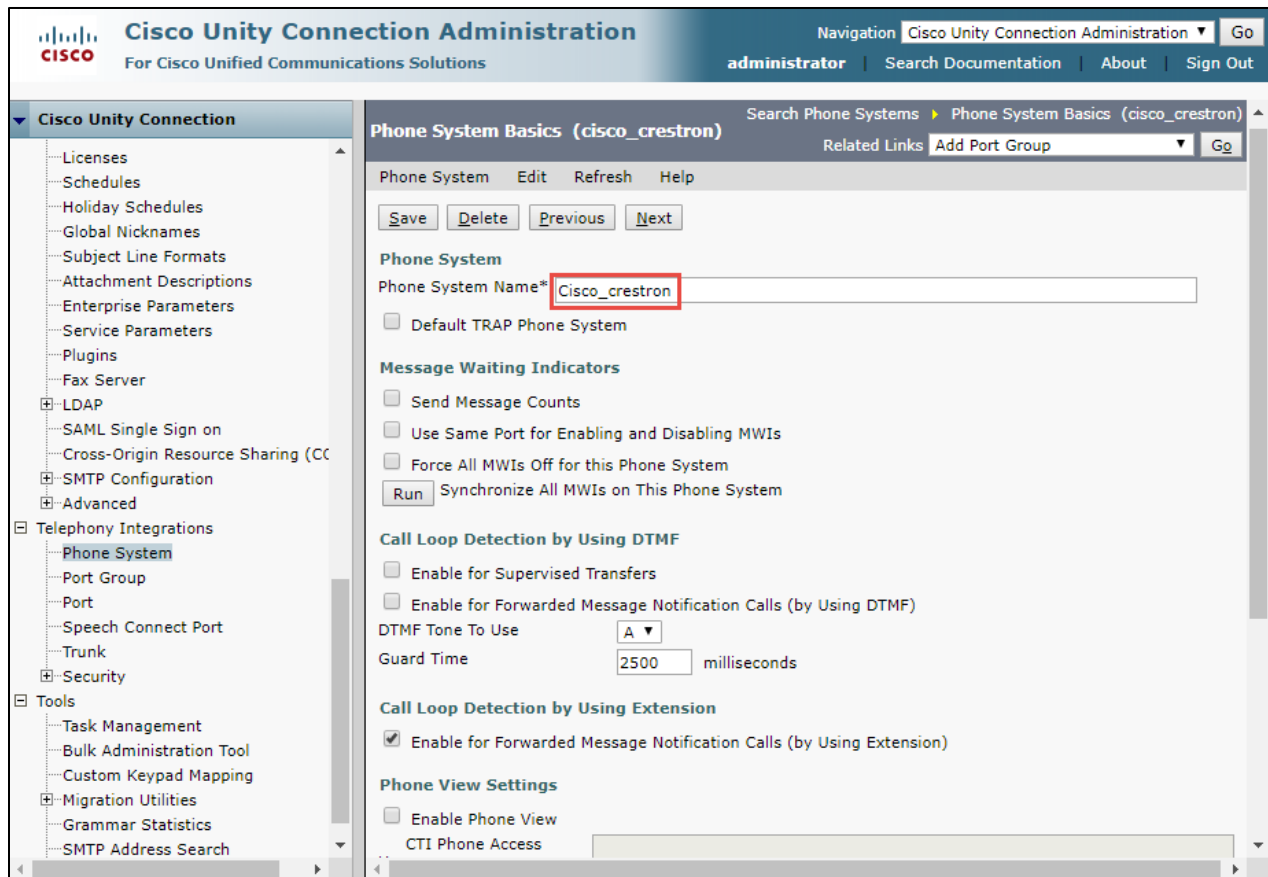


Figure 148 – SIP Trunk – Phone System – Contd.

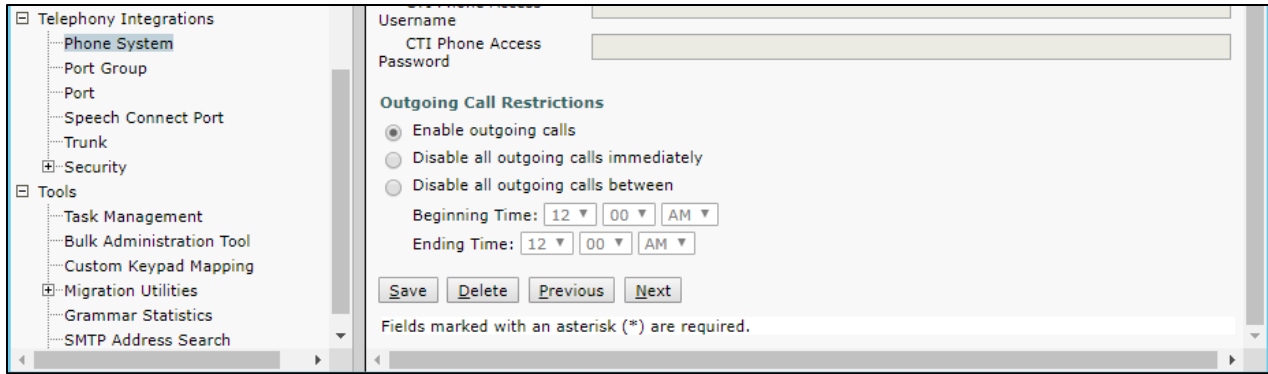


Figure 149 – SIP Trunk – Phone System – Contd.

4.8.2 Phone Group

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

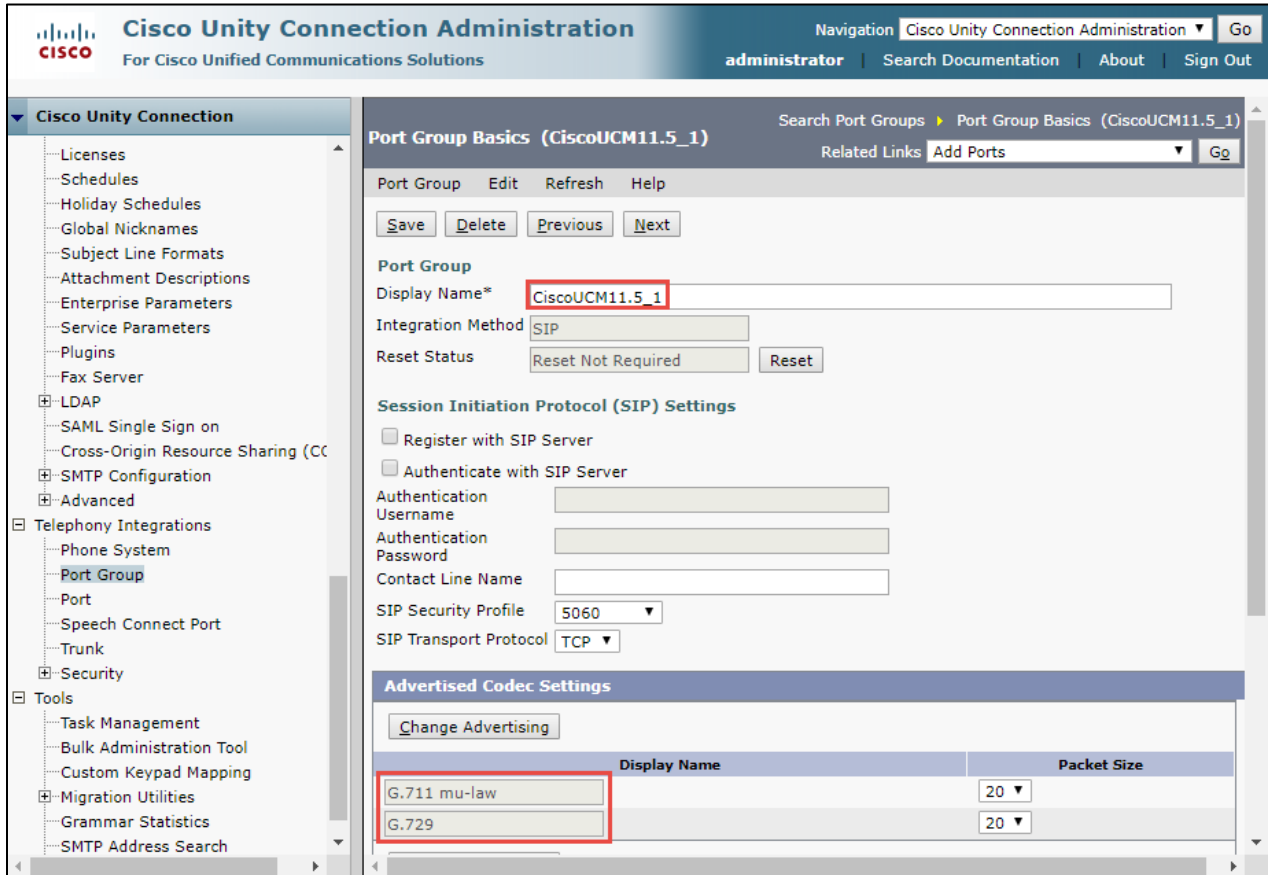


Figure 150 – Phone Group – Contd.

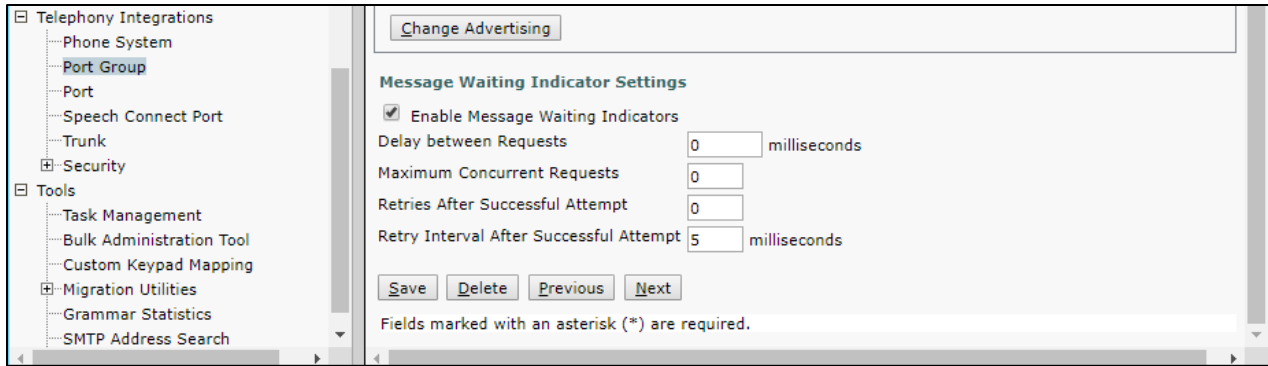


Figure 151 –Phone Group

4.8.3 Port

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

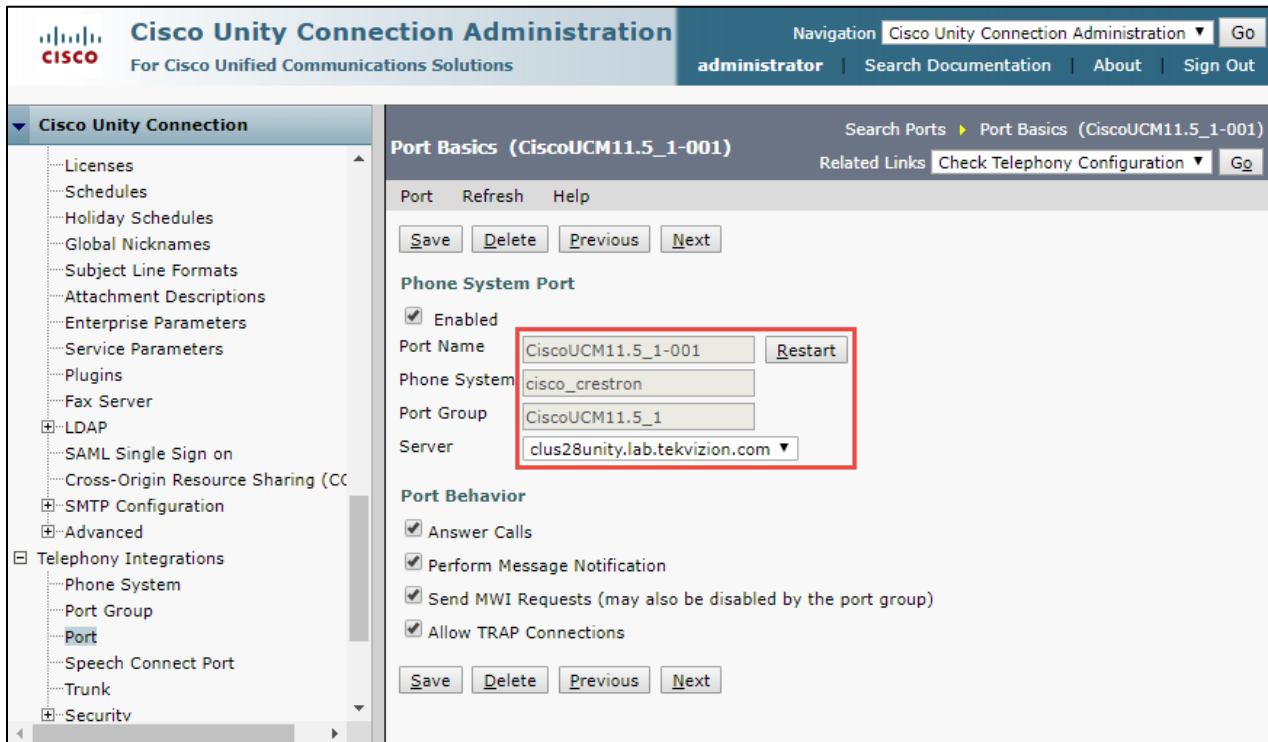


Figure 152 – Port

4.8.4 User

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

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

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

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

All other values are default.

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, and Networking. The main content area is titled 'Edit User Basics (6500)'. It features a header with 'User Edit Refresh Help' and buttons for 'Save', 'Delete', 'Previous', and 'Next'. The form fields are as follows:

Name	Alias*	6500
Name	First Name	CUCME
Name	Last Name	1
Name	Display Name	CUCME 1
Name	SMTP Address	6500@lab.tekvizion.com
Name	Initials	
Name	Title	
Name	Employee ID	
LDAP Integration Status	Integrate with LDAP Directory	<input type="radio"/>
LDAP Integration Status	Do Not Integrate with LDAP Directory	<input checked="" type="radio"/>
Phone	Extension*	6500
Phone	Cross-Server Transfer Extension or URI	
Phone	Outgoing Fax Number	
Phone	Outgoing Fax Server	--- Not Selected ---

Figure 153 – User

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

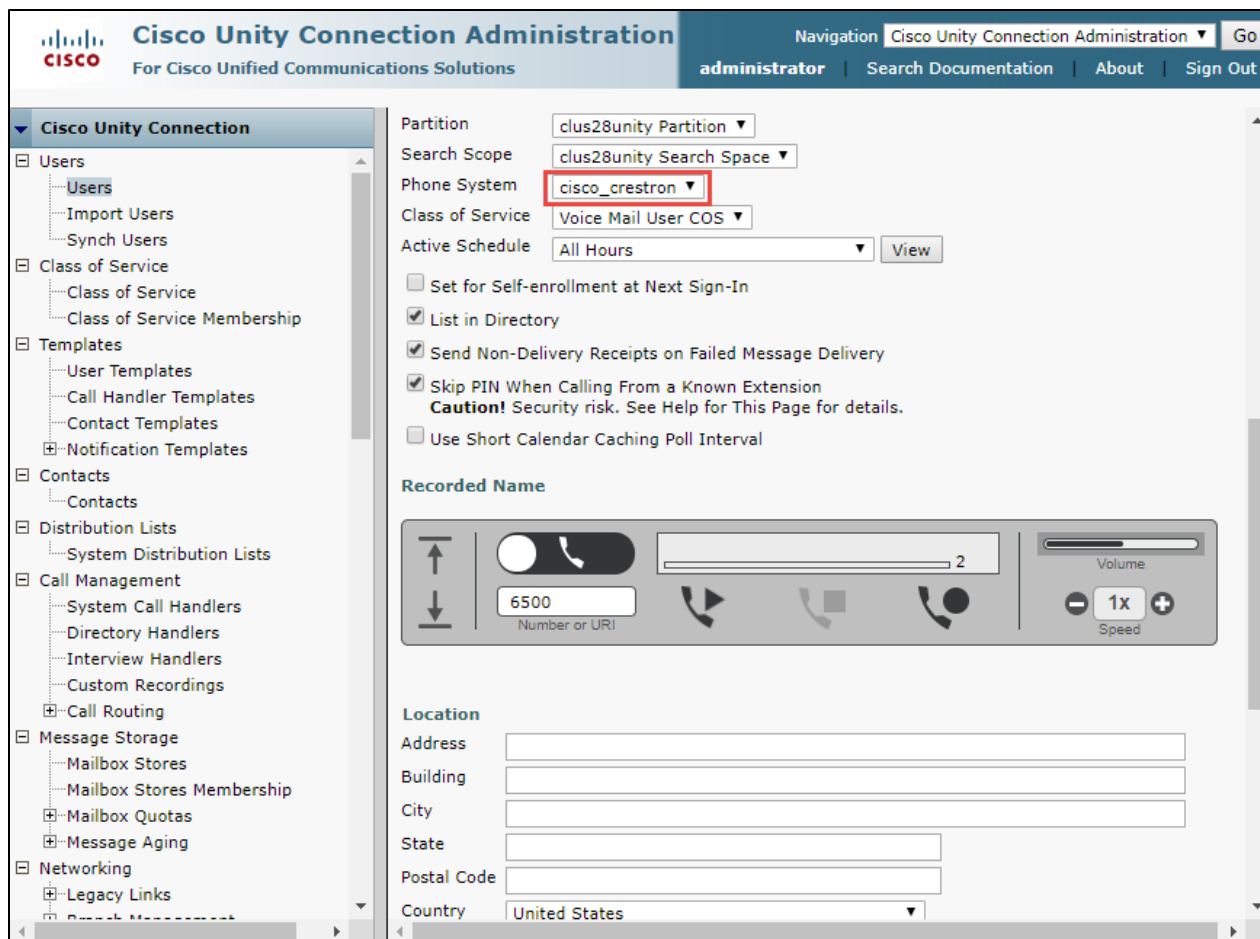


Figure 154 – User

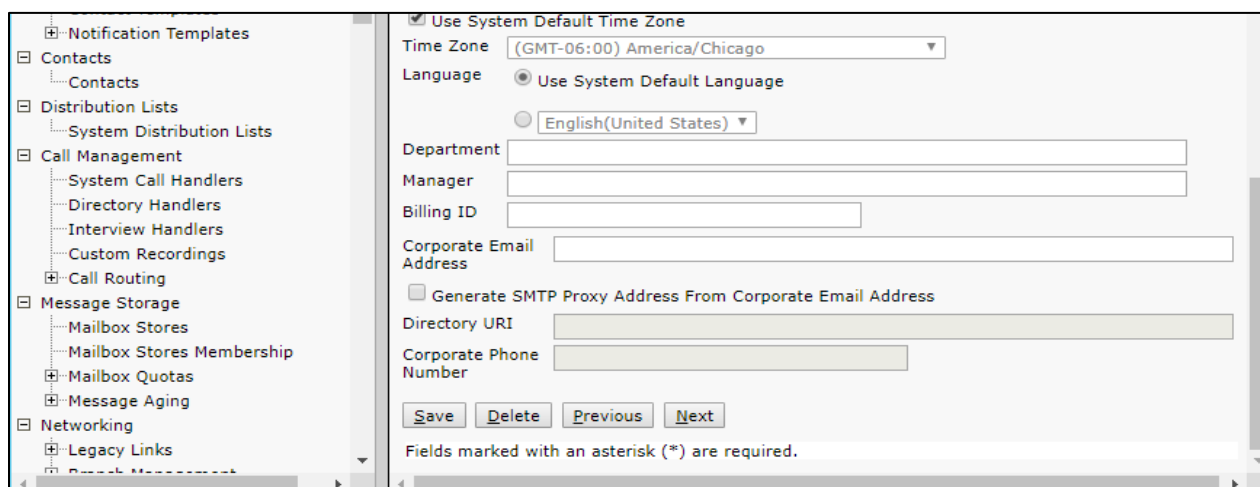


Figure 155 – User

5 Acronyms

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

6 Summary of Tests and Results

External ID	Title	Procedure	Expected Results	Status	Comments
1	Teams user Calls PBX A user	<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 	PASSED	
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 	<ol style="list-style-type: none"> 1. Call is connected with bi-directional audio, voice is clear, 	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		3. PSTN user answers the call 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. PSTN user hangs up the call 9. Verify call is cleared successfully	no echo 2. Call is disconnected		
4	Teams user Calls PBX A user and hangs up before answer	1. Make a voice call from Teams user to PBX A user 2. PBX A user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX A user is ringing 5. PBX A user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	
5	Teams user Calls PBX B user and hangs up before answer	1. Make a voice call from Teams user to PBX B user 2. PBX B user starts ringing 3. Teams user hears Ring back Tone 4. Teams user hangs up the call while PBX B user is ringing 5. PBX B user stops ringing 6. Verify call is cleared successfully	1. Call is disconnected before answer	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
6	Teams user Calls PSTN user and hangs up before answer	<ol style="list-style-type: none"> 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 5. PSTN user stops ringing 6. Verify call is cleared successfully 	<ol style="list-style-type: none"> 1. Call is disconnected before answer 	PASSED	
7	PBX A user Calls Teams 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. 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 	<ol style="list-style-type: none"> 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	<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. 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 	<ol style="list-style-type: none"> 1. Call is connected with bi-directional audio, voice is clear, no echo 2. Call is disconnected 	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
9	PSTN user Calls Teams user	<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 	PASSED	

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 	PASSED	

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 	FAILED	The UC-PHONE-PLUS desk phone is unable to resume a held call using Softkey, if the call has been answered by the phone using receiver or speaker button.
18	PSTN user Calls Teams user and Teams performs hold/resume	<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	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
		<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 	PASSED	
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 	PASSED	

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 	PASSED	
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 	PASSED	

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 	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		<ol style="list-style-type: none"> PBX B user 1 hangs up the call 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"> Make a voice call from PBX B user to Teams user PBX B user hears Ring back Tone Teams user answers the call Verify two way audio Teams user places a consultation call to PBX A user Verify PBX B user is placed on hold PBX A user answers the call Verify two way audio Teams user completes the transfer Verify two way audio between PBX B user and PBX A user PBX B user hangs up the call Verify call is cleared successfully 	<ol style="list-style-type: none"> Call is transferred successfully Two way audio present after call is transferred 	PASSED	
33	PBX B user Calls Teams user, Teams user performs Attended Transfer to PSTN user	<ol style="list-style-type: none"> Make a voice call from PBX B user to Teams user PBX B user hears Ring back Tone Teams user answers the call Verify two way audio Teams user places a consultation call to PSTN user Verify PBX B user is placed on hold PSTN user answers the call Verify two way audio Teams user completes the transfer 	<ol style="list-style-type: none"> Call is transferred successfully Two way audio present after call is transferred 	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
		<ul style="list-style-type: none"> 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	<ul 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 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 	<ul style="list-style-type: none"> 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	<ul 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 transfers the call to PBX A user 2 	<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
	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 	PASSED	

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 	PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
41	Teams user Calls PBX B user, Teams user performs Unattended Transfer to PBX A user	<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 	PASSED	
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 	PASSED	

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 	PASSED	
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 	PASSED	
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 	PASSED	
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 	PASSED	

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 	PASSED	
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 	PASSED	

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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a

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

External ID	Title	Procedure	Expected Results	Status	Comments
					Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
59	Teams user user Calls PBX B user, Teams user adds PBX B user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing.

External ID	Title	Procedure	Expected Results	Status	Comments
					Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
60	Teams user user Calls PBX B user, Teams user adds PBX A user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone

External ID	Title	Procedure	Expected Results	Status	Comments
					has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
61	Teams user user Calls PBX B user, Teams user adds PSTN user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to

External ID	Title	Procedure	Expected Results	Status	Comments
					add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
62	Teams user user Calls PSTN user, Teams user adds PBX B user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into

External ID	Title	Procedure	Expected Results	Status	Comments
					conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing
63	Teams user user Calls PSTN user, Teams user adds PBX A user to the ongoing call	<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

External ID	Title	Procedure	Expected Results	Status	Comments
					with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
64	Teams user user Calls PSTN 1 user, Teams user adds PSTN 2 user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

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

External ID	Title	Procedure	Expected Results	Status	Comments
68	PBX B user Calls Teams user, Teams user adds PBX B user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
69	PBX B user Calls Teams user, Teams user adds PBX A user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
70	PBX B user Calls Teams user, Teams user adds PSTN user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
71	PSTN user Calls Teams user, Teams user adds PBX B user to the ongoing call	<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 	FAILED	Crestron phone does not have an option to add a user into conference when its Teams user is assigned with E5 without Audio Conferencing license. Only on E5 without A/C license, audio conferencing a user works via Direct Routing. Currently phone has the option to add a user into conference only with E5 (with A/C) license. With the E5 license, conferencing a user works directly through Microsoft and not via Direct Routing

External ID	Title	Procedure	Expected Results	Status	Comments
72	PSTN user Calls Teams user, Teams user adds PBX A user to the ongoing call	<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. 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	PASSED	
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	PASSED	
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	PASSED	
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	PASSED	

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	PASSED	
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	PASSED	

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	
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	PASSED	

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	PASSED	
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	PASSED	

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	PASSED	
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	

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 	PASSED	
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
		5. PSTN user answers the call 6. Verify two way audio 7. Teams user hangs up the call 8. Verify call is cleared successfully			
97	PBX A user with restricted Caller ID Calls Teams user	1. Make a voice call from PBX A user with restricted caller ID to Teams user 2. PBX A user hears Ring back Tone 3. Teams user starts ringing 4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous 5. Teams user answers the call 6. Verify two way audio 7. PBX A user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to receive an inbound call with restricted caller ID 2. Call is successful with two way audio	PASSED	
98	PBX B user with restricted Caller ID Calls Teams user	1. Make a voice call from PBX B user with restricted caller ID to Teams user 2. PBX B user hears Ring back Tone 3. Teams user starts ringing 4. Verify caller ID displayed on Teams user is Unavailable/Private/Anonymous 5. Teams user answers the call 6. Verify two way audio 7. PBX B user hangs up the call 8. Verify call is cleared successfully	1. Teams user is able to receive an inbound call with restricted caller ID 2. Call is successful with two way audio	NOT SUPPORTED	SFB will not send Anonymous in the From header or PAI

External ID	Title	Procedure	Expected Results	Status	Comments
99	PSTN user with restricted Caller ID Calls Teams user	<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 	PASSED	

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

External ID	Title	Procedure	Expected Results	Status	Comments
104	Teams user Calls PBX B user and leaves voicemail	<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	PASSED	
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 PBX A
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 	PASSED	

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 A
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 	PASSED	
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 		PASSED	

External ID	Title	Procedure	Expected Results	Status	Comments
		Skype for Business user is internal and external			
111	Skype for Business user calls Teams user	<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 		PASSED	
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 		PASSED	

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 		PASSED	
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 		PASSED	
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 		PASSED	

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