Application Note

Public

Deploying a Multi-tenant Direct Routing SBC for Microsoft Phone System with Cisco Unified Border Element (CUBE-HA) [IOS-XE 17.9.1a]

August 08, 2023

Table of Contents

Table of Figures

Introduction

This application note describes a tested Cisco Unified Border Element (Cisco CUBE) High Availability (HA) configuration for connecting multiple Microsoft Phone System tenants to their PSTN trunking service (Lumen, Verizon) via Direct Routing, using tenant configurations. Please refer to provider documentation and the content provided at www.cisco.com/go/interoperability for guidance on how to adjust this tested configuration to meet the specific requirements of your trunking service.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco CUBE. The configuration settings specifically required for Microsoft Phone System Direct Routing and multi-tenancy are presented here. Feature configuration and most importantly the dial plan are customer specific and need an individual approach.

- This application note describes how to configure Direct Routing for Microsoft Phone System multi-tenant and Cisco CUBE on Catalyst 8300 Edge platform [IOS-XE – 17.9.1a] for connectivity to customer specific PSTN (Lumen, Verizon) SIP Trunking services.
- Testing was performed in accordance with Direct Routing for Microsoft Phone System Multi-tenant test methodology and features tested were – basic calls, DTMF transport, Music on Hold (MOH), consultative transfer and blind transfers, call forward and conference.
- The Cisco CUBE configuration presented in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between PSTN network and Direct Routing Microsoft Phone System. The configuration described in this document details the important configuration settings for successful interoperability and care must be taken by the network administrator configuring a Microsoft Phone System Direct Routing trunk in Microsoft Teams Admin page for a successful interworking with the service provider network.

Network Topology

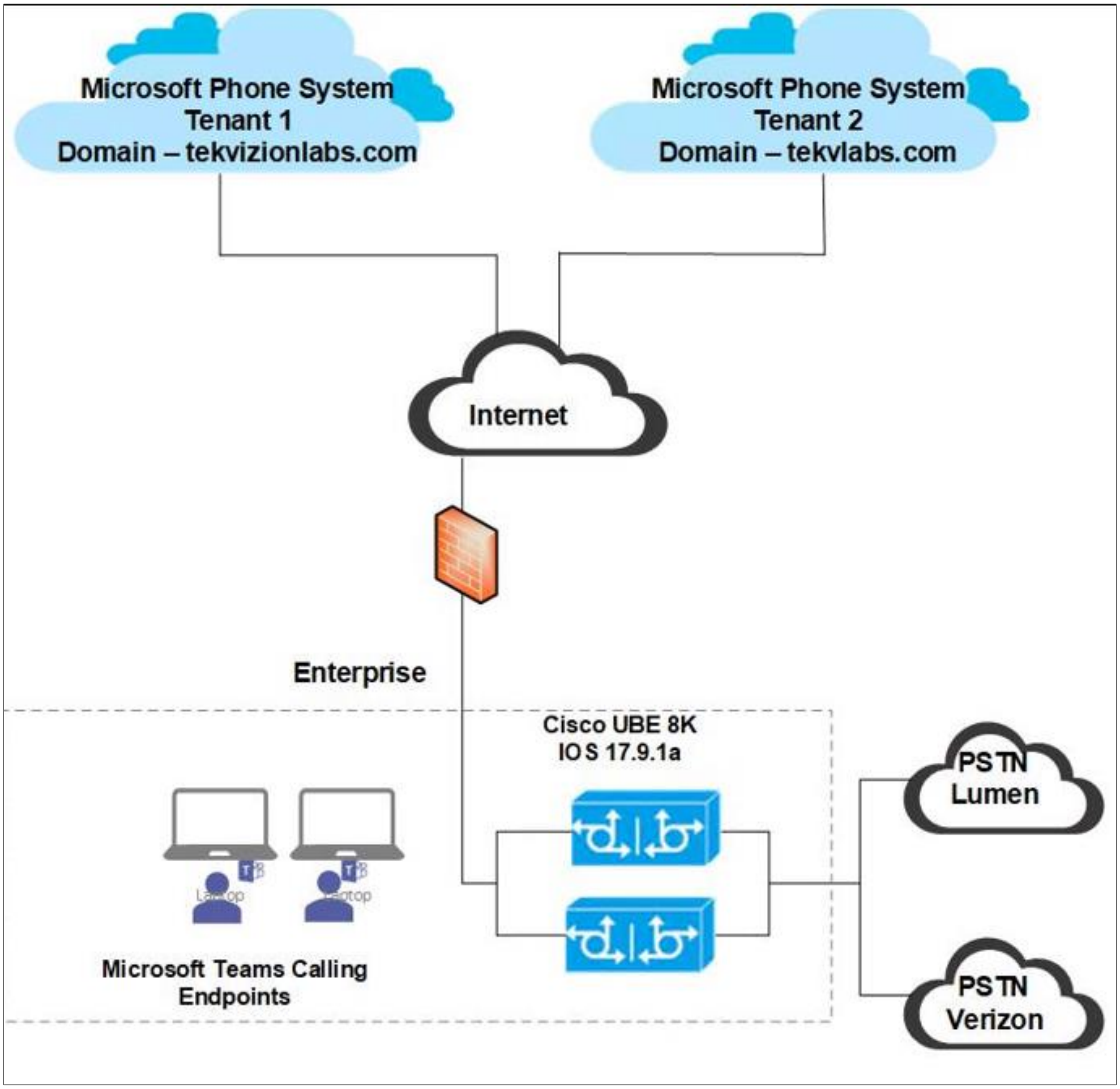

Figure 1: Network Topology

Cisco Webex Calling and Cisco CUBE Settings:

Setting	Value
Transport from Cisco CUBE to Direct Routing Microsoft phone system	TLS with SRTP
Transport from Cisco CUBE to PSTN Verizon	UDP with RTP
Transport from Cisco CUBE to PSTN Lumen	TCP with RTP

System Components

Hardware Requirements

- Cisco CUBE platform 8300-1N1S-6T

Software Requirements

- Cisco CUBE:
    - 14.6 running on IOS-XE 17.9.1a
    - Cisco IOS Software [Cupertino], c8000be Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 17.9.1a, RELEASE SOFTWARE (fc2)
- Microsoft Office 365 Tenant
- Microsoft Teams Desktop Client 1.6.00.11166

Features

Verified Features

- Incoming and outgoing calls using G711ulaw, G729 voice codecs
- Call Conference
- Auto Attendant
- Call hold & Resume
- Consultative transfer
- Blind Transfer
- Call forward
- DTMF (RFC2833)
- IP-PBX Calling number privacy

Features Not Supported

The following features are not supported by Microsoft Phone System

- FAX
- Voice mail pilot number to retrieve messages from PSTN

Features Not Verified

- Long duration calls in Cisco service provider test plan
- IP-PBX telephone number support
- PBX to PSTN international call
- Codec mid-call re-negotiation

Caveats

The following are the observations from Cisco CUBE.

- Cisco CUBE will forward RTCP packets but cannot generate them.
- Cisco CUBE will comfort noise packets but cannot generate them.
- Cisco CUBE does not support RTCP multiplexing.

The following are the observations from Microsoft Teams.

- When Teams users placed call on hold, the following behaviors are observed in signaling.
    - In Media Bypass disable mode, no Re-INVITE received from Microsoft Teams for hold and resume.
    - In Media Bypass enable mode, Microsoft Teams sends Re-INVITE for hold and no Re-INVITE for resume.

The following are the observations from Verizon.

- In blind transfer scenario, the Verizon trunk processes the call transfer only when SIP INVITE contains PAI header with Service provider's DID. To achieve this, Cisco CUBE is configured to send the PAI header whenever it receives Referred-by header from Microsoft Teams.

Configuration

1. Configuring Direct Routing for Microsoft Teams

The steps shown in this section are for Microsoft Phone Systems Tenant 1 – **tekvizionlabs.com**, where the FQDN of the SBC (CUBE) is **sbc6.tekvizionlabs.com**. Similar steps need to be taken for Tenant 2 – **tekvlabs.com** where the FQDN of the SBC (CUBE) is **sbc5.tekvlabs.com**.

1.1 Create User in Microsoft 365

Below are the steps to create a user in Microsoft 365 portal. These steps may be followed for each Microsoft tenant.

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

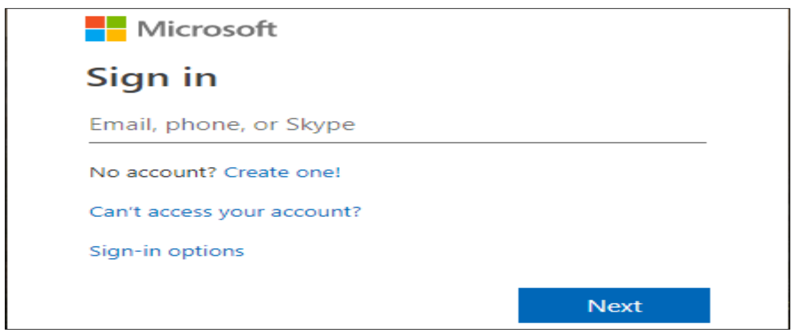

Figure 2 Office 365 Portal login

2. Select the Admin Icon in Office 365 to login Microsoft 365 Admin Center.

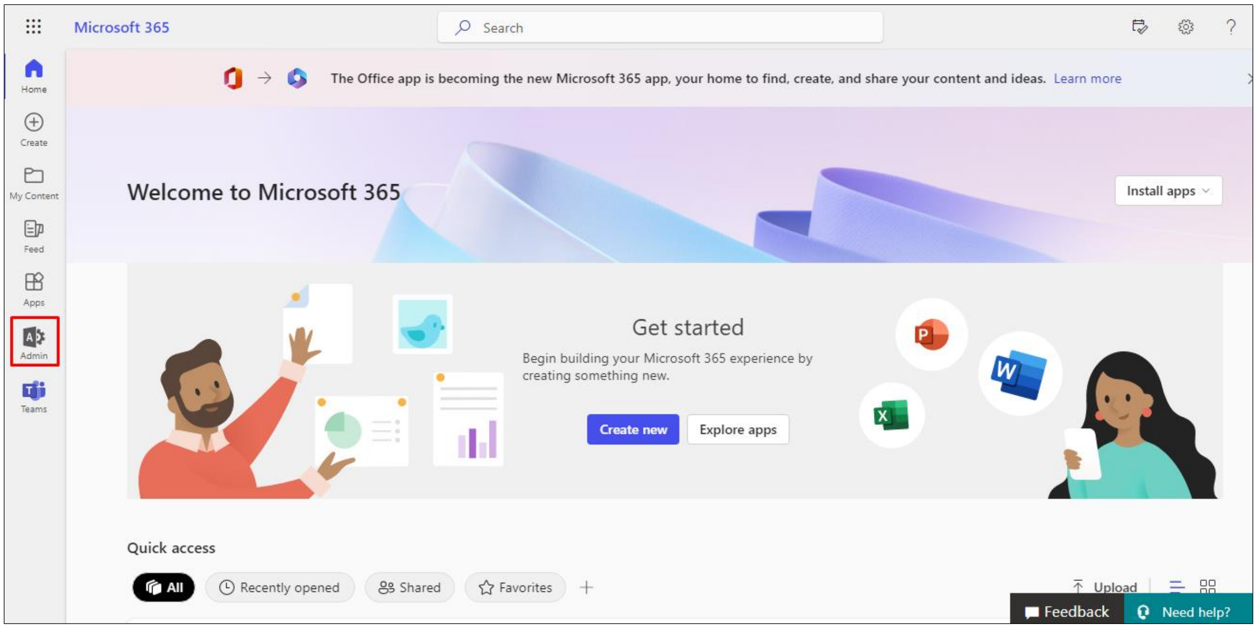

Figure 3 Navigating to Microsoft 365 Admin Center

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

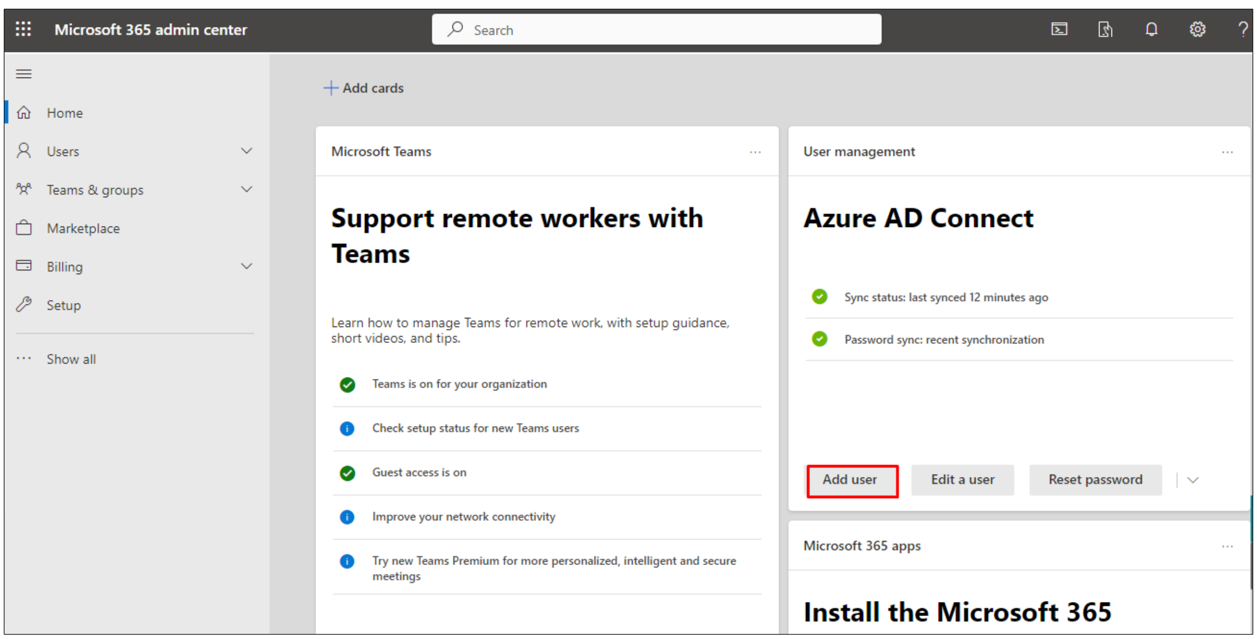

Figure 4 Microsoft 365 Admin Center

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

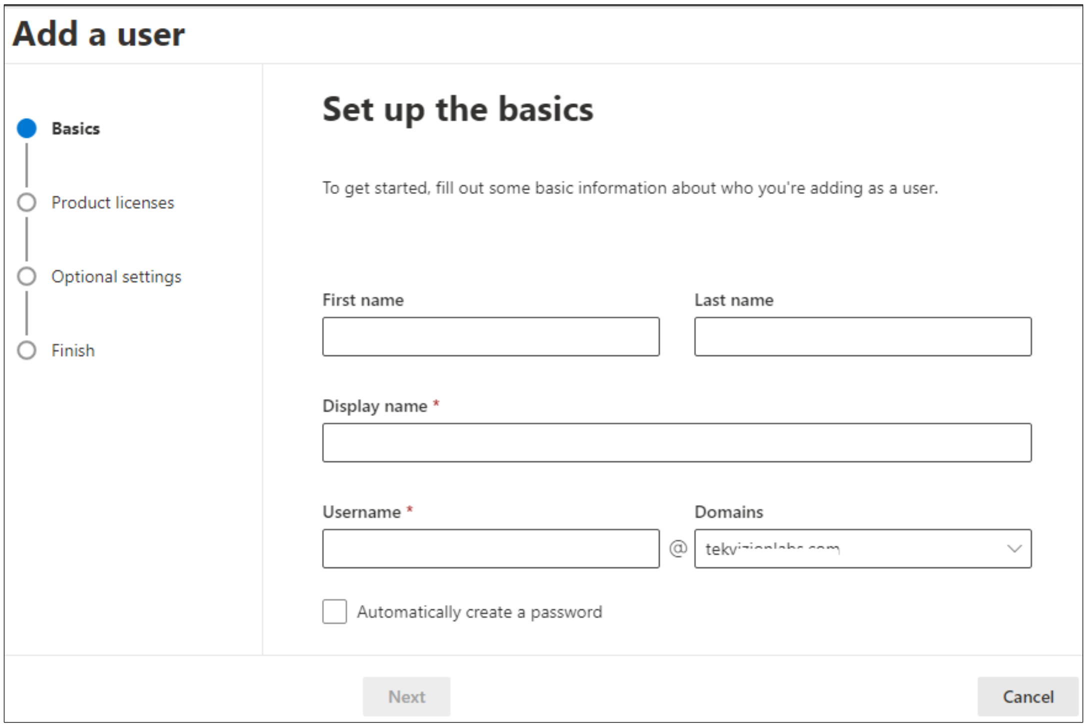

Figure 5 Teams user creation

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

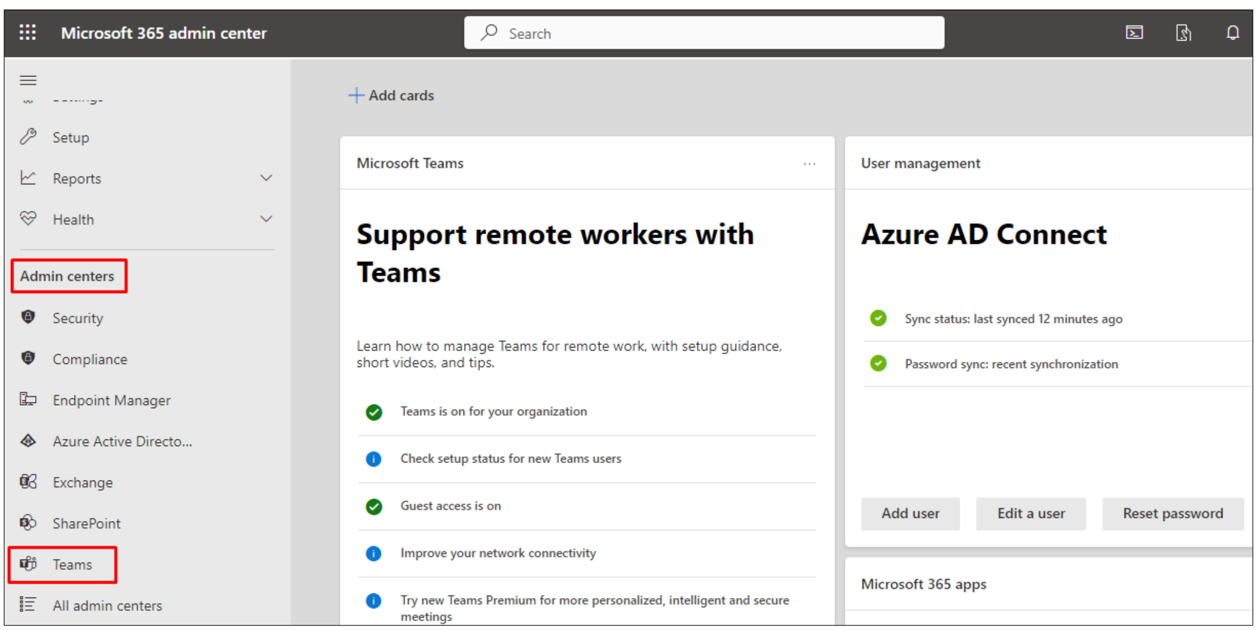

Figure 6 Microsoft 365 Admin Center to Teams Admin Center

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

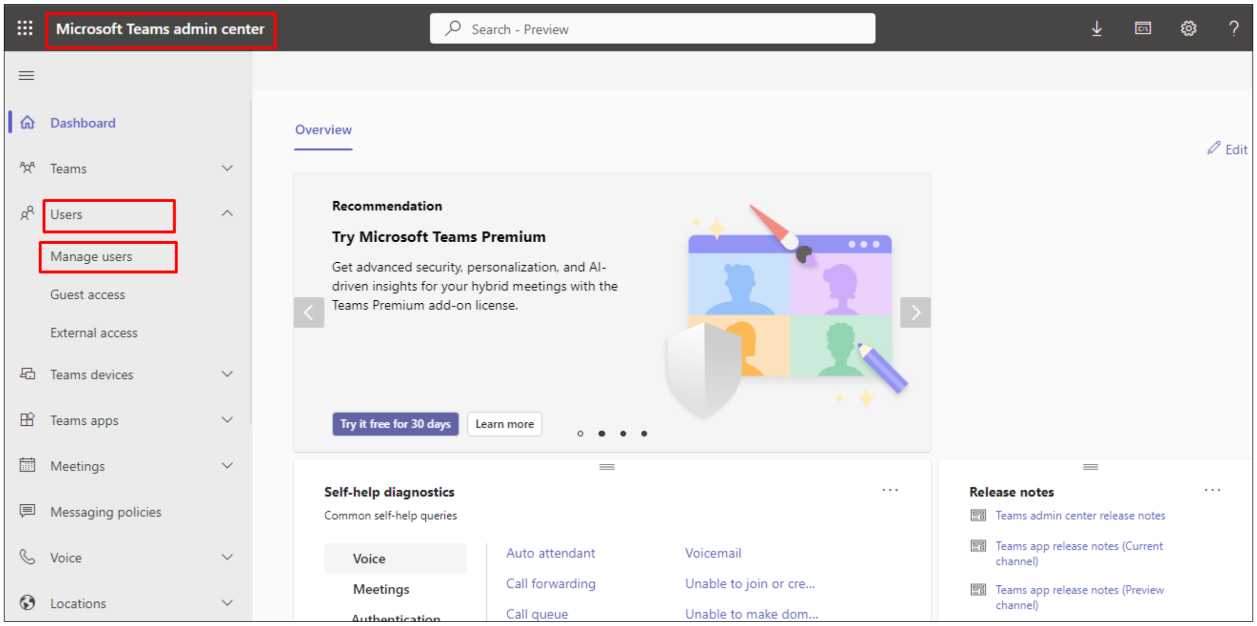

Figure 7 Users in Microsoft Teams Admin Center

7. Search for the user created and click on the user display name to view user properties as shown below.

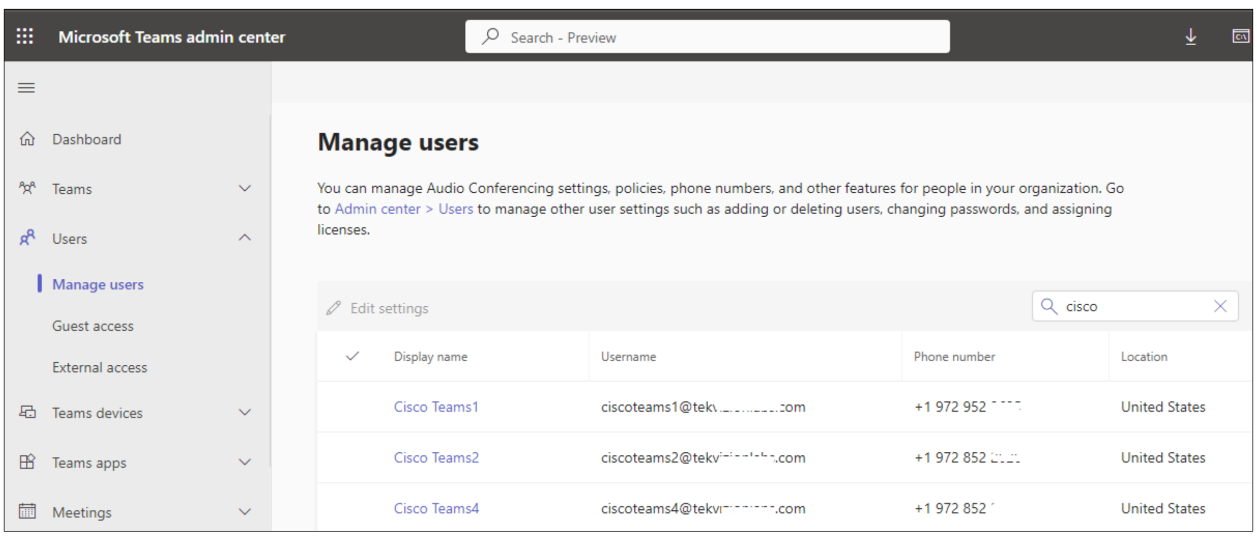

Figure 8 Users in Microsoft Teams Admin Center-cont.

8. Under user properties, navigate to Accounts and set the Teams upgrade mode to Teams only and Select Direct routing and assign phone number.

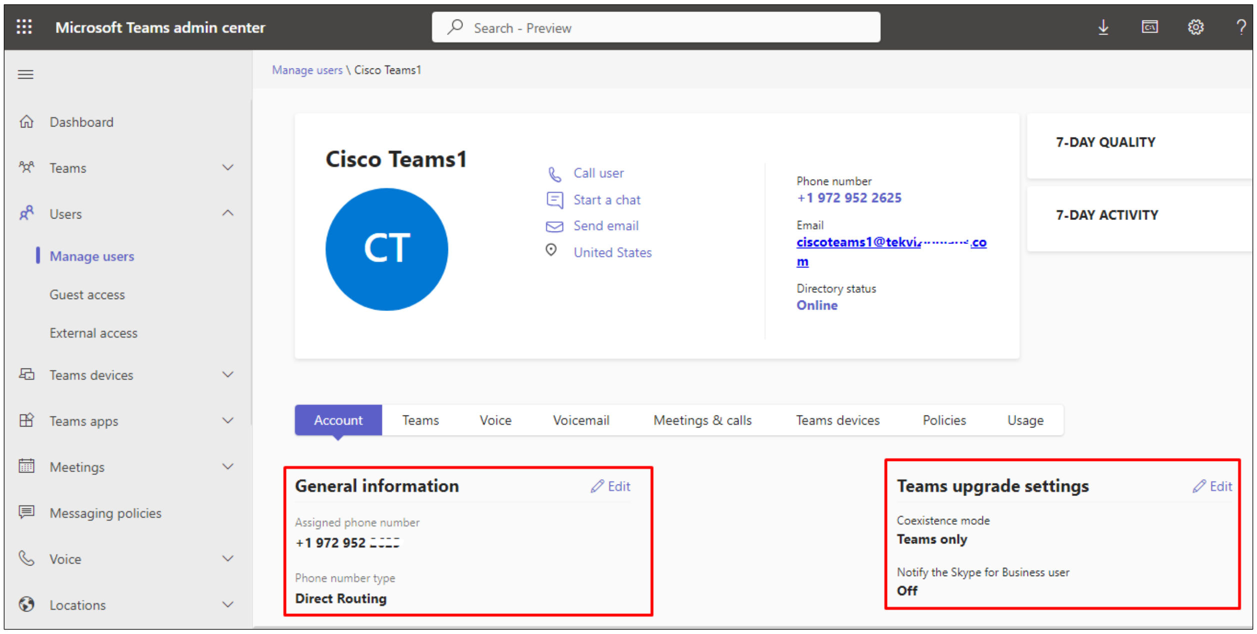

Figure 9 Upgrade Users to Teams only mode and Phone number

9. Under user properties, navigate to Policies and set the Calling Policy as shown below. Here in the below tenant1 custom policy “Busy on Busy enabled” is assigned to user. Procedure to create custom policy is shown in the next section.

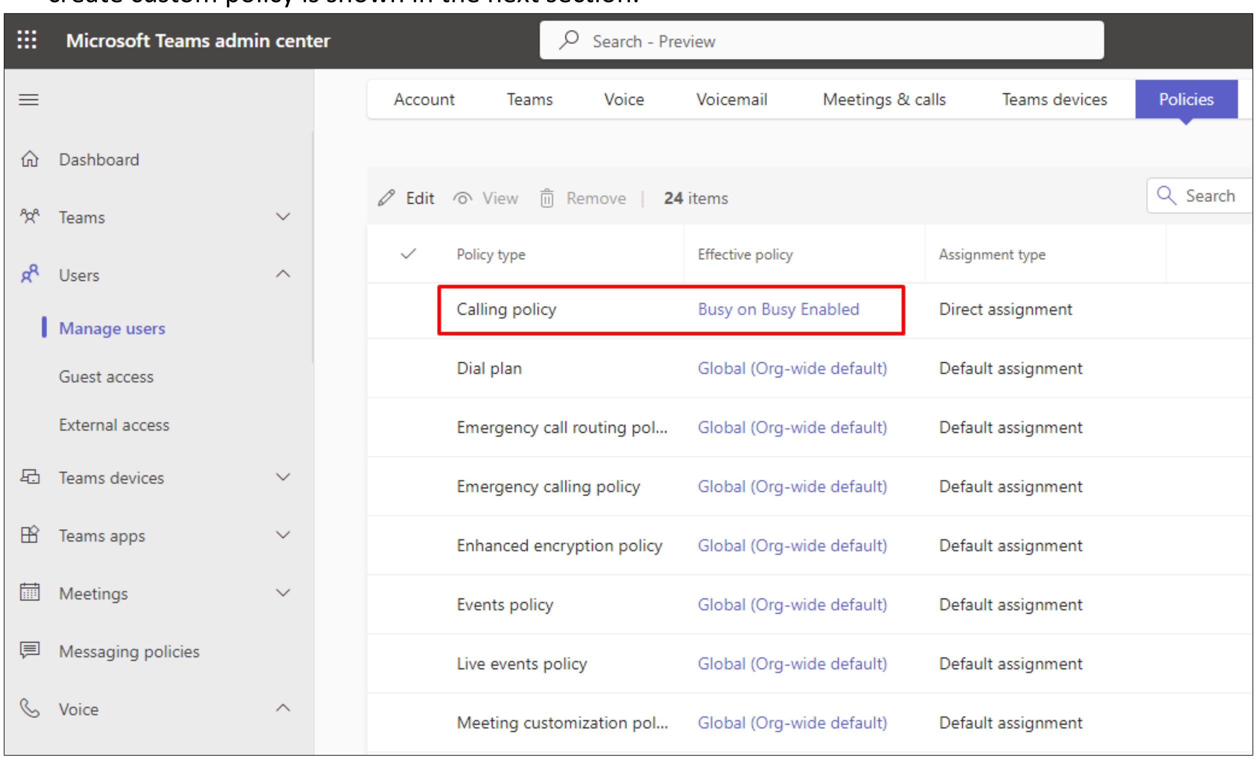

Figure 10 Assigning Calling Policies to Teams User

10. Under user properties, navigate to Policies and set the Caller ID Policy as shown below. Here in the below tenant1 caller ID policy "Anonymous" is assigned to user. Procedure to create custom policy is shown in the next section.

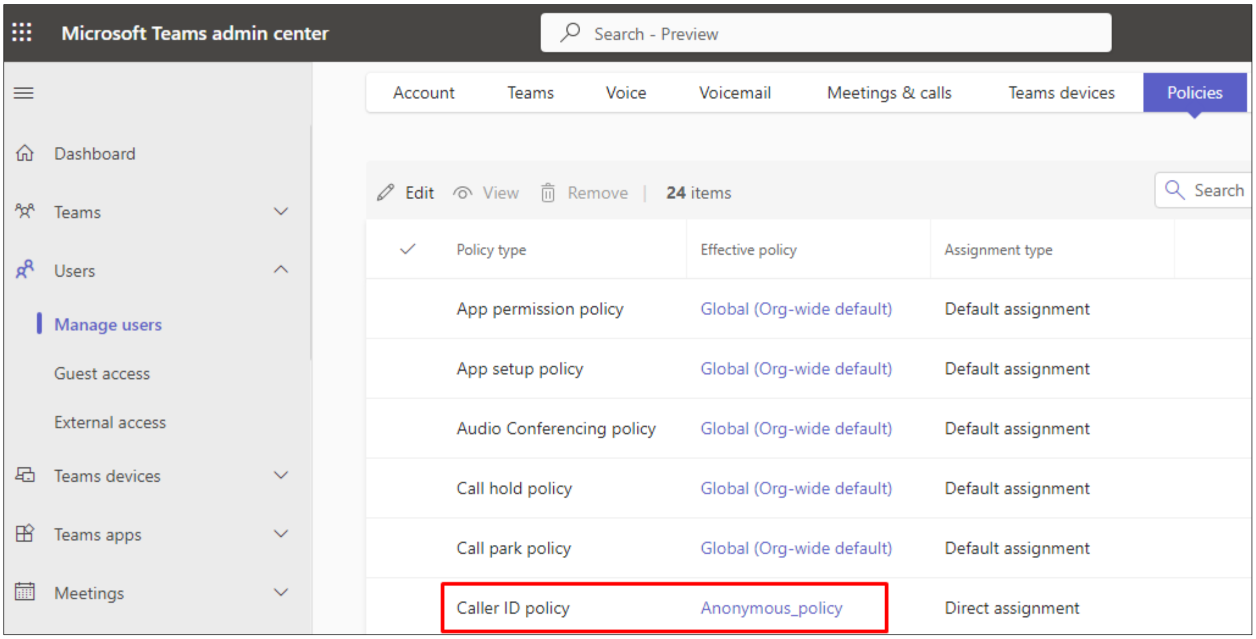

Figure 11 Assigning Caller ID policy to Teams User

1.2 Configure Calling policy in Microsoft Teams Admin Center.

1. To configure a custom policy, navigate to Microsoft Teams admin center > Voice > Calling Policies > New policy.

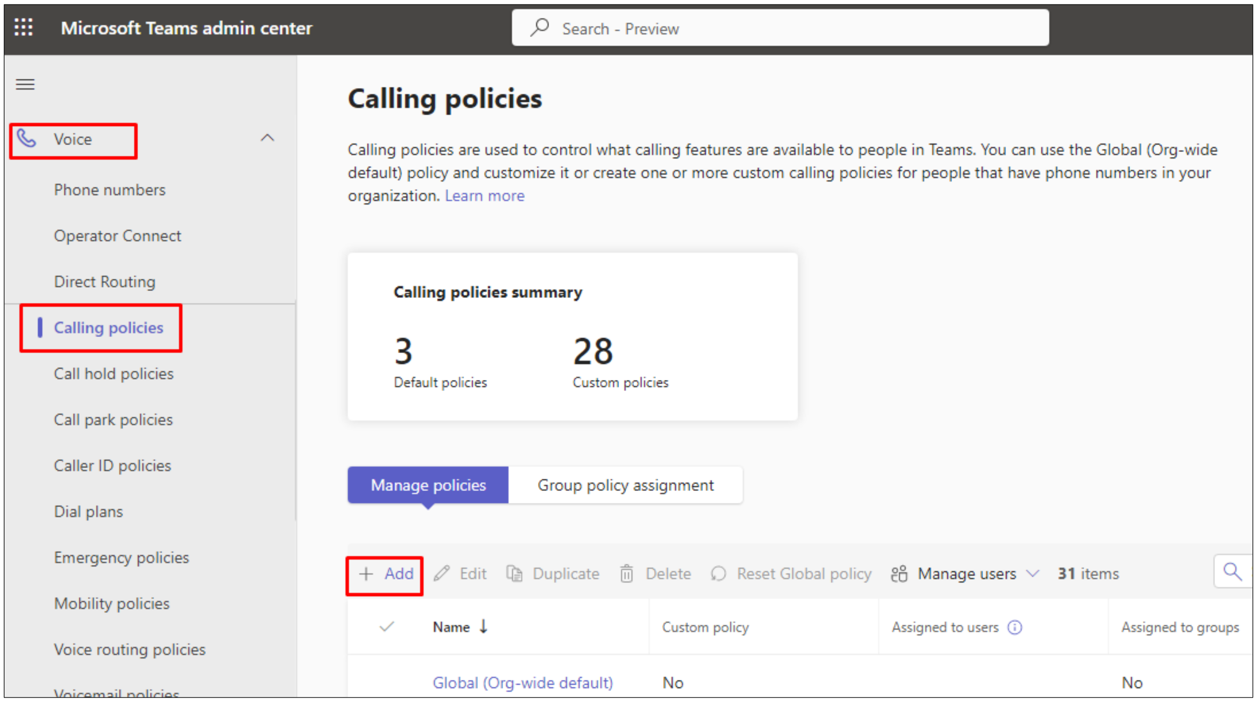

Figure 12 Calling Policies configuration

2. Create a calling policy to turn on Busy on Busy. Click save to complete the configuration

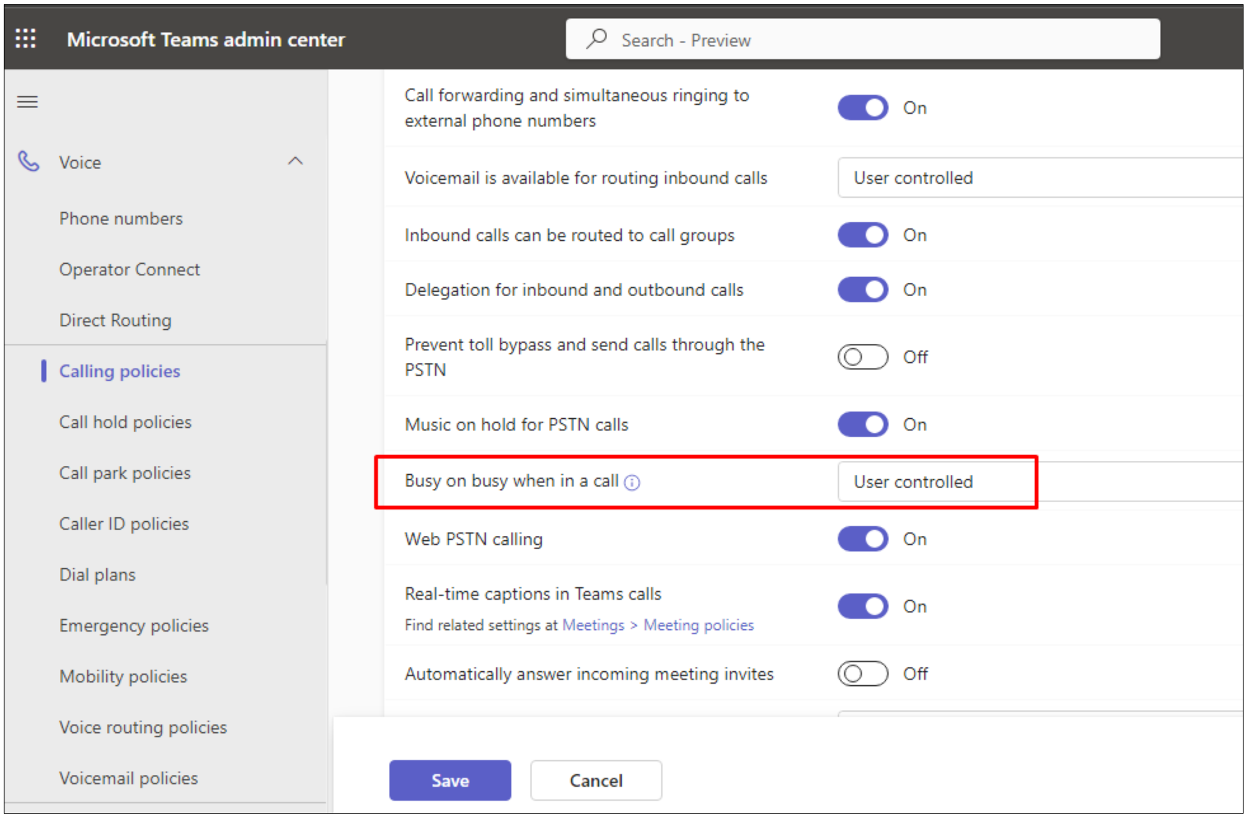

Figure 13 Enable Busy on Busy in Calling Policy

1.3 Configure Caller ID policy in Microsoft Teams Admin Center.

1. To configure a Caller ID policy, navigate to Microsoft Teams admin center > Voice > Caller ID Policies > Add

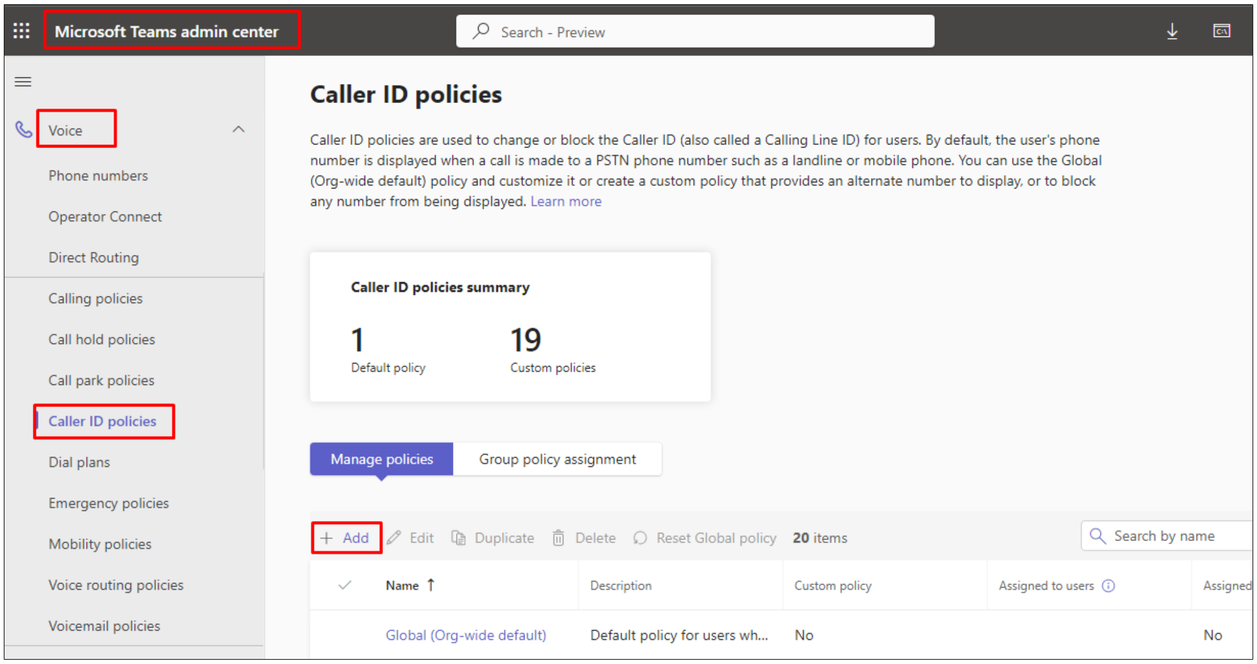

Figure 14 Caller ID Policies Configuration

2. Enter the caller ID policy Name and select the "Replace the Caller ID with Anonymous". Click save to complete the configuration

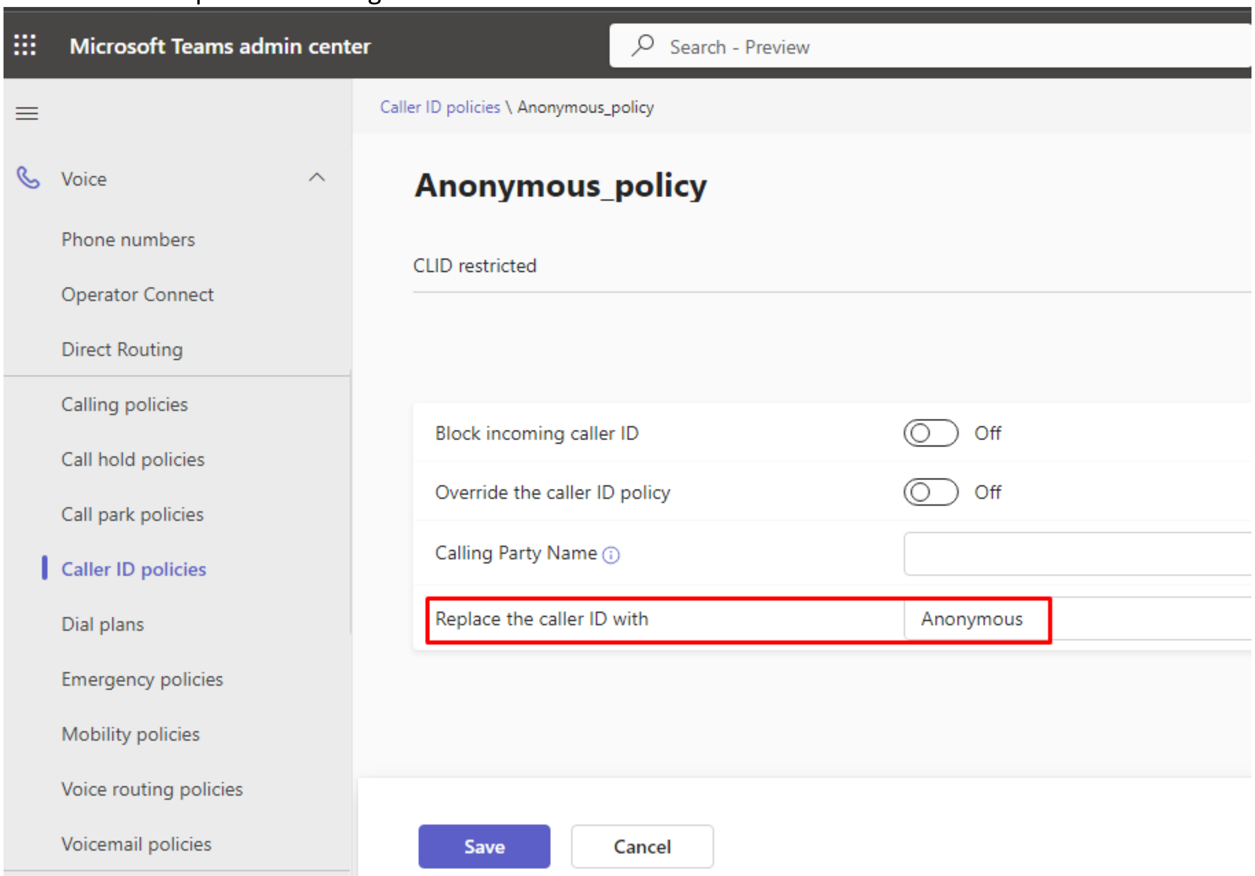

Figure 15 Select Anonymous for the Replace caller ID with

1.4 Create an Online PSTN Gateway

1. To configure a PSTN gateway, navigate to Microsoft Teams admin center > Voice > Direct Routing > SBC > Add

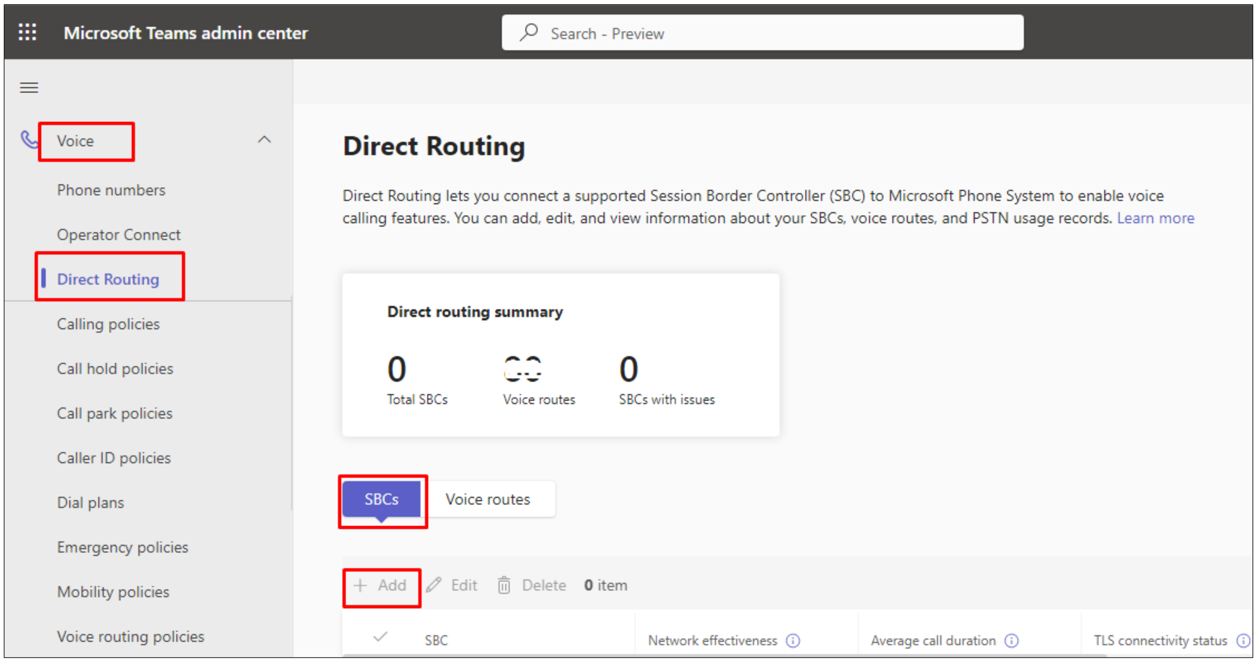

Figure 16 Create Online PSTN gateway

2. Enter the SBC trunk FQDN and enable the SIP Options, SIP signaling port, Send SIP Options, forward call history and Set Media Bypass ON/OFF

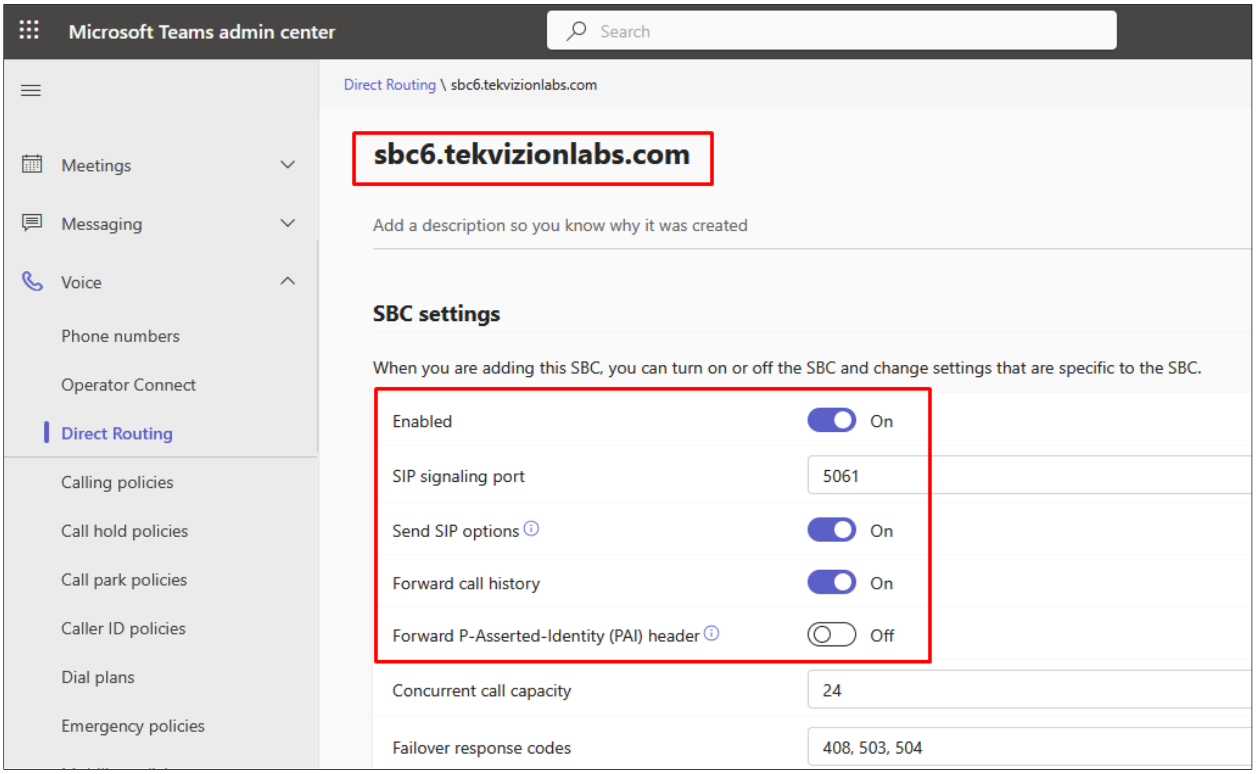

Figure 17 Create Online PSTN gateway

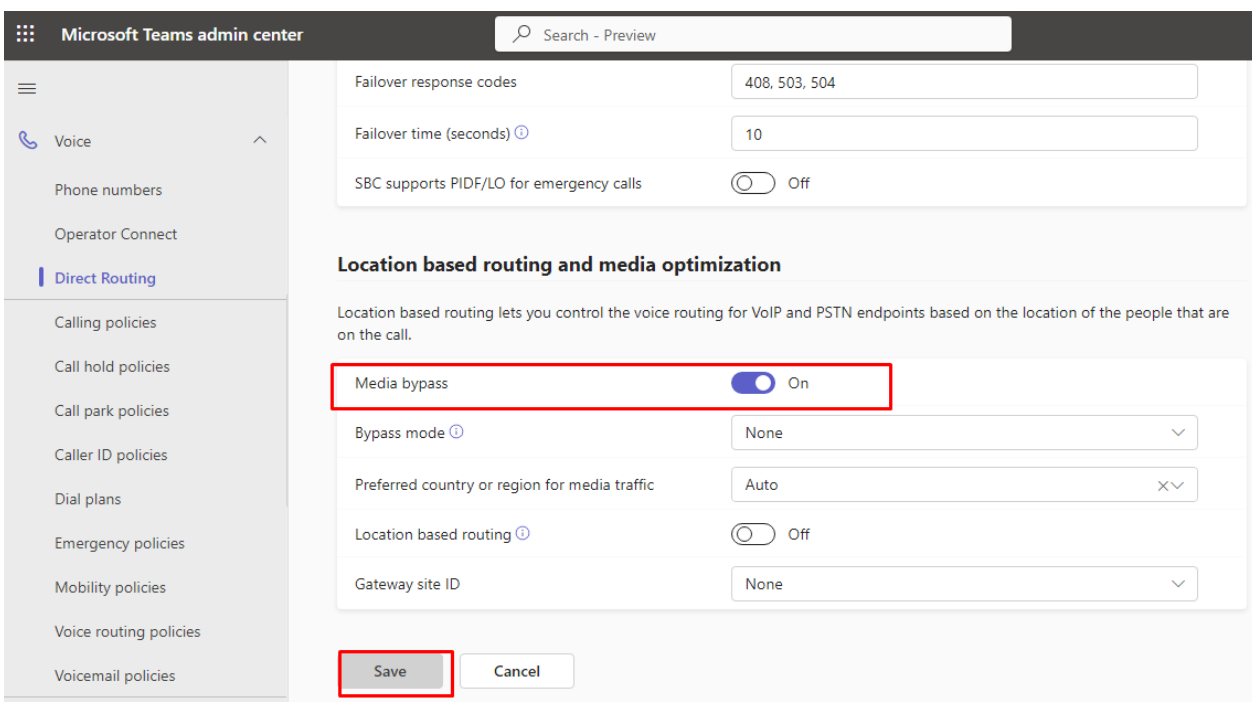

Figure 18 Create Online PSTN gateway

1.5 Configure Online PSTN usage

1. To configure a PSTN gateway, navigate to Microsoft Teams admin center > Voice > Direct Routing > Manage PSTN Usage

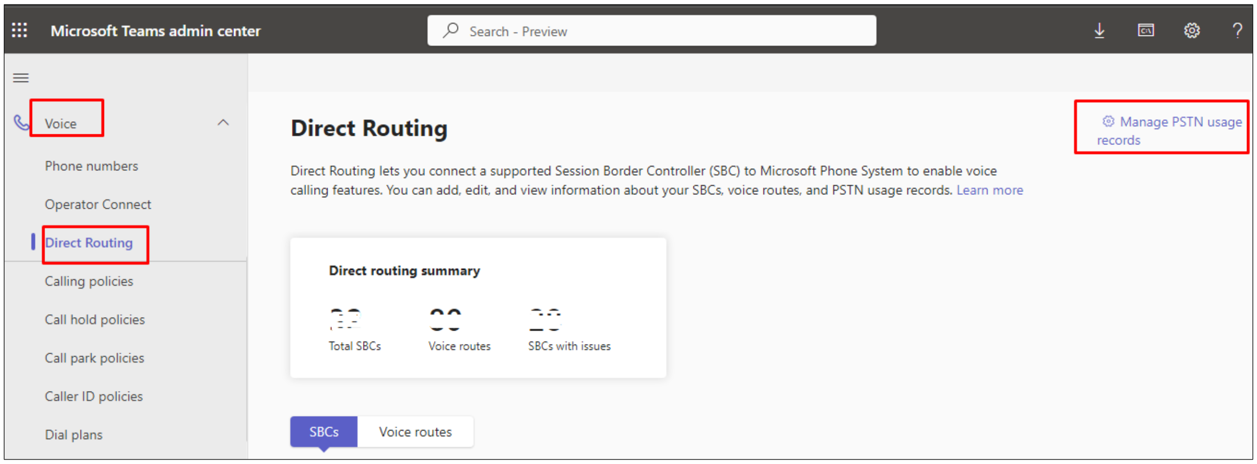

Figure 19 PSTN Usage

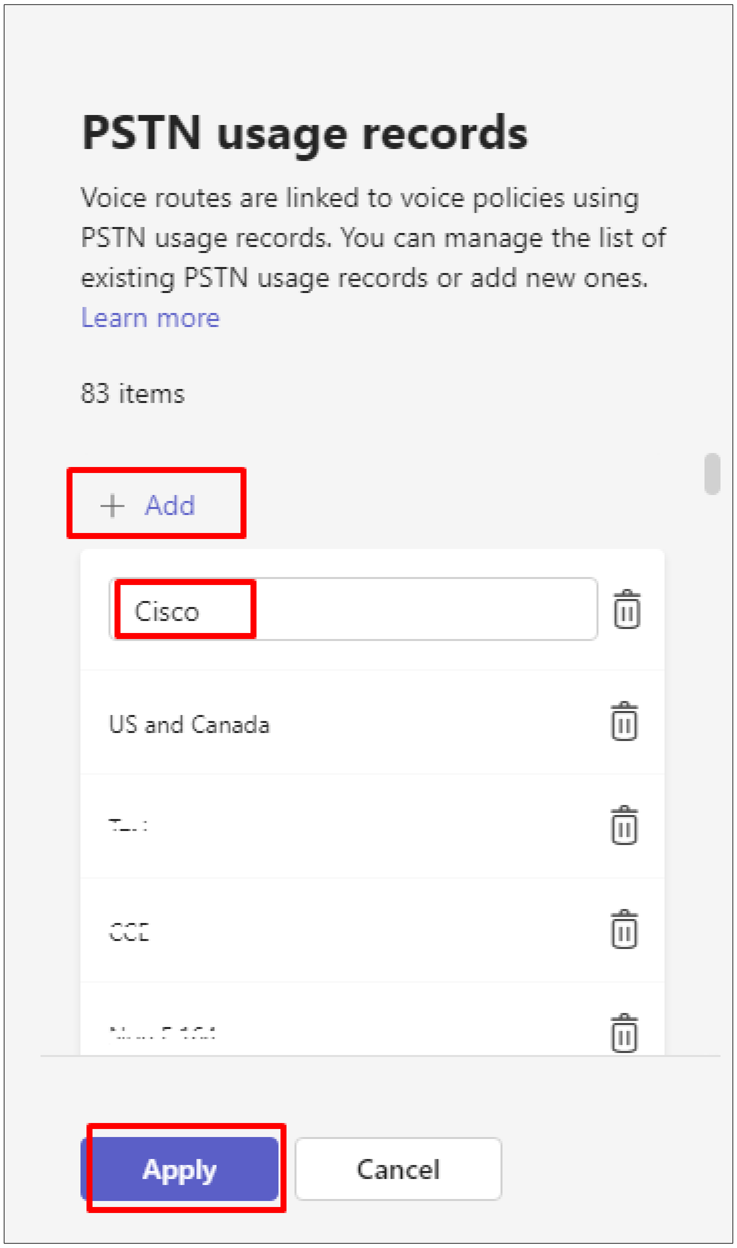

Figure 20 PSTN Usage

1.6 Configure Online Voice Route

1. To configure a Voice Route, navigate to Microsoft Teams admin center > Voice > Direct Routing > Voice Route > Add

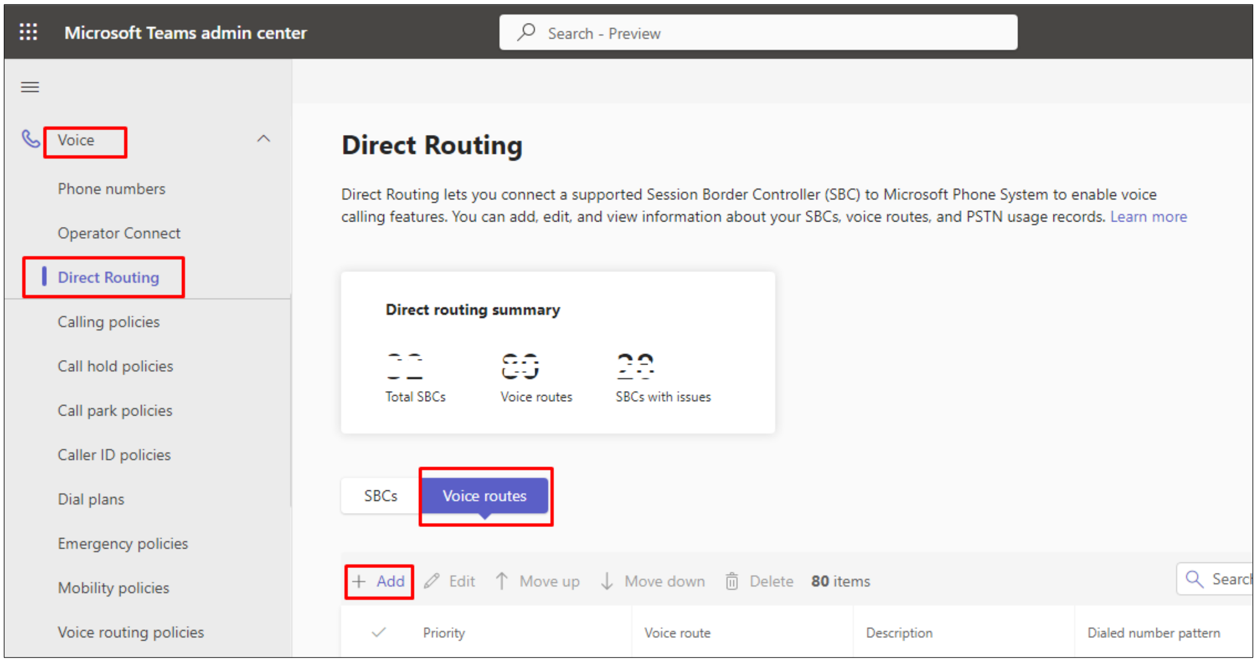

Figure 21 Voice route

2. Enter Voice name, assign SBC trunk FQDN and select PSTN usage.

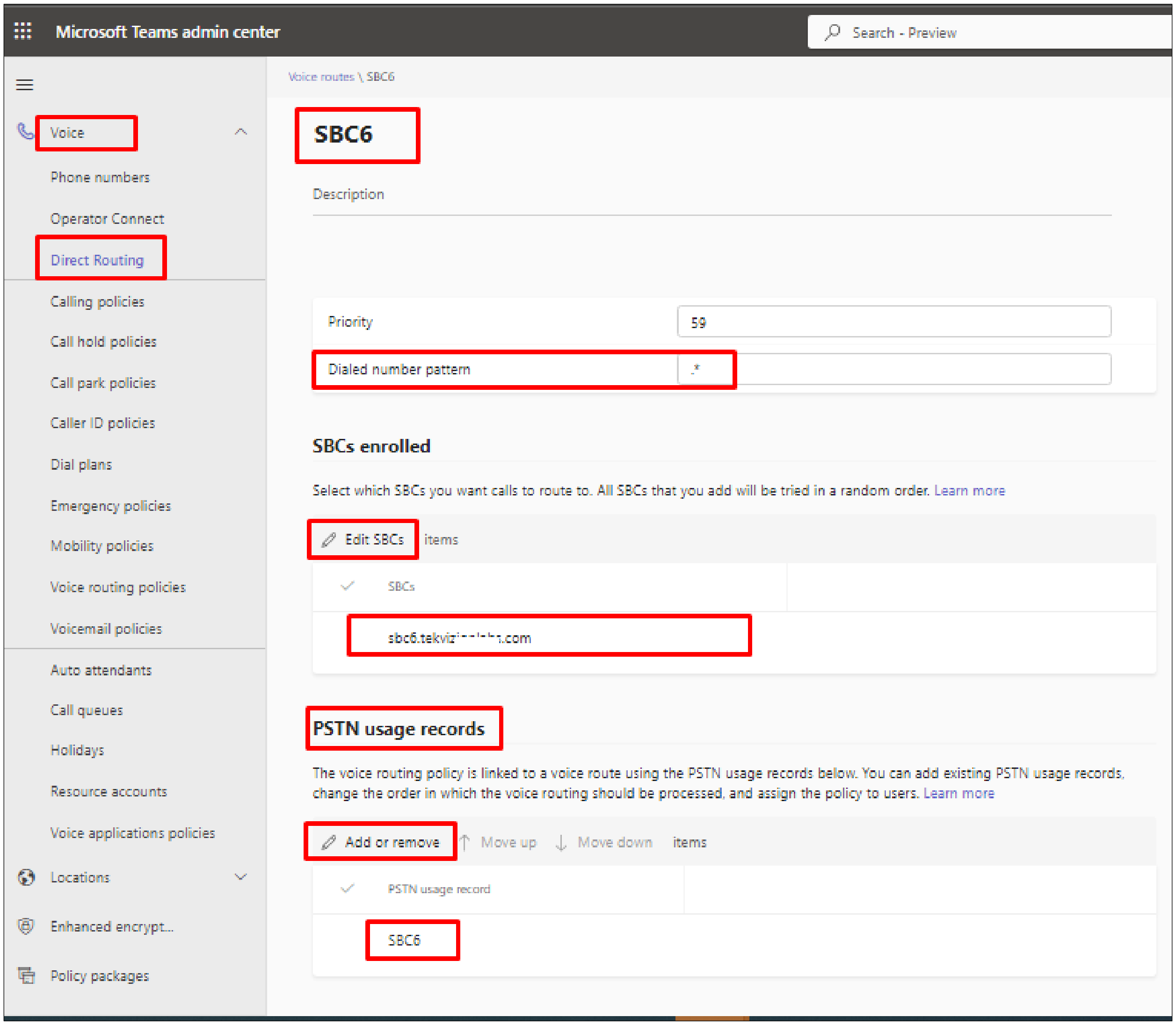

Figure 22 Voice Route

1.7 Configure Online Voice Routing Policy

1. To configure a Voice Routing policy, navigate to Microsoft Teams admin center > Voice > Voice Routing policy > Add

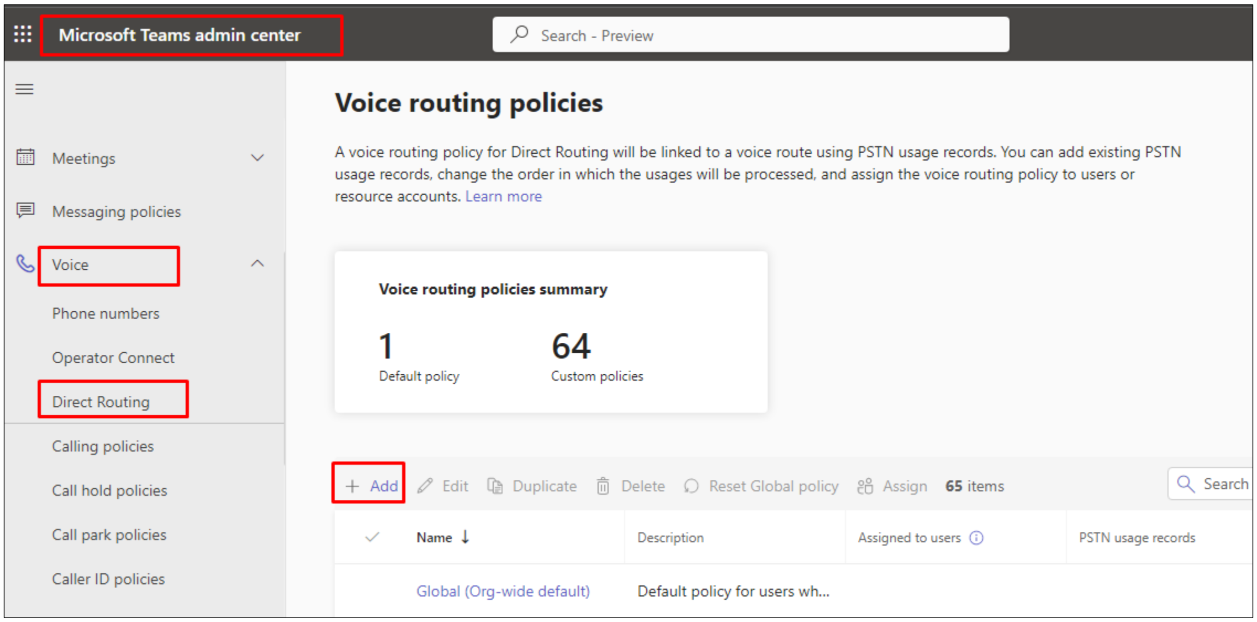

Figure 23 Voice routing policy

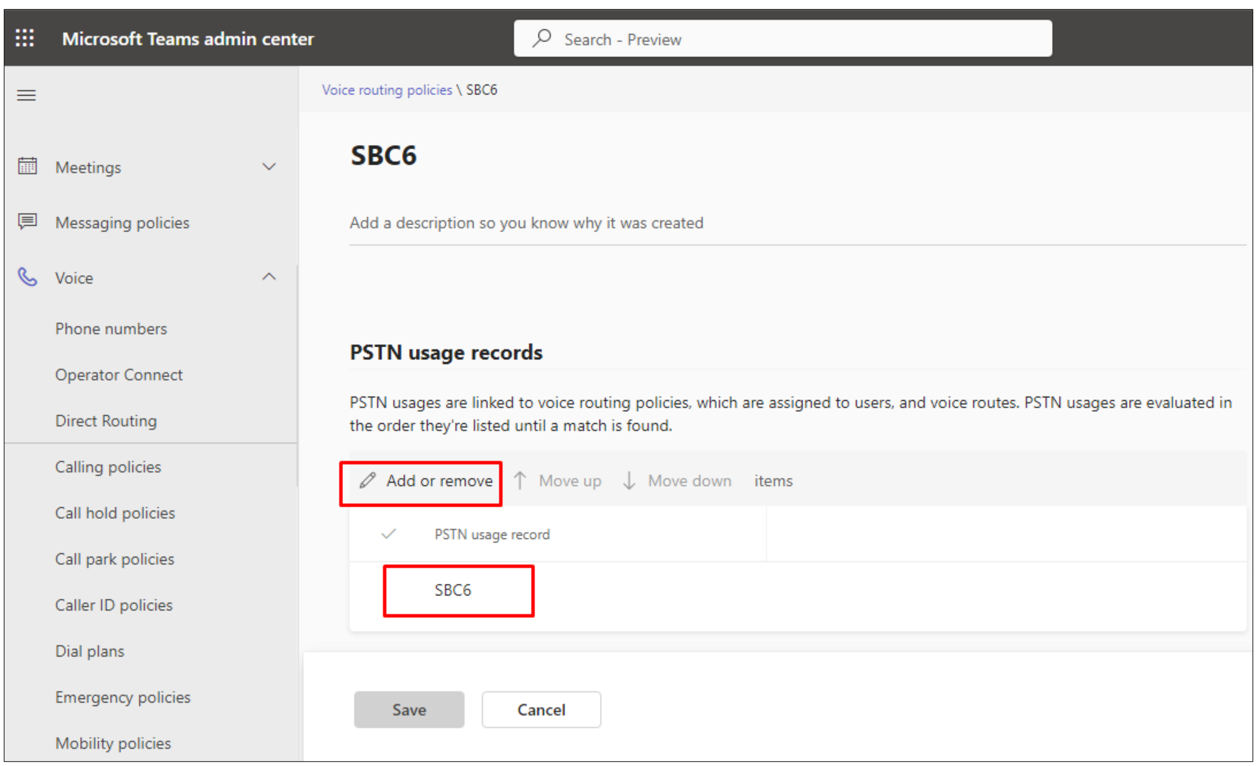

Figure 24 Voice routing policy

2. Cisco CUBE Configuration

The following configuration involves the CUBE High Availability (active/standby CUBEs for stateful failover of active calls).

2.1 IP Networking

```
interface GigabitEthernet0/0/0
 description To HA interface
 ip address 10.64.5.234 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/1.1
 description To PSTN Lumen
 encapsulation dot1Q 3811
 ip address 10.80.11.138 255.255.255.0
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/1.2
 description To PSTN Verizon
 encapsulation dot1Q 1506
 ip address 199.182.124.25x 255.255.255.192
 redundancy rii 18
 redundancy group 1 ip 199.182.124.24x exclusive
!
interface GigabitEthernet0/0/2
 description To Microsoft tenant
 ip address 192.65.79.11x 255.255.255.224
 negotiation auto
 redundancy rii 17
 redundancy group 1 ip 192.65.79.10x exclusive
```

Explanation

Command	Description
redundancy rii id	Redundant interface identifier to generate virtual MAC Same rii id to be used in CUBEs that has same virtual IP
redundancy group 1 ip x.x.x.x exclusive	Enable Redundancy group in physical interface with virtual IP towards PSTN and Microsoft Phone system
Interface GigabitEthernet0/0/1.x	Physical interface divided into multiple sub interfaces to use for 2 different PSTN terminations
encapsulation dot1Q xxxx	to configure VLAN tagging on each sub-interface to forward traffic.

2.2 Routing

To Direct Routing Tenants

```
ip route 0.0.0.0 0.0.0.0 192.65.79.x
```

To PSTN Lumen

```
ip route 10.64.0.0 255.255.0.0 10.80.11.1
```

To PSTN Verizon

```
ip route 152.188.28.0 255.255.255.0 199.182.124.19x
```

2.3 DNS Servers

DNS must be configured to resolve addresses for Microsoft Teams phone system.

```
ip name-server 8.8.8.8
```

2.4 Certificates

The following steps describe how to create and install a certificate.

Generate RSA key

```
crypto key generate rsa general-keys label sbc6 exportable redundancy modulus 2048
The name for the keys will be: sbc6

% The key modulus size is 2048 bits
% Generating 4096 bit RSA keys, keys will be exportable with redundancy...
[OK] (elapsed time was 1 seconds)
```

Create SBC Trustpoint

Hostname based certificate is used in CUBE for Multi-tenant.

```
crypto pki trustpoint sbc6
 enrollment terminal
 subject-name cn=sbc6.tekvizionlabs.com
 revocation-check crl
 rsakeypair sbc6
```

Generate Certificate Signing Request (CSR)

Use this CSR to request a certificate from one of the supported Certificate authorities.

```
crypto pki enroll sbc6
% Start certificate enrollment ..

% The subject name in the certificate will include: cn=sbc6.tekvizionlabs.com
% The subject name in the certificate will include: sbc6.tekvizionlabs.com
% Include the router serial number in the subject name? [yes/no]: no
% Include an IP address in the subject name? [no]: no
Display Certificate Request to terminal? [yes/no]: yes
Certificate Request follows:
```

Authenticate CA Certificate

Enter the following command, then paste the CA certificate that verifies the host certificate into the trust point (usually the intermediate certificate). Open the base 64 CER/PEM file with notepad, copy the text, and paste it into the terminal when prompted.

```
crypto pki authenticate sbc6

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
```

Import signed host certificate

Enter the following command then paste the host certificate into the trust point. Open the base 64 CER/PEM file with notepad, copy the text, and paste it into the terminal when prompted.

```
crypto pki import sbc6 certificate

Enter the base 64 encoded CA certificate.
End with a blank line or the word "quit" on a line by itself
```

Specify the TLS version

```
sip-ua
 transport tcp tls v1.2
```

Import Cisco CA bundle for Microsoft Baltimore certificate authentication

Create the CA certificate trust point used to validate Baltimore certificate from Microsoft in TLS messages:

```
crypto pki trustpool import clean url
http://www.cisco.com/security/pki/trs/ios_core.p7b
Reading file from http://www.cisco.com/security/pki/trs/ios_core.p7b
Loading http://www.cisco.com/security/pki/trs/ios_core.p7b
% PEM files import succeeded.
```

Exporting RSA key and certificate from CUBE 1 for High Availability

```
crypto pki export sbc6 pkcs12 ftp://<username>:<password>@x.x.x.x/ password xxxxx
Address or name of remote host [x.x.x.x]?
Destination filename [sbc6]?
Writing SAN Writing pkcs12 file to ftp://<username>@x.x.x.x/sbc6
!
CRYPTO_PKI: Exported PKCS12 file successfully.
```

Import RSA key and certificate in CUBE 2 for High Availability

Using the below command, import the certificate to CUBE 2. This will automatically create the trustpoint

```
crypto pki import sbc6 pkcs12 ftp://<username>:<password>@x.x.x.x/sbc6 password xxxx
% Importing pkcs12...
Address or name of remote host [x.x.x.x]?
Source filename [sbc6]?
Reading file from ftp://<username>@x.x.x.x/sbc5
[OK - 4931/4096 bytes]

CRYPTO_PKI: Imported PKCS12 file successfully.
```

Note: Repeat the same steps from 1.1.4.1 to 1.1.4.9 for the certificate of Direct Routing Microsoft Phone system tenant 2 sbc5.tekvlabs.com

2.5 Global CUBE settings

In order to enable CUBE with settings required to interwork with Microsoft Teams phone system Voice, the following commands must be entered:

```
voice service voip
 ip address trusted list
  ipv4 10.64.1.x
  ipv4 152.188.28.1xx
  ipv4 152.188.28.1x
  ipv4 172.16.29.0 255.255.255.0
  ipv4 52.0.0.0 255.0.0.0
 rtcp keepalive
 address-hiding
 mode border-element
 media bulk-stats
 media disable-detailed-stats
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711alaw
 h323
 trace
 sip
  listen-port secure  5067
  early-offer forced
  g729 annexb-all
  sip-profiles inbound
  no call service stop
```

Explanation

Command	**Description**
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
no supplementary-service sip refer no supplementary-service sip handle-replaces	Disable forwarding SIP REFER message for call transfers and replace the Dialog-ID in the Replaces header with the peer Dialog-ID
Redundancy group 1	Enable redundancy group
rtcp-keepalive	Enables the Cisco CUBE to send rtcp keepalive packets for the session keepalive
Listen-port secure 5067	To set 5061 as a secure listening port in Cisco CUBE tenant configuration, the global secure listen port needs to be changed other than the ports mentioned in tenants. Example: In a multi-tenant setup, sbc6.tekvizionlabs.com is listening on 5061 and sbc5.tekvlabs.com is listening on port 5063, hence, the global listen port is set to 5067.

2.6 Configure Redundancy group

```
redundancy
 mode none
 application redundancy
  group 1
   priority 150 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/0 protocol 1
   data GigabitEthernet0/0/0
   track 1 shutdown
   track 2 shutdown
   track 3 shutdown
!
track 1 interface GigabitEthernet0/0/1.1 line-protocol
!
track 2 interface GigabitEthernet0/0/1.2 line-protocol
!
track 3 interface GigabitEthernet0/0/2 line-protocol
```

Explanation

Command	Description
priority 150 failover threshold 75	Set priority weightage for CUBE 1 and CUBE 2. High priority CUBE turns Active and other StandBy
timers delay 30 reload 60	the amount of time to delay RG group's initialization and role negotiation after the interface comes up and reload
control GigabitEthernet0/0/0 protocol 1	interface used to exchange keepalive
data GigabitEthernet0/0/0	interface used for checkpointing of data traffic
Track x interface gigabit Ethernet x/x/x line-protocol	The track command is used in RG to track the voice traffic interface state so that the active router initiates switchover after the traffic interface is down.
Track x shutdown	Enable RG group tracking

2.7 SRTP crypto

Used to set the crypto cipher for the Direct Routing

```
voice class srtp-crypto 200
 crypto 1 AES_CM_128_HMAC_SHA1_80
```

2.8 STUN ICE-lite

Applies only when Media bypass is enabled in Microsoft phone system Teams trunk and assign this voice-class stun-usage to all inbound and outbound dialpeer Microsoft Teams.

```
voice class stun-usage 1
 stun usage ice lite
```

2.9 Codecs

To Direct Routing Teams/PSTN

```
voice class codec 100
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
```

2.10 Options keepalive to Direct Routing Teams

Enable SIP Options towards Direct Routing Teams trunk configured and to track the trunk status frequently set the interval and transport protocol. This keepalive profile is triggered from dial-peer configured towards Microsoft Teams.

To ensure that contact headers include the SBC fully qualified domain name, the following profile is used.

To Direct Routing Microsoft phone system Tenant 1

```
voice class sip-profiles 299
 rule 20 request OPTIONS sip-header Contact modify "<sip:(.*):" "<sip:sbc6.tekvizionlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-options-keepalive 100
 description Keepalive MStenant1
 up-interval 5
 transport tcp tls
 sip-profiles 299
```

To Direct Routing Microsoft phone system Tenant 2

```
voice class sip-profiles 399
 rule 10 request OPTIONS sip-header Contact modify "<sip:(.*):" "<sip:sbc5.tekvlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!!
voice class sip-options-keepalive 600
 description Keepalive MSTenant2
 up-interval 5
 transport tcp tls
 sip-profiles 399
```

2.11 Message Handling Rules

SIP Profiles: Manipulations for outbound messages to Direct Routing Teams

The following sip profile is required to:

1. **[Rules 10 and 20]** – To Modify Contact header from IP to SBC FQDN in SIP request and response.
2. **[Rule 30]** – To add user=Phone in SIP req URI
3. **[Rules 40 and 50]** – To add X-MS-SBC header with Cisco CUBE model information in request and response.
4. **[Rule 60**] – To modify send-only to inactive in SDP media attribute.
5. **[Rule 70]** – To modify connection info from 0.0.0.0 to interface IP.
6. **[Rules 80 and 90]** – To add lifetime value in crypto attribute.

To Tenant 1

```
voice class sip-profiles 200
 rule 10 request ANY sip-header Contact modify "@(.*):" "@sbc6.tekvizionlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):" "@sbc6.tekvizionlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061 (.*)" 
"sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: 
Cisco CUBE/C8300/\1"
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco 
CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly" "a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify "0.0.0.0" 
"192.65.79.10x"
 rule 80 request ANY sdp-header Audio-Attribute modify "(a=crypto:.*inline:[A-Za-z0-
9+/=]+)" "\1|2^31"
 rule 90 response ANY sdp-header Audio-Attribute modify "(a=crypto:.*inline:[A-Za-z0-
9+/=]+)" "\1|2^31"
!
```

To Tenant 2

```
voice class sip-profiles 600
 rule 10 request ANY sip-header Contact modify "@(.*):" "@sbc5.tekvlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):" "@sbc5.tekvlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061 (.*)" 
"sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: 
Cisco CUBE/C8300/\1"
```

```
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly" "a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify "0.0.0.0" "192.65.79.10x"
 rule 80 request ANY sdp-header Audio-Attribute modify "(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
 rule 90 response ANY sdp-header Audio-Attribute modify "(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
!
```

SIP Profiles: Manipulations for Inbound messages to Direct Routing Teams

The following sip profile is required to:

1. **[Rules 10 and 15]** - To Copy host information from Teams.
2. **[Rules 20 and 30]** – to append +AAA in Refer-to user part to match REFER dial-peer.
3. **[Rule 40]** – To add X-MS-SBC header with Cisco CUBE model information in request and response.
4. **[Rules 50 and 60]** – Require only when Media Bypass is disabled in Microsoft Teams phone system, to modify ICE attribute.

From Tenant 1

```
voice class sip-profiles 290
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AAA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AAA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco
CUBE/C8300V/\1"
rule 50 request ANY sdp-header Audio-Attribute modify "a=ice-.*" "a=label:main-audio"
rule 60 request ANY sdp-header Attribute modify "a=ice-.*" "a=label:main-audio"
!
```

From Tenant 2

```
voice class sip-profiles 690
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC: Cisco
CUBE/C8300V/\1"
rule 50 request ANY sdp-header Audio-Attribute modify "a=ice-.*" "a=label:main-audio"
rule 60 request ANY sdp-header Attribute modify "a=ice-.*" "a=label:main-audio"
!
```

SIP Profiles: Manipulations for REFER INVITE to Direct Routing Teams

The following sip profile is required to:

1. [Rules 1 and 2] - To remove +AAA from user part in SIP req URI.
2. Rules [5 and 8] – To modify IP to FQDN in Contact header in request and response.
3. [Rule 7] – To modify media attribute sendonly to inactive.
4. [Rules 10 and 20] – To add X-MS-SBC header with Cisco CUBE model information in request and response.
5. [Rules 30 and 40] – To Copy host information from SIP Req URI
6. [Rules 110, 120, 130, 140] – To modify host information in TO and Contact header.
7. [Rules 160 and 170] – To modify media connection info attribute IP.

From Tenant 1

```
voice class sip-profiles 5304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA@" "sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA" "sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):" "@sbc6.tekvizionlabs.com:"
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly" "a=inactive"
 rule 8 response ANY sip-header Contact modify "@(.*):" "@sbc6.tekvizionlabs.com:"
 rule 10 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC:Cisco
CUBE/C8300V/\1"
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC:
CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AAA@(.*)>" "<sip:\u01>"
 rule 120 request INVITE sip-header To modify "<sip:\+AAA(.*)@.*>" "<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
 rule 170 response 200 sdp-header Audio-Connection-Info modify "0.0.0.0" "\u03"
!
```

From Tenant 2

```
voice class sip-profiles 6304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AA@" "sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AA" "sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):" "@sbc5.tekvlabs.com:"
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly" "a=inactive"
 rule 10 request ANY sip-header User-Agent modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC:Cisco
CUBE/C8300V/\1"
```

```
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-MS-SBC:
CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AA@(.*)>" "<sip:\u01>"
 rule 120 request INVITE sip-header To modify "<sip:\+AA(.*)@.*>" "<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
 rule 170 response 200 sdp-header Audio-Connection-Info modify "0.0.0.0" "\u03"
 !
```

SIP Profiles: Manipulations for Outbound towards PSTN (Lumen/Verizon)

Message manipulations should be configured as required for the PSTN service being used. The following rule was required specifically for the Verizon and Lumen trunk used in this case:

1. [Rule 10] - Use SDP `inactive` instead of `sendonly`.
2. [Rule 20] – To add and use P-asserted-identity header with values of Referred-By header when Referred-by header is present in transfer scenarios

```
voice class sip-profiles 100
 rule 10 request ANY sdp-header Audio-Attribute modify "a=sendonly" "a=inactive"
 rule 20 request INVITE sip-header Referred-By modify "<sip:(.*)>"
"<sip:\1>\x0D\x0AP-ASSERTED-IDENTITY:<sip:\1>"
!
```

2.12 Specify the trust point in tls profile

To Direct Routing Teams Tenant 1

```
voice class tls-profile 100
 description MSTenant1
 trustpoint sbc6
```

To Direct Routing Teams Tenant 2

```
voice class tls-profile 600
 description MSTenant2
 trustpoint sbc5
```

Explanation

Command	Description
`trustpoint sbcx`	Associate the trunk FQDN trustpoint to Microsoft Teams Phone system tenants

2.13 Tenant

Mention secure listen-port in each tenant towards Direct routing tenant.

To Direct Routing Tenant 1

```
voice class tenant 200
  tls-profile 100
  listen-port secure 5061
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  session refresh
  no referto-passing
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
  no pass-thru content custom-sdp
  sip-profiles 200
  sip-profiles 290 inbound
  early-offer forced
  no privacy-policy passthru
!
```

To Direct Routing Tenant 2

```
voice class tenant 600
  tls-profile 600
  listen-port secure 5063
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc5.tekvlabs.com
  session transport tcp tls
  no session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
```

```
  no pass-thru content custom-sdp
  sip-profiles 600
  sip-profiles 690 inbound
  privacy-policy passthru
  !
```

Tenant to PSTN Lumen

```
voice class tenant 300
  session transport udp
  session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.2
  bind media source-interface GigabitEthernet0/0/1.2
  no pass-thru content custom-sdp
!
```

Tenant to PSTN Verizon

```
voice class tenant 100
  session transport tcp
  session refresh
  url sip
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.1
  bind media source-interface GigabitEthernet0/0/1.1
  no pass-thru content custom-sdp
  privacy-policy passthru
!
```

2.14 Number translation rules

To Direct Routing Teams

The following translation rule applies for non +E164 from PSTN to Direct Routing in E164.

```
voice translation-rule 100
 rule 1 /^\(.*\)/ /+1\1/
!
voice translation-profile 100
 translate calling 100
 translate called 100
```

From Direct Routing Tenant 1

The following translation rule applies from Direct Routing Tenant 1 in E164 (inbound dial-peer) to Lumen PSTN trunk with prefix routing dialpeer destination match 6

```
voice translation-rule 1001
 rule 1 /^\+1\(.*\)/ /6\1/
!
voice translation-profile 1001
 translate calling 1001
 translate called 1001
```

From Direct Routing Tenant 2

The following translation rule applies from Direct Routing Tenant 2 in E164 (inbound dial-peer) to Lumen PSTN trunk with prefix routing dialpeer destination match 5

```
voice translation-rule 6001
 rule 1 /^\+1\(.*\)/ /5\1/
!
voice translation-profile 6001
 translate calling 6001
 translate called 6001
```

To PSTN Verizon

The following translation rule applies towards Verizon dial-peer to remove 5 appended in calling and called number.

```
voice translation-rule 201
 rule 1 /^5\(.*\)/ /\1/
!
voice translation-profile 201
 translate calling 201
 translate called 201
```

To PSTN Lumen

The following translation rule applies towards Lumen dial-peer to remove 6 appended in calling and called number.

```
voice translation-rule 101
 rule 4 /^6\(.*\)/ /\1/
!
voice translation-profile 101
 translate calling 101
 translate called 101
```

2.15 Dial peers

Inbound calls from Microsoft Direct Routing tenant 1

```
voice class uri 200 sip
 pattern sbc6.tekvizionlabs.com
!
dial-peer voice 1001 voip
 description inbound from Microsoft Phone System
 translation-profile incoming 1001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 incoming uri to 200
 voice-class codec 100
 voice-class stun-usage 1--------------------(Only when Media Bypass enabled)
 voice-class sip tenant 200
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
```

Outbound calls to Microsoft Direct Routing tenant 1

```
voice class e164-pattern-map 1002
 description towards MSTeams tenant1
  e164 +19725980114
  e164 +19725980115
  e164 +19725980116
!
dial-peer voice 1002 voip
 description outbound to Teams1
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 1002
 voice-class codec 100
 voice-class stun-usage 1--------------------(Only when Media Bypass enabled)
```

```
 voice-class sip tenant 200
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
```

Inbound calls from Microsoft Direct Routing tenant 2

```
voice class uri 600 sip
 host sbc5.tekvlabs.com
!
dial-peer voice 6001 voip
 description Inbound MS teams tenant2
 translation-profile incoming 6001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session transport tcp tls
 incoming uri to 600
 voice-class codec 100
 voice-class stun-usage 1--------------------(Only when Media Bypass enabled)
 voice-class sip tenant 600
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
```

Outbound calls to Microsoft Direct Routing tenant 2

```
voice class e164-pattern-map 6002
 description towards MSTeams tenant2
  e164 +14698384746
  e164 +14698384747
 !
dial-peer voice 6002 voip
 description Outbound MS Teams tenant2
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 6002
 voice-class codec 100
 voice-class stun-usage 1--------------------(Only when Media Bypass enabled)
 voice-class sip rel1xx disable
 voice-class sip asserted-id pai
 voice-class sip tenant 600
 voice-class sip options-keepalive profile 600
 dtmf-relay rtp-nte
 srtp

!
```

Inbound calls from PSTN Lumen

```
voice class uri 100 sip
 host 10.64.1.x
!
dial-peer voice 100 voip
 description Incoming dial-peer from PSTNLumen
 translation-profile incoming 100
 rtp payload-type comfort-noise 13
 session protocol sipv2
 incoming uri from 100
 voice-class codec 100
 voice-class sip tenant 100
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
```

```
 dtmf-relay rtp-nte
 no vad
!
```

Outbound calls to PSTN Lumen

```
voice class e164-pattern-map 101
 description towardsPSTNLumen
  e164 6T
!
dial-peer voice 101 voip
 description outgoing dial-peer to IP PSTNLumen
 translation-profile outgoing 101
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target ipv4:10.64.1.72:5060
 session transport tcp
 destination e164-pattern-map 101
 voice-class codec 100
 voice-class sip profiles 100
 voice-class sip tenant 100
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
 dtmf-relay rtp-nte
 no vad
!
```

Inbound calls from PSTN Verizon

```
voice class uri 300 sip
 host ipv4:152.188.28.14x
 host ipv4:152.188.28.19x
!

dial-peer voice 200 voip
 description Incoming dial-peer from PSTN Verizon
 translation-profile incoming 200
 rtp payload-type comfort-noise 13
 session protocol sipv2
```

```
 session transport udp
 incoming uri request 300
 voice-class codec 100
 voice-class sip tenant 300
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
 no vad
```

Outbound calls to PSTN Verizon

```
voice class e164-pattern-map 201
 description towardsPSTNVerizon
  e164 5T
 !
dial-peer voice 201 voip
 description outgoing dial-peer to PSTN Verizon
 translation-profile outgoing 201
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target ipv4:152.188.28.147:5232
 session transport udp
 destination e164-pattern-map 201
 voice-class codec 100
 voice-class sip profiles 100
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
 no vad
!
```

Outbound REFER dial-peer towards Teams tenant 1

```
dial-peer voice 8333 voip
 description TEST for REFER_teams1
 destination-pattern +AAAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 5304
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
!
```

Outbound REFER dial-peer towards Teams tenant 2

```
dial-peer voice 6333 voip
 description TEST for REFER_teams2
 destination-pattern +AAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 6304
 voice-class sip tenant 600
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
```

3. Running Configuration

The following configuration snippet contains a sample configuration of Cisco CUBE with all parameters mentioned previously.

CUBE1

```
Building configuration...
Current configuration : 21493 bytes
!
version 17.9
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname ciscoTeams
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.09.01a.SPA.bin
boot-end-marker
!
logging buffered 10000000
no aaa new-model
!
ip name-server 8.8.8.8
ip domain name domain.com
!
login on-success log
!
subscriber templating
!
multilink bundle-name authenticated
!
password encryption aes
!
crypto pki trustpoint TP-self-signed-995020091
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-995020091
 revocation-check none
 rsakeypair TP-self-signed-995020091
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc6
```

```
 enrollment pkcs12
 revocation-check crl
 rsakeypair scb6
!
crypto pki trustpoint sbc5
 enrollment pkcs12
 revocation-check crl
 rsakeypair sbc5
!
!
crypto pki certificate chain TP-self-signed-995020091
 certificate self-signed 01
crypto pki certificate chain SLA-TrustPoint
 certificate ca 01
crypto pki certificate chain sbc6
 certificate 00BAB7A09A134933DF
 certificate ca 07
crypto pki certificate chain sbc5
 certificate 00DABCF4E9A4FB7CF7
 certificate ca 07
!
crypto pki certificate pool
 cabundle nvram:ios.p7b
!
voice service voip
 ip address trusted list
  ipv4 10.64.1.x
  ipv4 152.188.28.xxx
  ipv4 152.188.28.xxx
  ipv4 172.16.29.0 255.255.255.0
  ipv4 52.0.0.0 255.0.0.0
 rtcp keepalive
 address-hiding
 mode border-element
 media bulk-stats
 media disable-detailed-stats
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
 fax protocol pass-through g711alaw
 h323
 trace
 sip
  listen-port secure  5067
  early-offer forced
  g729 annexb-all
  sip-profiles inbound
  no call service stop
!
voice class uri 100 sip
 host 10.64.1.x
```

```
!
voice class uri 200 sip
 pattern sbc6.tekvizionlabs.com
!
voice class uri 600 sip
 host sbc5.tekvlabs.com
!
voice class uri 300 sip
 host ipv4:152.188.28.14x
 host ipv4:152.188.28.19x
!
voice class codec 100
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class stun-usage 1
 stun usage ice lite
!
voice class sip-profiles 200
 rule 10 request ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061
(.*)" "sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300/\1"
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "192.65.79.1xx"
 rule 80 request ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
 rule 90 response ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
!
voice class sip-profiles 290
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AAA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AAA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300V/\1"
 rule 50 request ANY sdp-header Audio-Attribute modify "a=ice-.*"
"a=label:main-audio"
```

```
 rule 60 request ANY sdp-header Attribute modify "a=ice-.*"
"a=label:main-audio"

!
voice class sip-profiles 299
 rule 20 request OPTIONS sip-header Contact modify "<sip:(.*):"
"<sip:sbc6.tekvizionlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 100
 rule 10 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 20 request INVITE sip-header Referred-By modify "<sip:(.*)>"
"<sip:\1>\x0D\x0AP-ASSERTED-IDENTITY:<sip:\1>"
!
voice class sip-profiles 5304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA@"
"sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA"
"sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 8 response ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"

 rule 10 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC:Cisco CUBE/C8300V/\1"
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AAA@(.*)>"
"<sip:\u01>"
 rule 120 request INVITE sip-header To modify "<sip:\+AAA(.*)@.*>"
"<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
 rule 170 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "\u03"
!
voice class sip-profiles 399
 rule 10 request OPTIONS sip-header Contact modify "<sip:(.*):"
"<sip:sbc5.tekvlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 600
```

```
 rule 10 request ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061
(.*)" "sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300/\1"
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "192.65.79.1xx"
 rule 80 request ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
 rule 90 response ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
!
voice class sip-profiles 690
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 6304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AA@"
"sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AA"
"sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 10 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC:Cisco CUBE/C8300V/\1"
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AA@(.*)>"
"<sip:\u01>"
 rule 120 request INVITE sip-header To modify "<sip:\+AA(.*)@.*>"
"<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
```

```
 rule 170 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "\u03"
!
voice class sip-hdr-passthrulist 290
 passthru-hdr Referred-By
!
voice class e164-pattern-map 101
 description towardsPSTNLumen
  e164 6T
 !
voice class e164-pattern-map 201
 description towardsPSTNVerizon
  e164 5T
 !
voice class e164-pattern-map 1002
 description towards MSTeams tenant1
  e164 +19725980114
  e164 +19725980115
  e164 +19725980116
 !
voice class e164-pattern-map 6002
 description towards MSTeams tenant2
  e164 +14698384746
  e164 +14698384747
 !
voice class sip-options-keepalive 100
 description Keepalive MStenant1
 up-interval 5
 transport tcp tls
 sip-profiles 299
!
voice class sip-options-keepalive 600
 description Keepalive MSTenant2
 up-interval 5
 transport tcp tls
 sip-profiles 399
!
voice class tenant 200
  tls-profile 100
  listen-port secure 5061
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  session refresh
  no referto-passing
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
  no pass-thru content custom-sdp
  sip-profiles 200
```

```
  sip-profiles 290 inbound
  early-offer forced

!
voice class tenant 100
  session transport tcp
  session refresh
  url sip
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.1
  bind media source-interface GigabitEthernet0/0/1.1
  no pass-thru content custom-sdp
  privacy-policy passthru
!
voice class tenant 600
  tls-profile 600
  listen-port secure 5063
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc5.tekvlabs.com
  session transport tcp tls
  no session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
  no pass-thru content custom-sdp
  sip-profiles 600
  sip-profiles 690 inbound
  privacy-policy passthru
!
voice class tenant 300
  session transport udp
  session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.2
  bind media source-interface GigabitEthernet0/0/1.2
  no pass-thru content custom-sdp
!
voice class srtp-crypto 200
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice class tls-profile 100
 description MSTenant1
 trustpoint sbc6
!
voice class tls-profile 600
 description MSTenant2
 trustpoint sbc5
!
voice translation-rule 100
 rule 1 /^\(.*\)/ /+1\1/
```

```
!
voice translation-rule 101
 rule 4 /^6\(.*\)/ /\1/
!
voice translation-rule 200
 rule 1 /^\(4.........\)/ /+1\1/
!
voice translation-rule 201
 rule 1 /^5\(.*\)/ /\1/
!
voice translation-rule 1001
 rule 1 /^\+1\(.*\)/ /6\1/
!
voice translation-rule 6001
 rule 1 /^\+1\(.*\)/ /5\1/
!
voice translation-profile 100
 translate calling 100
 translate called 100
!
voice translation-profile 1001
 translate calling 1001
 translate called 1001
!
voice translation-profile 101
 translate calling 101
 translate called 101
!
voice translation-profile 200
 translate calling 200
 translate called 200
!
voice translation-profile 201
 translate calling 201
 translate called 201
!
voice translation-profile 6001
 translate calling 6001
 translate called 6001
!
voice-card 0/1
 dsp services dspfarm
 no watchdog
!
no license feature hseck9
license udi pid C8300-1N1S-6T sn Fxxxxxxx
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 62769
!
diagnostic bootup level minimal
!
spanning-tree extend system-id
```

```
!
enable secret 9
$9$KsJDHUiGvUauWE$tF35FLGG2wpMEWyVl3vbRKFLqzdsdgQWLxkipeh.fIs
!
username admin privilege 15 password 7 0107030F6D5A1C5E7142
!
redundancy
 mode none
 application redundancy
  group 1
   name cube-ha
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/0 protocol 1
   data GigabitEthernet0/0/0
   track 1 shutdown
   track 2 shutdown
   track 3 shutdown
!
track 1 interface GigabitEthernet0/0/1.1 line-protocol
!
track 2 interface GigabitEthernet0/0/1.2 line-protocol
!
track 3 interface GigabitEthernet0/0/2 line-protocol
!
interface GigabitEthernet0/0/0
 description towards HA
 ip address 10.64.5.234 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/1.1
 encapsulation dot1Q 3811
 ip address 10.80.11.138 255.255.255.0
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/1.2
 description PSTN Verizon
 encapsulation dot1Q 1506
 ip address 199.182.124.25x 255.255.255.192
 redundancy rii 18
 redundancy group 1 ip 199.182.124.24x exclusive
!
interface GigabitEthernet0/0/2
 ip address 192.65.79.11x 255.255.255.224
 negotiation auto
 redundancy rii 17
 redundancy group 1 ip 192.65.79.10x exclusive
```

```
!
interface GigabitEthernet0/0/3
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/4
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/5
 no ip address
 negotiation auto
!
interface Service-Engine0/1/0
!
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/2
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 152.188.28.0 255.255.255.0 199.182.124.193
ip route 172.16.0.0 255.255.0.0 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
!
dspfarm profile 1 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 codec g722-64
 codec opus
 maximum sessions 9
 associate application CUBE
!
dial-peer voice 6001 voip
 description Inbound MS teams tenant2
 translation-profile incoming 6001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session transport tcp tls
 incoming uri to 600
 voice-class codec 100
```

```
 voice-class stun-usage 1
 voice-class sip tenant 600
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
dial-peer voice 201 voip
 description outgoing dial-peer to PSTN Verizon
 translation-profile outgoing 201
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target ipv4:152.188.28.147:5232
 session transport udp
 destination e164-pattern-map 201
 voice-class codec 100
 voice-class sip profiles 100
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 8333 voip
 description TEST for REFER_teams1
 destination-pattern +AAAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 5304
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
!
dial-peer voice 100 voip
 description Incoming dial-peer from PSTNLumen
 translation-profile incoming 100
 rtp payload-type comfort-noise 13
 session protocol sipv2
 incoming uri from 100
 voice-class codec 100
 voice-class sip tenant 100
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description outgoing dial-peer to IP PSTNLumen
 translation-profile outgoing 101
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target ipv4:10.64.1.72:5060
 session transport tcp
 destination e164-pattern-map 101
 voice-class codec 100
 voice-class sip tenant 100
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 6002 voip
 description Outbound MS Teams tenant2
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 6002
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip rel1xx disable
 voice-class sip asserted-id pai
 voice-class sip tenant 600
 voice-class sip options-keepalive profile 600
 dtmf-relay rtp-nte
 srtp
!
dial-peer voice 200 voip
 description Incoming dial-peer from PSTN Verizon
 translation-profile incoming 200
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session transport udp
 incoming uri request 300
 voice-class codec 100
 voice-class sip tenant 300
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 1001 voip
 description inbound from Microsoft Phone System
```

```
 translation-profile incoming 1001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 incoming uri to 200
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip tenant 200
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
dial-peer voice 6333 voip
 description TEST for REFER_teams2
 destination-pattern +AAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 6304
 voice-class sip tenant 600
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
!
dial-peer voice 1002 voip
 description outbound to Teams1
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 1002
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip tenant 200
 voice-class sip options-keepalive profile 100

 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
!
sip-ua
 retry invite 1
```

```
 transport tcp tls v1.2
 handle-replaces
!
!
line con 0
 exec-timeout 5 0
 password 7 111D1C0E
 logging synchronous
 login
 stopbits 1
line aux 0
line vty 0 4
 exec-timeout 60 0
 password 7 105A0
 logging synchronous
 login
 transport preferred telnet
 transport input telnet
line vty 5
 exec-timeout 60 0
 password 7 0010160D
 logging synchronous
 login
 transport input telnet
line vty 6 14
 login
 transport input ssh
!
call-home
 ! If contact email address in call-home is configured as sch-smart-
licensing@cisco.com
 ! the email address configured in Cisco Smart License Portal will be
used as contact email address to send SCH notifications.
 contact-email-addr sch-smart-licensing@cisco.com
 profile "CiscoTAC-1"
  active
  destination transport-method http
!
end
```

CUBE2

```
Building configuration...
Current configuration : 16889 bytes
!
version 17.9
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
!
hostname 8K_CUBE
!
boot-start-marker
boot system bootflash:c8000be-universalk9.17.09.01a.SPA.bin
boot-end-marker
!
!

logging buffered 10000000


no aaa new-model
!

!
ip name-server 8.8.8.8
ip domain name domain.com

!
!
!
login on-success log
!
!
subscriber templating
!

multilink bundle-name authenticated
!

password encryption aes
!
!
crypto pki trustpoint TP-self-signed-2307055185
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2307055185
 revocation-check none
 rsakeypair TP-self-signed-2307055185
```

```
!
crypto pki trustpoint SLA-TrustPoint
 enrollment pkcs12
 revocation-check crl
!
crypto pki trustpoint sbc5
 enrollment terminal
 fqdn accessedge.tenant2.com
 subject-name cn=accessedge.tenant2.com
 subject-alt-name sbc5.tekvlabs.com,accessedge.tenant2.com
 revocation-check crl
 rsakeypair sbc5
!
crypto pki trustpoint sbc6
 enrollment pkcs12
 revocation-check crl
 rsakeypair sbc6
!
!
crypto pki certificate chain TP-self-signed-2307055185
 certificate self-signed 01
crypto pki certificate chain SLA-TrustPoint
 certificate ca 01

crypto pki certificate chain sbc5
 certificate 00DABCF4E9A4FB7CF7
 certificate ca 07
crypto pki certificate chain sbc6
 certificate 00BAB7A09A134933DF
 certificate ca 07
!
crypto pki certificate pool
 cabundle nvram:ios.p7b

!
voice service voip
 ip address trusted list
  ipv4 10.64.1.x
  ipv4 152.188.28.xxx
  ipv4 152.188.28.xxx
  ipv4 172.16.29.0 255.255.255.0
  ipv4 52.0.0.0 255.0.0.0

 rtcp keepalive
 address-hiding
 mode border-element
 media bulk-stats
 media disable-detailed-stats
 allow-connections sip to sip
 redundancy-group 1
 no supplementary-service sip refer
 no supplementary-service sip handle-replaces
```

```
 fax protocol pass-through g711alaw
 h323
 trace
 sip
  listen-port secure 5067

  early-offer forced
  g729 annexb-all
  sip-profiles inbound
  no call service stop
!
!
voice class uri 100 sip
 host 10.64.1.x
!
voice class uri 200 sip
 pattern sbc6.tekvizionlabs.com
!
voice class uri 600 sip
 host sbc5.tekvlabs.com
!
voice class uri 300 sip
 host ipv4:152.188.28.14x
 host ipv4:152.188.28.19x

voice class codec 100
 codec preference 1 g729r8
 codec preference 2 g711ulaw
 codec preference 3 g711alaw
!
voice class stun-usage 1
 stun usage ice lite

!

voice class sip-profiles 200
 rule 10 request ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061
(.*)" "sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300/\1"
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "192.65.79.1xx"
 rule 80 request ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
```

```
 rule 90 response ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
!
voice class sip-profiles 290
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AAA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AAA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300V/\1"
 rule 50 request ANY sdp-header Audio-Attribute modify "a=ice-.*"
"a=label:main-audio"
 rule 60 request ANY sdp-header Attribute modify "a=ice-.*"
"a=label:main-audio"
!
voice class sip-profiles 299
 rule 20 request OPTIONS sip-header Contact modify "<sip:(.*):"
"<sip:sbc6.tekvizionlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 100
 rule 10 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 20 request INVITE sip-header Referred-By modify "<sip:(.*)>"
"<sip:\1>\x0D\x0AP-ASSERTED-IDENTITY:<sip:\1>"



!
voice class sip-profiles 5304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA@"
"sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AAA"
"sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 8 response ANY sip-header Contact modify "@(.*):"
"@sbc6.tekvizionlabs.com:"
 rule 10 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC:Cisco CUBE/C8300V/\1"
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AAA@(.*)>"
"<sip:\u01>"
```

```
 rule 120 request INVITE sip-header To modify "<sip:\+AAA(.*)@.*>"
"<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
 rule 170 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "\u03"
!
voice class sip-profiles 399
 rule 10 request OPTIONS sip-header Contact modify "<sip:(.*):"
"<sip:sbc5.tekvlabs.com:"
 rule 30 request OPTIONS sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 600
 rule 10 request ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
 rule 20 response ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
 rule 30 request ANY sip-header SIP-Req-URI modify "sip:(.*):5061
(.*)" "sip:\1:5061;user=phone \2"
 rule 40 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC: Cisco CUBE/C8300/\1"
 rule 50 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300/\1"
 rule 60 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 70 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "192.65.79.1xx"
 rule 80 request ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
 rule 90 response ANY sdp-header Audio-Attribute modify
"(a=crypto:.*inline:[A-Za-z0-9+/=]+)" "\1|2^31"
!
voice class sip-profiles 690
 rule 10 request REFER sip-header From copy "@(.*com)" u04
 rule 15 request REFER sip-header From copy "sip:(sip.*com)" u04
 rule 20 request REFER sip-header Refer-To modify "sip:\+(.*)@.*:5061"
"sip:+AA\1@\u04:5061"
 rule 30 request REFER sip-header Refer-To modify "<sip:sip.*:5061"
"<sip:+AA@\u04:5061"
 rule 40 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: Cisco CUBE/C8300V/\1"
!
voice class sip-profiles 6304
 rule 1 request INVITE sip-header SIP-Req-URI modify "sip:\+AA@"
"sip:"
 rule 2 request INVITE sip-header SIP-Req-URI modify "sip:\+AA"
"sip:+"
 rule 5 request ANY sip-header Contact modify "@(.*):"
"@sbc5.tekvlabs.com:"
```

```
 rule 7 request ANY sdp-header Audio-Attribute modify "a=sendonly"
"a=inactive"
 rule 10 request ANY sip-header User-Agent modify "(IOS.*)"
"\1\x0D\x0AX-MS-SBC:Cisco CUBE/C8300V/\1"
 rule 20 response ANY sip-header Server modify "(IOS.*)" "\1\x0D\x0AX-
MS-SBC: CiscoUBE/C8300V/\1"
 rule 30 request INVITE sip-header SIP-Req-URI copy "@(.*:5061)" u01
 rule 40 request INVITE sip-header From copy "@(.*)>" u02
 rule 110 request INVITE sip-header To modify "<sip:\+AA@(.*)>"
"<sip:\u01>"
 rule 120 request INVITE sip-header To modify "<sip:\+AA(.*)@.*>"
"<sip:+\1@\u01>"
 rule 130 request ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 140 response ANY sip-header Contact modify "@.*:" "@\u02:"
 rule 160 response 200 sdp-header Session-Owner copy "IN IP4 (.*)" u03
 rule 170 response 200 sdp-header Audio-Connection-Info modify
"0.0.0.0" "\u03"
!
voice class sip-hdr-passthrulist 290
 passthru-hdr Referred-By
!
!
voice class e164-pattern-map 101
 description towardsPSTNLumen
  e164 6T
 !
!
voice class e164-pattern-map 201
 description towardsPSTNVerizon
  e164 5T
 !
!
voice class e164-pattern-map 1002
 description towards MSTeams tenant1
  e164 +19725980114
  e164 +19725980115
  e164 +19725980116

 !
!
voice class e164-pattern-map 6002
 description towards MSTeams tenant2
  e164 +14698384746
  e164 +14698384747
 !
!
voice class sip-options-keepalive 100
 description Keepalive MStenant1
 up-interval 5
 transport tcp tls
 sip-profiles 299
!
```

```
voice class sip-options-keepalive 600
 description Keepalive MSTenant2
 up-interval 5
 transport tcp tls
 sip-profiles 399
!
voice class tenant 200
  tls-profile 100
  listen-port secure 5061
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc6.tekvizionlabs.com
  session transport tcp tls
  session refresh
  no referto-passing
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
  no pass-thru content custom-sdp
  sip-profiles 200
  sip-profiles 290 inbound
  early-offer forced

!
voice class tenant 100
  session transport tcp
  session refresh
  url sip
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.1

  bind media source-interface GigabitEthernet0/0/1.1
  no pass-thru content custom-sdp
  privacy-policy passthru
!
voice class tenant 600
  tls-profile 600
  listen-port secure 5063
  no remote-party-id
  srtp-crypto 200
  localhost dns:sbc5.tekvlabs.com
  session transport tcp tls
  no session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  pass-thru headers 290
  no pass-thru content custom-sdp
  sip-profiles 600
  sip-profiles 290 inbound
  privacy-policy passthru
```

```
!
voice class tenant 300
  session transport udp
  session refresh
  error-passthru
  bind control source-interface GigabitEthernet0/0/1.2

  bind media source-interface GigabitEthernet0/0/1.2
  no pass-thru content custom-sdp

!
voice class srtp-crypto 200
 crypto 1 AES_CM_128_HMAC_SHA1_80
!
voice class tls-profile 100
 description MS Tenant1
 trustpoint sbc6
!
voice class tls-profile 600
 description MSTenant2
 trustpoint sbc5
!
!
voice translation-rule 100
 rule 1 /^\(.*\)/ /+1\1/
!
voice translation-rule 101



 rule 1 /^6\(.*\)/ /\1/
!
voice translation-rule 200
 rule 1 /^\(4.........\)/ /+1\1/
!
voice translation-rule 201
 rule 1 /^5\(.*\)/ /\1/
!
voice translation-rule 1001
 rule 1 /^\+1\(.*\)/ /6\1/
!
voice translation-rule 6001
 rule 1 /^\+1\(.*\)/ /5\1/
!
!
voice translation-profile 100
 translate calling 100
 translate called 100
!
voice translation-profile 1001
 translate calling 1001
```

```
 translate called 1001
!
voice translation-profile 101
 translate calling 101
 translate called 101
!
voice translation-profile 200
 translate calling 200
 translate called 200
!
voice translation-profile 201
 translate calling 201
 translate called 201
!
voice translation-profile 6001
 translate calling 6001
 translate called 6001
!
!
!
!
voice-card 0/1
 dsp services dspfarm
 no watchdog
!
no license feature hseck9
license udi pid C8300-1N1S-6T sn Fxxxxxzz
license boot level network-essentials addon dna-essentials
memory free low-watermark processor 62769
!
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
enable secret 9
$9$OMYFKhiUaTSaeU$rHfTLyddXC7Z/Ch68qEBbbydLtS7ozoZg5rHalfC3so
!
!
username Admin privilege 15 password 7 111D1C0E2143115D5424
!
redundancy
 mode none
 application redundancy
  group 1
   name cube-ha
   priority 100 failover threshold 75
   timers delay 30 reload 60
   control GigabitEthernet0/0/0 protocol 1
   data GigabitEthernet0/0/0
   track 1 shutdown
   track 2 shutdown
   track 3 shutdown
```

```
track 1 interface GigabitEthernet0/0/1.1 line-protocol
!
track 2 interface GigabitEthernet0/0/1.2 line-protocol
!
track 3 interface GigabitEthernet0/0/2 line-protocol
!
!
interface GigabitEthernet0/0/0
 description connects to HA
 ip address 10.64.5.235 255.255.0.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/1.1
 encapsulation dot1Q 3811
 ip address 10.80.11.137 255.255.255.0
 redundancy rii 16
 redundancy group 1 ip 10.80.11.136 exclusive
!
interface GigabitEthernet0/0/1.2
 encapsulation dot1Q 1506
 ip address 199.182.124.25x 255.255.255.192
 redundancy rii 18
 redundancy group 1 ip 199.182.124.24x exclusive
!
interface GigabitEthernet0/0/2
 ip address 192.65.79.12x 255.255.255.224
 negotiation auto
 redundancy rii 17
 redundancy group 1 ip 192.65.79.10x exclusive
!
interface GigabitEthernet0/0/3
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/4
 no ip address
 negotiation auto
!
interface GigabitEthernet0/0/5
 no ip address
 negotiation auto
!
interface Service-Engine0/1/0
!
ip http server
ip http authentication local
```

```
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.80.11.1

ip route 152.188.28.0 255.255.255.0 199.182.124.193
ip route 172.16.0.0 255.255.0.0 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
!

dspfarm profile 1 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 codec g722-64
 codec opus
 maximum sessions 9
 associate application CUBE

!
dial-peer voice 6001 voip
 description Inbound MS teams tenant2
 translation-profile incoming 6001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session transport tcp tls
 incoming uri request 600
 incoming uri to 600
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip tenant 600
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
dial-peer voice 201 voip
 description outgoing dial-peer to PSTN Verizon
 translation-profile outgoing 201
 rtp payload-type comfort-noise 13
```

```
 session protocol sipv2
 session target ipv4:152.188.28.147:5232
 session transport udp
 destination e164-pattern-map 201
 voice-class codec 100
 voice-class sip profiles 100
 voice-class sip tenant 300
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 8333 voip
 description TEST for REFER_teams1
 destination-pattern +AAAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 5304
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2

 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
!
dial-peer voice 100 voip
 description Incoming dial-peer from PSTNLumen
 translation-profile incoming 100
 rtp payload-type comfort-noise 13

 session protocol sipv2
 incoming uri from 100
 voice-class codec 100
 voice-class sip tenant 100
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 101 voip
 description outgoing dial-peer to IP PSTNLumen
 translation-profile outgoing 101
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target ipv4:10.64.1.72:5060
```

```
 session transport tcp
 destination e164-pattern-map 101
 voice-class codec 100
 voice-class sip tenant 100
 voice-class sip options-keepalive
 voice-class sip bind control source-interface GigabitEthernet0/0/1.1
 voice-class sip bind media source-interface GigabitEthernet0/0/1.1
 dtmf-relay rtp-nte
no vad
!
dial-peer voice 6002 voip
 description Outbound MS Teams tenant2
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 6002
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip rel1xx disable
 voice-class sip asserted-id pai
 voice-class sip tenant 600
 voice-class sip options-keepalive profile 600
 dtmf-relay rtp-nte
 srtp
 !
dial-peer voice 200 voip
 description Incoming dial-peer from PSTN Verizon
 translation-profile incoming 200
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session transport udp
 incoming uri request 300
 voice-class codec 100
 voice-class sip tenant 300
 voice-class sip bind control source-interface GigabitEthernet0/0/1.2
 voice-class sip bind media source-interface GigabitEthernet0/0/1.2
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 1002 voip
 description outbound to Teams1
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target dns:sip.pstnhub.microsoft.com
 session transport tcp tls
 destination e164-pattern-map 1002
 voice-class codec 100
 voice-class sip tenant 200
 voice-class sip options-keepalive profile 100
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
```

```
 voice-class sip bind media source-interface GigabitEthernet0/0/2

 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
dial-peer voice 1001 voip
 description inbound from Microsoft Phone System
 translation-profile incoming 1001
 rtp payload-type comfort-noise 13
 session protocol sipv2
 incoming uri to 200
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip tenant 200
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 dtmf-relay rtp-nte
 rtcp keepalive
 srtp
!
dial-peer voice 6333 voip
 description TEST for REFER_teams2
 destination-pattern +AAT
 rtp payload-type comfort-noise 13
 session protocol sipv2
 session target sip-uri
 session transport tcp tls
 voice-class codec 100
 voice-class stun-usage 1
 voice-class sip profiles 6304
 voice-class sip tenant 600
 voice-class sip session refresh
 voice-class sip bind control source-interface GigabitEthernet0/0/2
 voice-class sip bind media source-interface GigabitEthernet0/0/2
 voice-class sip requri-passing
 dtmf-relay rtp-nte
 srtp
!
sip-ua
 retry invite 1
 transport tcp tls v1.2
 handle-replace
!
line con 0
 exec-timeout 5 0
 password 7 071B24
 logging synchronous
 login
 stopbits 1
line aux 0
```

```
line vty 0 4
 exec-timeout 60 0
 password 7 071B24
 logging synchronous
 login

 transport input telnet
line vty 5
 exec-timeout 5 0
 password 7 120D001
 logging synchronous
 login
 transport input telnet
line vty 6 14
 login
 transport input ssh
!
call-home
 ! If contact email address in call-home is configured as sch-smart-
licensing@cisco.com
 ! the email address configured in Cisco Smart License Portal will be
used as contact email address to send SCH notifications.
 contact-email-addr sch-smart-licensing@cisco.com
 profile "CiscoTAC-1"
  active
  destination transport-method http
!
end
```

Show commands

Dial-peer status to Microsoft Direct Routing tenant

```
8K_CUBE#sh dial-peer voip keepalive status
 TAG      TENANT   DESTINATION                    OOD-SessID   PRI     WT      STATUS
 6002     600      dns:sip.pstnhub.microsoft.com                               active
                    ipv4:52.114.132.46:5061       182565       -       -       active
                    ipv4:52.114.76.76:5061        182566       -       -       active
                    ipv4:52.114.32.169:5061       182567       -       -       active
                    ipv4:52.114.16.74:5061        182568       -       -       active
 1002        200   dns:sip.pstnhub.microsoft.com                               active
                    ipv4:52.114.132.46:5061       182561       -       -       active
                    ipv4:52.114.76.76:5061        182562       -       -       active
                    ipv4:52.114.32.169:5061       182563       -       -       active
                    ipv4:52.114.16.74:5061        182564       -       -       active
```

```
Note: Command introduced from 17.9.1a IOS
```

Dial-peer Summary

```
8K_CUBE#show dial-peer voice summary
dial-peer hunt 0
             AD                         PRE PASS SESS-SER-GRP\OUT
TAG  TYPE MIN OPER PREFIX DEST-PATTERN FER THRU SESS-TARGET  STAT PORT  KEEPALIVE VRF
6001 voip up  up                       0  syst                                    NA
201  voip up  up          map:201      0  syst ipv4:152.188.28.xxx:     active    NA
8333 voip up  up          +AAAT        0  syst sip-uri                            NA
100  voip up  up                       0  syst                                    NA
101  voip up  up          map:101      0  syst ipv4:10.64.1.x:5060      active    NA
6002 voip up  up          map:6002     0  syst dns:sip.pstnhub.micr     active    NA
200  voip up  up                       0  syst                                    NA
1002 voip up  up          map:1002     0  syst dns:sip.pstnhub.micr     active    NA
1001 voip up  up                       0  syst                                    NA
6333 voip up  up          +AAT         0  syst sip-uri                            NA
 For server-grp details please execute command:show voice class server-group <tag_id>
 To see complete session target for ipv6 use 'sh running-config | section dial-peer
<tag>
```

Voice class Keepalive sip Options

```
8K_CUBE#show voice class sip-options-keepalive
Voice class sip-options-keepalive: 100          AdminStat: Up
 Description: Keepalive MStenant1
 Transport: tcp tls             Sip Profiles: 299
 Interval(seconds) Up: 5                Down: 30
 Retry: 5

  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  1002                                          Active          19

 OOD SessID: 182761             OOD Stat: Active
  Target: ipv4:52.114.132.46:5061
  Transport: tcp tls            Sip Profiles: 299

 OOD SessID: 182762             OOD Stat: Active
  Target: ipv4:52.114.76.76:5061
  Transport: tcp tls            Sip Profiles: 299

 OOD SessID: 182763             OOD Stat: Active
  Target: ipv4:52.114.32.169:5061
  Transport: tcp tls            Sip Profiles: 299

 OOD SessID: 182764             OOD Stat: Active
  Target: ipv4:52.114.16.74:5061
  Transport: tcp tls            Sip Profiles: 299

------------------------------------------------------
Voice class sip-options-keepalive: 600          AdminStat: Up
 Description: Keepalive MSTenant2
 Transport: tcp tls             Sip Profiles: 399
 Interval(seconds) Up: 5                Down: 30
 Retry: 5

  Peer Tag      Server Group    OOD SessID      OOD Stat        IfIndex
  --------      ------------    ----------      --------        -------
  6002                                          Active          17
```

```
 OOD SessID: 182765              OOD Stat: Active
  Target: ipv4:52.114.132.46:5061
  Transport: tcp tls             Sip Profiles: 399

 OOD SessID: 182766              OOD Stat: Active
  Target: ipv4:52.114.76.76:5061
  Transport: tcp tls             Sip Profiles: 399

 OOD SessID: 182767              OOD Stat: Active
  Target: ipv4:52.114.32.169:5061
  Transport: tcp tls             Sip Profiles: 399

 OOD SessID: 182768              OOD Stat: Active
  Target: ipv4:52.114.16.74:5061
  Transport: tcp tls             Sip Profiles: 399

------------------------------------------------------
For destination configured as DNS - please execute: show dial-peer voip keepalive
status
```

SIP-ua connection details

```
8K_CUBE# show sip-ua connections tcp tls detail
Total active connections      : 12
No. of send failures          : 9
No. of remote closures        : 128
No. of conn. failures         : 66
No. of inactive conn. ageouts : 0
Max. tls send msg queue size of 1, recorded for 52.114.16.74:27328
TLS client handshake failures : 0
TLS server handshake failures : 39

---------Printing Detailed Connection Report---------
Note:
 ** Tuples with no matching socket entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port>'
      to overcome this error condition
 ++ Tuples with mismatched address/port entry
    - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port> id <connid>'
      to overcome this error condition
```

```
 * Connections with SIP OAuth ports

Remote-Agent:52.114.76.76, Connections-Count:4
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address
TLS-Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5061    5822 Established           0 192.65.79.1xx:47806
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        56448    5827 Established           0 192.65.79.1xx:5061
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        56449    5833 Established           0 192.65.79.1xx:5063
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600
         5061    5822 Established           0 192.65.79.1xx:47906
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600



Remote-Agent:52.114.132.46, Connections-Count:4
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address
TLS-Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5061    3901 Established           0 192.65.79.1xx:30800
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        52176    3908 Established           0 192.65.79.1xx:5061
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        52177    3916 Established           0 192.65.79.1xx:5063
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600
         5061    3901 Established           0 192.65.79.1xx:30900
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600



Remote-Agent:52.114.32.169, Connections-Count:4
  Remote-Port Conn-Id Conn-State  WriteQ-Size            Local-Address
TLS-Version Cipher                         Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5061    3902 Established           0 192.65.79.1xx:50336
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
         3272    3909 Established           0 192.65.79.1xx:5061
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
         3273    3918 Established           0 192.65.79.1xx:5063
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600
```

```
5061    3902 Established           0 192.65.79.1xx:50636
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600


Remote-Agent:52.114.16.74, Connections-Count:4
  Remote-Port Conn-Id Conn-State  WriteQ-Size           Local-Address
TLS-Version Cipher                          Curve Tenant
  =========== ======= =========== =========== ======================================
=========== ============================== ===== ======
         5061    3903 Established           0 192.65.79.1xx:55734
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        52160    3910 Established           0 192.65.79.1xx:5061
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   200
        52161    3920 Established           0 192.65.79.1xx:5061
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600
         5061    3903 Established           0 192.65.79.1xx:55994
TLSv1.2    ECDHE-RSA-AES256-GCM-SHA384 P-384   600




-------------- SIP Transport Layer Listen Sockets ---------------
  Conn-Id             Local-Address                      Tenant
 ==========    ===========================              ========
  0             [0.0.0.0]:5067:                               0
  6             [192.65.79.1xx]:5061:                       200
  10            [199.182.124.240]:5067:                       0
  11            [10.80.11.136]:5067:                          0
  13            [192.65.79.1xx]:5067:                         0
  295           [192.65.79.1xx]:5063:                       600
```

Show voip trace tenant

INVITE from SBC to Microsoft – Tenant 1

```
8K_CUBE#show voip trace tenant 200


------------------ Cover Buffer ---------------
Search-key      = +12145509073:+19725980114:11029010
  Timestamp       = *May 29 14:22:50.015
  Buffer-Id       = 997
  CallID          = 11029010
  Peer-CallID     = 11029009
  Correlator      = 527
```

```
  Called-Number  = +19725980114
  Calling-Number = +12145509073
  SIP CallID     = 2014899F-FD6311ED-881AF3EB-C64A5E9B@sbc6.tekvizionlabs.com
  SIP Session ID = f5baf63330445e07bd971900ac96c7fc
  GUID           = 359ABC13803A
  Tenant         = 200

Sent: SIP TLS message from 192.65.79.1xx:5061 to 52.114.132.46:5061
INVITE sip:+19725980114@sip.pstnhub.microsoft.com:5061;user=phone SIP/2.0
Via: SIP/2.0/TLS 192.65.79.1xx:5061;branch=z9hG4bK33118D0
From: "Javid Mohammad" <sip:+12145509073@sbc6.tekvizionlabs.com>;tag=A5F6B283-409
To: <sip:+19725980114@sip.pstnhub.microsoft.com>
Date: Mon, 29 May 2023 14:22:50 GMT
Call-ID: 2014899F-FD6311ED-881AF3EB-C64A5E9B@sbc6.tekvizionlabs.com
Supported: 100rel,timer,resource-priority,replaces
Min-SE:  1800
Cisco-Guid: 0899333139-4251128301-2151310358-2368317232

User-Agent: Cisco-SIPGateway/IOS-17.9.1a
X-MS-SBC: Cisco CUBE/C8300/IOS-17.9.1a
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1685370170
Contact: <sip:+12145509073@sbc6.tekvizionlabs.com:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 69
P-Asserted-Identity: "Javid Mohammad" <sip:+12145509073@sbc6.tekvizionlabs.com>
Session-ID: f5baf63330445e07bd971900ac96c7fc;remote=00000000000000000000000000000000
Session-Expires:  1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 609

v=0
o=CiscoSystemsSIP-GW-UserAgent 5505 8080 IN IP4 192.65.79.1xx
s=SIP Call
c=IN IP4 192.65.79.1xx
```

```
t=0 0
a=ice-lite
m=audio 8006 RTP/SAVP 0 8 101 13
c=IN IP4 192.65.79.1xx
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:13 CN/8000
a=ptime:20
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=candidate:1 1 UDP 2130706431 192.65.79.1xx 8006 typ host
a=candidate:1 2 UDP 2130706430 192.65.79.1xx 8007 typ host
a=rtcp:8007 IN IP4 192.65.79.1xx
a=ice-ufrag:JXKL
a=ice-pwd:5wbYTdbKeZz0KXDmat31vy
```

INVITE from SBC to Microsoft – Tenant 2

```
8K_CUBE#show voip trace tenant 600

------------------ Cover Buffer ---------------
Search-key     = +12145509073:+14698384746:11029277
  Timestamp      = *May 29 14:24:10.884
  Buffer-Id      = 1001
  CallID         = 11029277
  Peer-CallID    = 11029276
  Correlator     = 529
  Called-Number  = +14698384746
  Calling-Number = +12145509073
  SIP CallID     = 5048294F-FD6311ED-892AF3EB-C64A5E9B@sbc5.tekvlabs.com
  SIP Session ID = 0860c2d288d85d23b566c2c0a7962d07
  GUID           = 5047DB3B8924
  Tenant         = 600
-----------------------------------------------



Sent: SIP TLS message from 192.65.79.1xx:5063 to 52.114.132.46:5061
INVITE sip:+14698384746@sip.pstnhub.microsoft.com:5061;user=phone SIP/2.0
```

```
Via: SIP/2.0/TLS 192.65.79.1xx:5063;branch=z9hG4bK3BD34D
From: "214 5509073" <sip:+12145509073@sbc5.tekvlabs.com>;tag=A5F7EE93-1FCF
To: <sip:+14698384746@sip.pstnhub.microsoft.com>
Date: Mon, 29 May 2023 14:24:10 GMT
Call-ID: 5048294F-FD6311ED-892AF3EB-C64A5E9B@sbc5.tekvlabs.com
Supported: timer,resource-priority,replaces
Min-SE:  1800
Cisco-Guid: 1346886459-4251128301-2300900331-3326762651
User-Agent: Cisco-SIPGateway/IOS-17.9.1a
X-MS-SBC: Cisco CUBE/C8300/IOS-17.9.1a
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1685370250
Contact: <sip:+12145509073@sbc5.tekvlabs.com:5063;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
P-Asserted-Identity: "214 5509073" <sip:+12145509073@sbc5.tekvlabs.com>
Session-ID: 0860c2d288d85d23b566c2c0a7962d07;remote=00000000000000000000000000000000
Session-Expires:  1800
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 585

v=0
o=CiscoSystemsSIP-GW-UserAgent 2435 2402 IN IP4 192.65.79.1xx
s=SIP Call
c=IN IP4 192.65.79.1xx
t=0 0

a=ice-lite
m=audio 8014 RTP/SAVP 0 8 101
c=IN IP4 192.65.79.1xx
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

```
a=crypto:1 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=candidate:1 1 UDP 2130706431 192.65.79.1xx 8014 typ host
a=candidate:1 2 UDP 2130706430 192.65.79.1xx 8015 typ host
a=rtcp:8015 IN IP4 192.65.79.1xx
a=ice-ufrag:YRGj
a=ice-pwd:EVerXlyfNt20fKELvV2oss
```

INVITE from Microsoft to SBC – Tenant 1

```
8K_CUBE#show voip trace tenant 600
------------------ Cover Buffer ---------------
Search-key     = +19725980114:+12145509073:11028833
  Timestamp      = *May 29 14:21:58.344
  Buffer-Id      = 994
  CallID         = 11028833
  Peer-CallID    = 11028834
  Correlator     = 526
  Called-Number  = +12145509073
  Calling-Number = +19725980114
  SIP CallID     = d7f75eee23585f3f96324f3857d9a9e1
  SIP Session ID = 8c91622a89e156cf852dfc30db707c8c
  GUID           = 01482A518764
  Tenant         = 200
-----------------------------------------------
6694: *May 29 14:21:58.344: //11028833/01482A518764/CUBE_VT/SIP/Msg/ccsipDisplayMsg:
Received: SIP TLS message from 52.114.148.0:38848 to 192.65.79.1xx:5061
INVITE sip:+12145509073@sbc6.tekvizionlabs.com:5061;user=phone;transport=tls SIP/2.0
FROM: "Wxccuser
2"<sip:+19725980114@sip.pstnhub.microsoft.com:5061;user=phone>;tag=7da967eb135f4bb3ac
892b392119d08e
TO: <sip:+12145509073@sbc6.tekvizionlabs.com:5061;user=phone>
CSEQ: 1 INVITE
CALL-ID: d7f75eee23585f3f96324f3857d9a9e1
MAX-FORWARDS: 70
VIA: SIP/2.0/TLS 52.114.148.0:5061;branch=z9hG4bKb20b6368

RECORD-ROUTE: <sip:sip-du-a-us.pstnhub.microsoft.com:5061;transport=tls;lr>
CONTACT: <sip:api-du-a-asse.pstnhub.microsoft.com:443;x-i=42cf753d-5041-495a-a20d-
443b132386ed;x-
c=d7f75eee23585f3f96324f3857d9a9e1/d/8/beca0ec31edb40e4ba0b90f479bf5854>
```

```
CONTENT-LENGTH: 2277
SUPPORTED: histinfo
USER-AGENT: Microsoft.PSTNHub.SIPProxy v.2023.5.11.5 i.USWE2.3
CONTENT-TYPE: application/sdp
ALLOW: INVITE,ACK,OPTIONS,CANCEL,BYE,NOTIFY

v=0
o=- 0 0 IN IP4 52.113.11.112
s=session
c=IN IP4 52.113.11.112
b=CT:2500
t=0 0
m=audio 3479 RTP/SAVP 104 102 9 111 18 0 8 103 97 13 118 119 101
a=rtcp:3480
a=ice-ufrag:7IQL
a=ice-pwd:8ihj4Uo0fedz+Z/SOOd6nTea
a=rtcp-mux
a=candidate:6 1 UDP 1258288639 52.113.11.112 3479 typ relay raddr 14.142.185.162
rport 50016 MTURNID 16436065407760370922
a=candidate:6 2 UDP 1258288126 52.113.11.112 3480 typ relay raddr 14.142.185.162
rport 57697 MTURNID 24881398494987185
a=candidate:1 1 UDP 2130706431 172.16.29.81 50016 typ host
a=candidate:1 2 UDP 2130705918 172.16.29.81 50017 typ host
a=candidate:2 1 tcp-act 2121006078 172.16.29.81 50000 typ host
a=candidate:2 2 tcp-act 2121006078 172.16.29.81 50000 typ host
a=candidate:5 1 UDP 1694496767 14.142.185.162 50016 typ srflx raddr 172.16.29.81
rport 50016
a=candidate:5 2 UDP 1694496254 14.142.185.162 57697 typ srflx raddr 172.16.29.81
rport 50017
a=candidate:7 1 tcp-act 1684795902 14.142.185.162 50011 typ srflx raddr 172.16.29.81
rport 50011
a=candidate:7 2 tcp-act 1684795902 14.142.185.162 50011 typ srflx raddr 172.16.29.81
rport 50011
a=candidate:8 1 tcp-pass 1248194558 52.113.10.183 55578 typ relay raddr
14.142.185.162 rport 50011
a=candidate:8 2 tcp-pass 1248194558 52.113.10.183 55578 typ relay raddr
14.142.185.162 rport 50011
a=candidate:9 1 tcp-act 1248587262 52.113.10.183 55578 typ relay raddr 14.142.185.162
rport 50011
a=candidate:9 2 tcp-act 1248587262 52.113.10.183 55578 typ relay raddr 14.142.185.162
rport 50011
```

```
a=crypto:1 AES_CM_128_HMAC_SHA1_32
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:2 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:3 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=rtpmap:104 SILK/16000
a=fmtp:104 useinbandfec=0; usedtx=0
a=rtpmap:102 opus/48000/2
a=fmtp:102 useinbandfec=1; minptime=10
a=rtpmap:9 G722/8000

a=rtpmap:111 SIREN/16000
a=fmtp:111 bitrate=16000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:103 SILK/8000
a=fmtp:103 useinbandfec=0; usedtx=0
a=rtpmap:97 RED/8000
a=rtpmap:13 CN/8000
a=rtpmap:118 CN/16000
a=rtpmap:119 CN/24000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=maxptime:200
```

INVITE from Microsoft to SBC – Tenant 2

```
------------------ Cover Buffer ---------------
Search-key     = +14698384746:+12145509073:11029142
  Timestamp      = *May 29 14:23:32.179
  Buffer-Id      = 998
  CallID         = 11029142
  Peer-CallID    = 11029143
  Correlator     = 528
  Called-Number  = +12145509073
  Calling-Number = +14698384746
  SIP CallID     = bafd46fbaa0253fdac3525d5ea24d330
```

```
  SIP Session ID = e75e2c39028c5046a47bc8f362b01a26
  GUID           = 39361943889B
  Tenant         = 600
-----------------------------------------------
7282: *May 29 14:23:32.179: //11029142/39361943889B/CUBE_VT/SIP/Msg/ccsipDisplayMsg:
Received: SIP TLS message from 52.114.148.0:48768 to 192.65.79.1xx:5063
INVITE sip:+12145509073@sbc5.tekvlabs.com:5063;user=phone;transport=tls SIP/2.0
FROM: "Cisco
user2"<sip:+14698384746@sip.pstnhub.microsoft.com:5061;user=phone>;tag=a47de96b131243
fcb9b61f263725a71e
TO: <sip:+12145509073@sbc5.tekvlabs.com:5063;user=phone>
CSEQ: 1 INVITE
CALL-ID: bafd46fbaa0253fdac3525d5ea24d330
MAX-FORWARDS: 70
VIA: SIP/2.0/TLS 52.114.148.0:5061;branch=z9hG4bKf317a47b
RECORD-ROUTE: <sip:sip-du-a-us.pstnhub.microsoft.com:5061;transport=tls;lr>
CONTACT: <sip:api-du-a-jawe.pstnhub.microsoft.com:443;x-i=2aef5cd0-f2e6-4510-a1a7-
ecbb361d20b3;x-
c=bafd46fbaa0253fdac3525d5ea24d330/d/8/d1c33ba89f6b48d8a4003efa30d1ad35>
CONTENT-LENGTH: 2520
SUPPORTED: histinfo
USER-AGENT: Microsoft.PSTNHub.SIPProxy v.2023.5.11.5 i.USWE2.6
CONTENT-TYPE: application/sdp
ALLOW: INVITE,ACK,OPTIONS,CANCEL,BYE,NOTIFY
P-ASSERTED-IDENTITY: <tel:+14698384746>,<sip:ciscouser2@tenant2.com>
PRIVACY: id

v=0
o=- 0 0 IN IP4 52.115.240.168
s=session
c=IN IP4 52.115.240.168
b=CT:2500
t=0 0
m=audio 51168 RTP/SAVP 104 102 9 111 18 0 8 103 97 13 118 119 101
a=rtcp:53745
a=ice-ufrag:IQXL
a=ice-pwd:8ihUo0vfdz+Z/SOOd6nuZTe7

a=rtcp-mux
```

```
a=candidate:4 1 UDP 1258289663 52.115.240.168 51168 typ relay raddr 14.142.185.162
rport 5165
a=candidate:4 2 UDP 1258289150 52.115.240.168 53745 typ relay raddr 14.142.185.162
rport 50015
a=candidate:1 1 UDP 2130706431 172.16.31.219 50014 typ host
a=candidate:1 2 UDP 2130705918 172.16.31.219 50015 typ host
a=candidate:2 1 tcp-act 2121006078 172.16.31.219 50000 typ host
a=candidate:2 2 tcp-act 2121006078 172.16.31.219 50000 typ host
a=candidate:3 1 UDP 1694497791 14.142.185.162 5165 typ srflx raddr 172.16.31.219
rport 50014
a=candidate:3 2 UDP 1694497278 14.142.185.162 50015 typ srflx raddr 172.16.31.219
rport 50015
a=candidate:5 1 tcp-act 1684796926 14.142.185.162 50002 typ srflx raddr 172.16.31.219
rport 50002
a=candidate:5 2 tcp-act 1684796926 14.142.185.162 50002 typ srflx raddr 172.16.31.219
rport 50002
a=candidate:6 1 tcp-pass 1248195582 52.115.240.161 56132 typ relay raddr
14.142.185.162 rport 50002
a=candidate:6 2 tcp-pass 1248195582 52.115.240.161 56132 typ relay raddr
14.142.185.162 rport 50002
a=candidate:7 1 tcp-act 1248588286 52.115.240.161 56132 typ relay raddr
14.142.185.162 rport 50002
a=candidate:7 2 tcp-act 1248588286 52.115.240.161 56132 typ relay raddr
14.142.185.162 rport 50002
a=x-candidate-info:4 network-type=WLAN
a=x-candidate-info:1 network-type=WLAN
a=x-candidate-info:2 network-type=WLAN
a=x-candidate-info:3 network-type=WLAN
a=x-candidate-info:5 network-type=WLAN
a=x-candidate-info:6 network-type=WLAN
a=x-candidate-info:7 network-type=WLAN
a=crypto:1 AES_CM_128_HMAC_SHA1_32
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:2 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=crypto:3 AES_CM_128_HMAC_SHA1_80
inline:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
a=rtpmap:104 SILK/16000
a=fmtp:104 useinbandfec=0; usedtx=0
a=rtpmap:102 opus/48000/2
a=fmtp:102 useinbandfec=1; minptime=10
a=rtpmap:9 G722/8000
```

```
a=rtpmap:111 SIREN/16000
a=fmtp:111 bitrate=16000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:103 SILK/8000
a=fmtp:103 useinbandfec=0; usedtx=0
a=rtpmap:97 RED/8000
a=rtpmap:13 CN/8000
a=rtpmap:118 CN/16000
a=rtpmap:119 CN/24000
a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16
a=ptime:20
a=maxptime:200
```

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