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# Ribbon SBC Edge SWe Lite R9.0 Interop with Anywhere365 and Microsoft Teams Direct Routing : Interoperability Guide

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# Interoperable Vendors

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## Document Overview

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This document depicts the configuration details for Ribbon's SBC SWe Lite interworking & compliance with Anywhere365 and Microsoft Teams.

### About Ribbon SBC SWe Lite

The Ribbon Session Border Controller Software Edition Lite (SBC SWe Lite) provides best-in class communications security. The SBC SWe Lite dramatically simplifies the deployment of robust communications security services for SIP Trunking, Direct Routing, and Cloud UC services. SBC SWe Lite operates natively in the Azure and AWS Cloud as well as on virtual machine platforms including Microsoft Hyper-V, VMware and Linux KVM.

### About Anywhere365

Anywhere365 Contact Center and Enterprise Dialogue Management is able to leverage the capabilities of Microsoft Teams to route calls to Teams powered agents with all the rich features of Anywhere365, such as: Call recording, Real-time Translation, IVR, Supervisor, Reporting, Wallboards and many more.

## Scope

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This document provides configuration best practices for deploying Ribbon SBC SWe Lite when connecting with Anywhere365 and Microsoft Teams. Note that these are configuration best practices, and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design which best meets their requirements.

## Non-Goals

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It is not the goal of this guide to provide detailed configurations that will meet the requirements of every customer. Use this guide as a starting point and build the SBC configurations in consultation with network design and deployment engineers.

## Audience

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This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC and the third-party product. Navigating the third-party product as well as the Ribbon Ribbon SBC Edge GUI is required. Understanding the basic concepts of TLS/TCP/UDP, IP/Routing, SIP/RTP and SIP/SRTP is also necessary to complete the configuration and any required troubleshooting.

## Prerequisites

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The following aspects are required before proceeding with the interop:

- Ribbon SBC SWe Lite on VMware
- SBC SWe Lite License
- Public IP Addresses and Port
- Anywhere365 SIP trunks
  - Contact Anywhere365 for Domain, IP and Port information
- TLS Certificates for SBC SWe Lite

- For details refer "[License and TLS Certificates](#)" section in the document



#### Note

During this interop, the SIP trunk between Anywhere365 and Ribbon SBC SWe Lite has been configured with TLS and SRTP

## Product and Device Details

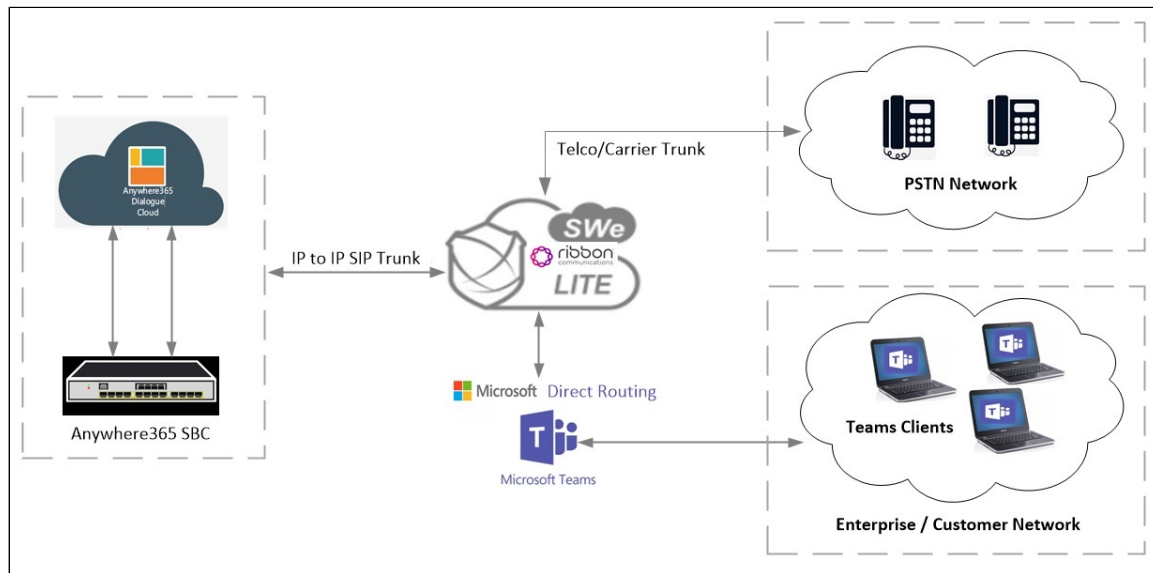
The configuration uses the following equipment and software:

**Table 1:** Requirements

Product	Equipment/ Devices	Software/ Firmware Version
Ribbon Communications	Ribbon SBC SWe Lite	9.0.1
Microsoft Corporation	Microsoft Teams ClientDesktop app	1.4.00.11161
	Microsoft Teams ClientMobile app	1416
Third Party Phones	Kapanga Softphone	1.00
	Phonerlite	2.93
Anywhere365	Anywhere365SIP trunks	NA
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

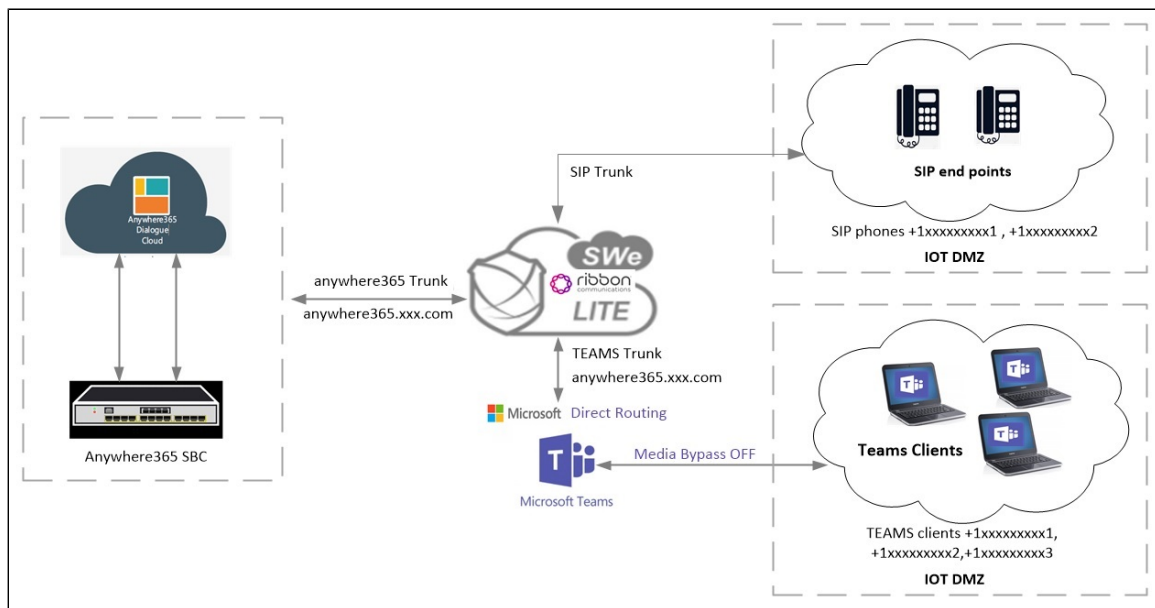
## Network Topology

### Deployment Topology

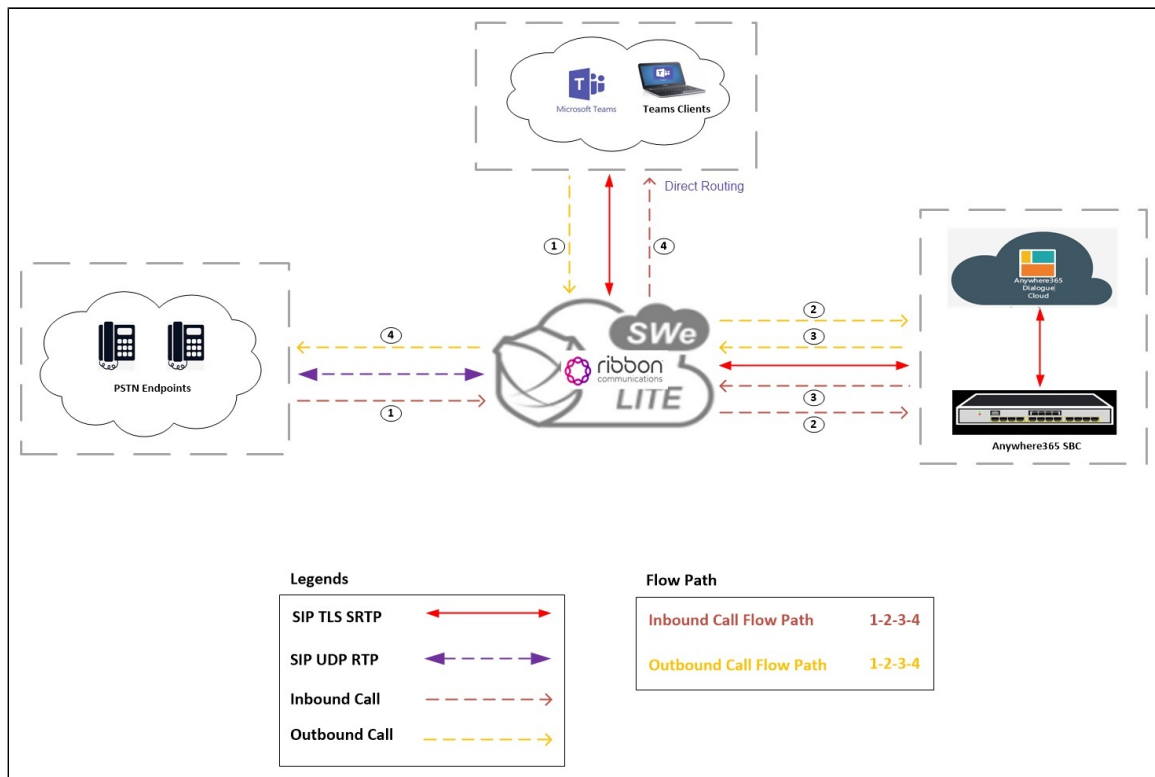


### IOT Lab Topology



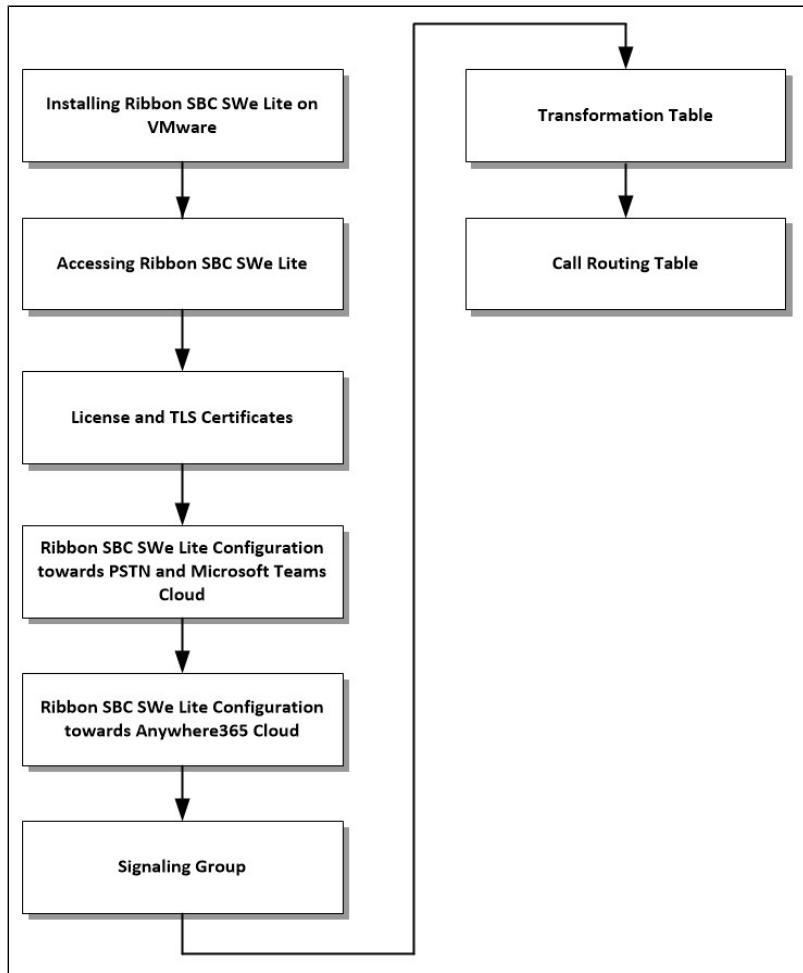


## Signaling and Media Flow



## Document Workflow

The sections in this document follow the sequence below. The reader is advised to complete each section for the successful configuration.



## Section A: Ribbon SBC SWe Lite Configuration

### Installing Ribbon SBC SWe Lite on VMware

The SBC SWe Lite is available for deployment on VMware. To deploy an SBC SWe Lite instance, refer to <https://doc.rbbn.com/display/UXDOC90/Installing+SBC+SWe+Lite+on+VMware+ESXi>. Once SWe Lite instance is successfully created on VMware, retrieve the allocated NAT Public IPs, Ethernet IPs and Management IP. For more details, visit the link given in the References section.

### Accessing Ribbon SBC SWe Lite

Open any browser and enter the SBC IP address.

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Click **Enter** and log in with a valid User ID and Password.



[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
- Signaling Groups
- Networking Interfaces
  - Logical Interfaces
    - Admin IP
    - Ethernet 1 IP**
    - Ethernet 2 IP
    - Ethernet 3 IP
    - Ethernet 4 IP
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration
- Emergency Services

Interface Name **Ethernet 1 IP**  
I/F Index **8**  
Alias   
Description   
Admin State **Enabled**

### Networking

MAC Address   
IP Addressing Mode **IPv4**

#### IPv4 Information

IP Assign Method **Static**  
Primary Address 172.  \* X.X.X.X  
Primary Netmask 255.255.255.0 \* X.X.X.X  
Media Next Hop IP 172.  \* X.X.X.X

## Ethernet 2 IP

Configure this Ethernet 2 interface as follows as per the requirement. This interface will face the Anywhere365 and Microsoftteams interface.

[Expand All](#) | [Collapse All](#) | [Reload](#)

- Call Routing
- Signaling Groups
- Networking Interfaces
  - Logical Interfaces
    - Admin IP
    - Ethernet 1 IP
    - Ethernet 2 IP**
    - Ethernet 3 IP
    - Ethernet 4 IP
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration
- Emergency Services

Interface Name **Ethernet 2 IP**  
I/F Index **9**  
Alias   
Description   
Admin State **Enabled**

### Networking

MAC Address **00:0c:29:ec:f8:22**  
IP Addressing Mode **IPv4**

#### IPv4 Information

IP Assign Method **Static**  
Primary Address 115.  \* X.X.X.X  
Primary Netmask 255.  \* X.X.X.X  
Media Next Hop IP 115.  \* X.X.X.X

## Configure Static Routes

Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network that you can only access through one point or one interface (single path access or default route).

### Destination IP

Specifies the destination IP address.

## Mask

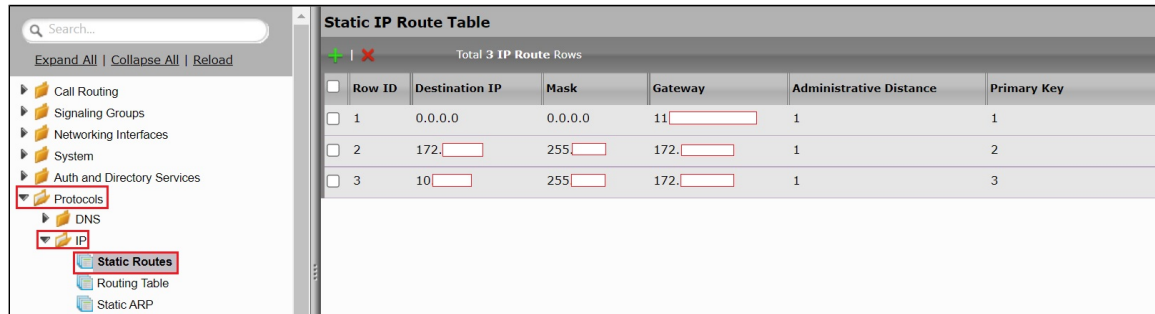
Specifies the network mask of the destination host or subnet. If the 'Destination IP Address' field and 'Mask' field are both 0.0.0.0, the static route is called the 'default static route'.

## Gateway

Specifies the IP address of the next-hop router to use for this static route.

## Metric

Specifies the cost of this route, and therefore indirectly specifies the preference of the route. Lower values indicate more preferred routes. The typical value is 1 for most static routes, indicating that static routes are preferred to dynamic routes.



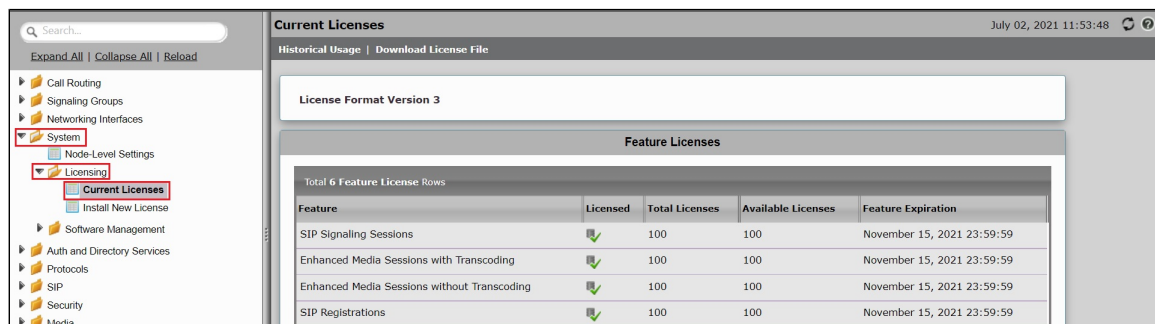
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key
1	0.0.0.0	0.0.0.0	11	1	1
2	172.	255.	172.	1	2
3	10	255	172.	1	3

# License and TLS Certificates

## View License

This section describes how to view the status of each license along with a copy of the license keys installed on your SBC. The **Feature Licenses** panel enables you to verify whether a feature is licensed, along with the number of remaining licenses available for a given feature at run-time.

From the **Setting** tab, navigate to **System > Licensing > Current Licenses**.



Feature	Licensed	Total Licenses	Available Licenses	Feature Expiration
SIP Signalling Sessions	✓	100	100	November 15, 2021 23:59:59
Enhanced Media Sessions with Transcoding	✓	100	100	November 15, 2021 23:59:59
Enhanced Media Sessions without Transcoding	✓	100	100	November 15, 2021 23:59:59
SIP Registrations	✓	100	100	November 15, 2021 23:59:59

For more details on Licenses, refer to [Ribbon SBC SWe Lite Licenses](#).

## Import Trusted Root CA Certificates

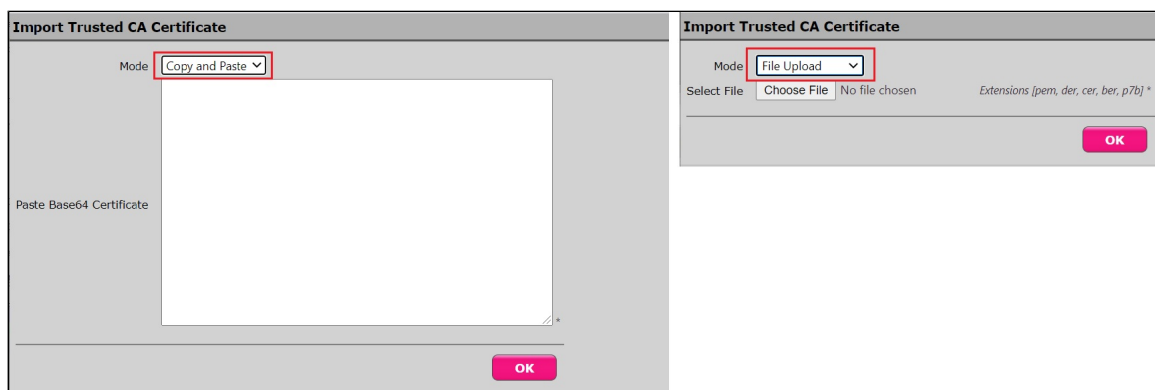
A Trusted CA Certificate is a certificate issued by a trusted certificate authority. Trusted CA Certificates are imported to the SBC SWe Lite to establish its authenticity on the network.

From the **Setting** tab, navigate to **Security > SBC Certificates > Trusted CA Certificates**.



This section describes the process of importing Trusted Root CA Certificates, using either the File Uploader Copy and Paste methods.

1. To import a Trusted CA Certificate, click the Import Trusted CA Certificate (📁) icon.
2. Select either Copy and Paste or File Upload from the Mode menu.
3. If you choose File Upload, use the Select File button to find the file.
4. Click OK.



Follow the steps above to import certificates for this interop.



#### Note

Anywhere365 certificates: <https://sectigo.com/resource-library/sectigo-root-intermediate-certificate-files>

For this interop certificates has been downloaded using the link above.

For the root chain, see the following information:

- **Intermediate 1:**  
CN = Sectigo RSA Domain Validation Secure Server CA  
Serial number: 7d5b5126b476ba11db74160bbc530da7
- **Intermediate 2:**  
CN = USERTrust RSA Certification Authority  
Serial number: 3972443af922b751d7d36c10dd313595
- **RootCa:**  
CN = AAA Certificate Services  
Serial number: 01

For more details on Certificates, refer to [Working with Certificates](#).



#### Note

When the **Verify Status** field in the Certificate panel indicates Expired or Expiring Soon, replace the Trusted CA Certificate. You must delete the old certificate before importing a new certificate successfully.

**Note**

Most Certificate Vendors sign the SBC Edge certificate with an intermediate certificate authority. There is at least one, but could be several intermediate CAs in the certificate chain. When importing the Trusted Root CA Certificates, import the root CA certificate and all Intermediate CA certificates. Failure to import all certificates in the chain causes the import of the SBC Edge certificate to fail. Refer to [Unable To Get Local Issuer Certificate](#) for more information.

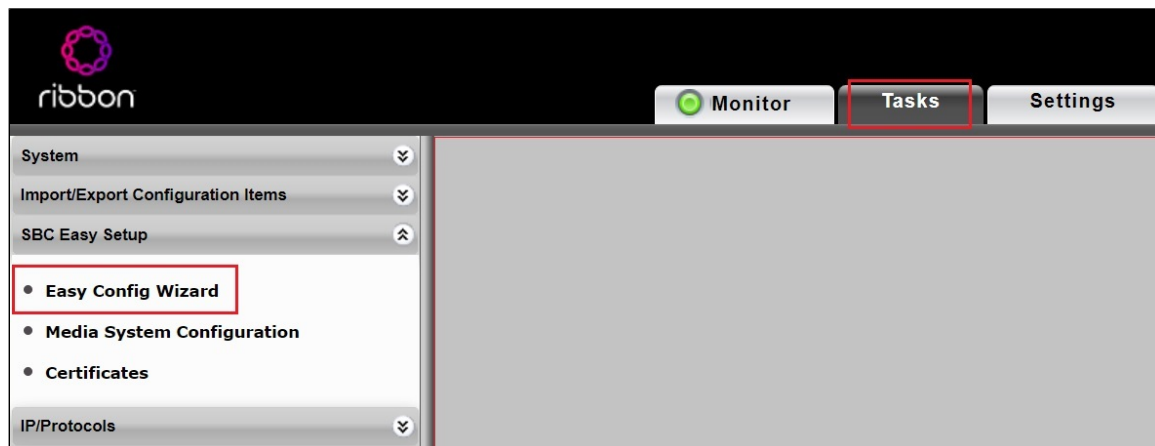
## Ribbon SBC SWe Lite Configuration towards PSTN and Microsoft Teams Cloud

This configuration uses web-based Easy Configuration Wizard for quick start provisioning towards the PSTN/carrier network as well as towards Microsoft Teams cloud.

### Access the Easy Configuration Wizard

1. In the WebUI, click the **Task** tab.
2. In the left navigation pane, navigate to **SBC Easy Setup > Easy Config Wizard**. The Easy Configuration screen opens.

The SBC SWe Lite WebUI provides a built-in Easy Configuration wizard that lets you quickly and easily deploy the SBC for operation with provider endpoints (SIP trunk, ISDN PSTN trunk, or IP PBX trunk) and user endpoints (Microsoft Teams, Microsoft On Premises - Skype for Business/Lync, IP Phones, or ISDN PBX or IP PBX).



### Navigating the Wizard

As the wizard runs, it directs you through three configuration steps:

**Step 1:** Set the following parameters to describe the topology for the telephony service provider and user ends of the scenario.

- **Application:** Click the drop-down arrow, then select the Service Provider and user endpoint types that the SBC is to connect to.
- **Scenario Description:** Type up to 32 characters to describe the connectivity scenario.
- **Telephone Country:** Click the drop-down arrow, then select the country where the telephone services operate.
- **Emergency Services:** Choose **ELIN Identifier**, **E911/E112**, or **None** as the emergency services type.
- **SIP Sessions:** Type a number from 1-960 to indicate the SIP sessions to allocate for the scenario.

**Step 2:** Configure the items required for the endpoints selected, fields display based on the endpoint selection in Step 1.

**Step 3:** The Easy Config validates the final parameters and displays a read-only summary of the configuration that the wizard will apply when you click **Finish** at Step 3. Before you click **Finish**, you can return to previous steps to make adjustments to the data summarized.

The wizard displays the following buttons for navigation:

- **Previous:** Moves back to the previous step.
- **Next:** Advances to the next step when the current step is validated and complete.
- **Finish:** Submits the data to the SBC.
- **Cancel:** Cancels the Easy Configuration data entered and redirects to the main WebUI.

### Configure SBC SWe Lite for SIP Trunk and for Microsoft Teams

During this interop:

- Multi-legged approach was used to configure PSTN SIP Trunk and Microsoft Teams (Application: SIP Trunk Microsoft Teams).

**Step 1:** Configure SIP trunk along with Microsoft Teams using Multi-legged approach by following the steps below:

1. Choose **SIP Trunk <> Microsoft Teams** from the Application dropdown.
2. Provide the Description.
3. Select **United States** in the **Telephone Country** field.
4. Type a number from 1-960 against **SIP Sessions** field.
5. Select SIP Trunk Name as **Other SIP Trunk** for SIP Trunk and Microsoft Teams Connection as **Teams Direct Routing**.
6. Click **Next**.

**Easy Configuration** July 06, 2021 14:02:37

**Step 1** Step 2 Step 3 This step takes input about the topology

**Scenario Parameters**

Application: SIP Trunk <> Microsoft Teams  
Scenario Description: TEAMS  
Telephone Country: United States  
Emergency Services: None

**SIP Properties**

SIP Sessions: 100 [1..960]

**SIP Trunk** Name: Other SIP Trunk

**Microsoft Teams** Teams Connection: Teams Direct Routing

Cancel Previous Next Finish

**Step 2:** After selecting the scenario in Step 1, the following template displays. Complete this step by performing the below actions:

1. Provide the FQDNs for Primary Element servers. The traffic is sent to these FQDNs from the SBC SWe Lite.
2. Use UDP with port number 5060 for SIP trunk configuration.
3. For MS Teams configuration, select the **External interface** (in this case Ethernet 2). After selecting Signaling/Media source IP. Check if the IP is correct and proceed by clicking **Next**.

**Step 1** **Step 2** Step 3 This step takes input about the Provider and User side configuration

**SIP Trunk: Other SIP Trunk**

Border Element Server: [FQDN or IP]  
Protocol: UDP  
Port Number: 5060 [1024..65535]  
Use Secondary Border Element Server: Disabled

**Microsoft Teams: Teams Direct Routing**

Teams Connection Type: Standalone Direct Connection  
Signaling/Media Source IP: Ethernet 2 IP [External I/F]  
Outbound NAT Traversal: Disable  
Apply ACL: ACL already applied  
Protocol: TLS  
Server Port Number: 5061  
Listening Port Number: 5061 [Port Number]

**Step 3:** This step displays a read-only summary of the configuration.



1. Check if the information entered in the previous steps is correct. If the information entered is wrong, return to the previous step by clicking **Previous** and modify the required field.
2. Click **Finish** to complete the configuration.

This step is a summary of what will be configured

### SBC Setup Configuration Summary

#### Scenario Parameters

Application	SIP Trunk <-> Microsoft Teams
Scenario Description	TEAMS
Telephone Country	United States
Emergency Services	None

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#### SIP Properties

SIP Sessions	100
--------------	-----

#### SIP Trunk: Other SIP Trunk

Border Element Server	<input type="text"/>
Protocol	UDP
Port Number	5060
Use Secondary Border Element Server	Disabled

#### Microsoft Teams: Teams Direct Routing

Teams Connection Type	Standalone Direct Connection
Signaling/Media Source IP	Ethernet 2 IP <input type="text"/>
Outbound NAT Traversal	Disable
Apply ACL	ACL already applied
Protocol	TLS
Server Port Number	5061
Listening Port Number	5061

- A pop up window appears once all the 3 steps are completed. Click **OK** to continue.
- Wait for the configuration to complete and click **OK** on the next window. This will complete the configuration of SIP Trunk and Microsoft Teams.

## Modify Media List

Modify the media list that was created using easy wizard configuration.

Select **Settings > Media > Media List > TEAMS: Teams Direct Routing List** and remove SILK media profile and apply.

Media List View

Total 3 Media List Rows

Description
Default Media List
<b>TEAMS: Teams Direct Routing List</b>

Description: TEAMS: Teams Direct Routing List

Media Profiles List

- TEAMS (Teams): SILK
- TEAMS (Teams): G.711 A-Law
- TEAMS (Teams): G.711 Mu-Law

Up

Down

Add/Edit

Remove

SDES-SRTP Profile: TEAMS: Teams Direct Routing SRT1 Associated SIP SG Listen Ports should be TLS only.

Media DSCP: 46 \* [0..63]

Dead Call Detection: Disabled

Silence Suppression: Enabled

Select **Settings > Media > Media List > TEAMS: SIP Trunk List** and remove Fax media profile and apply.

Expand All | Collapse All | Reload

- Call Routing
- Signaling Groups
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
  - Media System Configuration
  - Media Profiles
    - SDES-SRTP Profiles
    - Media List
      - Default Media List
      - TEAMS: Teams Direct Routing Li...
      - TEAMS: SIP Trunk List
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms

### Media List View

+ - | ✕
Total 3 Media List Rows

Description
Default Media List
TEAMS: Teams Direct Routing List
TEAMS: SIP Trunk List

Description

Media Profiles List

TEAMS (Trunk): G.711 A-Law  
TEAMS (Trunk): G.711 Mu-Law  
TEAMS (Trunk): Fax

Up  
Down  
Add/Edit  
Remove

SDES-SRTP Profile: None
Associated SIP SG Listen Ports should be TLS only. +

Media DSCP: 46
\* [0..63]

Dead Call Detection: Disabled

## Ribbon SBC SWe Lite Configuration towards Anywhere365 Cloud

This section describes the steps to configure the SBC SWe Lite with TLS/SRTP towards Anywhere365.

### Media Profile

Select **Settings > Media > Media Profiles**.

- Select Create Media Profile > Voice Codec Profile.
- Create a Media profile with description "**Anywhere365: G.711-A-Law**" and "**Anywhere365: G.711 Mu-Law**" with **G.711-A-Law** and **G.711-Mu-Law** codec. Since dialogue cloud supports G.711 codec only.

- System
- Auth and Directory Services
- Protocols
- SIP
- Security
- Media
  - Media System Configuration
  - Media Profiles
    - Default G711A
    - Default G711u
    - TEAMS (Teams): G.711 A-Law
    - TEAMS (Teams): G.711 Mu-Law
    - TEAMS (Teams): SILK
    - TEAMS (Trunk): G.711 A-Law
    - TEAMS (Trunk): G.711 Mu-Law
    - TEAMS (Trunk): G.711 Mu-Law
    - Anywhere365: G.711-A-Law
    - Anywhere365: G.711 Mu-Law
    - TEAMS (Trunk): Fax
- SDES-SRTP Profiles
- Media List
- Tone Tables
- Telephony Mapping Tables
- SNMP/Alarms
- Logging Configuration

### Media Profiles

Create Media Profile + - | ✕
Total 10 Media Profile Rows

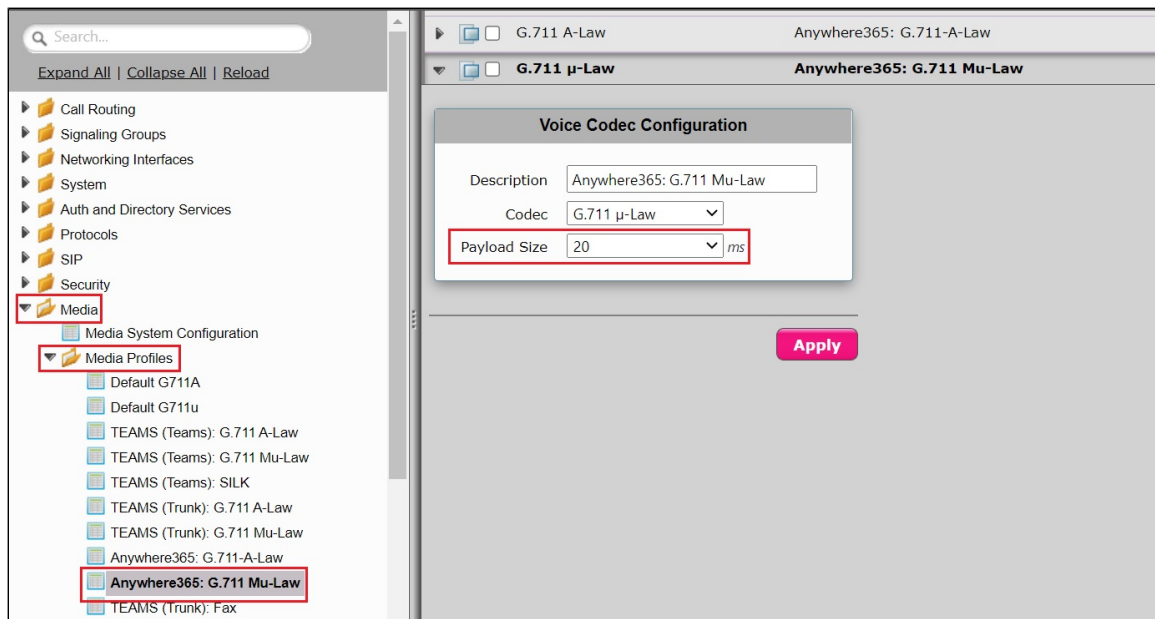
Codec	Description
G.711 A-Law	Default G711A
G.711 μ-Law	Default G711u
G.711 A-Law	TEAMS (Teams): G.711 A-Law
G.711 μ-Law	TEAMS (Teams): G.711 Mu-Law
SILK	TEAMS (Teams): SILK
G.711 A-Law	TEAMS (Trunk): G.711 A-Law
G.711 μ-Law	TEAMS (Trunk): G.711 Mu-Law
G.711 A-Law	Anywhere365: G.711-A-Law

Voice Codec Configuration

Description: Anywhere365: G.711-A-Law

Codec: G.711 A-Law

Payload Size: 20 ms



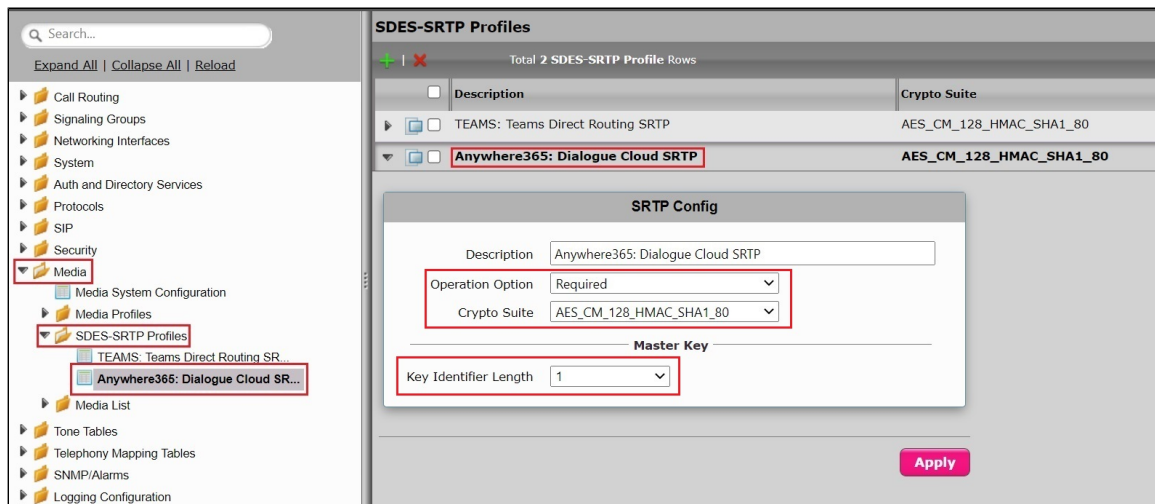
It is recommended to use a maximum packet time (max pTime) of 20ms for all Voice Codecs.

## Create SRTP Profile

SDES-SRTP Profiles define a cryptographic context that is used in SRTP negotiation. SDES-SRTP Profiles are required for enabling encryption and SRTP are applied to Media Lists. SDES-SRTP Profiles was previously named Media Crypto Profiles.

From the **Setting** tab, navigate to **Media > SDES-SRTP Profiles**.

- Click the **+** icon to create a new SRTP profile with description **"Anywhere365: Dialogue Cloud SRTP"**.



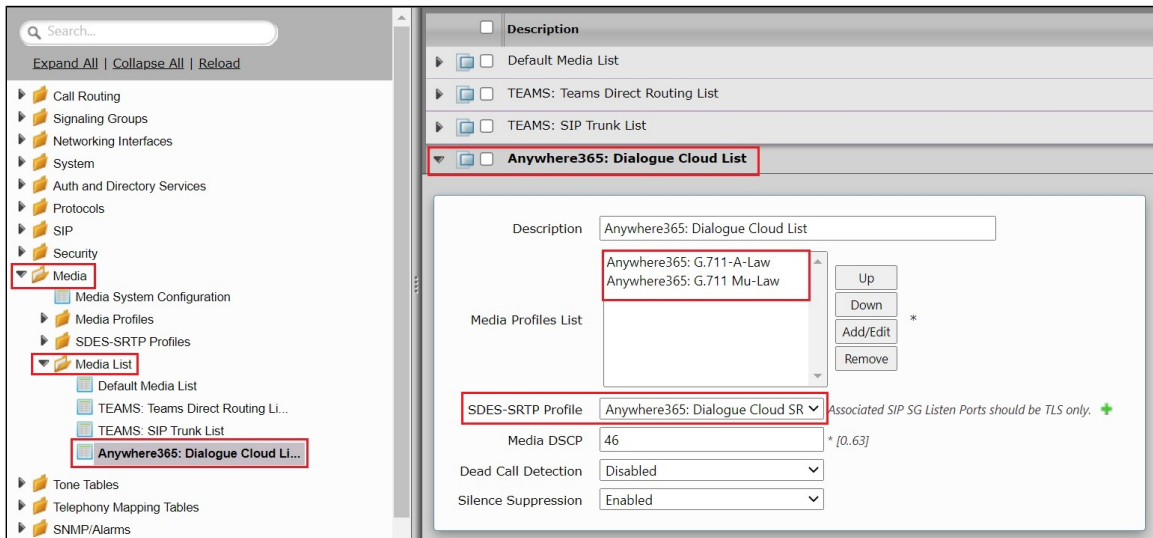
Follow the steps below to complete the configuration:

- Provide the desired description for the profile.
- Set Operation Option as "Required". This setting permits call connections only if you can use encryption for the call. If the peer device does not support SRTP (Secure Real Time Protocol) for voice encryption over the IP network, the call setup will fail.
- Attach the Crypto suite "AES\_CM\_128\_HMAC\_SHA1\_80" - A crypto suite algorithm that uses the 128 bit AES-CM encryption key and a 80 bit HMAC\_SHA1 message authentication tag length.
- Key Identifier Length set to "1".
- Click **OK**.

## Media List

Media Profiles specify the individual voice and fax compression codecs and their associated settings for inclusion into a Media List. Different codecs provide varying levels of compression, allowing the reduction of bandwidth requirements.

- Select **Settings > Media > Media List**.
- Create a media list with desired description "**Anywhere365: Dialogue Cloud List**", add the media profile List and attach the SDES-SRTP Profile "**Anywhere365: Dialogue Cloud SRTP**".



Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

Networking Interfaces

System

Auth and Directory Services

Protocols

SIP

Security

Media

Media System Configuration

Media Profiles

SDES-SRTP Profiles

Media List

Default Media List

TEAMS: Teams Direct Routing Li...

TEAMS: SIP Trunk List

Anywhere365: Dialogue Cloud Li...

Tone Tables

Telephony Mapping Tables

SNMP/Alarms

Description

Anywhere365: Dialogue Cloud List

Media Profiles List

Anywhere365: G.711-A-Law

Anywhere365: G.711 Mu-Law

Up

Down

Add/Edit

Remove

SDES-SRTP Profile

Anywhere365: Dialogue Cloud SR

Media DSCP

46

Dead Call Detection

Disabled

Silence Suppression

Enabled

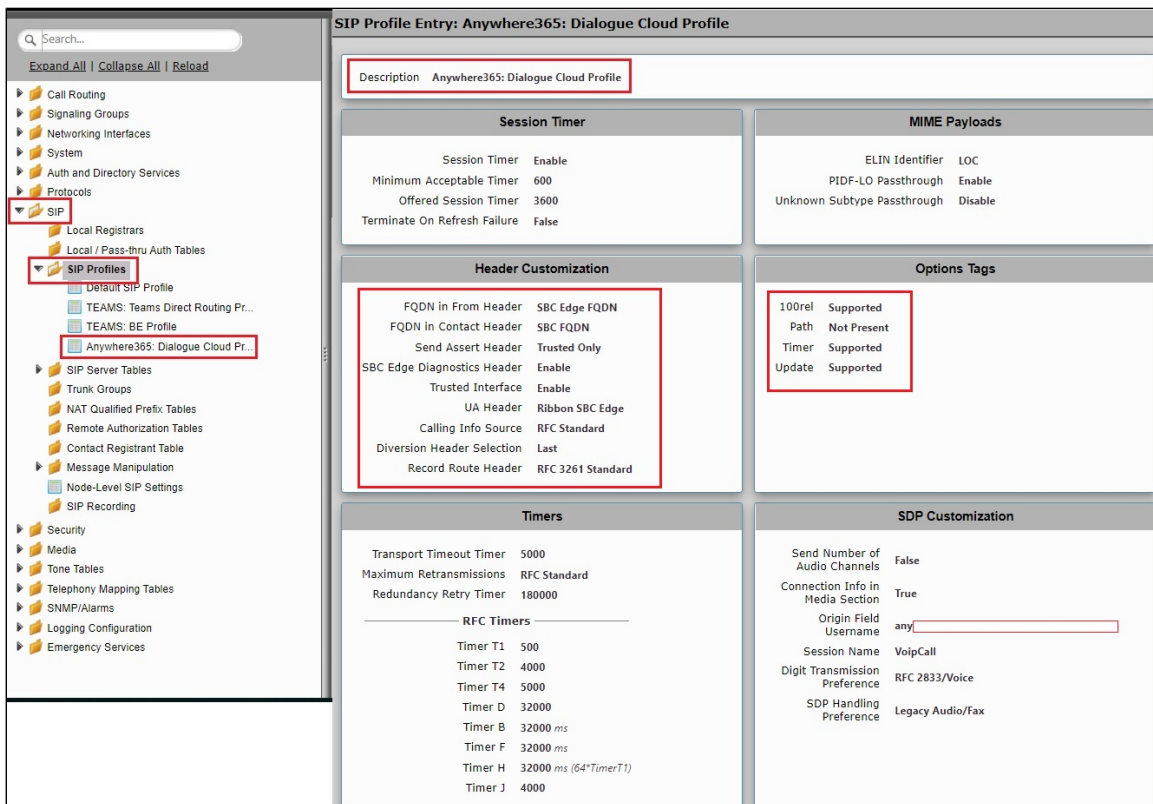
Associated SIP SG Listen Ports should be TLS only.

## SIPProfile

SIP Profiles control how the SBC SWe Lite communicates with SIP devices. The SIP Profile controls important characteristics, such as the following: session timers, SIP header customization, SIP timers, MIME payloads, and option tags

Select **Settings > SIP > SIP Profiles**.

- Create a new SIP profile with desired name "**Anywhere365: Dialogue Cloud Profile**" and with the session timer enabled. The Minimum Acceptable Timer is 600, and the Offered Session Timer is 3600.
- Configure the required fields as shown in the screenshot below.



Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

Networking Interfaces

System

Auth and Directory Services

Protocols

SIP

Local Registrars

Local / Pass-thru Auth Tables

SIP Profiles

Default SIP Profile

TEAMS: Teams Direct Routing Pr...

TEAMS: BE Profile

Anywhere365: Dialogue Cloud Pr...

SIP Server Tables

Trunk Groups

NAT Qualified Prefix Tables

Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Node-Level SIP Settings

SIP Recording

Security

Media

Tone Tables

Telephony Mapping Tables

SNMP/Alarms

Logging Configuration

Emergency Services

SIP Profile Entry: Anywhere365: Dialogue Cloud Profile

Description

Anywhere365: Dialogue Cloud Profile

Session Timer

Session Timer

Enable

Minimum Acceptable Timer

600

Offered Session Timer

3600

Terminate On Refresh Failure

False

MIME Payloads

ELIN Identifier

LOC

PIDF-LO Passthrough

Enable

Unknown Subtype Passthrough

Disable

Header Customization

FQDN in From Header

SBC Edge FQDN

FQDN in Contact Header

SBC FQDN

Send Assert Header

Trusted Only

SBC Edge Diagnostics Header

Enable

Trusted Interface

Enable

UA Header

Ribbon SBC Edge

Calling Info Source

RFC Standard

Diversion Header Selection

Last

Record Route Header

RFC 3261 Standard

Options Tags

100rel

Supported

Path

Not Present

Timer

Supported

Update

Supported

Timers

Transport Timeout Timer

5000

Maximum Retransmissions

RFC Standard

Redundancy Retry Timer

180000

RFC Timers

Timer T1

500

Timer T2

4000

Timer T4

5000

Timer D

32000

Timer B

32000 ms

Timer F

32000 ms

Timer H

32000 ms (64\*TimerT1)

Timer J

4000

SDP Customization

Send Number of Audio Channels

False

Connection Info in Media Section

True

Origin Field Username

any

Session Name

VoipCall

Digit Transmission Preference

RFC 2833/Voice

SDP Handling Preference

Legacy Audio/Fax

## TLS Profile

The TLS profile defines the crypto parameters for the SIP protocol.

Select **Settings > Security > TLS Profile**. Click the **+** icon to create a new TLS profile.

- Create a profile with desired name **"Anywhere365: Dialogue Cloud TLS Profile"** and Set TLS Protocol to TLS 1.2 Only.
- Select the correct valid public certificate from the drop down menu and Finally, ensure Validate Client FQDN is Disabled.

Search...

Expand All | Collapse All | Reload

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Default TLS Profile

TEAMS: Teams Direct Routing TL...

Anywhere365: Dialogue Cloud TL...

Change Password

Ribbon Protect Bad Actors

Media

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SNMP/Alarms

Description Anywhere365: Dialogue Cloud TLS Profile

TLS Parameters

Common Attributes

TLS Protocol TLS 1.2 Only

Mutual Authentication Enabled

Handshake Inactivity Timeout 10

Certificate SBC Edge Certificate

Client Attributes

Client Cipher List

TLS\_ECDHE\_RSA\_WITH\_AES\_256\_CBC\_SHA384

TLS\_ECDHE\_RSA\_WITH\_AES\_128\_CBC\_SHA256

TLS\_ECDHE\_RSA\_WITH\_3DES\_EDE\_CBC\_SHA

TLS\_RSA\_WITH\_AES\_256\_CBC\_SHA256

TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA256

TLS\_RSA\_WITH\_AES256\_CBC\_SHA

TLS\_RSA\_WITH\_AES128\_CBC\_SHA

Validate Server FQDN Enabled

Server Attribute

Validate Client FQDN Disabled

## SIP Server Table

Select **Settings > SIP > SIP Server Tables**

SIP Server Tables contain information about the SIP devices connected to the SBC SWe Lite. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The table entries also contain links to counters that are useful for troubleshooting.

- Create a SIP Server Table with desired name **"Anywhere365: Dialogue Cloud SIP Servers"**
- Add domain name provided by the Anywhere365.
- Enter the correct signaling port number this will be 5061 for TLS or 5060 for TCP/UDP.
- Select the signaling protocol you require. The TLS is shown.
- For TLS, you will need to select the TLS Profile you created earlier.

Search...

Expand All | Collapse All | Reload

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TEAMS: Teams Direct Routing Se...

TEAMS: Border Element

Anywhere365: Dialogue Cloud SI...

PSTN 2

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NAT Qualified Prefix Tables

Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Create SIP Server

Total 1 SIP Server Row

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
emea-weu-t-sbc01.emea-weu.anywhere365.cloud	IP/FQDN	5061	TLS	Counters	1	1

Server Host

Server Lookup IP/FQDN

Priority 1

Host FQDN/IP emea-weu-t-sbc01.emea-weu.ar \*

Host IP Version IPv4

Port 5061 \* [1..65535]

Protocol TLS

TLS Profile Anywhere365: Dialogue Clou \*

Transport

Monitor SIP Options

Keep Alive Frequency 60 \* secs [30..300]

Recover Frequency 5 \* secs [5..300]

Local Username Anonymous \* Local Username of SBC Edge

Peer Username Anonymous \* Peer Username of sip server

Remote Authorization and Contacts

Remote Authorization Table None

Contact Registrant Table None

Session URI Validation Liberal

Connection Reuse

Reuse True

Sockets 4

Reuse Timeout Forever

## Enable OPTIONS



An OPTIONS message is sent to the server. When this option is selected, additional configuration items are displayed:

### Keep Alive Frequency

Specifies how often, in seconds, theSBC Edge queries the server with an OPTIONS message to determine the server's availability. Visible only whenSIP Optionsis selected from theMonitorfield. If the server does not respond, theSBC Edge marks the Signaling Group as down. When the server begins to respond to the OPTIONS messages again, it is marked as up.In this case, Keep Alive Frequency is set to 30 seconds.

### Recover Frequency

Specifies frequency in seconds to check server to determine whether it has become available.Recovery Frequency is set to 5 seconds for this interop.

### Local Username

Local user name of theSBC Edge system. Default entry:**Anonymous**. Visible only whenSIP Optionsis selected from theMonitorfield.

### Peer Username

User name of the SIP Server. Visible only whenSIP Optionsis selected from theMonitorfield. The user can changeLocal and Peer Usernames according to their wishes.

Host / Domain	Server Lookup	Port	Protocol	Display Counters	Priority	Primary Key
emea-weu-t-sbc01.emea-weu.anywhere365.cloud	IP/FQDN	5061	TLS	Counters	1	1

**Server Host**  
Server Lookup: IP/FQDN  
Priority: 1  
Host FQDN/IP: emea-weu-t-sbc01.emea-weu.anywhere365.cloud  
Host IP Version: IPv4  
Port: 5061  
Protocol: TLS  
TLS Profile: Anywhere365: Dialogue Cloud

**Transport**  
Monitor: SIP Options  
Keep Alive Frequency: 60 \* secs [30..300]  
Recover Frequency: 5 \* secs [5..300]  
Local Username: Anonymous \* Local Username of SBC Edge  
Peer Username: Anonymous \* Peer Username of sip server

Remote Authorization and Contacts

Connection Reuse



#### Note

Enable OPTIONS if required for the other SIP Server Tables.

## Message Manipulation

The Message Manipulation feature work in concert to modify and add SIP messages. The Message Manipulation examples below are used to save /store the headers and add to the request.

### The SMM performs the following actions:

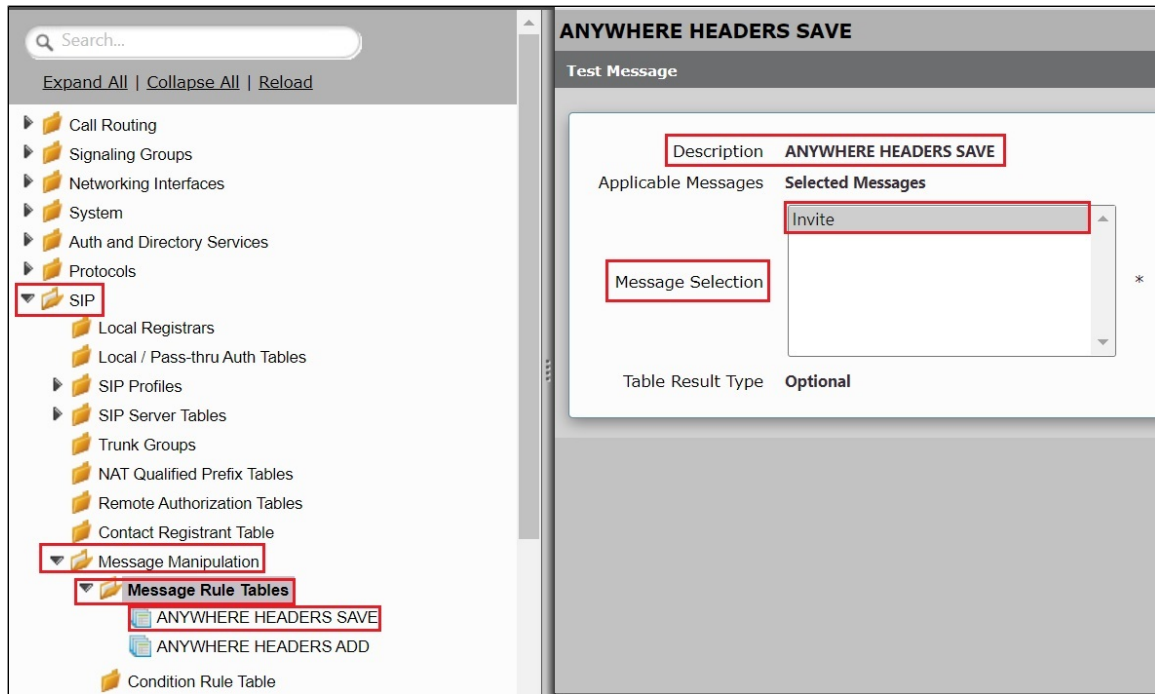
Saves the following headers of the inbound SIP messages.

- A365-AgentHunt
- A365-TenantId
- X-MS-RoutingPolicies

SelectSettings >SIP > Message Manipulation > Message Rule Table

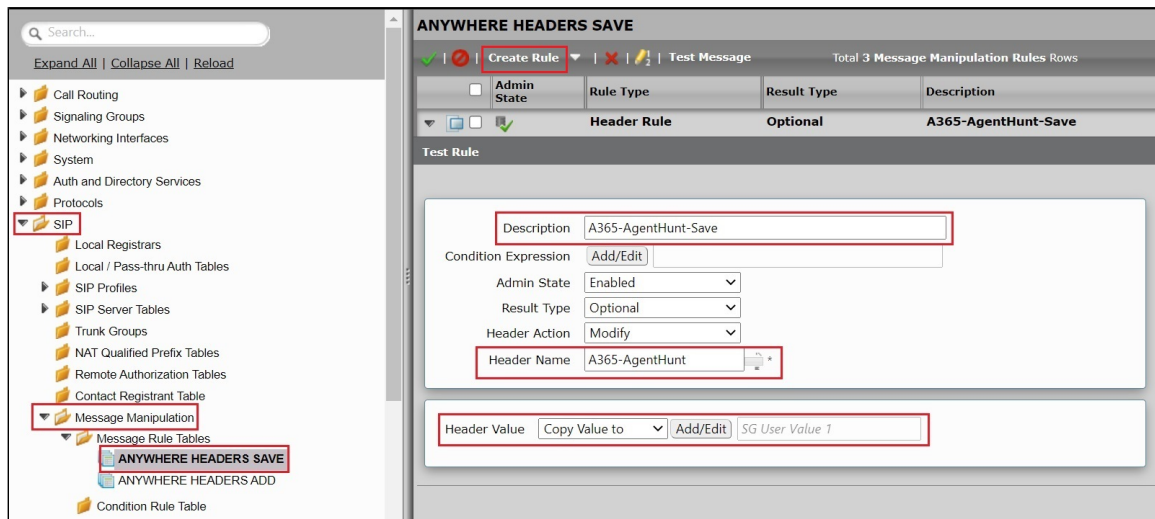
Click theCreate Message Rule Table (+) icon.

- Provide the desired name"**ANYWHERE HEADERS SAVE**"
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click**OK**.



#### Message Manipulation - Create three header rules for the newly created rule table "ANYWHERE HEADERS SAVE"

- Click the **expand** ( ) icon next to the Rule Table entry created.
- From the Create Rule drop down box, select Header Rule.
- Provide a description as required "A365-AgentHunt-Save".
- Header Action "Modify".
- Header Name "A365-AgentHunt".
- Header value 'Copy value to' with SG User Value 1 and apply.



- From the Create Rule drop down box, again select Header Rule.
- Provide a description as required "A365-TenantId-Save".
- Header Action "Modify".
- Header Name "A365-TenantId".
- Header value 'Copy value to' with SG User Value 2 and apply.

**ANYWHERE HEADERS SAVE**

Expand All | Collapse All | Reload

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Local / Pass-thru Auth Tables

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NAT Qualified Prefix Tables

Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Message Rule Tables

ANYWHERE HEADERS SAVE

ANYWHERE HEADERS ADD

Condition Rule Table

**ANYWHERE HEADERS SAVE**

✓ | ✗ | Create Rule | Test Message | Total 3 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
Enabled	Header Rule	Optional	A365-AgentHunt-Save
Enabled	Header Rule	Optional	A365-TenantId-Save
Enabled	Header Rule	Optional	X-MS-RoutingPolicies-Save

**Test Rule**

Description: A365-TenantId-Save

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Modify

Header Name: A365-TenantId

Header Value: Copy Value to | Add/Edit | SG User Value 2

- From the Create Rule drop down box, again select Header Rule.
- Provide a description as required "X-MS-RoutingPolicies-Save".
- Header Action "Modify".
- Header Name "X-MS-RoutingPolicies".
- Header value 'Copy value to' with SG User Value 3 and apply.

**ANYWHERE HEADERS SAVE**

Expand All | Collapse All | Reload

Search...

SIP

Local Registrars

Local / Pass-thru Auth Tables

SIP Profiles

SIP Server Tables

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Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Message Rule Tables

ANYWHERE HEADERS SAVE

ANYWHERE HEADERS ADD

Condition Rule Table

**ANYWHERE HEADERS SAVE**

✓ | ✗ | Create Rule | Test Message | Total 3 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
Enabled	Header Rule	Optional	A365-AgentHunt-Save
Enabled	Header Rule	Optional	A365-TenantId-Save
Enabled	Header Rule	Optional	X-MS-RoutingPolicies-Save

**Test Rule**

Description: X-MS-RoutingPolicies-Save

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Modify

Header Name: X-MS-RoutingPolicies

Header Value: Copy Value to | Add/Edit | SG User Value 3

The SMM performs the following actions:

Adds the following headers to the outbound SIP messages.

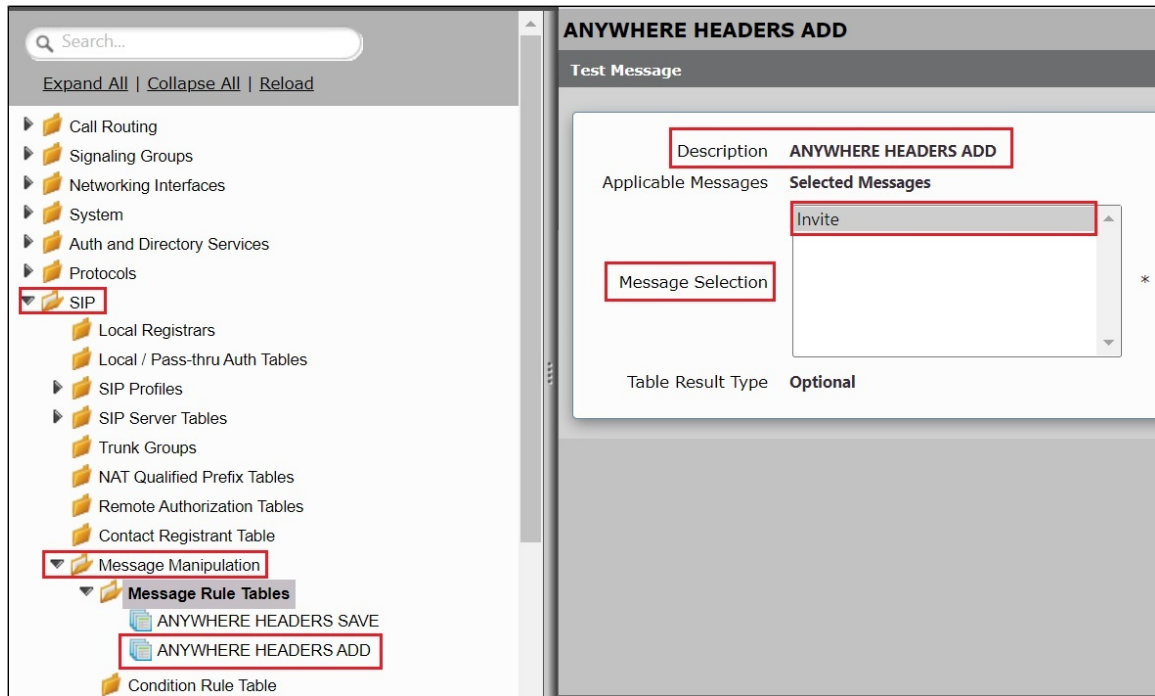
- A365-AgentHunt
- A365-TenantId
- X-MS-RoutingPolicies

SelectSettings > SIP > Message Manipulation > Message Rule Table

Click the Create Message Rule Table (+) icon.

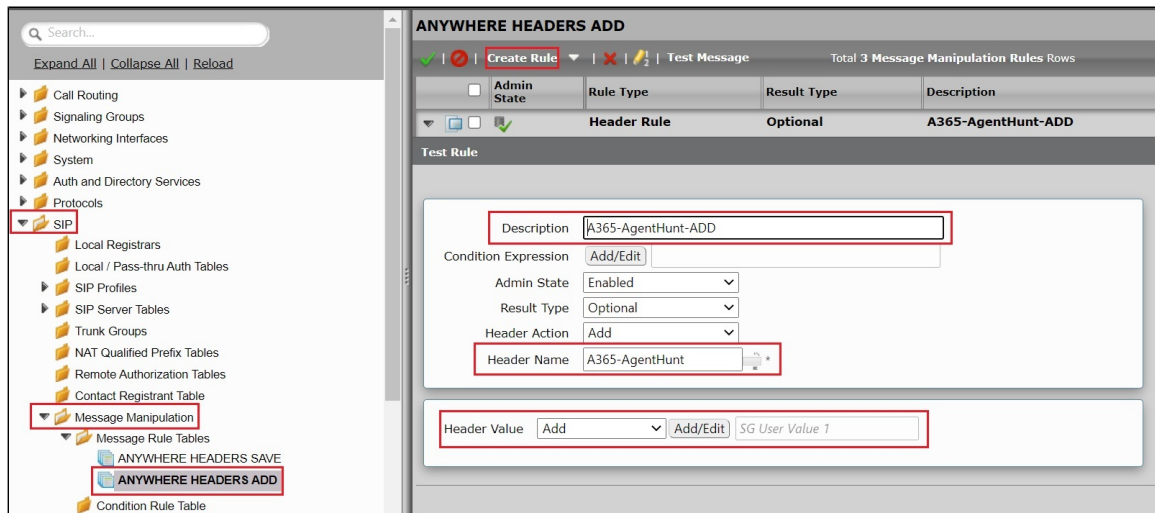
- Provide the desired name "ANYWHERE HEADERS ADD".
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK.





#### Message Manipulation - Create three header rules for the newly created rule tablesay "ANYWHERE HEADERS ADD"

- Click the **expand** (▶) icon next to the Rule Table entry created.
- From the Create Rule drop down box, select Header Rule.
- Provide a description as required "A365-AgentHunt-ADD".
- Header Action "Add".
- Header Name "A365-AgentHunt".
- Header value 'Add' with **SG User Value 1** and apply.



- From the Create Rule drop down box, again select Header Rule.
- Provide a description as required "A365-TenantId-ADD".
- Header Action "Add".
- Header Name "A365-TenantId".
- Header value 'Add' with **SG User Value 2** and apply.

**ANYWHERE HEADERS ADD**

Total 3 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	A365-AgentHunt-ADD
<input type="checkbox"/>	Header Rule	Optional	A365-TenantId-ADD
<input type="checkbox"/>	Header Rule	Optional	X-MS-RoutingPolicies-ADD

**Test Rule**

Description: A365-TenantId-ADD

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Add

Header Name: A365-TenantId

Header Value: Add Add/Edit SG User Value 2

- From the Create Rule drop down box, again select Header Rule.
- Provide a description as required "X-MS-RoutingPolicies-ADD".
- Header Action "Add".
- Header Name "X-MS-RoutingPolicies".
- Header value "Add" with SG User Value 3 and apply.

**ANYWHERE HEADERS ADD**

Total 3 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	A365-AgentHunt-ADD
<input type="checkbox"/>	Header Rule	Optional	A365-TenantId-ADD
<input type="checkbox"/>	Header Rule	Optional	X-MS-RoutingPolicies-ADD

**Test Rule**

Description: X-MS-RoutingPolicies-ADD

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Add

Header Name: X-MS-RoutingPolicies

Header Value: Add Add/Edit SG User Value 3

## Signaling Group

Signaling Groups allow grouping telephony channels together for the purposes of routing and shared configuration. They are the entity where calls are routed, as well as the location that the Call Routes are selected.

### SelectSettings > Signaling Groups

- Create an entry in signaling group named **"Anywhere365: Dialogue Cloud"**.
- Choose **"Anywhere365: Dialogue Cloud Profile"** under Sip Profile.
- Choose Call Routing as **"Anywhere365: From Dialogue Cloud"**.



Initially choose Default call Route. Create the Route, as shown in the call Routing section, and then update the call Route to **"Anywhere365: From Dialogue Cloud"**.

- Choose Agent type as "Back-to-Back user agent" and media list as **"Anywhere365: Dialogue Cloud List"**.
- Choose SIP Server Table as **"Anywhere365: Dialogue Cloud SIP Servers"**.

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) TEAMS: Teams Direct Rout...

(SIP) TEAMS: Border Element

(SIP) Anywhere365: Dialogue CL...

(SIP) PSTN2

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Emergency Services

Description

Anywhere365: Dialogue Cloud

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table

None

Call Routing Table

Anywhere365: From Dialogue Cloud

No. of Channels

60

SIP Profile

Anywhere365: Dialogue Cloud Profile

SIP Mode

Basic Call

Agent Type

Back-to-Back User Agent

SIP Server Table

Anywhere365: Dialogue Cloud SIP Servers

Load Balancing

Priority: Register All

Channel Hunting

Most Idle

Notify Lync CAC Profile

Disable

Challenge Request

Disable

Outbound Proxy IP/FQDN

Outbound Proxy Port

5060

Call Setup Response Timer

255

Call Proceeding Timer

180

Use Register as Keep Alive

Enable

Forked Call Answered Too Soon

Disable

SIP Recording

SIP Recording Status

Disabled

Media Information

Supported Audio Modes

DSP

Proxy

Direct

Supported Video/Application Modes

Media List ID

Anywhere365: Dialogue Cloud List

Play Ringback

Auto on 180/183

Tone Table

TEAMS: United States

Play Congestion Tone

Disable

Early 183

Enable

Allow Refresh SIP

Enable

Music on Hold

Disabled

RTCP Multiplexing

Enable

Mapping Tables

SIP To Q.850 Override Table

Default (RFC4497)

Q.850 To SIP Override Table

Default (RFC4497)

Pass-thru Peer SIP Response Code

Enable

- Update the Federated IP/FQDN, i.e. the FQDN of the anywhere365 as provided by anywhere365 .
  - Add a listening port for TLS.
  - Add message manipulation under the inbound section that we created earlier to save headers.
- Enable Message Manipulation.
  - Click **Add/Edit** Inbound Message Manipulation.
  - This displays a drop-down list of available message tables. Select an entry and click **Apply**.

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) TEAMS: Teams Direct Rout...

(SIP) TEAMS: Border Element

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SIP IP Details

Teams Local Media Optimization

Disable

Signaling/Media Source IP

Ethernet 2 IP

Signaling DSCP

40

NAT Traversal

ICE Support

Disabled

Static NAT - Outbound

Outbound NAT Traversal

None

Static NAT - Inbound

Detection

Disabled

Listen Ports

Total 3 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5061	TLS	Anywhere365: Dialogue...
5060	UDP	N/A
5060	TCP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
	255.255.255.255

Message Manipulation

Enabled

Inbound Message Manipulation

ANYWHERE HEADERS SAVE

Message Table List

Outbound Message Manipulation

Message Table List

## Cause Code Reroutes

Select **Telephony Mapping Tables >Cause Code Reroutes**.

- Click the Cause Code Reroute table (+) icon for new entry creation.
- Provide the required description.
- Add the **Q.850 Cause Codes** as '102: Recovery on Timer Expiry'.
- Click **Apply**.

Expand All | Collapse All | Reload

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  - SIP to Q.850 Override Tables
  - Q.850 to SIP Override Tables
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Cause Code Reroute Table

+

×

Total 1 Reroute Table Row

Description	Primary Key
408 Request Timeout	1

Description

408 Request Timeout

Q.850 Cause Codes

102: Recovery on Timer Expiry

Add/Edit

Remove



Cause code reroute must be used in the call routing table **"TEAMS: From SIP Trunk"** and **"TEAMS: From Microsoft Teams Direct Routing"** to reroute call when receiving **"408 Request Timeout"** response from dialogue cloud.

## Q.850 to SIP Override Tables

Select **Telephony Mapping Tables** > **Q.850 to SIP Override Tables**.

- Click the Q.850 to SIP Override tables ( ) icon for new entry creation.
- Provide the required description.
- Click **OK**.
- Select the entry and update **Q.850 Cause Code** from the drop down list '34: No Circuit/Channel Available'.
- Select **SIP Response** from drop down as '486 - Busy Here'.
- Click **Apply**.

Expand All | Collapse All | Reload

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  - Cause Code Reroutes
  - Message Translations
  - SIP to Q.850 Override Tables
  - Q.850 to SIP Override Tables

cause=34 , map to 486

+

×

Total 1 Q.850 Cause Code to SIP Mapping Row

Q.850 Cause Code	SIP Response
34: No Circuit/Channel Available	486 - Busy Here

Q.850 Cause Code

34: No Circuit/Channel Available

SIP Response

486 - Busy Here

Apply

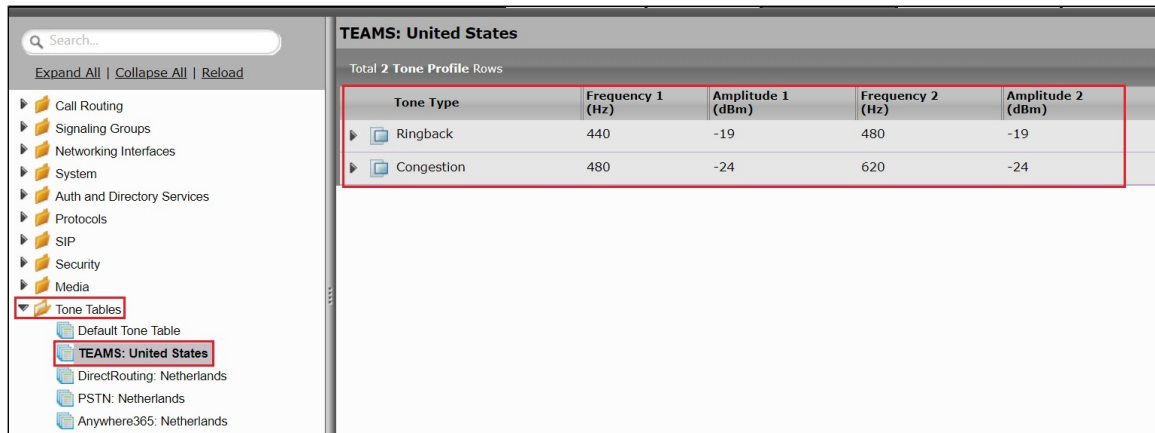


In order to avoid multiple REFER from TEAMS, choose the "Q.850 to SIP Override Tables" record for transfer reject scenario when we get '486 response' for reject from PSTN. Update the Q.850 to SIP Override Table in **"TEAMS: Teams Direct Routing"** signaling group.

## Tone Table (Optional)

When using the Easy Config Wizard, we get **"TEAMS: United States"** record.

- Click the **Tone Table** ( ) icon for any new creation if required. Signaling group can be attached with the tone table.



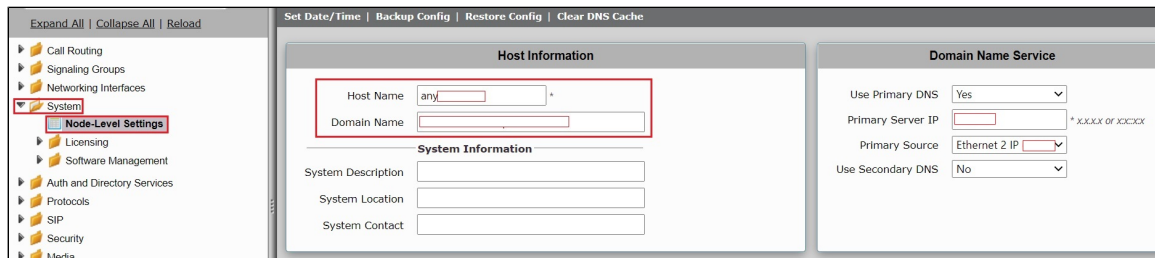
TEAMS: United States				
Total 2 Tone Profile Rows				
Tone Type	Frequency 1 (Hz)	Amplitude 1 (dBm)	Frequency 2 (Hz)	Amplitude 2 (dBm)
Ringback	440	-19	480	-19
Congestion	480	-24	620	-24

## Node level Settings

In the Sip profile (Anywhere365: Dialogue Cloud Profile), we have "FQDN in From Header" and "FQDN in Contact Header" configured as "SBC Edge FQDN" and "SBC FQDN". As a result, Node level settings need to be configured.

From the **System > Node-Level Settings**

- Using the FQDN update the **"Host name"** and **"Domain Name"**.



Expand All | Collapse All | Reload

Set Date/Time | Backup Config | Restore Config | Clear DNS Cache

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Default Tone Table

**TEAMS: United States**

DirectRouting: Netherlands

PSTN: Netherlands

Anywhere365: Netherlands

Node-Level Settings

Licensing

Software Management

System Description

System Location

System Contact

Host Information

Domain Name Service

## Signaling Group

Signaling Groups allow grouping telephony channels together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected.

## Update Signaling Group

Update the signaling group which was created using easy wizard configuration.

**Settings > Signaling Groups.** Click the expand ( ) icon next to the entry.

- Update the signalinggroup **"TEAMS: Teams Direct Routing"**.
- Enable **"Message Manipulation"** and attach the rule **"ANYWHERE365\_HEADERS\_ADD"** rule to outbound Message Manipulation.
- Update the **"Q.850 to SIP Override"** with the rule created.
- Verify all the required changes has been taken care as per the snap for **"TEAMS: Teams Direct Routing"** signalinggroup.
- Click **Apply**.





Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) TEAMS: Teams Direct Rout...

(SIP) TEAMS: Border Element

(SIP) Anywhere365: Dialogue Cl...

(SIP) PSTN2

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Logging Configuration

Emergency Services

Description

TEAMS: Border Element

Admin State

Enabled

Service Status

Up

SIP Channels and Routing

Action Set Table	None
Call Routing Table	TEAMS: From SIP Trunk
No. of Channels	100
SIP Profile	TEAMS: BE Profile
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
SIP Server Table	TEAMS: Border Element
Load Balancing	Priority: Register All
Channel Hunting	Standard
Notify Lync CAC Profile	Disable
Challenge Request	Disable
Outbound Proxy IP/FQDN	
Outbound Proxy Port	
Call Setup Response Timer	180
Call Proceeding Timer	180
Use Register as Keep Alive	Enable
Forked Call Answered Too Soon	Disable

SIP Recording

SIP Recording Status Disabled

Media Information

Supported Audio Modes	DSP
Supported Video/Application Modes	
Media List ID	TEAMS: SIP Trunk List
Play Ringback	Auto on 180
Tone Table	TEAMS: United States
Play Congestion Tone	Disable
Early 183	Enable
Allow Refresh SDP	Enable
Music on Hold	Disable
RTCP Multiplexing	Disable

Mapping Tables

SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	Default (RFC4497)

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

(SIP) TEAMS: Teams Direct Rout...

(SIP) TEAMS: Border Element

(SIP) Anywhere365: Dialogue Cl...

(SIP) PSTN2

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Teams Local Media Optimization

Disable

Signaling/Media Source IP

Ethernet 1 IP (172.255.255.255)

Signaling DSCP

40

NAT Traversal

ICE Support Disabled

Static NAT - Outbound

Outbound NAT Traversal None

Static NAT - Inbound

Detection Disabled

Listen Ports

Total 1 SIP Listen Port Row		
Port	Protocol	TLS Profile ID
5060	UDP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row	
IP/FQDN	Netmask/Prefix
172.255.255.255	255.255.255.255

Message Manipulation

Enabled

Inbound Message Manipulation

Message Table List

Outbound Message Manipulation

ANYWHERE HEADERS ADD

## Transformation Table

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables are configurable as a reusable pool that an Action set can reference.

From the **Settings > Call Routing > Transformation**.

## Create Transformation Table

Each Transformation Table contains a list of entries considered as routing rules to execute on. Each rule is executed in order until the end of the table is reached or when a mandatory entry fails to execute.

- Create transformation table as **"To Dialogue Cloud"**.

Follow the procedure described below to configure Transformation Tables and the Entries.

- Click the **Create (+)** icon.
  - Enter a descriptive name in the **Description** text field.
  - Click **OK**.
- Creating an Entry to a Message Transformation Table.

For this interop, the entries are created based on the numbers associated with each endpoint. Users are free to select their own variables or Regular expressions. UCC prime number can be configured.

1. Click the **Create** (+) icon next to the table created in the previous step.
2. Provide the below details:

**Admin State:**

Enabled-The default state is Enabled.

**Match Type:**

Optional: Optional entries must match at least one of that Input Field type.

When a call arrives at a Transformation Table, the incoming message contains a number of Informational Elements (IEs). These IEs include important call information such as: Called Address/Number, Called Extension, Calling Name, Redirecting Number and others.

Each Informational Element is processed row by row in the Transformation Table.

**Value (Input/Output):**

Specifies the value to match against for the selected type. Depending on the type selected, values are free-form or selected from a menu.

3. Click **Apply**.

**To Dialogue Cloud**

Total 2 Transformation Entry Rows

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
Enabled	Called Address/Number	^( \+ [redacted] \$)	Called Address/Number	\1	Optional (Match One)

Description: DialogueCloudNumbers

Admin State: Enabled

Match Type: Optional (Match One)

**Input Field**

Type: Called Address/Number

Value: ^(\+ [redacted] \$)

**Output Field**

Type: Called Address/Number

Value: \1

Apply



For details on Transformation Table Entry configuration, refer to [Creating and Modifying Entries to Transformation Tables](#). For call digit matching and manipulation through the use of regular expressions, refer to [Creating Call Routing Logic with Regular Expressions](#).

## Modify Transformation Table

The Transformation Tables created for MS Teams and SIP Trunk through Easy Config Wizard are to be modified.

- Modify transformation table for **"TEAMS: From Microsoft Teams Direct Routing: Passthrough"** as **"To Teams Direct Routing"**.
- TEAMS user agent numbers can be configured as required.

1. Click the **expand** (▸) icon next to the entry you wish to modify.
2. Modify the table's **Description** as desired.
3. Modify the Values from **Input field** and **Output field** as required.
4. Set the Match Type as **Optional (Match one)**.
5. Click **OK**.



**To Teams Direct Routing**

Total 1 Transformation Entry Row

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input type="checkbox"/>	Called Address/Number	^(\\+ [1-3])\$	Called Address/Number	\\1	Optional (Match One)

Description: TeamsDirectRoutingNumbers

Admin State: Enabled

Match Type: Optional (Match One)

**Input Field**

Type: Called Address/Number

Value: ^(\+|[1-3])\$

**Output Field**

Type: Called Address/Number

Value: \1

Apply

- Modify transformation table for **"TEAMS: From SIP Trunk: Passthrough"** as **"To PSTN"**.
- PSTN user numbers can be configured as required.

1. Click the **expand** ( ) icon next to the entry you wish to modify.
2. Modify the table's **Description** as desired.
3. Modify the Values from **Input field** and **Output field** as required.
4. Set the Match Type as **Mandatory (Must Match)**.
5. Click **OK**.

**To PSTN**

Total 1 Transformation Entry Row

Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type
<input type="checkbox"/>	Called Address/Number	^(\\+ [1-3])\$	Called Address/Number	\\1	Mandatory

Description: To PSTN

Admin State: Enabled

Match Type: Mandatory (Must Match)

**Input Field**

Type: Called Address/Number

Value: ^(\+|[1-3])\$

**Output Field**

Type: Called Address/Number

Value: \1

Apply



While modifying the transformation table, description and the entries as been modified as required

## Call Routing Table

Call Routing allows carrying of calls between Signaling Groups. Routes are defined by Call Routing Tables, which allow for a flexible configuration that calls carry, and how to translate them.

Select **Settings > Call Routing > Call Routing Table**.

### Create Call Routing Table

Call Routing Tables are one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

- Create Call Routing table as **"Anywhere365: From Dialogue Cloud"**

In the SBC SWe Lite, call routing occurs between **Signaling Groups**.

In order to route any call to or from a call system connected to the SBC, you must first configure a Signaling Group to represent that device or system. The following list illustrates the hierarchical relationships of the various Telephony routing components of a SBC call system:


- Signaling Group describes the source call and points to a routing definition known as a Call Route Table
- Call Route Table contains one or more Call Route Entries
- Call Route Entries points to the destination Signaling Group(s)

Each call routing entry describes how to route the call and also points to a Transformation Table that defines the conversion of names, numbers and other fields when routing a call.

To create an entry:

1. Click the **Create Routing Entry** (+) icon.
2. Set the following fields:

**Admin State:**

Enabled - Enables the call route entry for routing the call, displays in configuration header as .

**Route Priority:**

Priority of the route from 1 (highest) to 10 (lowest). Higher priority routes are matched against before lower priority routes, regardless of the order of the routes in the table.

**Number/Name Transformation Table:**

Specifies the Transformation Table to use for this routing entry. This drop-down list is populated from the entries in the Transformation Table.

**Destination Signaling Groups:**

Specifies the Signaling Groups used as the destination of calls. The first operational Signaling Group from the list is chosen to place the call. Click the **Add/Edit** button to select the destination signaling group.

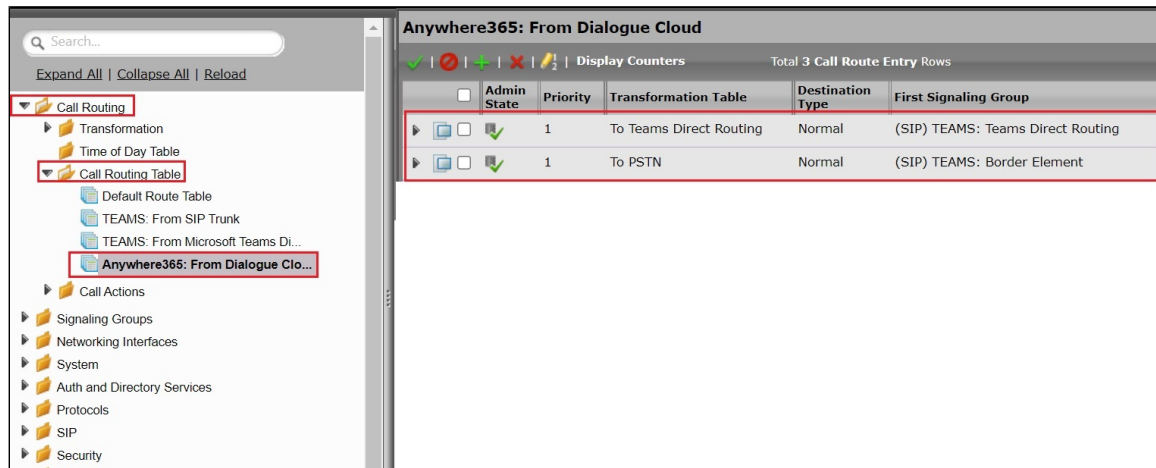
**Audio Stream Mode:**



DSP(default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support DSP are attempted in order.

**Media Transcoding:**

Enabled: Enable Transcoding on SIP-to-SIP calls.

3. Click **Apply**.



Admin State	Priority	Transformation Table	Destination Type	First Signaling Group
	1	To Teams Direct Routing	Normal	(SIP) TEAMS: Teams Direct Routing
	1	To PSTN	Normal	(SIP) TEAMS: Border Element

**Route1 details:** Number/Name Transformation Table with "To Teams Direct Routing" and Destination Signaling Groups with "TEAMS: Teams Direct Routing".

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Route Details

Description

To Teams Direct Routing

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To Teams Direct Routing

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) TEAMS: Teams Direct Routing

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

**Route2 details:**Number/Name Transformation Table with "To PSTN" and Destination Signaling Groups with "TEAMS: Border Element".

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Route Details

Description

To PSTN

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To PSTN

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) TEAMS: Border Element

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

## Modify Call Routing Table

The Call routing Tables created for MS Teams and SIP Trunk through Easy Config Wizard are to be modified.

- Modify Call Routing table for "TEAMS: From SIP Trunk".

The screenshot shows the 'Call Routing' configuration page. On the left, a tree view shows the hierarchy: Call Routing > Transformation > Time of Day Table > Call Routing Table > Default Route Table > **TEAMS: From SIP Trunk**. The main area displays a table titled 'TEAMS: From SIP Trunk' with 3 rows. The table has columns: Admin State, Priority, Transformation Table, Destination Type, and First Signaling Group. The rows are:

Admin State	Priority	Transformation Table	Destination Type	First Signaling Group
<input checked="" type="checkbox"/>	1	To Teams Direct Routing	Normal	(SIP) Anywhere365: Dialogue Cloud
<input checked="" type="checkbox"/>	1	To Teams Direct Routing	Normal	(SIP) TEAMS: Teams Direct Routing
<input checked="" type="checkbox"/>	1	To Dialogue Cloud	Normal	(SIP) Anywhere365: Dialogue Cloud

**Route1 details:** Number/Name Transformation Table with "To Teams Direct Routing" and Destination Signaling Groups with "Anywhere365: Dialogue Cloud", cause code reroute configured.

The screenshot shows the 'Route Details' configuration page for the 'TEAMS: From SIP Trunk' route. The left sidebar shows the same hierarchy as the previous screenshot. The main area is divided into several sections:

- Route Details:**
  - Description: **To Teams Direct Routing**
  - Admin State: **Enabled**
  - Route Priority: **1**
  - Call Priority: **Normal**
  - Number/Name Transformation Table: **To Teams Direct Routing**
  - Time of Day Restriction: **None**
- Destination Information:**
  - Destination Type: **Normal**
  - Message Translation Table: **None**
  - Cause Code Reroutes: **408 Request Timeout**
  - Cancel Others upon Forwarding: **Disabled**
  - Fork Call: **No**
  - Destination Signaling Groups: **(SIP) Anywhere365: Dialogue Cloud**
  - Enable Maximum Call Duration: **Disabled**
- Media:**
  - Audio Stream Mode: **DSP**
  - Video/Application Stream Mode: **Disabled**
  - Media Transcoding: **Enabled**
  - Media List: **None**
- Quality of Service:**
  - Quality Metrics Number of Calls: **10**
  - Quality Metrics Time Before Retry: **10**
  - Min. ASR Threshold: **0**
  - Enable Min MOS Threshold: **Disabled**
  - Enable Max. R/T Delay: **Enabled**
  - Max. R/T Delay: **65535**
  - Enable Max. Jitter: **Enabled**

**Route2 details:** Reroute the call based on 408 response to (TEAMS: Teams Direct Routing) signaling group.

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Route Details

Description

to teams if rerouted 408

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To Teams Direct Routing

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) TEAMS: Teams Direct Routing

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

**Route3 details:**Number/Name Transformation Table with "To Dialogue Cloud" and Destination Signaling Groups with "Anywhere365: Dialogue Cloud".

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Route Details

Description

To Dialogue Cloud

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To Dialogue Cloud

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) Anywhere365: Dialogue Cloud

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled



- Modify Call Routing table for "TEAMS: From Microsoft Teams Direct Routing".

Search...

Expand All

 | 

Collapse All

 | 

Reload

Call Routing

▶

Transformation

▶

Time of Day Table

Call Routing Table

▶

Default Route Table

▶

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Direct Routing

▶

Anywhere365: From Dialogue Cloud

▶

Call Actions

TEAMS: From Microsoft Teams Direct Routing

✓

✗

+

-

⚙

Display Counters

Total 5 Call Route Entry Rows

<div><div></div><div></div></div>	Admin State	Priority	Transformation Table	Destination Type	First Signaling Group
<div><div>▶</div><div><div><div></div><div></div></div><div><div></div><div></div></div></div></div>	<div><div></div><div></div></div>	1	To Teams Direct Routing	Normal	(SIP) TEAMS: Teams Direct Routing
<div><div>▶</div><div><div><div></div><div></div></div><div><div></div><div></div></div></div></div>	<div><div></div><div></div></div>	1	To PSTN	Normal	(SIP) Anywhere365: Dialogue Cloud
<div><div>▶</div><div><div><div></div><div></div></div><div><div></div><div></div></div></div></div>	<div><div></div><div></div></div>	1	To PSTN	Normal	(SIP) TEAMS: Border Element

**Route1 details:**Number/Name Transformation Table "To Teams Direct Routing" and Destination Signaling Groups "TEAMS: Teams Direct Routing".

<div>Search...</div> <div>Expand All   Collapse All   Reload</div> <ul style="list-style-type: none"> <li>Call Routing           <ul style="list-style-type: none"> <li>Transformation</li> <li>Time of Day Table</li> <li>Call Routing Table               <ul style="list-style-type: none"> <li>Default Route Table</li> <li>TEAMS: From SIP Trunk</li> <li>TEAMS: From Microsoft Teams Di...</li> <li>Anywhere365: From Dialogue Clo...</li> </ul> </li> <li>Call Actions</li> </ul> </li> <li>Signaling Groups</li> <li>Networking Interfaces</li> <li>System</li> <li>Auth and Directory Services</li> <li>Protocols</li> <li>SIP</li> <li>Security</li> <li>Media</li> <li>Tone Tables</li> <li>Telephony Mapping Tables</li> <li>SNMP/Alarms</li> <li>Logging Configuration</li> <li>Emergency Services</li> </ul>	<div>Route Details</div> <table> <tr> <td>Description</td><td>To Teams</td></tr> <tr> <td>Admin State</td><td>Enabled</td></tr> <tr> <td>Route Priority</td><td>1</td></tr> <tr> <td>Call Priority</td><td>Normal</td></tr> <tr> <td>Number/Name Transformation Table</td><td>To Teams Direct Routing</td></tr> <tr> <td>Time of Day Restriction</td><td>None</td></tr> </table> <div>Destination Information</div> <table> <tr> <td>Destination Type</td><td>Normal</td></tr> <tr> <td>Message Translation Table</td><td>None</td></tr> <tr> <td>Cause Code Reroutes</td><td>None</td></tr> <tr> <td>Cancel Others upon Forwarding</td><td>Disabled</td></tr> <tr> <td>Fork Call</td><td>No</td></tr> <tr> <td>Destination Signaling Groups</td><td>(SIP) TEAMS: Teams Direct Routing</td></tr> <tr> <td>Enable Maximum Call Duration</td><td>Disabled</td></tr> </table> <div>Media</div> <table> <tr> <td>Audio Stream Mode</td><td>DSP</td></tr> <tr> <td>Video/Application Stream Mode</td><td>Disabled</td></tr> <tr> <td>Media Transcoding</td><td>Enabled</td></tr> <tr> <td>Media List</td><td>None</td></tr> </table> <div>Quality of Service</div> <table> <tr> <td>Quality Metrics Number of Calls</td><td>10</td></tr> <tr> <td>Quality Metrics Time Before Retry</td><td>10</td></tr> <tr> <td>Min. ASR Threshold</td><td>0</td></tr> <tr> <td>Enable Min MOS Threshold</td><td>Disabled</td></tr> <tr> <td>Enable Max. R/T Delay</td><td>Enabled</td></tr> <tr> <td>Max. R/T Delay</td><td>9999</td></tr> <tr> <td>Enable Max. Jitter</td><td>Enabled</td></tr> <tr> <td>Max. Jitter</td><td>3000</td></tr> </table>	Description	To Teams	Admin State	Enabled	Route Priority	1	Call Priority	Normal	Number/Name Transformation Table	To Teams Direct Routing	Time of Day Restriction	None	Destination Type	Normal	Message Translation Table	None	Cause Code Reroutes	None	Cancel Others upon Forwarding	Disabled	Fork Call	No	Destination Signaling Groups	(SIP) TEAMS: Teams Direct Routing	Enable Maximum Call Duration	Disabled	Audio Stream Mode	DSP	Video/Application Stream Mode	Disabled	Media Transcoding	Enabled	Media List	None	Quality Metrics Number of Calls	10	Quality Metrics Time Before Retry	10	Min. ASR Threshold	0	Enable Min MOS Threshold	Disabled	Enable Max. R/T Delay	Enabled	Max. R/T Delay	9999	Enable Max. Jitter	Enabled	Max. Jitter	3000
Description	To Teams																																																		
Admin State	Enabled																																																		
Route Priority	1																																																		
Call Priority	Normal																																																		
Number/Name Transformation Table	To Teams Direct Routing																																																		
Time of Day Restriction	None																																																		
Destination Type	Normal																																																		
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Cause Code Reroutes	None																																																		
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Fork Call	No																																																		
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Enable Maximum Call Duration	Disabled																																																		
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Enable Min MOS Threshold	Disabled																																																		
Enable Max. R/T Delay	Enabled																																																		
Max. R/T Delay	9999																																																		
Enable Max. Jitter	Enabled																																																		
Max. Jitter	3000																																																		

**Route2 details:**Number/Name Transformation Table with "To PSTN" and Destination Signaling Groups with "Anywhere365: Dialogue Cloud", cause code reroute configured.

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Emergency Services

Route Details

Description

to PSTN1 newly added

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To PSTN

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

408 Request Timeout

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) Anywhere365: Dialogue Cloud

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

**Route3 details:** Based on 408 response reroute to (TEAMS: Border Element) signaling group.

Search...

Expand All | Collapse All | Reload

Call Routing

Transformation

Time of Day Table

Call Routing Table

Default Route Table

TEAMS: From SIP Trunk

TEAMS: From Microsoft Teams Di...

Anywhere365: From Dialogue Clo...

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Logging Configuration

Emergency Services

Route Details

Description

to pstn if rerouted 408

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

To PSTN

Time of Day Restriction

None

Destination Information

Destination Type

Normal

Message Translation Table

None

Cause Code Reroutes

None

Cancel Others upon Forwarding

Disabled

Fork Call

No

Destination Signaling Groups

(SIP) TEAMS: Border Element

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP

Video/Application Stream Mode

Disabled

Media Transcoding

Enabled

Media List

None

Quality of Service

Quality Metrics Number of Calls

10

Quality Metrics Time Before Retry

10

Min. ASR Threshold

0

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

Enable Max. Jitter

Enabled

## Section B: Anywhere365 Configuration

For anywhere365 related configurations and queries, please contact the Anywhere365 technical support team.

## Supplementary Services and Features Coverage

The following checklist depicts the set of services/features covered through the configuration defined in this Interop Guide.

Sr. No.	Supplementary Services/ Features	Coverage
1	Call Setup and Termination over TLS	✓
2	Call Transfer (Blind/Unattended/Cold)	✓
3	Call Transfer (consultative/Attended/Warm)	✓
4	Supervisor actions	✓
5	Quality Management using DTMF	✓
6	Call hold and Resume (with MOH enable)	✓
7	Call hold and Resume (without MOH enable)	✓
8	AnonymousCall	✓
9	Long Duration	✓
10	OPTIONS validation	✓
11	DTMF handling	✓
12	Session Audits	✓

#### Legend

Supported	✓
Not Supported	✗

## Caveats

- Two way audio issue (both sides not audible) has been observed when blind transfer initiated by Teams is rejected from PSTN side. This is a known issue for Ribbon and will be addressed in the upcoming SBC releases.

## Support

For any support related queries about this guide, please contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: <https://ribboncommunications.com/about-us>

## References

For detailed information about Ribbon products and solutions, please visit:

<https://ribboncommunications.com/products>

For detailed information about Anywhere365 products and solutions, please visit:

<https://anywhere365.io/>

## Conclusion

This Interoperability Guided describe the configuration steps required for **Ribbon SBC SWe Lite** to successfully interoperate with **Anywhere365**. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in Test Results.

All features and capabilities tested are detailed within this document - any limitations, notes or observations are also recorded in order to provide the reader with an accurate understanding of what is/is not covered.



Configuration guidance is provided to enable the reader to replicate the same base setup additional configuration changes are possibly required to suit the exact deployment environment.

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