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# Ribbon SBC Edge Configuration with OBS (UDP)

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## Document Overview

This document provides a configuration guide for Ribbon SBC Edge Series (Session Border Controller) when connecting to OBS Business Talk (BTIP) SIP trunk.

This configuration guide supports features given in the BTIP North Profile Compliancy and Compliance tests documents.

Ribbon has configured the BTIP side in a such manner it doesn't matter the 3rd party system connected on the SBC.

The SBC Edge is certified by Orange Business Services as a '**certified Enterprise SBC**'.

- For additional information on OBS, please visit <https://www.orange-business.com/en/products/business-talk>
- For additional information on Ribbon SBC Edge, please visit <https://ribboncommunications.com/>

## Introduction

The interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC Edge and OBS.

## Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC and the third-party product. There will be steps that require navigating the third-party and the Ribbon SBC Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

**i** This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information in this guide.

## Requirements

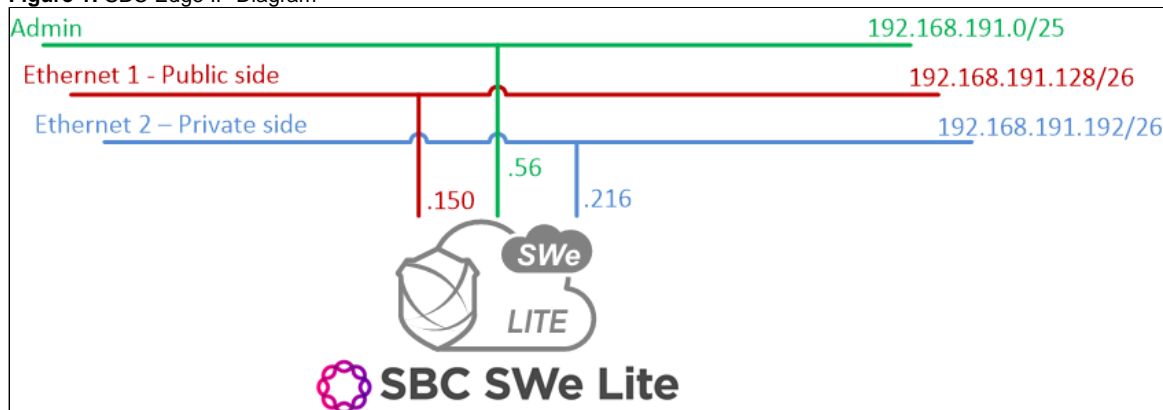
The following equipment and software were used for the sample configuration provided:

	Equipment	Software Version
<b>Ribbon Communications</b>	Ribbon SBC SWE-Lite	9.0.0
<b>Third-party Equipment</b>	CISCO CUCM	12.5
<b>Other software</b>	VentaFax	7.3.233.582l

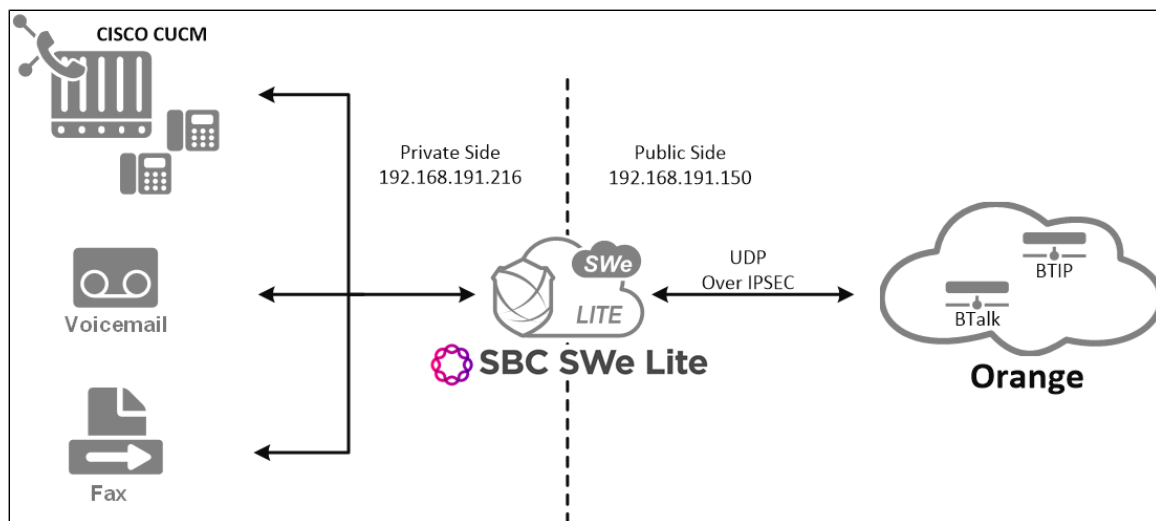
## Reference Configuration

The following reference configuration shows the connectivity between the third-party and Ribbon SBC Edge.

**Figure 1:** SBC Edge IP Diagram



**Figure 2:** Topology



## Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

## Third-Party Product Features

**Table 1:** Product Features

- Basic Call
- Long Duration Call + CLIR
- Call Cancellation
- DTMF + Voicemail
- Transfer
  - Supervised + MOH
  - Blind
- Forward
  - Unconditional
  - Busy
  - No Answer
- Busy Call
- Not Answered Call
- Conference X3
- Prehook
  - With Transfer
  - With Forward
- Call Parking
- Call Pickup
- Hunt Group
- Second Line
- CAC
- Emergency Number
- Fax

## Prerequisites

**Table 2:** Prerequisites

- A Valid SBC Edge License
- VentaFax Software
- Voicemail Software

## Verify License

A valid SWE-Lite (Key Based) license with the features necessary to run the tests.

## Cisco Unified Communications Manager (CUCM)

The following new configurations are included in this section:

1. [SIP Trunk](#)
2. [Route Group](#)
3. [Route List](#)
4. [Route Pattern](#)

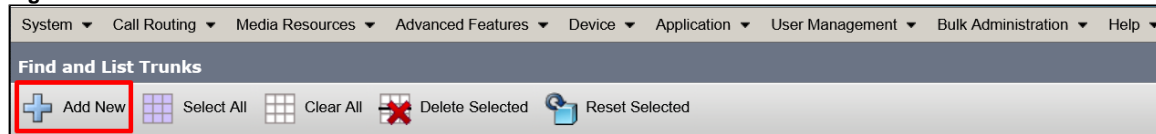
## SIP Trunk

SIP trunks allow administrators to connect the Cisco Unified Communications Manager to external devices such as SIP gateways, SIP Proxy Servers, Unified Communications applications, remote clusters, or a Session Management Edition. Ribbon uses the SIP trunk to connect the CUCM to the Ribbon SBC SWE-Lite.

Login to the CUCM as an *admin* user and go to the following path: *Device > Trunk*

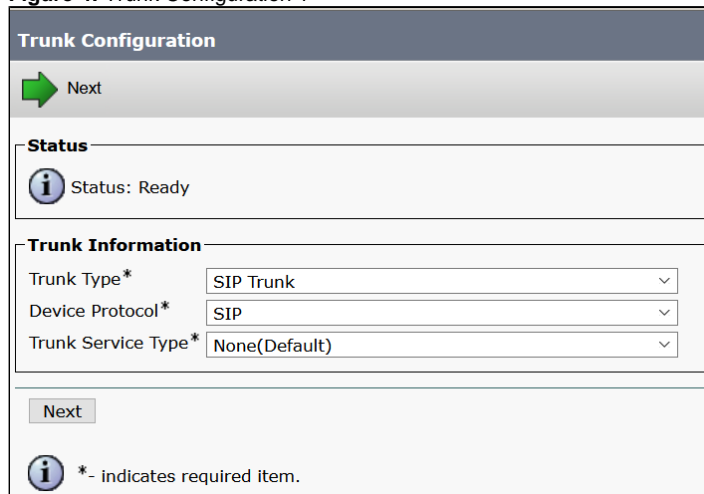
Click on the *Add New* icon to add a new Trunk.

**Figure 3:** Add New Trunk



Set the trunk configuration as shown in the figure below.

**Figure 4:** Trunk Configuration 1

The image shows a 'Trunk Configuration' form. At the top, there is a 'Next' button with a green arrow icon. Below this is a 'Status' section with an information icon and the text 'Status: Ready'. Underneath is a 'Trunk Information' section with three dropdown menus: 'Trunk Type\*' (set to 'SIP Trunk'), 'Device Protocol\*' (set to 'SIP'), and 'Trunk Service Type\*' (set to 'None(Default)'). At the bottom of the form, there is another 'Next' button and an information icon with the text '\* - indicates required item.'.

After clicking the *Next* button, a new screen will be shown. Set the device (trunk) name, the profiles, and the destination IP address used by the trunk. Below is an example.

**Figure 5:** Trunk Configuration 2

Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	OrangeSBCLite
Description	Trunk to Orange SBC Lite
Device Pool*	Sonus_DP
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

The IP address on the *SWE-Lite* towards the *CUCM* is 192.168.191.216.

**Figure 6: Trunk Configuration 3**

SIP Information						
<input type="checkbox"/> Destination Address is an SRV						
1*	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
	192.168.191.216		5060	N/A	N/A	N/A
MTP Preferred Originating Codec*	711ulaw					
BLF Presence Group*	Standard Presence group					
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile					
Rerouting Calling Search Space	< None >					
Out-Of-Dialog Refer Calling Search Space	< None >					
SUBSCRIBE Calling Search Space	< None >					
SIP Profile*	Standard SIP Profile <a href="#">View Details</a>					
DTMF Signaling Method*	No Preference					

Click the **Save** button.

## Route Group

In Cisco Unified Communications Manager, use the *Call Routing > Route/Hunt > Route Group* menu path to configure route groups.

A route group allows you to designate the order in which gateways and trunks are selected. It allows you to prioritize a list of gateways and ports for outgoing trunk selection.

Click on the **Add New** icon to add a Route Group.

**Figure 7: Add New Route Group**

Find and List Route Groups	
<input type="button" value="+ Add New"/>	
Route Group	
Find Route Group where Route Group Name	<input type="text" value="begins with"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="-"/>
No active query. Please enter your search criteria using the options above.	
<input type="button" value="Add New"/>	

Specify the *Route Group name* and select the *devices* that are going to be used by this *Route Group*.

In the example below, the device selected is OrangeSBCLite, created in [SIP Trunk above](#).

**Figure 8: Route Group Configuration**

<b>Route Group Information</b>	
Route Group Name*	OrangeSBCLite
Distribution Algorithm*	Circular
<b>Route Group Member Information</b>	
<b>Find Devices to Add to Route Group</b>	
Device Name contains	<input type="text"/> Find
Available Devices**	<div> <div>CUBE</div> <div>CharterSpectrum</div> <div>FM2900</div> <div>OrangeSBCLite</div> <div>Plusnet</div> </div>
Port(s)	None Available
Add to Route Group	
<b>Current Route Group Members</b>	
Selected Devices (ordered by priority)*	OrangeSBCLite (All Ports)
Reverse Order of Selected Devices	

Click the Save button.

## Route List

In Cisco Unified Communications Manager Administration, use the *Call Routing > Route/Hunt > Route List* menu path to configure route lists.

A route list associates a set of route groups in a specified priority order. The route list associates the route groups with one or more route patterns and determines the order in which those route groups are accessed. The order controls the progress of the search for available devices for outgoing calls.

A route list can contain only route groups. Each route list should have at least one route group. Each route group includes at least one device.

Click on the *Add New* icon to add a Route List.

**Figure 9: Add New Route List**

<b>Find and List Route Lists</b>	
+ Add New	
<b>Route List</b>	
Find Route List where	Name begins with <input type="text"/> Find Clear Filter + -

In the next screen, specify the Route List name, description, and select the Cisco Unified Communications Manager Group used by the Route List.

**Figure 10: Route List Configuration 1**

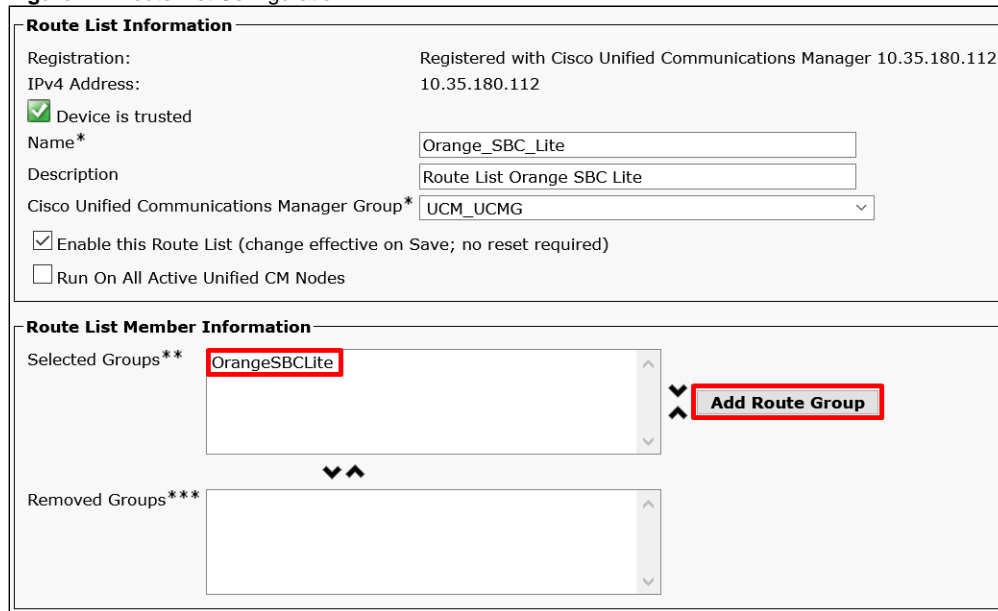
<b>Route List Configuration</b>	
Save	
<b>Status</b>	
Status: Ready	
<b>Route List Information</b>	
<input checked="" type="checkbox"/> Device is trusted	
Name*	Orange_SBC_Lite
Description	Route List Orange SBC Lite
Cisco Unified Communications Manager Group*	UCM_UCMG

Click the Save button.

After saving, you will be able to add the *Route Group* to the *Route List*.

Click the *Add Route Group* icon to add the *Route Group* to the *Route List*.

**Figure 11: Route List Configuration 2**



The screenshot shows the 'Route List Configuration 2' interface. It is divided into two main sections: 'Route List Information' and 'Route List Member Information'.

**Route List Information:**

- Registration: Registered with Cisco Unified Communications Manager 10.35.180.112
- IPv4 Address: 10.35.180.112
- ☒ Device is trusted
- Name\*: Orange\_SBC\_Lite
- Description: Route List Orange SBC Lite
- Cisco Unified Communications Manager Group\*: UCM\_UCMG
- ☒ Enable this Route List (change effective on Save; no reset required)
- ☐ Run On All Active Unified CM Nodes

**Route List Member Information:**

- Selected Groups\*\*: OrangeSBCLite (highlighted with a red box)
- Removed Groups\*\*\*: (empty)
- An 'Add Route Group' button (highlighted with a red box) is located next to the Selected Groups list.

Click the *Save* button. At this point the new Routing List has been created.

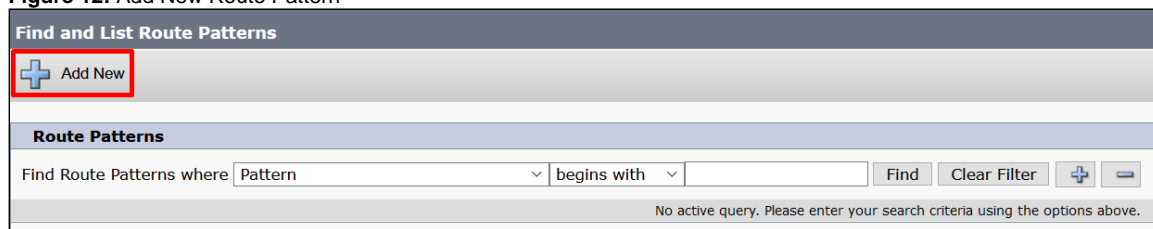
## Route Pattern

In Cisco Unified Communications Manager Administration, use the *Call Routing > Route/Hunt > Route Pattern* menu path to configure route patterns.

A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that can be assigned to a route list or a gateway. Route patterns provide flexibility in network design. They work in conjunction with route filters and route lists to direct calls to specific devices and to include, exclude, or modify specific digit patterns.

Click the *Add New* icon to add a Route Pattern.

**Figure 12: Add New Route Pattern**



The screenshot shows the 'Find and List Route Patterns' interface. It features a search bar and a list of route patterns.

**Find and List Route Patterns:**

- A red box highlights the '+ Add New' button.
- The 'Route Patterns' section shows a search bar with the text 'Find Route Patterns where Pattern' and a dropdown menu set to 'begins with'.
- Buttons for 'Find', 'Clear Filter', and a '+ -' icon are visible.
- A message at the bottom states: 'No active query. Please enter your search criteria using the options above.'

In the next screen, specify the *Route Pattern*, *Description*, and *Gateway/Route List*.

**Figure 13: Route Pattern Configuration**



Route Pattern Configuration

Save

Delete

Copy

Add New

Pattern Definition

Route Pattern\*

6XXXXXXXXXXXXX

Route Partition

< None >

Description

Route Pattern to Orange SBC Lite

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence\*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class\*

Default

Gateway/Route List\*

Orange\_SBC\_Lite

(Edit)

Route Option

☒ Route this pattern
 ☐ Block this pattern
 

No Error

Call Classification\*

OffNet

External Call Control Profile

< None >

☐ Allow Device Override

☒ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level\*

0

☐ Require Client Matter Code



#### Cisco Wildcard

The X wildcard matches any single digit in the range 0 through 9. For instance, the route pattern 9XXX routes or blocks all numbers in the range 9000 through 9999.

For more information regarding special characters and settings on the CISCO CUCM, go to [Appendix A](#).

Click the **Save** button. At this point the new Route Pattern has been created.

All traffic matching this route pattern will be routed through the route list *Orange\_SBC\_Lite*.

## SBC SWE-Lite Configuration

The following new configurations are included in this section:

1. [System Settings](#)
2. [Network Interfaces](#)
3. [Static Routes](#)
4. [SIP Profiles](#)
5. [SIP Server Tables](#)
6. [Message Manipulations](#)
7. [Media Profiles](#)
8. [Media Lists](#)
9. [Q.850 to SIP Override Table](#)
10. [Signaling Groups](#)
11. [Transformations Tables](#)
12. [Call Routing Tables](#)

### System Settings

The *System > Node-Level settings* menu path allows you to set the *Host name*, the *Domain name service*, and *Time management*. Below is an example of the system settings.

Figure 14: System Node-level settings

Host Information	Domain Name Service
Host Name <input type="text" value="Orange"/> *	Use Primary DNS <input type="text" value="Yes"/>
Domain Name <input type="text"/>	Primary Server IP <input type="text" value="172.16.21.230"/> * XXX.XX.XX.XX
<hr/>	
<b>System Information</b>	Primary Source <input type="text" value="Auto"/>
System Description <input type="text"/>	Use Secondary DNS <input type="text" value="Yes"/>
System Location <input type="text"/>	Secondary Server IP <input type="text" value="172.16.21.231"/> * XXX.XX.XX.XX
System Contact <input type="text"/>	Secondary Source <input type="text" value="Auto"/>
<hr/>	
<b>Time Management</b>	<b>EdgeView</b>
Time Zone <input type="text" value="(GMT-6:00) Central (US/Canada)"/>	EdgeView <input type="text" value="No"/>
<hr/>	
<b>Network Time Protocol</b>	
Use NTP <input type="text" value="No"/>	


## Network Interfaces

The *Networking Interfaces > Logical Interfaces* menu path allows you to configure the IP addresses (both IPv4 and IPv6) for the Ethernet ports and VLANs.

The SBC SWe Lite supports five system-created logical interfaces (known as **Administrative IP**, **Ethernet 1 IP**, **Ethernet 2 IP**, **Ethernet 3 IP**, and **Ethernet 4 IP**). In addition to the system created logical interfaces, the Ribbon SBC SWe supports user-created VLAN logical sub-interfaces.

### Administrative IP

The SBC SWe Lite system supports a logical interface called the Admin IP (Administrative IP, also known as the Management IP). A Static IP or DHCP is used for running Initial Setup of the SBC SWe Lite system.

**Admin IP**  
The Administrative IP interface should be used for [Running Initial Setup](#), as well as all management related functionality from the web browser.

### Ethernet IP

The SBC SWe Lite system has four logical interfaces. In most deployments, one of these logical interfaces (typically **Ethernet 1 IP**) is assigned an IP address used for transporting all the VOIP media packets (e.g., RTP, SRTP) and all protocol packets (e.g., SIP, RTCP, TLS). DNS servers in the customer's network should map the SBC SWe Lite system hostname to this IP address. The hostname, or these IP addresses, may be used by UC-enabling systems such as SIP-phones, IP-PBX and Microsoft Lync Servers and for accessing the SBC SWe Lite WebUI.

In the default software, **Ethernet 1 IP** is enabled and an IPv4 IP address is acquired via a connected DHCP server. This IP address is used for performing Initial Setup on the SBC SWe Lite. See [Running Initial Setup](#) for more information. The default IP address for the logical interface named **Ethernet 2 IP** is 192.168.129.2. After initial configuration, you may configure this logical interface using Settings or Tasks tabs in the WebUI.

Below are examples of the Admin and Ethernet IP interfaces configuration.

Figure 15: Admin IP Configuration

Identification/Status	
Interface Name	Admin IP
I/F Index	7
Alias	<input type="text"/>
Description	<input type="text"/>
Admin State	Enabled <input type="button" value="v"/>

Networking	
MAC Address	00:0c:29:a6:bb:b1
IP Addressing Mode	IPv4 <input type="button" value="v"/>

IPv4 Information	
IP Assign Method	Static <input type="button" value="v"/>
Primary Address	192.168.191.56 * X.X.X.X
Primary Netmask	255.255.255.128 * X.X.X.X

**Figure 16:** Ethernet IP Configuration

Identification/Status	
Interface Name	Ethernet 1 IP
I/F Index	8
Alias	<input type="text"/>
Description	<input type="text"/>
Admin State	Enabled <input type="button" value="v"/>

Networking	
MAC Address	00:0c:29:a6:bb:d9
IP Addressing Mode	IPv4 <input type="button" value="v"/>

IPv4 Information	
IP Assign Method	Static <input type="button" value="v"/>
Primary Address	192.168.191.150 * X.X.X.X
Primary Netmask	255.255.255.192 * X.X.X.X
Media Next Hop IP	192.168.191.129 * X.X.X.X

## Static Routes

The *Protocols > IP > Static Route Table* menu path allows you to manually specify the next hop routers used to reach other networks. This is also where you specify the default routes for the connected IP networks (which use 0 . 0 . 0 . 0 as the Destination and Mask).



### DHCP Configuration

When DHCP is configured on an interface, the default Static Route (0.0.0.0/0) will be removed and configured dynamically. To view the dynamically created default route, access the WebUI and navigate to **Protocols > IP > Routing Table**.

To add a new *Static Route*, click on the *plus (+)* icon.

**Figure 17:** Add New Static Route

Static IP Route Table				
<div> <div>+</div> <div>×</div> </div> Total 12 IP Route Rows				
<input type="checkbox"/>	Row ID	Destination IP	Mask	Gateway
<input type="checkbox"/>	1	0.0.0.0	0.0.0.0	192.168.191.1

Figure 18: Create Static IP Route Entry

Row ID	13
Destination IP	<input type="text" value="172.22.244.209"/> * X.X.X.X
Mask	<input type="text" value="255.255.255.255"/> * X.X.X.X
Gateway	<input type="text" value="192.168.191.129"/> * X.X.X.X
Administrative Distance	<input type="text" value="1"/> [1..255]
<div>OK</div>	

- **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 '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 indirectly specifies the preference of the route. Lower values indicate more preferred routes. Typical value is 1 for most static routes, indicating that static routes are preferred to dynamic routes.

## SIP Profiles

The *SIP > SIP Profiles* menu path controls how the SBC Edge communicates with SIP devices. They control important characteristics such as: session timers, SIP header customization, SIP timers, MIME payloads, and option tags.

To add a new *SIP Profile*, click on the *plus (+)* icon.

Figure 19: New SIP Profile

SIP Profile Table	
<div> <div>+</div> <div>×</div> </div> Total 5 SIP Profile Rows	
<input type="checkbox"/>	Description
<input type="checkbox"/>	Default SIP Profile

## OBS SIP Profile Configuration

Figure 20: OBS SIP Profile Configuration 1

Description: Orange_SIPProfile-UDP	
<b>Session Timer</b> Session Timer: <span style="border: 2px solid red;">Disable</span>	<b>MIME Payloads</b> ELIN Identifier: LOC PIDF-LO Passthrough: Enable Unknown Subtype Passthrough: Disable
<b>Header Customization</b> FQDN in From Header: Disable FQDN in Contact Header: Disable Send Assert Header: <span style="border: 2px solid red;">Trusted Only</span> SBC Edge Diagnostics Header: <span style="border: 2px solid red;">Disable</span> Trusted Interface: <span style="border: 2px solid red;">Enable</span> UA Header: Ribbon SBC EDGE Calling Info Source: RFC Standard Diversion Header Selection: Last Record Route Header: RFC 3261 Standard	<b>Options Tags</b> 100rel: <span style="border: 2px solid red;">Not Present</span> Path: <span style="border: 2px solid red;">Not Present</span> Update: Supported

**Figure 21: OBS SIP Profile Configuration 2**

<b>Timers</b> Transport Timeout Timer: 5000 ms [5000..32000] Maximum Retransmissions: RFC Standard Redundancy Retry Timer: 180000 ms [5000..180000] <hr/> <b>RFC Timers</b> Timer T1: 500 ms [100..10000] Timer T2: 4000 ms [1000..80000](>= T1) Timer T4: 5000 ms [1000..100000] Timer D: 32000 ms [5000..640000] Timer B: 32000 ms Timer F: 32000 ms Timer H: 32000 ms (64*TimerT1) Timer J: 4000 ms [4000..640000]	<b>SDP Customization</b> Send Number of Audio Channels: False Connection Info in Media Section: True Origin Field Username: SBC default: SBC Session Name: VoipCall default: VoipCall Digit Transmission Preference: RFC 2833/Voice SDP Handling Preference: Legacy Audio/Fax
---	---



**OBS SIP Profile Configuration**

The configuration in red was modified to fulfill OBS requirements. The rest of the features use the default settings.

## CUCM SIP Profile Configuration

**Figure 22: CUCM SIP Profile Configuration 1**

DescriptionCUCM\_SIPProfile

Session Timer

Session TimerEnable

Minimum Acceptable Timer600\* secs [90..7200]

Offered Session Timer3600\* secs [90..7200]

Terminate On Refresh FailureFalse

MIME Payloads

ELIN IdentifierLOC

PIDF-LO PassthroughEnable

Unknown Subtype PassthroughDisable

Header Customization

FQDN in From HeaderDisable

FQDN in Contact HeaderDisable

Send Assert HeaderTrusted Only

SBC Edge Diagnostics HeaderEnable

Trusted InterfaceEnable

UA HeaderRibbon SBC Edge

Calling Info SourceRFC Standard

Diversion Header SelectionLast

Record Route HeaderRFC 3261 Standard

Options Tags

100relSupported

PathNot Present

TimerSupported

UpdateSupported

Figure 23: CUCM SIP Profile Configuration 2

Timers

Transport Timeout Timer5000ms [5000..32000]

Maximum RetransmissionsRFC Standard

Redundancy Retry Timer180000ms [5000..180000]

RFC Timers

Timer T1500ms [100..10000]

Timer T24000ms [1000..80000](>= T1)

Timer T45000ms [1000..100000]

Timer D32000ms [5000..640000]

Timer B32000ms

Timer F32000ms

Timer H32000ms (64\*TimerT1)

Timer J4000ms [4000..640000]

SDP Customization

Send Number of Audio ChannelsFalse

Connection Info in Media SectionTrue

Origin Field UsernameSBCdefault: SBC

Session NameVoipCalldefault: VoipCall

Digit Transmission PreferenceRFC 2833/Voice

SDP Handling PreferenceLegacy Audio/Fax

CUCM SIP Profile Configuration

The CUCM SIP Profile uses the default settings

Ventafax SIP Profile Configuration

Figure 24: Ventafax SIP Profile Configuration 1

Description <input type="text" value="Ventafax_SIPProfile"/>	
<b>Session Timer</b>	<b>MIME Payloads</b>
Session Timer <input type="text" value="Enable"/>	ELIN Identifier <input type="text" value="LOC"/>
Minimum Acceptable Timer <input type="text" value="600"/> * secs [90..7200]	PIDF-LO Passthrough <input type="text" value="Enable"/>
Offered Session Timer <input type="text" value="3600"/> * secs [90..7200]	Unknown Subtype Passthrough <input type="text" value="Disable"/>
Terminate On Refresh Failure <input type="text" value="False"/>	
<b>Header Customization</b>	<b>Options Tags</b>
FQDN in From Header <input type="text" value="Disable"/>	100rel <input type="text" value="Supported"/>
FQDN in Contact Header <input type="text" value="Disable"/>	Path <input type="text" value="Not Present"/>
Send Assert Header <input type="text" value="Trusted Only"/>	Timer <input type="text" value="Supported"/>
SBC Edge Diagnostics Header <input type="text" value="Enable"/>	Update <input type="text" value="Supported"/>
Trusted Interface <input type="text" value="Enable"/>	
UA Header <input type="text" value="Ribbon SBC Edge"/>	
Calling Info Source <input type="text" value="RFC Standard"/>	
Diversion Header Selection <input type="text" value="Last"/>	
Record Route Header <input type="text" value="RFC 3261 Standard"/>	

**Figure 25: Ventafax SIP Profile Configuration 2**

<b>Timers</b>	<b>SDP Customization</b>
Transport Timeout Timer <input type="text" value="5000"/> ms [5000..32000]	Send Number of Audio Channels <input type="text" value="False"/>
Maximum Retransmissions <input type="text" value="RFC Standard"/>	Connection Info in Media Section <input type="text" value="True"/>
Redundancy Retry Timer <input type="text" value="180000"/> ms [5000..180000]	Origin Field Username <input type="text" value="SBC"/> default: SBC
<b>RFC Timers</b>	Session Name <input type="text" value="VoipCall"/> default: VoipCall
Timer T1 <input type="text" value="500"/> ms [100..10000]	Digit Transmission Preference <input type="text" value="RFC 2833/Voice"/>
Timer T2 <input type="text" value="4000"/> ms [1000..80000] (>= T1)	SDP Handling Preference <input type="text" value="Legacy Audio/Fax"/>
Timer T4 <input type="text" value="5000"/> ms [1000..100000]	
Timer D <input type="text" value="32000"/> ms [5000..640000]	
Timer B <input type="text" value="32000"/> ms	
Timer F <input type="text" value="32000"/> ms	
Timer H <input type="text" value="32000"/> ms (64*TimerT1)	
Timer J <input type="text" value="4000"/> ms [4000..640000]	

## SIP Server Tables

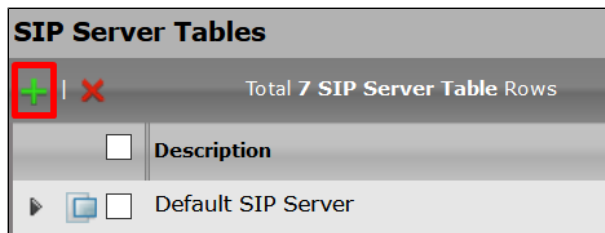
The *SIP > SIP Server Table* menu path contains information about the SIP devices connected to the SBC Edge. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting. The SIP Server supports either an FQDN or IP Address (V4 or V6).

### OBS SIP Server Table

#### BTIP

To add a new *SIP Server Table*, click on the *plus (+)* icon.

**Figure 26: New SIP Server Table**



Set a *Description* and select *SIP Server*.

**Figure 27:** New SIP Server Table Description

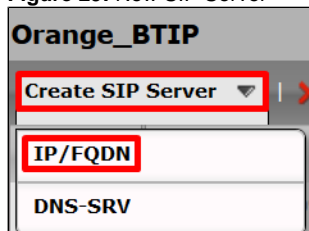
Select the *SIP server table* that has just been created.

**Figure 28:** SIP Server Table Path



Click on the *Create SIP Server > IP/FQDN* icon to add a new SIP Server.

**Figure 29:** New SIP Server



**Orange BTIP SIP Servers**

BTIP uses two SIP Servers, the first one works as an active server and the second one as a backup.

**Figure 30:** BTIP SIP Server 1



Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	300 * secs [30..300]
Host FQDN/IP	[Redacted]	Recover Frequency	5 * secs [5..300]
Port	5060 * [1..65535]	Local Username	Ribbon * Local Username of SBC Edge
Protocol	UDP	Peer Username	Ribbon * Peer Username of sip server

Remote Authorization and Contacts	
Remote Authorization Table	None
Contact Registrant Table	None
Session URI Validation	Liberal

**Figure 31: BTIP SIP Server 2**

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	2	Keep Alive Frequency	300 * secs [30..300]
Host FQDN/IP	[Redacted]	Recover Frequency	5 * secs [5..300]
Port	5060 * [1..65535]	Local Username	Ribbon * Local Username of SBC Edge
Protocol	UDP	Peer Username	Ribbon * Peer Username of sip server

Remote Authorization and Contacts	
Remote Authorization Table	None
Contact Registrant Table	None
Session URI Validation	Liberal



#### BTIP SIP Servers Configuration

The configuration in red was modified to fulfill OBS requirements. The rest of the features use the default settings.

## BTalk

To add a new *SIP Server Table*, click on the *plus (+)* icon.

**Figure 32: New SIP Server Table**

SIP Server Tables	
Total 7 SIP Server Table Rows	
<input type="checkbox"/>	Description
<input type="checkbox"/>	Default SIP Server

Set a *Description* and select *SIP Server*.

**Figure 33: New SIP Server Table Description**

**Create SIP Server Table**
November 09, 2020

Row ID
2

Description

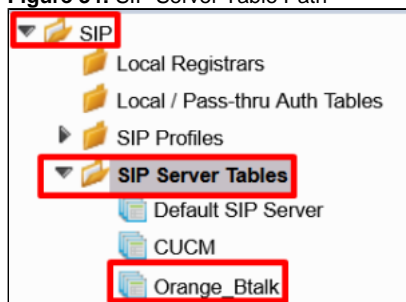
Type

SIP Server ▾

OK

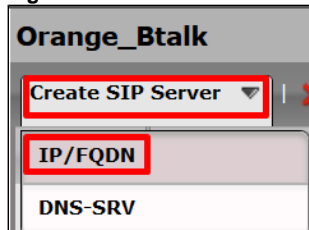
Select the *SIP server table* that has just been created.

**Figure 34:** SIP Server Table Path



Click on the *Create SIP Server > IP/FQDN* icon to add a new SIP Server.

**Figure 35:** New SIP Server



**Orange BTalk SIP Servers**

BTalk uses two SIP Servers, the first one works as an active server and the second one as a backup.

**Figure 36:** BTalk SIP Server 1

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options ▾
Priority	1 ▾	Keep Alive Frequency	300 * secs [30..300]
Host FQDN/IP	<div style="border: 1px solid #ff0000; background-color: black; width: 100px; height: 15px;"></div> *	Recover Frequency	5 * secs [5..300]
Port	5060 * [1..65535]	Local Username	Ribbon * Local Username of SBC Edge
Protocol	UDP ▾ *	Peer Username	Ribbon * Peer Username of sip server
<b>Remote Authorization and Contacts</b>			
Remote Authorization Table	None ▾ +		
Contact Registrant Table	None ▾ +		
Session URI Validation	Liberal ▾		

**Figure 37:** BTalk SIP Server 2

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	2	Keep Alive Frequency	300 * secs [30..300]
Host FQDN/IP		Recover Frequency	5 * secs [5..300]
Port	5060 * [1..65535]	Local Username	Ribbon * Local Username of SBC Edge
Protocol	UDP *	Peer Username	Ribbon * Peer Username of sip server

Remote Authorization and Contacts	
Remote Authorization Table	None +
Contact Registrant Table	None +
Session URI Validation	Liberal



**BTalk SIP Servers Configuration**

The configuration in red was modified to fulfill OBS requirements. The rest of the features use the default settings.

## CUCM SIP Server Table

To add a new *SIP Server Table*, click on the *plus (+)* icon.

**Figure 38:** New SIP Server Table

SIP Server Tables	
	Total 7 SIP Server Table Rows
<input type="checkbox"/>	Description
<input type="checkbox"/>	Default SIP Server

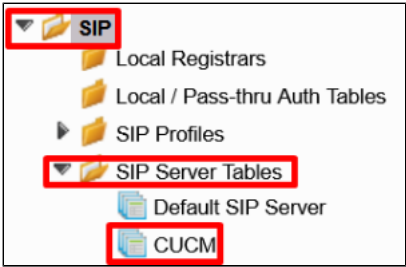
Set a *Description* and select *SIP Server*.

**Figure 39:** New SIP Server Table Description

Create SIP Server Table		November 09, 2020
Row ID	2	
Description	<input type="text" value="CUCM"/>	
Type	SIP Server	
<div>OK</div>		

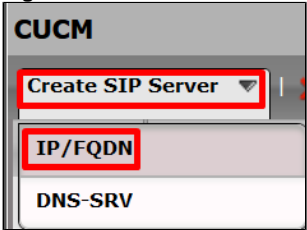
Select the *SIP server table* that has just been created.

**Figure 40:** SIP Server Table Path



Click on the *Create SIP Server > IP/FQDN* icon to add a new SIP Server.

Figure 41: New SIP Server



The figure below depicts the CUCM SIP Server Configuration.

Figure 42: CUCM SIP Server Configuration

Server Host	Transport
<div>Server Lookup <b>IP/FQDN</b></div> <div>Priority <div>1</div></div> <div>Host FQDN/IP <div>10.35.180.112</div> *</div> <div>Port <div>5060</div> * [1..65535]</div> <div>Protocol <div>UDP</div> *</div>	

Remote Authorization Table 

None

 +

Contact Registrant Table 

None

 +

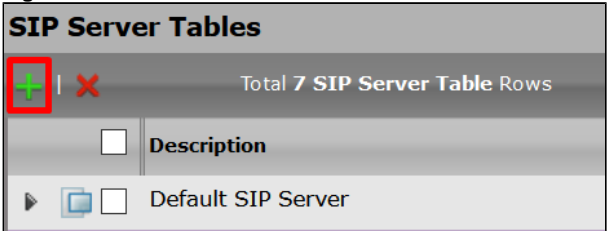
Session URI Validation 

Liberal

Ventafax SIP Server Table

To add a new *SIP Server Table*, click on the *plus (+)* icon.

Figure 43: New SIP Server Table



Set a *Description* and select *SIP Server*.

Figure 44: New SIP Server Table Description

Create SIP Server Table

November 09, 2020

Row ID

2

Description

Ventafax

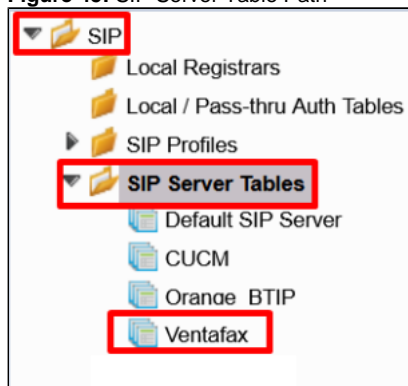
Type

SIP Server

OK

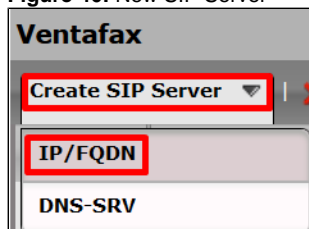
Select the *SIP server table* that has just been created.

**Figure 45:** SIP Server Table Path



Click on the *Create SIP Server > IP/FQDN* icon to add a new SIP Server.

**Figure 46:** New SIP Server



The figure below depicts the Ventafax SIP Server Configuration.

**Figure 47:** Ventafax SIP Server Configuration

Server Host	Transport
<div> <div>Server Lookup</div> <div>IP/FQDN</div> </div> <div> <div>Priority</div> <div>1</div> </div> <div> <div>Host FQDN/IP</div> <div>10.35.137.105</div> </div> <div> <div>Port</div> <div>5060</div> </div> <div> <div>Protocol</div> <div>UDP</div> </div>	<div> <div>Monitor</div> <div>SIP Options</div> </div> <div> <div>Keep Alive Frequency</div> <div>300</div> </div> <div> <div>Recover Frequency</div> <div>5</div> </div> <div> <div>Local Username</div> <div>Ribbon</div> </div> <div> <div>Peer Username</div> <div>Ribbon</div> </div>
<div>Remote Authorization and Contacts</div> <div> <div>Remote Authorization Table</div> <div>None</div> </div> <div> <div>Contact Registrant Table</div> <div>None</div> </div> <div> <div>Session URI Validation</div> <div>Liberal</div> </div>	

## Message Manipulations

The *SIP > Message Manipulation* menu path is used to manipulate the incoming or outgoing messages. This feature is intended to enhance interoperability with different vendor equipment and applications, and for correcting any fixable protocol errors in SIP messages on the fly without any changes to firmware/software.

Although SIP is considered a mature protocol, there are still devices running old firmware and systems that interpret the standard in a non-conforming way. In addition, there are cases where a compliant message may be modified to adapt to an application specific requirement.

This capability consists of two components, condition rules and message rules. Condition rules provide a means to identify which messages and what components in the message must be present before any modifications are performed. For example, I want to modify all INVITE messages with a from uri host of "ribbon.net".

The message rule does the actual modification of a message. Once the conditions of a rule have been met, the message rule(s) are applied. Continuing with the example above, a message rule may change the from uri display name to "Ribbon".



### SIP Message Manipulation

For more information regarding SIP Message Manipulation go to [Appendix A](#)

## Condition Rule Table

Condition rules are simple rules that apply to a specific component of a message (e.g., diversion.uri.host, from.uri.host, etc.) the value of the field specified in the Match Type list box can be matched against a literal value, token, or REGEX.

The Condition Rule Table stores a collection of all the user created Condition Rules.

### Match\_Content-Type

This Condition Rule matches only if *SG User Value 1 = application/sdp*. This condition is created to identify whether the SDP is present or not in the SIP messages.



### SG User Value 1

The *SG User Value 1* is stored using a Message Rule (*Store\_Content-Type*) that will be defined in the next section.

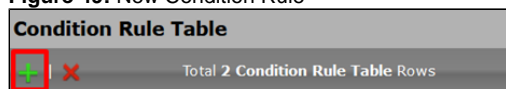
*SG User Value 1* is the predefined name used by the SBC.

To add a new *Condition Rule*, go to *SIP > Message Manipulation > Condition Rule Table* menu path and click on the *plus (+)* icon.

**Figure 48:** Condition Rule Table menu path



**Figure 49:** New Condition Rule



Set the new entry as per the following picture.

**Figure 50:** Match\_Content-Type

Description **Match\_Content-Type**

**Match Type**

Match Type **SG User Value 1** \*

Operation **Equals** ✓

Match Value Type **Literal** ✓

Match Value **application/sdp** \*

## Match\_Anonymous

This *Condition Rule* matches only if *from.displayname = Anonymous*

It compares whether the *display name* that is in the *From* header is equal to *Anonymous*.



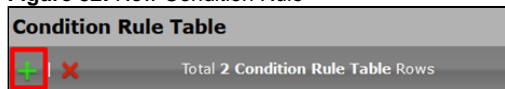
This condition is used by a Message Rule (*Modify\_From\_Anonymous*) defined in the next section. That rule is used to set the format requested by OBS (*sip:anonymous@anonymous.invalid*)

To add a new *Condition Rule*, go to *SIP > Message Manipulation > Condition Rule Table* menu path and click on the *plus (+)* icon.

**Figure 51:** Condition Rule Table menu path



**Figure 52:** New Condition Rule



Set the new entry as per the following picture.

**Figure 53:** Match\_Anonymous

Description

Match Type

Match Type

Operation

Match Value Type

Match Value

## Message Rule Tables

Message Rule Tables are simply sets of Condition Rules and are applied in SIP Signaling Groups when Message Manipulation is enabled.

The *Message Rule Tables* collect *SIP Messages Manipulations Rules* that are applied according to the *Message Type* set in the Message Rule Tables. The following tables are defined to set the format requested by OBS.

Table Description	Rules	Result Type	Message Type	Comments
Add_P-Early-Media	Add P-Early-Media supported	Optional	180, 183	This table applies only to 180 and 183 respond messages.  This table collects the rules used to insert the P-Early-Media header requested by OBS.
	Del_P-Early-Media			
	Add_P-Early-Media sendrecv			
Store_Content-Type	Store Content-Type	Optional	180, 183	This table applies only to 180 and 183 respond messages.  This table collects the rules used to store the Content-type header value. This value is used to know whether the SIP message contains an SDP or not.
Store_User-Agent_Value	Store_User-Agent_Value	Optional	All	This table applies to all messages.  This table collects the rules used to store the IPPBX User-Agent and Server headers values
	Store_Server_Value			
OBS_SIP_Profile_Adaptation_01	Remove_SGI D_From_Hea der	Optional	All	This table applies to all messages.  This table collects rules used to set the format requested by OBS.
	Remove_SGI D_To_Header			
	Modify_User-Agent_Head er			
	Modify_Serv er_header			
	Modify_Allow _header			
OBS_SIP_Profile_Adaptation_02	Modify_PA I	Optional	Requests	This table applies only to request messages.  This table collects rules used to set the format requested by OBS.
	Add plus P-Asserted-Identity			
	Modify_From _Anonymous			



	<a href="#">Modify_Diver sion</a>		
--	---------------------------------------	--	--

Add\_P-Early-Media Table

This table collects the rules that are used to add the *P-Early-Media* header in the SIP 180 and SIP 183 responses.

To add a new *Message Rule Table*, go to the *SIP > Message Manipulation > Message Rule Tables* menu path and click on the *plus (+)* icon.

Figure 54: Message Rule Table menu path



Figure 55: New Message Rule Table



Set the new entry as per the following picture.

Figure 56: Add\_P-Early-Media Table

Create Message Rule Table

Row ID3

DescriptionAdd\_P-Early-Media

Applicable MessagesSelected Messages

Message Selection

180 Ringing  
183 Session Progress

Add/Edit \*  
Remove

Table Result TypeOptional

Add\_P-Early-Media Rules

Description	Rule Type	Result Type	Comments
Add P-Early-Media supported	Header Rule	Optional	It adds the P-Early-Media header value = supported
Del_P-Early-Media	Header Rule	Optional	It deletes the P-Early-Media header to avoid duplicate headers
Add_P-Early-Media sendrecv	Header Rule	Optional	It adds the P-Early-Media header value = sendrecv

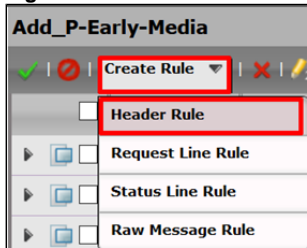
Add P-Early-Media supported

To add a new *Message Rule*, go to the left menu path and click on the *Add\_P-Early-Media* table you created, then click on the *Create Rule > Header Rule* icon.

Figure 57: Add\_P-Early-Media menu path



**Figure 58: New Rule**



Set the new entry as per the following picture.

**Figure 59: Add P-Early-Media supported 1**

<b>Description</b>	Add P-Early-Media supported
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
<b>Header Action</b>	Add
<b>Header Name</b>	P-Early-Media

Once you select *Add* in the *Header Action* field, the bottom section will change its options.

Select *Add* in the *Header Value* field and click on the *Add/Edit* icon.

**Figure 60: Add P-Early-Media supported 2**

<b>Header Value</b>	Add	<b>Add/Edit</b>	'supported'
---------------------	-----	-----------------	-------------

Once you click on the *Add/Edit* icon, a popup screen appears.

Set the configuration as per the following picture.

**Figure 61: Add P-Early-Media supported 3**

<b>Edit Message Field</b>	
<b>Type of Value</b>	Literal
<b>Value</b>	supported

Click on the *Apply* icon to save the changes.

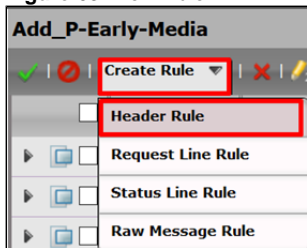
## Del\_P-Early-Media

To add a new *Message Rule*, go to the left menu path and click on the *Add\_P-Early-Media* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 62:** Add\_P-Early-Media menu path



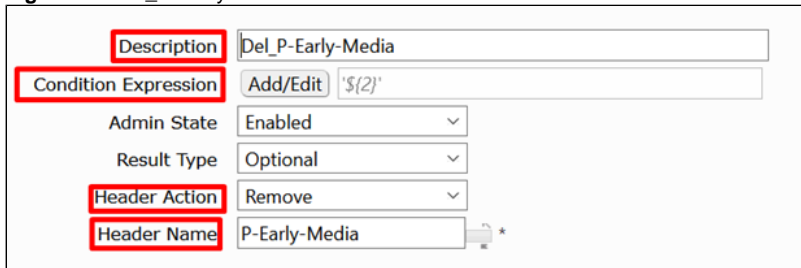
**Figure 63:** New Rule



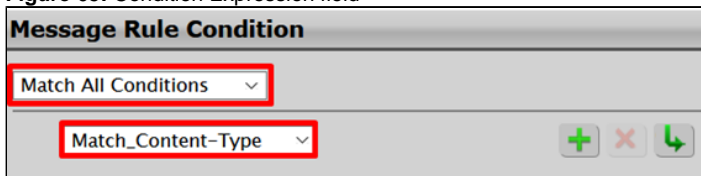
Set the new entry as per the following pictures.

For the *Condition Expression* field, click on the *Add/Edit* icon. A popup screen appears. Set the configuration as per the following picture.

**Figure 64:** Del\_P-Early-Media rule 1



**Figure 65:** Condition Expression field

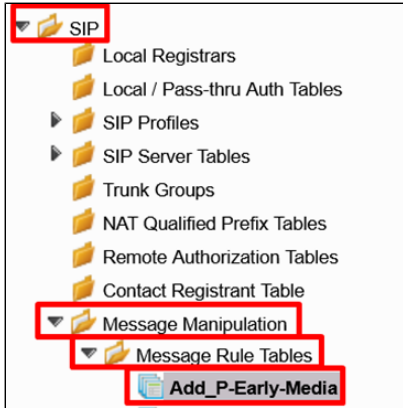


Click on the *Apply* icon to save the changes.

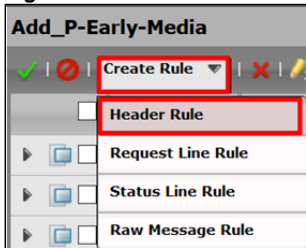
## Add\_P-Early-Media sendrecv

To add a new *Message Rule*, go to the left menu path and click on the *Add\_P-Early-Media* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 66:** Add\_P-Early-Media menu path



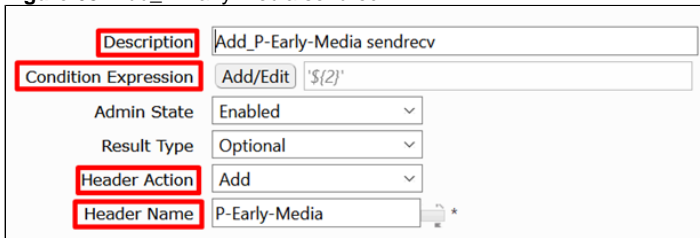
**Figure 67: New Rule**



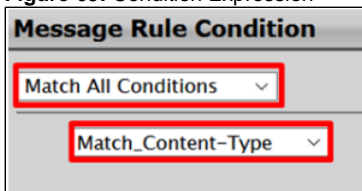
Set the new entry as per the following pictures.

For the *Condition Expression* field, click on the *Add/Edit* icon. A popup screen appears. Set the configuration as per the following picture.

**Figure 68: Add\_P-Early-Media sendrecv 1**



**Figure 69: Condition Expression**



Once you select *Add* in the *Header Action* field, the bottom section will change its options. Select *Add* in the *Header Value* field and click on the *Add/Edit* icon.

**Figure 70: Header Value**



Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 71: Message Field**

Edit Message Field

Type of Value

Literal

Value

sendrecv

Click on the *Apply* icon to save the changes.

Store\_Content-Type Table

This table collects the rule that is used to store the *Content-Type* value in the *SG User Value 1*. It applies only to *180* and *183* respond messages.

This table must be applied on the Signaling Group facing the *IPPBX*. Set it as Inbound Message Manipulation.

To add a new *Message Rule Table*, go to the *SIP > Message Manipulation > Message Rule Tables* menu path and click on the *plus (+)* icon.

Figure 72: Message Rule Table menu path

SIP

Local Registrars

Local / Pass-thru Auth Tables

SIP Profiles

SIP Server Tables

Trunk Groups

NAT Qualified Prefix Tables

Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Message Rule Tables

Figure 73: New Message Rule Table

SIP Message Rule Table

Test Selected Tables

Set the new entry as per the following picture.

Figure 74: Store\_Content-Type Table

Create Message Rule Table

Row ID

3

Description

Store\_Content-Type

Applicable Messages

Selected Messages

Message Selection

180 Ringing

183 Session Progress

Add/Edit

Remove

Table Result Type

Optional

Store\_Content-Type Rules

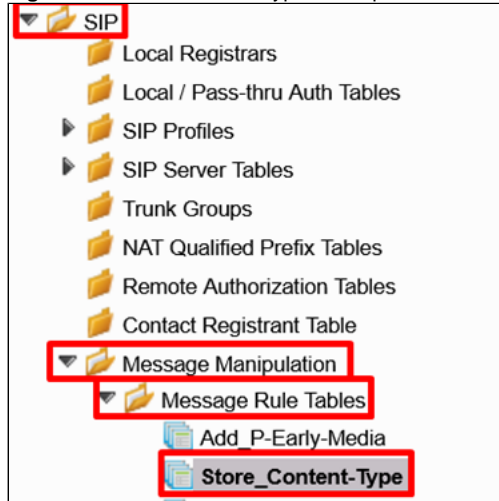
Description	Rule Type	Result Type	Comments
-------------	-----------	-------------	----------

Store Content-Type	Header Rule	Optional	It stores the Content-Type value in the SG User Value 1
--------------------	-------------	----------	---

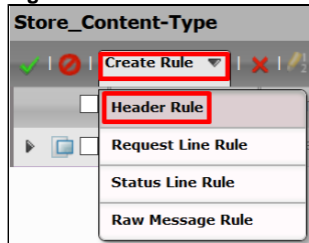
## Store Content-Type

To add a new *Message Rule*, go to the left menu path and click on the *Store\_Content-Type* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 75:** Store\_Content-Type menu path



**Figure 76:** Create Rule



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Copy Value to* in the Header Value field and click on the *Add/Edit* icon.

**Figure 77:** Store Content-Type

Description	Store Content-Type		
Condition Expression	Add/Edit		
Admin State	Enabled		
Result Type	Optional		
Header Action	Modify		
Header Name	Content-Type		
Header Value	Copy Value to	Add/Edit	SG User Value 1

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 78:** Edit Message Field

**Edit Message Field**

Value SG User Value 1

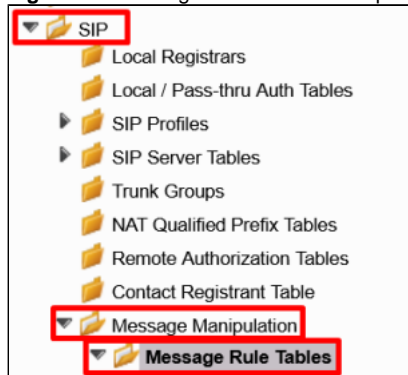
Click on the *Apply* icon to save the changes.

## Store\_User-Agent Table

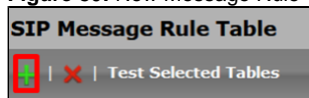
This table collects the rules that are used to store the User-Agent and Server headers values sent by the IPPBX.

To add a new *Message Rule Table*, go to the *SIP > Message Manipulation > Message Rule Tables* menu path and click on the *plus (+)* icon.

**Figure 79:** Message Rule Table menu path



**Figure 80:** New Message Rule Table



Set the new entry as per the following picture.

**Figure 81:** Store\_User-Agent Table

Description	<span>Store_User-Agent</span>
Applicable Messages	<span>All Messages</span>
Table Result Type	<span>Optional</span>

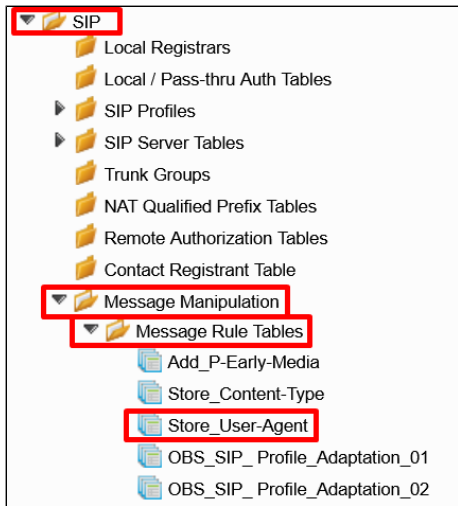
## Store\_User-Agent Rules

Description	Rule Type	Result Type	Comments
<a href="#">Store_User-Agent_Value</a>	Header Rule	Optional	It stores the User-Agent value in the SG User Value 2
<a href="#">Store_Server_Value</a>	Header Rule	Optional	It stores the Server value in the SG User Value 3

## Store\_User-Agent\_Value

To add a new *Message Rule*, go to the left menu path and click on the *Store\_User-Agent* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 82:** Left Menu Path



**Figure 83: CreateRule**



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Copy Value to* in the Header Value field and click on the *Add/Edit* icon.

**Figure 84: Store\_User-Agent\_Value**

Description	Store_User-Agent_Value
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	User-Agent *
Header Value	Copy Value to Add/Edit SG User Value 2

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 85: Edit Message Field**

Value	SG User Value 2
-------	-----------------

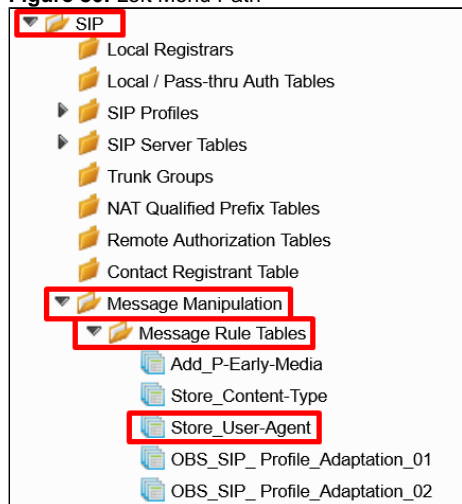
Click on the *Apply* icon to save the changes



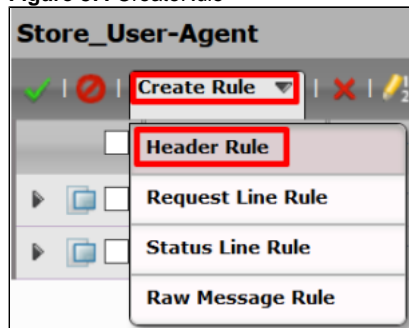
## Store\_Server\_Value

To add a new *Message Rule*, go to the left menu path and click on the *Store\_User-Agent* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 86:** Left Menu Path



**Figure 87:** CreateRule



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Copy Value to* in the Header Value field and click on the *Add/Edit* icon.

**Figure 88:** Store\_Server\_Value

A screenshot of a configuration form for 'Store\_Server\_Value'. The form has several fields: 'Description' (Store\_Server\_Value), 'Condition Expression' (Add/Edit), 'Admin State' (Enabled), 'Result Type' (Optional), 'Header Action' (Modify), and 'Header Name' (Server). Below these fields, there is a section for 'Header Value' with a dropdown menu set to 'Copy Value to' and an 'Add/Edit' button. The 'Copy Value to' dropdown is highlighted with a red box.

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 89:** Edit Message Field

**Edit Message Field**

Value SG User Value 3

Click on the *Apply* icon to save the changes.

## OBS\_SIP\_Profile\_Adaptation\_01 Table

This table collects rules used to set the format requested by OBS. It applies to all messages.

To add a new *Message Rule Table*, go to the *SIP > Message Manipulation > Message Rule Tables* menu path and click on the *plus (+)* icon.

**Figure 90:** Message Rule Table menu path



**Figure 91:** New Message Rule Table



Set the new entry as per the following picture.

**Figure 92:** OBS\_SIP\_Profile\_Adaptation\_01

Description	<span style="border: 1px solid red; padding: 2px;">OBS_SIP_Profile_Adaptation_01</span>
Applicable Messages	<span style="border: 1px solid red; padding: 2px;">All Messages</span>
Table Result Type	<span style="border: 1px solid red; padding: 2px;">Optional</span>

## OBS\_SIP\_Profile\_Adaptation\_01 Rules

Description	Rule Type	Result Type	Comments
<a href="#">Remove_SGID_From_Header</a>	Header Rule	Optional	It removes the <i>SGDI</i> parameter from the <i>FROM</i> header
<a href="#">Remove_SGID_To_Header</a>	Header Rule	Optional	It removes the <i>SGDI</i> parameter from the <i>TO</i> header
<a href="#">Modify_User-Agent_Header</a>	Header Rule	Optional	It sets the <i>User-Agent</i> value as per OBS requirements
<a href="#">Modify_Server_header</a>	Header Rule	Optional	It sets the <i>Server</i> value as per OBS requirements
<a href="#">Modify_Allow_header</a>	Header Rule	Optional	It sets the <i>Allow</i> value as per OBS requirements

## Remove\_SGID\_From\_Header

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_01* table you created, then click on the *Create Rule > Header Rule* icon.

Figure 93: Left Menu Path

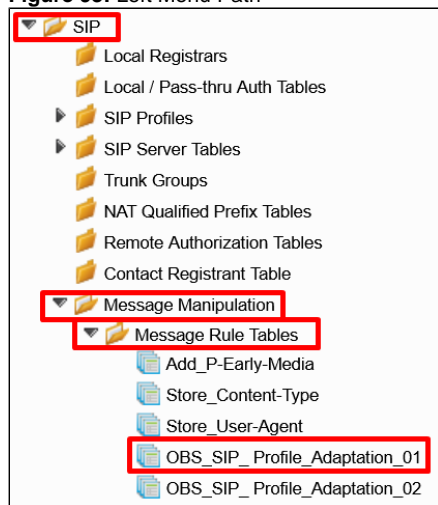


Figure 94: Create Rule



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Ignore* in the Header Value field and click the *plus* (+) icon in the *Header Parameters* section.

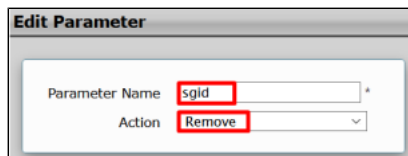
Figure 95: Remove\_SGID\_From\_Header

The screenshot shows the configuration form for 'Remove\_SGID\_From\_Header'. The fields are: Description (Remove\_SGID\_From\_Header), Condition Expression (Add/Edit), Admin State (Enabled), Result Type (Optional), Header Action (Modify), Header Name (From), and Header Value (Ignore). The 'Header Parameters' section is expanded, showing a table with 1 row.

Header Parameters	
+	Total 1 SPRHeaderParam Row

Once you click on the *plus* (+) icon, a popup screen appears. Set the configuration as per the following picture.

Figure 96: Edit Parameter



**Edit Parameter**

Parameter Name:  \*

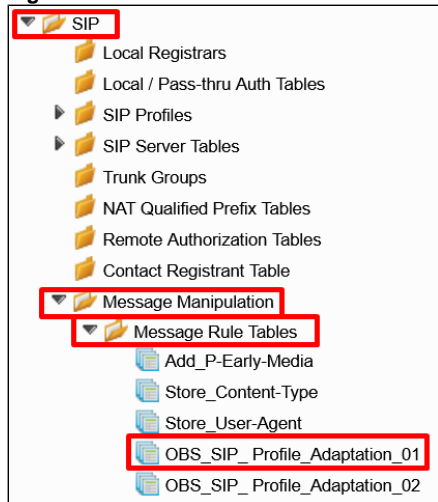
Action:

Click on the *Apply* icon to save the changes.

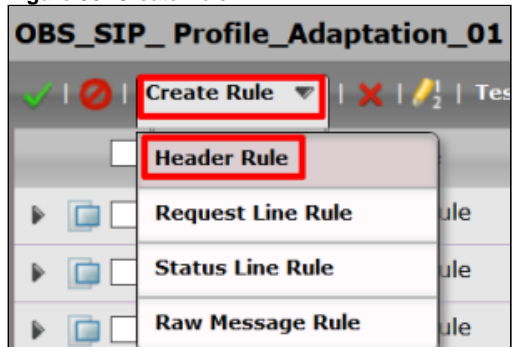
## Remove\_SGID\_To\_Header

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_01* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 97:** Left Menu Path



**Figure 98:** Create Rule



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Ignore* in the Header Value field and click the *plus* (+) icon in the *Header Parameters* section.

**Figure 99:** Remove\_SGID\_To\_Header

Description	Remove_SGID_To_Header
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	To

Header Value	Ignore
--------------	--------

Header Parameters	
<div> <div></div> <div></div> </div>	Total 1 SPRHeaderParam Row

Once you click on the *plus (+)* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 100: Edit Parameter**

Edit Parameter	
Parameter Name	sgld
Action	Remove

Click on the *Apply* icon to save the changes.

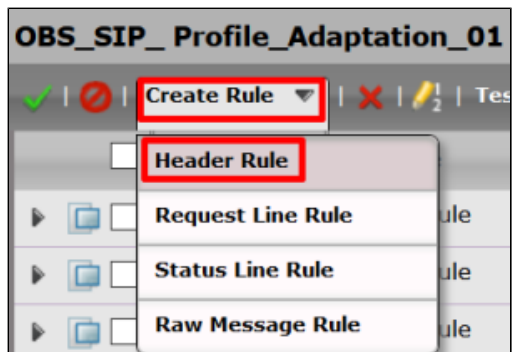
## Modify\_User-Agent\_Header

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_01* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 101: Left Menu Path**

<div> <div></div> <div>SIP</div> </div> <div> <div></div> <div>Local Registrars</div> </div> <div> <div></div> <div>Local / Pass-thru Auth Tables</div> </div> <div> <div></div> <div>SIP Profiles</div> </div> <div> <div></div> <div>SIP Server Tables</div> </div> <div> <div></div> <div>Trunk Groups</div> </div> <div> <div></div> <div>NAT Qualified Prefix Tables</div> </div> <div> <div></div> <div>Remote Authorization Tables</div> </div> <div> <div></div> <div>Contact Registrant Table</div> </div> <div> <div></div> <div>Message Manipulation</div> </div> <div> <div></div> <div>Message Rule Tables</div> </div> <div> <div></div> <div>Add_P-Early-Media</div> </div> <div> <div></div> <div>Store_Content-Type</div> </div> <div> <div></div> <div>Store_User-Agent</div> </div> <div> <div></div> <div>OBS_SIP_Profile_Adaptation_01</div> </div> <div> <div></div> <div>OBS_SIP_Profile_Adaptation_02</div> </div>
--

**Figure 102: Create Rule**



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 103:** Modify\_User-Agent\_Header

 A screenshot of the 'Modify\_User-Agent\_Header' configuration form. The form contains several fields: 'Description' (Modify\_User-Agent\_Header), 'Condition Expression' (Add/Edit), 'Admin State' (Enabled), 'Result Type' (Optional), 'Header Action' (Modify), and 'Header Name' (User-Agent). Below these is a section for 'Header Value' with a dropdown set to 'Modify' and an 'Add/Edit' button. The 'Add/Edit' button and the 'Modify' dropdown are highlighted with red boxes.

Once you click on the *Add/Edit* icon a popup screen appears. Set the configuration as per the following picture.

**Figure 104:** Edit Message Field

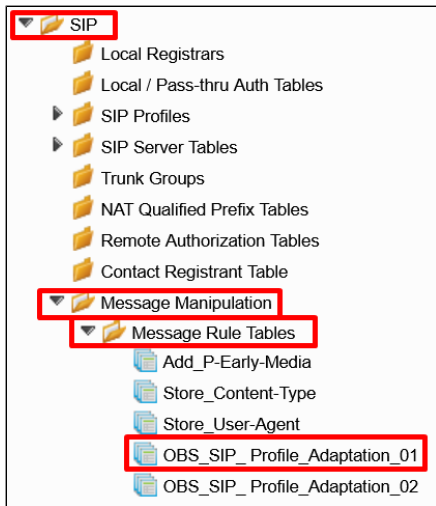
 A screenshot of the 'Edit Message Field' configuration form. The form contains several fields: 'Type of Value' (Token), 'Value' (SG User Value 2), 'Prefix' (IPBX\_), and 'Suffix' (\_SBC Ribbon V9.0.0). The 'Token' dropdown, 'SG User Value 2' text field, 'IPBX\_' text field, and '\_SBC Ribbon V9.0.0' text field are all highlighted with red boxes.

Click on the *Apply* icon to save the changes.

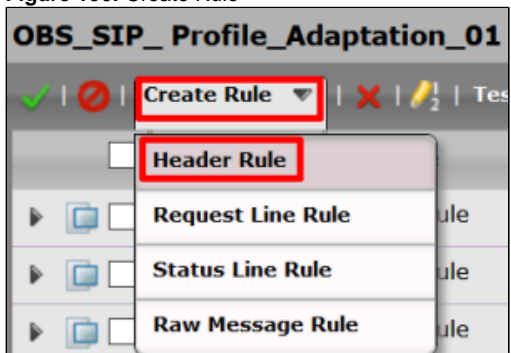
## Modify\_Server\_Header

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_01* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 105:** Left Menu Path



**Figure 106:** Create Rule



Set the new entry as per the following picture.

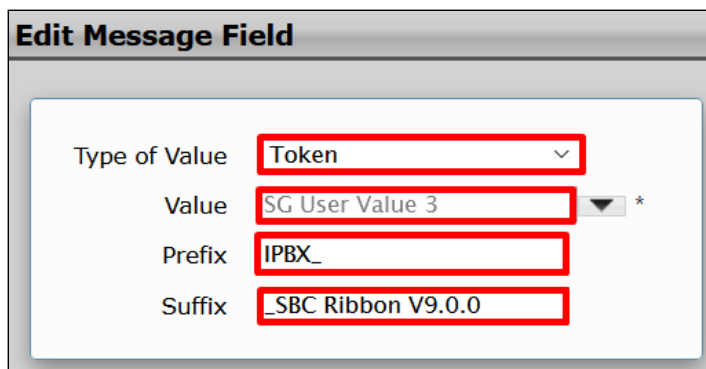
Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 107:** Modify\_Server\_Header

Description	Modify_Server_header	
Condition Expression	Add/Edit	
Admin State	Enabled	
Result Type	Optional	
Header Action	Modify	
Header Name	Server	
Header Value	Modify	Add/Edit 'IPBX_' + SG User Value 3 + '_SE'

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 108:** Edit Message Field



**Edit Message Field**

Type of Value: **Token**

Value: **SG User Value 3**

Prefix: **IPBX\_**

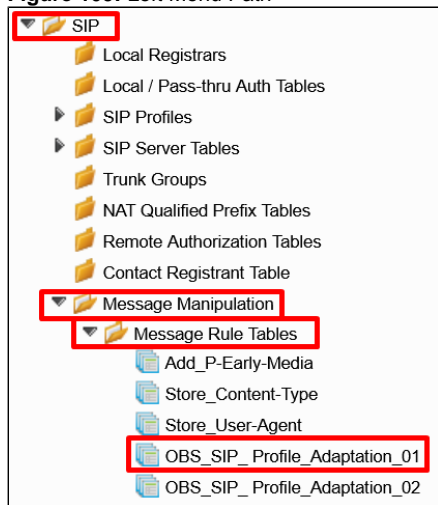
Suffix: **\_SBC Ribbon V9.0.0**

Click on the *Apply* icon to save the changes.

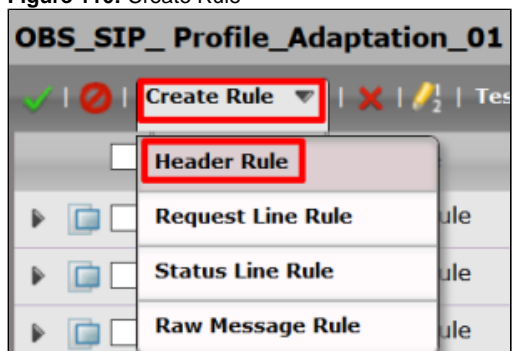
## Modify\_Allow\_header

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_01* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 109:** Left Menu Path



**Figure 110:** Create Rule



Set the new entry as per the following picture.

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 111:** Modify\_Allow\_Header



Description	Modify-Allow_header
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	Allow
Header Value	Modify Add/Edit 'INVITE, ACK, BYE, CANCEL, OP;

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 112: Edit Message Field**

### Edit Message Field

Type of Value	Literal
Value	INVITE, ACK, BYE, CANCEL, *



**Edit Message Field**

The *Value* field should contains the following information:

INVITE, ACK, BYE, CANCEL, OPTIONS, UPDATE

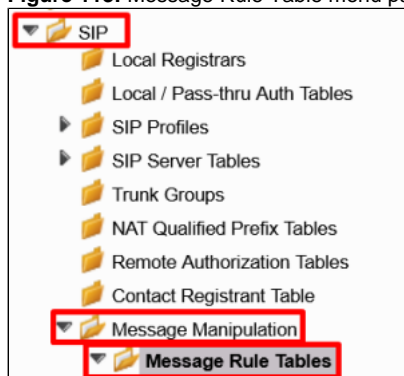
Click on the *Apply* icon to save the changes.

## OBS\_SIP\_Profile\_Adaptation\_02 Table

This table collects rules used to set the format requested by OBS. It applies to all messages.

To add a new *Message Rule Table*, go to the *SIP > Message Manipulation > Message Rule Tables* menu path and click on the *plus (+)* icon.

**Figure 113: Message Rule Table menu path**



**Figure 114: New Message Rule Table**

### SIP Message Rule Table

| 
 | Test Selected Tables

Set the new entry as per the following picture.

**Figure 115:** OBS\_SIP\_Profile\_Adaptation\_02

Description	OBS_SIP_Profile_Adaptation_02
Applicable Messages	All Requests
Table Result Type	Optional

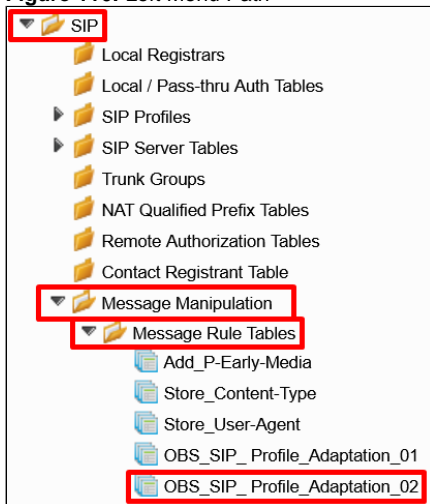
## OBS\_SIP\_Profile\_Adaptation\_02 Rules

Description	Rule Type	Result Type	Comments
Modify_PA1	Header Rule	Optional	It sets the host part of the URI as the local SWeLite IP address
Add plus P-Asserted-Identity	Header Rule	Optional	It adds a <i>plus</i> (+) in the user part of the URI
Modify_From_Anonymous	Header Rule	Optional	When the SBC receives an <i>anonymous</i> call, the FROM header is modified according to OBS requirements
Modify_Diversion	Header Rule	Optional	It adds a <i>plus</i> (+) in the user part of the URI and adds the <i>counter</i> parameter

## Modify\_PA1

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_02* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 116:** Left Menu Path



**Figure 117:** Create Rule



Set the new entry as per the following picture

**Figure 118: Modify\_PA1**

Description	Modify_PA1
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	P-Asserted-Identity
Header Ordinal Number	1st

Once you select *Modify* in the Header Action field, the bottom section will change its options. Click on the arrow next to *Header Value*, then click on the arrow next to *URI*. Select *Modify* in the *URI Host* field and click on the *Add/Edit* icon

**Figure 119: URI Host**

Header Value	Ignore
URI Scheme	Ignore
URI User Info	Ignore
URI Host	Modify
URI Port	Ignore

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 120: Edit Message Field**

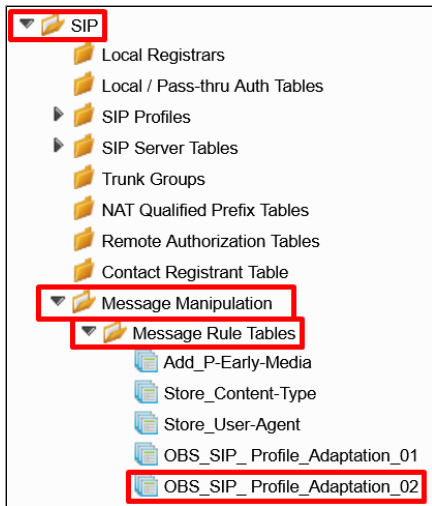
Type of Value	Token
Value	from.uri.host
Prefix	
Suffix	

Click on the *Apply* icon to save the changes.

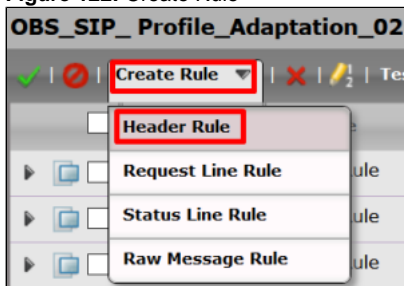
### Add plus P-Asserted-Identity

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_02* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 121: Left Menu Path**



**Figure 122: Create Rule**



Set the new entry as per the following picture

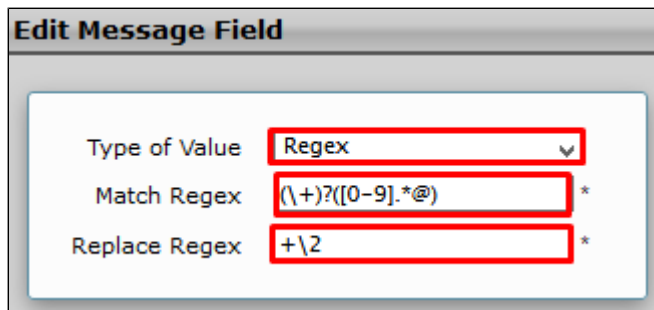
Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 123: Add plus P-Asserted-Identity**

Description	Add plus P-Asserted-Identity		
Condition Expression	Add/Edit		
Admin State	Enabled		
Result Type	Optional		
Header Action	Modify		
Header Name	P-Asserted-Identity		
Header Ordinal Number	1st		
<div> ▶ Header Value Modify Add/Edit Match: (\+)?([0-9].*@) </div>			

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 124: Edit Message Field**



**Edit Message Field**

Type of Value: **Regex**

Match Regex: **(\+)?([0-9].\*@)**

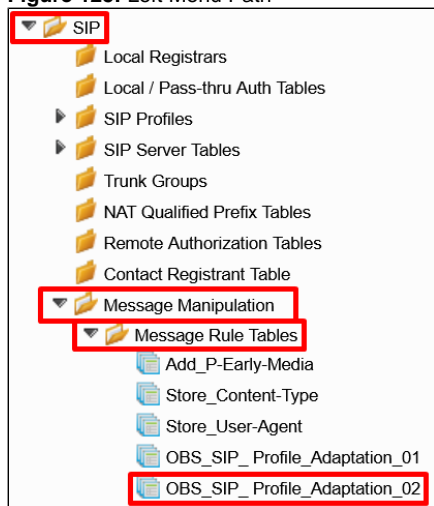
Replace Regex: **+\2**

Click on the *Apply* icon to save the changes.

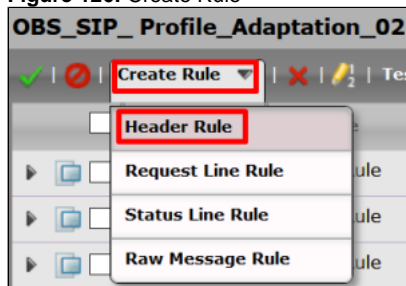
## Modify\_From\_Anonymous

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_02* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 125: Left Menu Path**



**Figure 126: Create Rule**



Set the new entry as per the following picture.

**Figure 127: Modify\_From\_Anonymous**

Description	Modify_From_Anonymous
Condition Expression	<span>Add/Edit</span> \${3}
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From

Click on the *Add/Edit* icon next to *Condition Expression* to set the *Message Rule Condition*.

**Figure 128:** Condition Expression

Description	Modify_From_Anonymous
Condition Expression	<span>Add/Edit</span> \${3}
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	From

Set the *Message Rule Condition* as per the following picture

**Figure 129:** Message Rule Condition

Message Rule Condition	
Match All Conditions	▼
Match_Anonymous	▼

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 130:** Header Value

▶ Header Value	Modify	<span>Add/Edit</span>	Match: <.*>
----------------	--------	-----------------------	-------------

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 131:** Edit Message Field

Type of Value	Regex
Match Regex	<.*>
Replace Regex	<sip:anonymous@anonym



### Edit Message Field

The *Replace Regex* field should contain the following value:

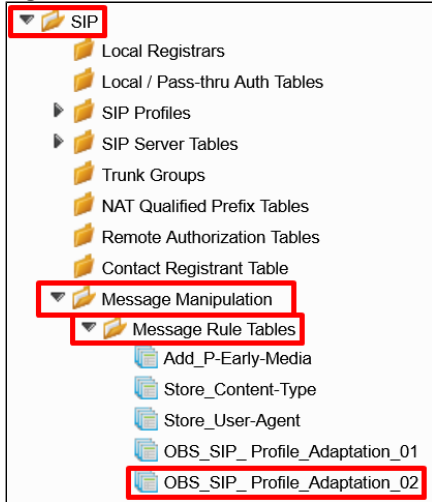
<sip:anonymous@anonymous.invalid>

Click on the *Apply* icon to save the changes.

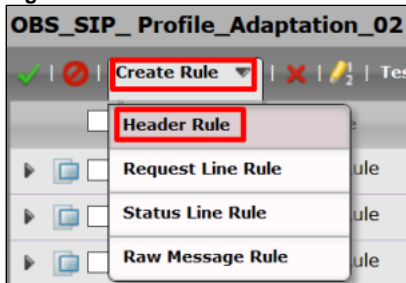
## Modify\_Diversion

To add a new *Message Rule*, go to the left menu path and click on the *OBS\_SIP\_Profile\_Adaptation\_02* table you created, then click on the *Create Rule > Header Rule* icon.

**Figure 132: Left Menu Path**



**Figure 133: Create Rule**



Set the new entry as per the following picture

Once you select *Modify* in the Header Action field, the bottom section will change its options. Select *Modify* in the Header Value field and click on the *Add/Edit* icon.

**Figure 134: Add plus P-Asserted-Identity**

Description	Modify_Diversion
Condition Expression	Add/Edit
Admin State	Enabled
Result Type	Optional
Header Action	Modify
Header Name	Diversion
Header Ordinal Number	1st
Header Value	Modify Add/Edit Match: (\+)?([0-9].*@)

Once you click on the *Add/Edit* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 135:** Edit Message Field

**Edit Message Field**

Type of Value: Regex

Match Regex: (\+)?([0-9].\*@)\*

Replace Regex: +\2\*

Click on the *plus (+)* icon in the *Header Parameters* section.

**Figure 136:** Header Parameters

**Header Parameters**

+ - Total 1 SPRHeaderParam Row

Once you click on the *plus (+)* icon, a popup screen appears. Set the configuration as per the following picture.

**Figure 137:** Counter Parameter

**Edit Parameter**

Parameter Name: counter\*

Action: Add

Type of Value: Literal

Value: 1\*

Click on the *Apply* icon to save the changes.

## Media Profiles

The *Media > Media Profiles* menu path allows you to specify the individual voice and fax compression codecs and their associated settings, for inclusion in a Media List. Different codecs provide varying levels of compression, allowing one to reduce bandwidth requirements at the expense of voice quality.

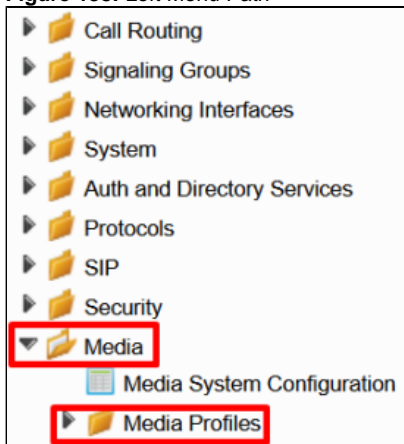
**Table 3:** OBS codecs

Description	Codec	Payload Size
<a href="#">G.722</a>	G.722	20 ms
<a href="#">Default G711A</a>	G.711 A-Law	20 ms
<a href="#">G.729</a>	G.729	20 ms
<a href="#">Default G711U</a>	G.711 U-Law	20 ms
<a href="#">T38</a>	T.38 Fax	



To create a new *Media Profile*, go to *Media > Media Profiles* on the left menu path.

**Figure 138:** Left Menu Path



## G.722

Click on the *Create Media Profile > Voice Codec Profile* icon.

**Figure 139:** Create Media Profile



Set the new G.722 codec as per the following picture.

**Figure 140:** G722

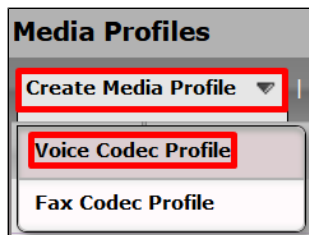
Voice Codec Configuration	
Description	G.722
Codec	G.722
Rate	64000 b/s
Payload Size	20 ms

Click on the *Apply* icon to save the changes.

## Default G711A

Click on the *Create Media Profile > Voice Codec Profile* icon.

**Figure 141:** Create Media Profile



Set the G.711 A-law codec as per the following picture.

Figure 142: G711A

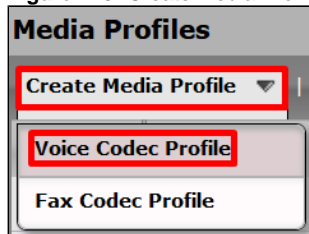
Voice Codec Configuration	
Description	Default G711A
Codec	G.711 A-Law
Payload Size	20 ms

Click on the *Apply* icon to save the changes.

## G.729

Click on the *Create Media Profile > Voice Codec Profile* icon.

Figure 143: Create Media Profile



Set the G.729 codec as per the following picture.

Figure 144: G729

Voice Codec Configuration	
Description	G.729
Codec	G.729
Payload Size	20 ms

Click on the *Apply* icon to save the changes.

## Default G711U

Click on the *Create Media Profile > Voice Codec Profile* icon.

Figure 145: Create Media Profile

**Media Profiles**

- Create Media Profile ▼
- Voice Codec Profile
- Fax Codec Profile

Set the *G.711 U-law* codec as per the following picture.

**Figure 146:** G711U

**Voice Codec Configuration**

Description	Default G711u
Codec	G.711 μ-Law ▼
Payload Size	20 ▼ ms

Click on the *Apply* icon to save the changes.

## T38

Click on the *Create Media Profile > Fax Codec Profile* icon.

**Figure 147:** Create Media Profile

**Media Profiles**

- Create Media Profile ▼
- Voice Codec Profile
- Fax Codec Profile

Set the *T.38* codec as per the following picture.

**Figure 148:** T38

**Fax Codec Configuration**

Description	T38
Codec	T.38 Fax
Maximum Rate	14400 ▼ b/s
Signaling Packet Redundancy	3 [0..7]
Payload Packet Redundancy	0 [0..3]
Error Correction Mode	Enabled ▼
Training Confirmation Procedure	Send Over Network
Fallback to Passthrough	Enabled ▼

Click on the *Apply* icon to save the changes.

## Media Lists

The *Media > Media List* menu path allows you to specify a set of codecs and fax profiles that are allowed on a given SIP Signaling Group. They contain one or more Media Profiles, which must first be defined in *Media Profiles*. These lists allow you to accommodate specific transmission requirements, and SIP devices that only implement a subset of the available voice codecs.

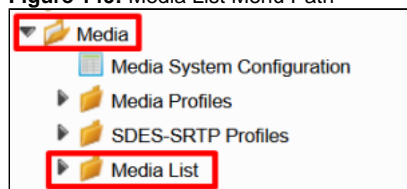
**Table 4:** Media Lists

Description	Media Profiles List	SDES-SRTP Profile	Media DSCP	Silence Suppression	Modem Passthrough	Fax Passthrough	Fax Tone Detection
<a href="#">CUCM_MediaList</a>	Default G711A G.729	None	46	Disabled	Enabled	Enabled	Enabled
<a href="#">Orange_MediaList-UDP</a>	Default G711A G.729 T38	None	46	Disabled	Enabled	Enabled	Enabled

### CUCM\_MediaList

To add a *Media List*, go to *Media > Media List* on the left menu path.

**Figure 149:** Media List Menu Path



Click on the *plus (+)* icon to add a new entry.

**Figure 150:** New Media List



Configure the new entry as per the following picture.

**Figure 151:** CUCM\_MediaList

Description: **CUCM\_MediaList**

Media Profiles List: **Default G711A**  
**G.729**

SDES-SRTP Profile: **None**

Media DSCP: **46**

Dead Call Detection: **Disabled**

Silence Suppression: **Disabled**

Buttons: Up, Down, Add/Edit, Remove

Associated SIP SG Listen Ports should be TLS only. +

\* [0..63]

Digit Relay	
Digit (DTMF) Relay Type	RFC 2833
Digit Relay Payload Type	101 [96..127]

Passthrough/Tone Detection	
Modem Passthrough	Enabled
Fax Passthrough	Enabled
Fax Tone Detection	Enabled

Click on the *Apply* icon to save the changes.

## Orange\_MediaList-UDP

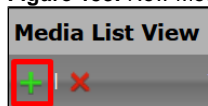
To add a *Media List*, go to *Media > Media List* on the left menu path.

**Figure 152:** Media List Menu Path



Click on the *plus (+)* icon to add a new entry.

**Figure 153:** New Media List



Configure the new entry as per the following picture.

**Figure 154:** Orange\_MediaList-UDP

Description	Orange_MediaList-UDP	
Media Profiles List	<div> Default G711A  G.729  T38 </div>	<div> Up  Down  Add/Edit  Remove </div>
SDES-SRTP Profile	None	Associated SIP SG Listen Ports should be TLS only.
Media DSCP	46	* [0..63]
Dead Call Detection	Disabled	
Silence Suppression	Disabled	

Digit Relay	
Digit (DTMF) Relay Type	RFC 2833
Digit Relay Payload Type	101 [96..127]

Passthrough/Tone Detection	
Modem Passthrough	Enabled
Fax Passthrough	Enabled
Fax Tone Detection	Enabled

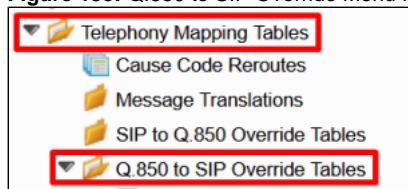
Click on the *Apply* icon to save the changes.

## Q.850 to SIP Override Table

SIP and ISDN use different response messages to communicate why a call failed or could not be connected (Q.850 for ISDN and SIP Responses for SIP). By default, the SBC Edge uses RFC 4497 to map these to each other. The *Telephony Mapping Tables > Q.850 to SIP Override Tables* menu path allows you to override one or more of these mappings to a different message, which is useful for inter-operating with nonstandard equipment.

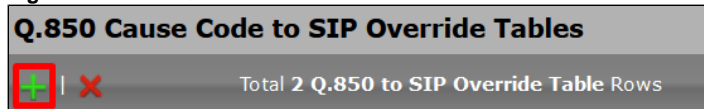
To add a new *Q.850 to SIP Override Table*, go to *Telephony Mapping Tables > Q.850 to SIP Override Tables* on the left menu path.

**Figure 155:** Q.850 to SIP Override Menu Path



To add a new *Q.850 to SIP Override Table*, click on the *plus (+)* icon.

**Figure 156:** New Q.850 to SIP Override Table



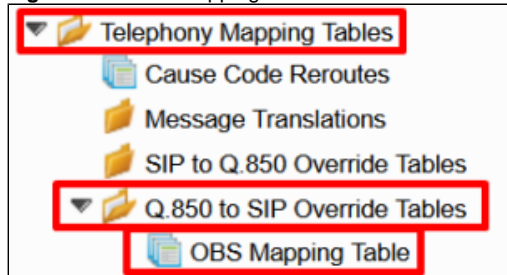
Set the *Description* as per the following picture.

**Figure 157:** Table Description

Description	OBS Mapping Table
-------------	-------------------

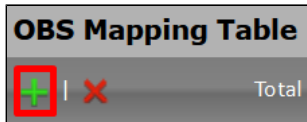
On the left menu path, click on the *OBS Mapping Table*.

**Figure 158:** OBS Mapping Table Menu Path



To add a new entry click on the *plus (+)* icon.

Figure 159: New Entry



Configure the new entry as per the following picture.

Figure 160: Q.850 to SIP

Q.850 Cause Code	27: Destination Out of Order
SIP Response	480 - Temporarily Unavailable

Click on the *Apply* icon to save the changes.

## Signaling Groups

Signaling groups allow telephony channels to be grouped 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. They are also the location from which *Tone Tables* and *Action Sets* are selected.

### From-To\_CUCM

Table 5: CUCM Signaling Group Parameters

Description	Call Routing Table	SIP Profile	SIP Server Table	Media List ID	Federated IP/FQDN	Signaling DSCP	Inbound Message Manipulation	Outbound Message Manipulation
From-To_CUCM	To_Orange	CUCM_SIP Profile	CUCM	CUCM_MediaList	<CUCM IP Address>	40	Store_Content-Type Store_User-Agent	

To add a new *Signaling Group*, go to the *Signaling Groups* menu path.

Figure 161: Signaling Groups Menu Path

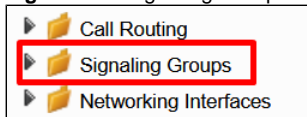
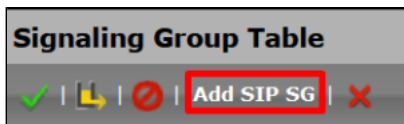


Figure 162: New Signaling Group



Set the new entry as per the following picture.

Figure 163: From-To\_CUCM

Description	From-To_CUCM
Admin State	Enabled
Service Status	Up

### SIP Channels and Routing

Action Set Table	None
Call Routing Table	To_Orange
No. of Channels	60
SIP Profile	CUCM_SIPProfile
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
SIP Server Table	CUCM
Load Balancing	Round Robin
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 Proxy with Local SRTP	*
Supported Video/Application Modes	Proxy Direct	*
Media List ID	CUCM_MediaList	
Proxy Local SRTP	None	
Crypto Profile ID		
Play Ringback	Auto on 180	
Tone Table	Default Tone Table	
Play Congestion Tone	Disable	
Early 183	Enable	
Allow Refresh SDP	Enable	
Music on Hold	Disabled	
RTCP Multiplexing	Disable	

### 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

### SIP IP Details

Teams Local Media Optimization	Disable
Signaling/Media Source IP	
Signaling DSCP	40
NAT Traversal	
ICE Support	Disabled
Static NAT - Outbound	
Outbound NAT Traversal	None
Static NAT - Inbound	
Detection	Disabled

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Listen Ports

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
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

Message Table List

Store\_Content-Type

Store\_User-Agent

\*

Outbound Message Manipulation

Message Table List

\*

Click on the *Apply* icon to save the changes.

From-To\_OrangeBTIP

Table 6: From-To\_OrangeBTIP Signaling Group Parameters

Description	Call Routing Table	SIP Profile	SIP Server Table	Media List ID	Federated IP/FQDN	Signaling DSCP	Q.850 to SIP Override Table	Outbound Message Manipulation
From-To_OrangeBTIP	To_Private	Orange_SIPProfile-UDP	Orange_BTIP	Orange_Media List-UDP	<OBS IP Addresses>	46	OBS Mapping Table	OBS_SIP_Profile_Adaptation_02 OBS_SIP_Profile_Adaptation_01 Add_P-Early-Media

To add a new *Signaling Group*, go to *Signaling Groups* menu path.

Figure 164: Signaling Groups Menu Path

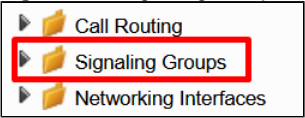
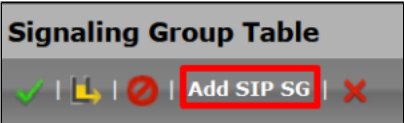


Figure 165: New Signaling Group



Set the new entry as per the following picture.

Figure 166: From-To\_OrangeBTIP

Description	From-To_OrangeBTIP
Admin State	Enabled
Service Status	Up

---

### SIP Channels and Routing

Action Set Table	None
Call Routing Table	To_Private
No. of Channels	10
SIP Profile	Orange_SIPProfile-UDP
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
SIP Server Table	Orange_BTIP
Load Balancing	First
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 Proxy with Local SRTP	*
Supported Video/Application Modes	Proxy Direct	*
Media List ID	Orange_MediaList-UDP	
Proxy Local SRTP	None	
Crypto Profile ID		
Play Ringback	Auto on 180	
Tone Table	Default Tone Table	
Play Congestion Tone	Disable	
Early 183	Disable	
Allow Refresh SDP	Enable	
Music on Hold	Disabled	
RTCP Multiplexing	Disable	

### Mapping Tables

SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	OBS Mapping Table
Pass-thru Peer SIP Response Code	Disable

### SIP IP Details

Teams Local Media Optimization	Disable
Signaling/Media Source IP	
Signaling DSCP	46
<b>NAT Traversal</b>	
ICE Support	Disabled
<b>Static NAT - Outbound</b>	
Outbound NAT Traversal	None
<b>Static NAT - Inbound</b>	
Detection	Disabled

Listen Ports

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
5060	UDP	N/A
5060	TCP	N/A

Federated IP/FQDN

Total 2 SIP Federated IP Rows

IP/FQDN	Netmask/Prefix
	255.255.255.255
	255.255.255.255

Message Manipulation Enabled

Inbound Message Manipulation

Message Table List

\*

Outbound Message Manipulation

Message Table List

OBS\_SIP\_Profile\_Adaptation\_02  
OBS\_SIP\_Profile\_Adaptation\_01  
Add\_P-Early-Media

\*

Click on the *Apply* icon to save the changes.

From-To\_OrangeBtalk

Table 7: From-To\_OrangeBtalk Signaling Group Parameters

Description	Call Routing Table	SIP Profile	SIP Server Table	Media List ID	Federated IP/FQDN	Signaling DSCP	Q.850 to SIP Override Table	Outbound Message Manipulation
From-To_OrangeBtalk	To_Private	Orange_SIPProfile-UDP	Orange_Btalk	Orange_Media List-UDP	<OBS IP Addresses>	46	OBS Mapping Table	OBS_SIP_Profile_Adaptation_02  OBS_SIP_Profile_Adaptation_01  Add_P-Early-Media

To add a new *Signaling Group*, go to *Signaling Groups* menu path.

Figure 167: Signaling Groups Menu Path

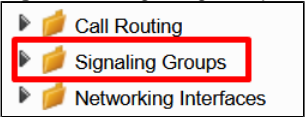


Figure 168: New Signaling Group



Set the new entry as per the following picture.

Figure 169: From-To\_OrangeBtalk

Description	From-To_OrangeStalk
Admin State	Enabled
Service Status	Up

### SIP Channels and Routing

Action Set Table	None
Call Routing Table	To_Private
No. of Channels	10
SIP Profile	Orange_SIPProfile-UDP
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
SIP Server Table	Orange_Btalk
Load Balancing	First
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 Proxy with Local SRTP	*
Supported Video/Application Modes	Proxy Direct	*
Media List ID	Orange_MediaList-UDP	
Proxy Local SRTP Crypto Profile ID	None	
Play Ringback	Auto on 180	
Tone Table	Default Tone Table	
Play Congestion Tone	Disable	
Early 183	Disable	
Allow Refresh SDP	Enable	
Music on Hold	Disabled	
RTCP Multiplexing	Disable	

### Mapping Tables

SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	OBS Mapping Table
Pass-thru Peer SIP Response Code	Disable

### SIP IP Details

Teams Local Media Optimization	Disable
Signaling/Media Source IP	
Signaling DSCP	46
<b>NAT Traversal</b>	
ICE Support	Disabled
<b>Static NAT - Outbound</b>	
Outbound NAT Traversal	None
<b>Static NAT - Inbound</b>	
Detection	Disabled

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Listen Ports			Federated IP/FQDN	
Total 2 SIP Listen Port Rows			Total 2 SIP Federated IP Rows	
Port	Protocol	TLS Profile ID	IP/FQDN	Netmask/Prefix
5060	UDP	N/A		255.255.255.255
5060	TCP	N/A		255.255.255.255
Message Manipulation <b>Enabled</b>				
Inbound Message Manipulation			Outbound Message Manipulation	
Message Table List <div>             *           </div>			Message Table List <div>             OBS_SIP_Profile_Adaptation_02              OBS_SIP_Profile_Adaptation_01              Add_P-Early-Media           </div> *	

Click on the *Apply* icon to save the changes.

## Transformations Tables

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.



### Regular Expressions

Regular Expressions (REGEX) is a basic topic to set Transformation Tables. To know more about REGEX go to [Appendix A](#)

**Table 8:** Transformation Tables

Transformation Table	Transformation Entries
CUCM_Prefixes	To_CUCM
Orange_Btalk	Add plus calling number To_Btalk
Orange_BTIP	Add plus Calling number To_BTIP

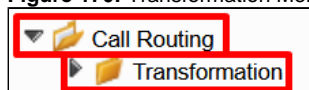
## CUCM\_Prefixes

**Table 9:** CUCM\_Prefixes entries

Description	Match Type	Input Field		Output Field	
		Type	Value	Type	Value
To_CUCM	Optional	Called Address / Number	(\+?)(33.*)	Called Address / Number	\2

To add a new *Transformation Table*, go to *Call Routing > Transformation* on the left menu path.

**Figure 170:** Transformation Menu Path



Set the *Description* as per the following picture.

Figure 171: Table Description

Description

CUCM\_Prefixes

On the left menu path, click on the *CUCM\_Prefixes Table*.

Figure 172: CUCM\_Prefixes Table

Call Routing

Transformation

CUCM\_Prefixes

To\_CUCM

To add a new entry click on the *plus (+)* icon.

Figure 173: New Entry

CUCM\_Prefixes

✓

✗

✚

✖

🔧

Configure the new entry as per the following picture.

Figure 174: Transformation Entry

Description

To\_CUCM

Admin State

Enabled

Match Type

Optional (Match One)

Input Field

Type

Called Address/Number

Value

(\+)?(33.\*)

Output Field

Type

Called Address/Number

Value

\2

Click on the *Apply* icon to save the changes.

Orange\_Btalk

Table 10: Orange\_Btalk entries

Description	Match Type	Input Field		Output Field	
		Type	Value	Type	Value
Add plus calling number	Optional	Calling Address / Number	(\+)?(.*)	Calling Address / Number	+2
To_Btalk	Optional	Called Address / Number	(\+)?(63 64 65 66 67)(.*)	Called Address / Number	+2\3

To add a new *Transformation Table*, go to *Call Routing > Transformation* on the left menu path.

Figure 175: Transformation Menu Path

Call Routing

Transformation

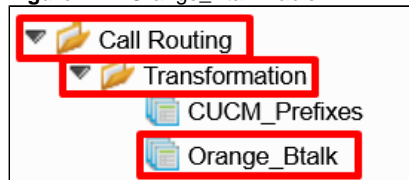
Set the *Description* as per the following picture.

**Figure 176:** Table Description

Description	Orange_Btalk
-------------	--------------

On the left menu path, click on the *Orange\_Btalk Table*.

**Figure 177:** Orange\_Btalk Table



Add plus calling number

To add a new entry click on the *plus (+)* icon.

**Figure 178:** New Entry



Configure the new entry as per the following picture.

**Figure 179:** Transformation Entry

Description	Add plus calling number		
Admin State	Enabled		
Match Type	Optional (Match One)		
<b>Input Field</b>		<b>Output Field</b>	
Type	Calling Address/Number	Type	Calling Address/Number
Value	(\+)?(.*)	Value	+\2

Click on the *Apply* icon to save the changes.

To\_Btalk

To add a new entry click on the *plus (+)* icon.

**Figure 180:** New Entry



Configure the new entry as per the following picture.

**Figure 181:** Transformation Entry

Description

To\_Btalk

Admin State

Enabled

Match Type

Optional (Match One)

Input Field

Type

Called Address/Number

Value

(\+)?(63|64|65|66|67)(.\*)

Output Field

Type

Called Address/Number

Value

+\2\3

Click on the *Apply* icon to save the changes.

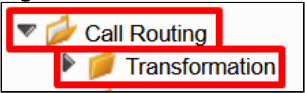
Orange\_BTIP

Table 11: Orange\_BTIP entries

Description	Match Type	Input Field		Output Field	
		Type	Value	Type	Value
Add plus Calling number	Optional	Calling Address / Number	(\+)?(.*)	Calling Address / Number	+\2
To_BTIP	Optional	Called Address / Number	(\+)?(96 33 039910)(.*)	Called Address / Number	+\2\3

To add a new *Transformation Table*, go to *Call Routing > Transformation* on the left menu path.

Figure 182: Transformation Menu Path



Set the *Description* as per the following picture.

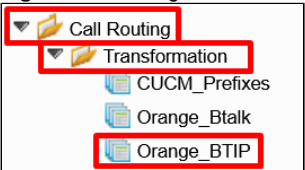
Figure 183: Table Description

Description

Orange\_BTIP

On the left menu path, click on the *Orange\_BTIP Table*.

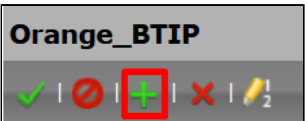
Figure 184: Orange\_BTIP Table



Add plus Calling number

To add a new entry click on the *plus (+)* icon.

Figure 185: New Entry



Configure the new entry as per the following picture.



Figure 186: Transformation Entry

DescriptionAdd plus Calling number

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeCalling Address/Number

Value(\+)?(.\*)

Output Field

TypeCalling Address/Number

Value+\2

Click on the *Apply* icon to save the changes.

To\_BTIP

To add a new entry click on the *plus (+)* icon.

Figure 187: New Entry



Configure the new entry as per the following picture.

Figure 188: Transformation Entry

DescriptionTo\_BTIP

Admin StateEnabled

Match TypeOptional (Match One)

Input Field

TypeCalled Address/Number

Value(\+)?(96|33|039910)(.\*)

Output Field

TypeCalled Address/Number

Value+\2\3

Click on the *Apply* icon to save the changes.

Call Routing Tables

Call Routing allows calls to be carried between signalling groups, thus allowing calls to be carried between ports, and between protocols. Routes are defined by *Call Routing Tables*, which allow for flexible configuration of which calls are carried, and how they are translated. These tables are one of the central connection points of the system, linking *Transformation Tables*, *Message Translations*, *Cause Code Reroute Tables*, *Media Lists* and *Signaling Groups*.

Table 12: Call Routing Tables

Description	Transformation Table
To_Private	CUCM_Prefixes
To_Orange	Orange_Btalk
	Orange_BTIP

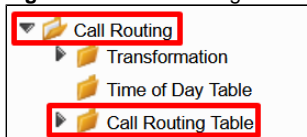
## To\_Private

**Figure 189:** Call Route Entry Parameters

Description	Number/Name Transformation Table	Destination Signaling Groups	Audio Stream Mode	Media List
To_CUCM	CUCM_Prefixes	From-To_CUCM	DSP Preferred over Proxy	CUCM_MediaList

To add a new *Call Routing Table*, go to *Call Routing > Call Routing Table* on the left menu path.

**Figure 190:** Call Routing Table Menu Path



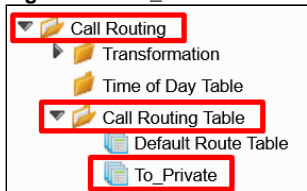
Set the *Description* as per the following picture.

**Figure 191:** Table Description

Description	To_Private
-------------	------------

On the left menu path, click on the *To\_Private Table*.

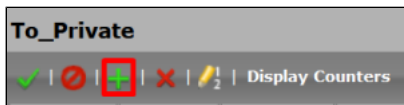
**Figure 192:** To\_Private Table



## To\_CUCM

To add a new entry click on the *plus (+)* icon.

**Figure 193:** New Entry



Configure the new entry as per the following picture.

**Figure 194:** Call Route Entry

Route Details	
Description	To_CUCM
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	CUCM_Prefixes
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) From-To\_CUCM

Up

Down

Add/Edit

Remove

\*

Enable Maximum Call Duration

Disabled

Media

Quality of Service

Audio Stream Mode

DSP preferred over Proxy

Video/Application Stream Mode

Disabled

Proxy SRTP Handling

Relay

Media Transcoding

Disabled

Media List

CUCM\_MediaList

+

Quality Metrics Number of Calls

10

[1..100]

Quality Metrics Time Before Retry

10

[1-60] min.

Min. ASR Threshold

0

% [0..100]

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

ms [1..65535]

Enable Max. Jitter

Enabled

Max. Jitter

3000

ms [1..3000]

Click on the *Apply* icon to save the changes.

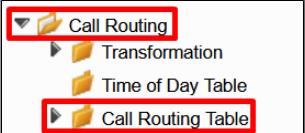
To\_Orange

Figure 195: Call Route Entry Parameters

Description	Number/Name Transformation Table	Destination Signaling Groups	Audio Stream Mode	Media Transcoding	Media List
To_OrangeBtalk	Orange_Btalk	From-To_OrangeBtalk	DSP Preferred over Proxy	Enabled	Orange_MediaList-UDP
To_OrangeBTIP	Orange_BTIP	From-To_OrangeBTIP	DSP Preferred over Proxy	Enabled	Orange_MediaList-UDP

To add a new *Call Routing Table*, go to *Call Routing > Call Routing Table* on the left menu path.

Figure 196: Call Routing Table Menu Path



Set the *Description* as per the following picture.

**Figure 197:** Table Description

Description	To_Orange
-------------	-----------

On the left menu path, click on the *To\_Orange Table*.

**Figure 198:** To\_Orange Table

A navigation menu with the following items: Call Routing (expanded), Transformation, Time of Day Table, Call Routing Table (expanded), Default Route Table, To\_Private, and To\_Orange. Red boxes highlight the 'Call Routing' icon, the 'Call Routing Table' icon, and the 'To\_Orange' icon.

To\_OrangeBtalk

To add a new entry click on the *plus (+)* icon.

**Figure 199:** New Entry

The header bar for the 'To\_Orange' table. It contains a green checkmark, a red circle with a white 'X', a green plus sign (highlighted with a red box), a red circle with a white 'X', and a yellow lightning bolt icon. To the right of these icons is the text 'Display Counters'.

Configure the new entry as per the following picture.

**Figure 200:** Call Route Entry

Route Details	
Description	To_OrangeBtalk
Admin State	Enabled
Route Priority	1
Call Priority	Normal
Number/Name Transformation Table	Orange_Btalk
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) From-To\_OrangeBtalk

Up

Down

Add/Edit

Remove

\*

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP preferred over Proxy

Video/Application Stream Mode

Disabled

Proxy SRTP Handling

Relay

Media Transcoding

Enabled

Media List

Orange\_MediaList-UDP

+

Quality of Service

Quality Metrics Number of Calls

10

[1..100]

Quality Metrics Time Before Retry

10

[1-60] min.

Min. ASR Threshold

0

% [0..100]

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

ms [1..65535]

Enable Max. Jitter

Enabled

Max. Jitter

3000

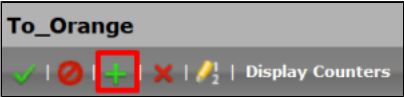
ms [1..3000]

Click on the *Apply* icon to save the changes.

To\_OrangeBTIP

To add a new entry click on the *plus (+)* icon.

Figure 201: New Entry



Configure the new entry as per the following picture.

Figure 202: Call Route Entry

Route Details

Description

To\_OrangeBTIP

Admin State

Enabled

Route Priority

1

Call Priority

Normal

Number/Name Transformation Table

Orange\_BTIP

+

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) From-To\_OrangeBTIP

Up

Down

Add/Edit

Remove

\*

Enable Maximum Call Duration

Disabled

Media

Audio Stream Mode

DSP preferred over Proxy

Video/Application Stream Mode

Disabled

Proxy SRTP Handling

Relay

Media Transcoding

Enabled

Media List

Orange\_MediaList-UDP

+

Quality of Service

Quality Metrics Number of Calls

10

[1..100]

Quality Metrics Time Before Retry

10

[1-60] min.

Min. ASR Threshold

0

% [0..100]

Enable Min MOS Threshold

Disabled

Enable Max. R/T Delay

Enabled

Max. R/T Delay

65535

ms [1..65535]

Enable Max. Jitter

Enabled

Max. Jitter

3000

ms [1..3000]

Click on the *Apply* icon to save the changes.

## Test Results

Table 13: Test Results

Preliminary Phase				
Use Case	Test ID	Test Case	Test Result	Comments

Basic Call		BC01	BC01_[Phone_1]_[OFFNET]	OK	
		BC02	BC02_[OFFNET]_[Phone_1]	OK	
Long Duration Call + CLIR		LCLIR01	LCLIR01_[Phone_1]_[OFFNET]	OK	
		LCLIR02	LCLIR02_[OFFNET]_[Phone_1]	OK	
Call Cancellation		CANC01	CANC01_[Phone_1]_[OFFNET]	OK	
		CANC02	CANC02_[OFFNET]_[Phone_1]	OK	
DTMF + Voicemail		DTMF03	DTMF03_[Phone_1]_[OFFNET-IVR]	OK	
		DTMF04	DTMF04_[OFFNET-voicemail]_[Phone_1]	OK	
Transfer	Supervised + MOH	TRMOH01	TRMOH01_[OFFNET]_[Phone_1]_[OFFNET]	OK	
	Blind	TRAB01	TRAB01_[OFFNET]_[Phone_1]_[OFFNET]	OK	
		TRAB04	TRAB04_[OFFNET]_[Phone_1]_[OFFNET-IVR]	OK	
Forward	Unconditional	FWDU01	FWDU01_[OFFNET]_[Phone_1]_[OFFNET]	OK	
		FWDU04	FWDU04_[OFFNET]_[Phone_1]_[OFFNET-IVR]	OK	
	No Answer	FWDNA01	FWDNA01_[OFFNET]_[Phone_1]_[OFFNET]	OK	
SVAIP		SVAIP02	SVAIP02_[IPBX-ForcedONNET]_[SAN-SVAIP+33296084273]	N/A	Out of scope as it is currently not possible to perform those tests
Advanced Phase					
Use Case		Test ID	Test Case	Test Result	Comments
Busy Call		BUSY01	BUSY01_[Phone_1]_[OFFNET]	OK	
		BUSY02	BUSY02_[OFFNET]_[Phone_1]	OK	
Not Answered Call		NA01	NA01_[Phone_1]_[OFFNET]	OK	
		NA02	NA02_[OFFNET]_[Phone_1]	OK	
Transfer	Supervised	TRAS02	TRAS02_[OFFNET]_[Phone_1]_[Phone_2]	OK	
		TRAS03	TRAS03_[Phone_1]_[Phone_2]_[OFFNET]	OK	
	Blind	TRAB02	TRAB02_[OFFNET]_[Phone_1]_[Phone_2]	OK	
		TRAB03	TRAB03_[Phone_1]_[Phone_2]_[OFFNET]	OK	
Forward	Unconditional	FWDU02	FWDU02_[OFFNET]_[Phone_1]_[Phone_2]	OK	
	Busy	FWDB02	FWDB02_[OFFNET]_[Phone_1]_[Phone_2]	OK	
	No Answer	FWDNA02	FWDNA02_[OFFNET]_[Phone_1]_[Phone_2]	OK	
		FWDNA03	FWDNA03_[Phone_1]_[Phone_2]_[OFFNET]	OK	
		FWDNA04	FWDNA04_[OFFNET]_[Phone_1]_[OFFNET-IVR]	OK	
Conference X3		CONF01	CONF01_[OFFNET]_[Phone_1]_[OFFNET]	OK	
Prehook	(with) Transfer Sup.	PREHOK01	PREHOOK01_[OFFNET]_[Phone_1]_[OFFNET]	N/A	
		PREHOK02	PREHOOK02_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
		PREHOK03	PREHOOK03_[Phone_1]_[Phone_2]_[OFFNET]	N/A	
	(with) Forward	PREHOK04	PREHOOK04_[OFFNET]_[Phone_1]_[OFFNET]	N/A	

<b>Call Features</b>	<b>Call Parking</b>	CPA01	CPA01_[Phone_1]_[OFFNET]_[Phone_2]	OK	
	<b>Call Pickup</b>	PKU01	PKU01_[OFFNET]_[Phone_1]_[Phone_2]	OK	
	<b>Hunt Group</b>	HUG01	HUG01_[OFFNET]_[Phone_1]	OK	
	<b>Second Line</b>	SL01	SL01_[OFFNET]_[Phone_1]	OK	
<b>DTMF</b>		DTMF03	DTMF03_[Phone_1]_[OFFNET-IVR]	OK	
<b>E2E Overflow</b>		OVF01	OVF01_[NBI-Int+670012144326845]_[cSBCRibbon+33296031233]_[]	OK	
		OVF02	OVF02_[Offnet-Devil+ +960012144326845]_[cSBCRibbon+33296086974]_[]	OK	
		OVF03	OVF03_[NBI-Fr+33399106845]_[cSBCRibbon+33296031233]_[]	OK	
		OVF04	OVF04_[OFFNET]_[select device]_[]	N/A	
		OVF05	OVF05_[OFFNET]_[select device]_[]	N/A	
		OVF06	OVF06_[OFFNET]_[select device]_[]	N/A	
		OVF07	OVF07_[NBI-Int+670012144326845]_[cSBCRibbon+33296039150]_[]	OK	
		OVF08	OVF08_[cSBCRibbon+33296031233]_[Offnet-NBI-Fr+33399106845]_[]	OK	
		OVF09	OVF09_[cSBCRibbon+33296031233]_[Offnet-NBI-Fr+33399106845]_[]	OK	
<b>CAC</b>		CAC01	CAC01_[OFFNET]_[Phone_1]	N/A	
		CAC02	CAC02_[Phone_1]_[OFFNET]	N/A	
		CAC03	CAC03_[Phone_1]_[Phone_2]	N/A	
		CAC04	CAC04_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
		CAC05	CAC05_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
<b>Emergency Number</b>		EMN01	EMN01_[Phone_1]_[OFFNET-EMN]	OK	
		EMN02	EMN02_[Phone_1]_[OFFNET-EMN]	N/A	
		EMN03	EMN03_[Phone_1]_[OFFNET-EMN]	OK	
<b>Attendant Console</b>		AC01	AC01_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
		AC02	AC02_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
		AC03	AC03_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
		AC04	AC04_[OFFNET]_[Phone_1]_[Phone_2]	N/A	
<b>Fax Tests</b>					
<b>Use Case</b>		<b>Test ID</b>	<b>Test Case</b>	<b>Test Result</b>	<b>Comments</b>
<b>Fax</b>	<b>Offnet -&gt; HQ</b>	Fax_01	Devil+_IPTEL_G3	OK	
		Fax_02	Devil+_IPTEL_SG3	N/A	
		Fax_03	Neo_IPTEL_G3	OK	
		Fax_04	Neo_IPTEL_SG3	N/A	
		Fax_05	NBI-France_IPTEL_G3	OK	
		Fax_06	NBI-France_IPTEL_SG3	N/A	
		Fax_07	NBI-International_IPTEL_G3	OK	
		Fax_08	NBI-International_IPTEL_SG3	N/A	
	<b>HQ -&gt; Offnet</b>	Fax_09	IPTEL_Devil+_G3	OK	
		Fax_10	IPTEL_Devil+_SG3	N/A	
		Fax_11	IPTEL_Neo_G3	OK	



	Fax_12	IPTEL_Neo_SG3	N/A	
	Fax_13	IPTEL_NBI-France_G3	OK	
	Fax_14	IPTEL_NBI-France_SG3	N/A	
	Fax_15	IPTEL_NBI-International_G3	OK	
	Fax_16	IPTEL_NBI-International_SG3	N/A	

## Conclusion

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These Application Notes describe the configuration steps required for Ribbon to successfully interoperate with OBS. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in [Test Results](#).

## Appendix A

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- [Cisco CUCM - Special Characters and Settings](#)
- [Ribbon SBC Edge - Understanding Regular Expressions](#)
- [Ribbon SBC Edge - SIP Message Manipulation](#)