# **Ribbon SBC Edge R9.0 Interop with IP-PBX for Deutsche** Telekom CompanyFlex SIP Trunk : Interoperability Guide

# ribbon

# **Table of Contents**

- Interoperable Vendors
- Copyright
- Document Overview
  - About Ribbon SBC Edge
    - About Deutsche Telekom
- Scope
- Non-Goals
- Audience
- Prerequisites
- Product and Device Details
- Network Topology Deployment Topology

  - IOT Lab Topology
- Section A: Ribbon SBC Edge Configuration
  - Installing Ribbon SBC Edge
  - Accessing Ribbon SBC Edge
  - License and TLS Certificates
    - View License Import Trusted Root CA Certificates
  - Configure Static Routes
  - Ribbon SBC SWe Lite Configuration towards
    - Deutsche Telekom End
      - Remote Authorization Table
      - Contact Registration Table
      - Create TLS Profile
      - SIP Server Table
      - Create SRTP Profile
      - Media Profile
      - SIP Profile
  - Signaling Group **Transformation Table**
  - Call Routing Table
  - SWe Lite Configuration Towards IP-PBX CUCM
    - SIP Server Table
      - Signaling Group Table
  - Message Manipulation
    - Updating Signaling Group with Message
    - Manipulation
- Section B: CUCM (IP-PBX) Configuration
  - Accessing CUCM (Cisco Unified CM Administration)
  - Configure SIP Trunk Security Profile
  - Configure SIP Profiles
    - Configure Normalization Script
    - Trunk Configuration
    - Configure Call Routing
  - Configure End Users
  - Phone Setup
  - Device Association
- Supplementary Services and Features Coverage
- Caveats
- Support
- References
- Conclusion

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# Interoperable Vendors

#### **Deutsche Telekom**

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# **Document Overview**

This document depicts the configuration details for Ribbon SBC Edge interworking & compliance against Deutsche Telekom CompanyFlex SIP Trunking solution.

# About Ribbon SBC Edge

The Ribbon Session Border Controller provides best-in class communications security. The Ribbon SBC Edge dramatically simplifies the deployment of robust communications security services for SIP Trunking.

# About Deutsche Telekom

Deutsche Telekom is a telecommunications company that offers a range of fixed-network services, such as voice and data communication services based on fixed-network and broadband technology, and sells terminal equipment, other hardware, and services to resellers.

# Scope

This document provides configuration best practices for deploying Ribbon's SBC 1000/2000 and SWe Lite series when connecting with Deutsche Telekom CompanyFlex. Note that these are configuration best practices, and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design which best meets their requirements.

# Non-Goals

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

# Audience

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

# Prerequisites

The following aspects are required before proceeding with the interop:

- Ribbon SBC Edge
- SBC License
- IP-PBX SIP Connect 2.0 Compliant
- Deutsche Telekom "CompanyFlex" SIP trunks
  - Contact Deutsche Telekom for Domain, Outbound proxy, Registrar, SIP trunk Registration number, SIP trunk password and block of numbers for the end points.
  - For more information, visit https://hilfe.companyflex.de/de/einrichtung/einrichtung-sip-trunk

Any IP-PBX which is SIP Connect 2.0 Compliant can be deployed with Ribbon SBC Edge. For this interop testing we have used CUCM 12.5 which is SIP Connect 2.0 Compliant.

Note

During this interop, the SIP Trunk between Deutsche Telekom and Ribbon SBC Edge has been configured with TLS and SRTP.

# **Product and Device Details**

The configuration uses the following equipment and software:

#### Table 1: Requirements

Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC SWe Lite	9.0.3
Third-party Equipment	Cisco Unified Communication Manager	12.5.1.11900-146
Deutsche Telekom	Deutsche Telekom "CompanyFlex"	NA
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

# Network Topology

# **Deployment Topology**



# **IOT Lab Topology**



# Installing Ribbon SBC Edge

Refer to the following document for installing the Ribbon SBC Edge: Installing SBC 1000/2000.

# Accessing Ribbon SBC Edge

Open any browser and enter the SBC IP address.

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

Welcome to Ribbon SBC SWe Lite					
	Users (authorized or unauthorized) have no explicit or implicit expectation of privacy. Any or all uses of this system and all files on this system may be intercepted, monitored, recorded, copied, audited, inspected, and disclosed to authorized site, customer administrative, and law enforcement personnel, as well as authorized officials of government agencies, both domestic and foreign. By using this system, the user consents to such interception, monitoring, recording, copying, auditing, inspection, and disclosure at the discretion of authorized personnel. Unauthorized or improper use of this system may result in administrative disciplinary action and civil and criminal penalties. By continuing to use this system you indicate your awareness of and consent to these terms and conditions of use. CANCEL YOUR LOGIN IMMEDIATELY if you do not agree to the conditions stated in this warning.				
	User Name admin Password Login Cancel Copyright © 2010-2021 <u>Ribbon Communications Operating Company, Inc.</u> All Rights Reserved				

# License and TLS Certificates

# **View License**

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

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

Ô							Welcome
noddin		🔘 Monitor	Tasks	Settings	Diagnostics	System	
Application Solution Module	Total 19 Feature License Ro	ws					
Node-Level Settings	Feature		Licensed	Total Lice	nses	Available Licenses	
CoE	SIP Calls			100		100	
System Timing	SIP Registrations			200		199	
System Companding Law	DSP Resources			Unlimited		Unlimited	
V Zicensing	Forking		R/	Unlimited		Unlimited	
License Keys	SBA			Unlimited		Unlimited	
Install New License			~				
🕨 🥖 Software Management	Active Directory		W	Unlimited		Unlimited	
Auth and Directory Services	Transcoding			Unlimited		Unlimited	
Active Directory	REST			Unlimited		Unlimited	
P Praturala	CAS		₹	Unlimited		Unlimited	
DNS	CDR		U	Unlimited		Unlimited	
▼ 龙 IP	OSPF		R/	Unlimited		Unlimited	
E Static Routes			~				
E Routing Table	RIP			Unlimited		Unlimited	
E Static ARP	IPsec		₽⁄	Unlimited		Unlimited	
decess Control Lists	RBA			Unlimited		Unlimited	

For more details on Licenses, refer to Ribbon SBC Edge Licenses.

# Import Trusted Root CA Certificates

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

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

	SBC Certificates Index     Generate SBC Edge CSR     SBC Primary Certificate     SBC Supplementary Certificates     Trusted CA Certificates
SBC Primary Certificate     SBC Supplementary Certificates     Trusted CA Certificates	

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

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

Mode Copy and Paste V Mode File Upload V Select File Choose File No file chosen Extensions (pern, der, cer, ber, p76) *	port Trusted CA Certificate	Import Trusted CA Certificate
Paste Base64 Certificate	Mode Copy and Paste	Mode File Upload Select File Choose File No file chosen Extensions (pern, der, cer, ber, p7b) *

Follow the above steps to import the Service Provider's (Deutsche Telekom) Root and Intermediate certificates of their Public CA.

Note
 Deutsche Telekom Root certificate: https://corporate-pki.telekom.de/en/GlobalRootClass2.html

For more details on Certificates, refer to Working with Certificates.

£	Note
~	

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

#### Note

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

# **View Networking Interfaces**

The Ribbon SBC Edge 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 Edge supports user-created VLAN logical sub-interfaces.

Admin IP, Ethernet 2 IP, Ethernet 1 IP are used for this interop.

From the Settings tab, navigate to Networking Interfaces > Logical 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.



#### Ethernet 1 IP

Ethernet 1 IP is assigned an IP address used for transporting all the VOIP media packets (for example, RTP, SRTP) and all protocol packets (for example, SIP, RTCP, TLS). In the default software, Ethernet 1 IP is enabled, and an IPv4 address is acquired via a connected DHCP server. You can assign a static IP as well. This interface will face the Deutsche Telekom.

Q. Search Expand All   Collapse All   Reload	v C Ethernet 1 IP
Call Routing	Identification/Status
Signaling Groups  Networking Interfaces  Admin IP  Ethernet 1 IP  Ethernet 2 IP  System	Interface Name Ethernet 1 IP I/F Index 8 Alias Description Admin State Enabled
<ul> <li>Auth and Directory Services</li> <li>Protocols</li> <li>SIP</li> </ul>	Networking
Local Registrars     Local / Pass-thru Auth Tables     SIP Profiles     Default SIP Profile     TELEKOM SIP PROFILE	MAC Address 0
♥ SIP Server Tables In Default SIP Server In telekom sip server table In cucm	IP Assign Method Static
<ul> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Emote Authorization Tables</li> </ul>	Primary Address 1 *xxxx Primary Netmask 255.255.0 *xxxx
Gontact Registrant Table     Message Manipulation     Made Lowel CIR Settings	

#### Ethernet 2 IP

Configure this Ethernet 2 interface as follows according to the requirement. This interface will face the IP-PBX (CUCM).

Q Search_	Logical Interfaces
Expand All   Collapse All   Reload	V   🖉   Create VLAN I/F   🗶 Total 3 LogicalInterface Rows
🕨 💋 Call Routing	Interface IPv4 Address
Sionalino Groups	Admin IP
Vetworking Interfaces	Fthernet 1 ID
<ul> <li>Logical Interfaces</li> <li>Logical Interfaces</li> </ul>	
Ethernet 1 IP	v 📄 🗆 Ethernet 2 IP 1
Ethernet 2 IP	
	Identification/Status
🕨 📁 System	
Multi Auth and Directory Services	Interface Name Ethernet 2 IP
Protocols	I/F Index 9
<ul> <li>SIP</li> <li>I coal Registrars</li> </ul>	Alias
Local / Pass-thru Auth Tables	Description
V SIP Profiles	Admin State Eachlod
Default SIP Profile	
TELEKOM SIP PROFILE	
💌 🧀 SIP Server Tables	Networking
♥ SIP Server Tables ☐ Default SIP Server	Networking
♥ SIP Server Tables           Image: Default SIP Server           Image: Default SIP Server table	Networking
♥ SIP Server Tables Image: Default SIP Server Image: Table Image: Table Image: Table Image: Table Image: Table	MAC Address 0
♥ <sup>™</sup> SIP Server Tables	Networking MAC Address 0 IP Addressing Mode IPv4
SIP Server Tables     Im Default SIP Server     Im telekom sip server table     Im cucm     Irunk Groups     NAT Qualified Prefix Tables	Networking MAC Address 0
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Memote Authorization Tables</li> </ul>	MAC Address 0
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 IPv4 Information
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 IP Addressing Mode IPv4 IP Assign Method Stratic
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 IPv4 Information IP Assign Method Static
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> </ul>	Networking          MAC Address       0         IP Addressing Mode       IPv4         IPv4 Information         IP Assign Method       Static         Primary Address       1         *xxxxx
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> <li>Security</li> </ul>	Networking          MAC Address       0         IP Addressing Mode       IPv4         IPv4 Information         IP Assign Method       Static         Primary Address       1         Primary Netmask       255.255.255.0
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> <li>Security</li> <li>Media</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 IP Addressing Mode IPv4 IP Assign Method Static Primary Address 1 Primary Netmask 255.255.255.0 * X.X.X.X Media Next Hop IP Media Next Hop IP Media Next Hop IP * X.X.X
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> <li>Security</li> <li>Media</li> <li>Media System Configuration</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 V IPv4 Information IP Assign Method Static V Primary Address 1 * XXXX Primary Netmask 255.255.255.0 * XXXX Media Next Hop IP 1 * XXXX
<ul> <li>SIP Server Tables</li> <li>Default SIP Server</li> <li>telekom sip server table</li> <li>cucm</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> <li>Security</li> <li>Media</li> <li>Media System Configuration</li> <li>Media Profiles</li> </ul>	Networking MAC Address 0 IP Addressing Mode IPv4 V IPv4 Information IP Assign Method Static V Primary Address 1 Primary Netmask 255.255.255.0 * XXXX Media Next Hop IP 1 * XXXX

# Attention

If you are migrating from SIP Trunk DeutschlandLAN towards CompanyFlex, ensure that you configure either a second (different) interface IP address on SBC1000 / SBC2000, or in case of SBC SWe Lite, a second interface with a different IP address.

Do not use the same IP for DeutschlandLAN and CompanyFlex on the SBC.

# **Configure Static Routes**

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

#### **Destination IP**

Specifies the destination IP address.

#### Mask

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

#### Gateway

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

#### Metric

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

Q Search	Static IP Route Table							
Expand All   Collapse All   Reload	+ I X	Total 27 IP Route Rows						
🕨 🥖 Call Routing	Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key		
Signaling Groups	1	0.0.0.0	0.0.0.0	10.0.	1	1		
System	2	157.49.	255.255.255.255	10.0.	1	2		
Auth and Directory Services	3	157.49.	255.255.255.255	10.0.	1	3		
Protocols	4	115.110.	255.255.255.255	10.0.	1	4		
	5	115.110.	255.255.255.255	10.0.	1	5		
Static Routes	6	157.49.	255.255.255.255	10.0.	1	6		
Table	7	157.49.	255.255.255.255	10.0.	1	7		

# **Ribbon SBC SWe Lite Configuration towards Deutsche Telekom End**

This section describes the steps to configure SBC SWe Lite with TLS/SRTP towards Deutsche Telekom SIP Trunk.

### **Remote Authorization Table**

#### Select Settings > SIP > Remote Authorization Tables.

Remote Authorization Tables entries contain information for responses to request message challenges by an upstream server.

- Create a new entry "SipTrunk2" under "Remote Authorization Table" .
- Add domain name provided by Deutsche Telekom under "Realm".
- Add SIP Trunk number under Authentication ID.
- Add password provided by Deutsche Telekom under "Password" and confirm it.
- Choose regex under "From URI User Match" and add ".\* " for "Match regex".

System     Auth and Directory Services	TELEKOM-REMOTE-AUTH-TABLE	
Protocols	🕂   🗶   🥂 Total 1 SIP Remote Authorization	Row
V SIP V placal Registrars	Realm	Authentication ID
📁 Local / Pass-thru Auth Tables	🔻 🔲 🗌 tel.t-online.de	+49
🕨 💋 SIP Profiles		
🕨 🥩 SIP Server Tables		
💋 Trunk Groups	Realm tel.t-online.de	
💋 NAT Qualified Prefix Tables	Authorization ID	
💌 🤣 Remote Authorization Tables		
TELEKOM-REMOTE-AUTH-TABLE	Password Setting Use Current 🗸	
🕨 🍺 Contact Registrant Table	From URI User Match Regex 🗸	
🕨 🍺 Message Manipulation	Match Regex .*	
Node-Level SIP Settings		
g SIP Recording		
Security		_
🔻 🥻 Media	Apply	
Media System Configuration		

### **Contact Registration Table**

#### Select Settings > SIP > Contact Registration Table.

The Contact Registrant Tables manage contacts that are registered to a SIP server. The SIP Server Configuration can specify a Contact Registrant Table. The username portion of the table is used for outbound calls.

- Create a new entry "Telekom contact reg" under Contact Registrant table.
- · Choose local as "Type of address of record".
- Provide the SIP Trunk number provided by Deutsche Telekom under the "Address of record URI".
- Provide 600 seconds for Global Timer to Live and 120 seconds for Failed Registration Retry Timer.
- · Create an entry under "SIP Contacts".
- Provide the SIP Trunk number provided by Deutsche Telekom under "Contact URI Username" and set TTL value as "Inherited".

Q Search	CONTACT REG TABLE
Expand All   Collapse All   Reload	-   X Total 1 SIP Contact Registrant Entry Row
Call Routing	Address of Record
Signaling Groups	▼ □ +49
System	
Auth and Directory Services	Type of Address of Record
Protocols	
SIP	
Local / Pass-thru Auth Tables	
🕨 📁 SIP Profiles	Failed Registration Retry Timer 120 * secs (30.86400)
SIP Server Tables	
Trunk Groups	SIP Contacts
Remote Authorization Tables	Total 1 STP User Contact Row
💌 💋 Contact Registrant Table	
CONTACT REG TABLE	Contact URI Username TTL (secs) Priority (Q)
Message Manipulation	/ +49 Inherited 0
Node-Level SIP Settings SIP Recording	
Security	
🔻 🧀 Media	Apply
Media System Configuration	

Click on Registration status under the "Contact Registration profile" to see the status of SIP Trunk registration with Deutsche Telekom.

ribbon			i Monitor	Tasks	Settings	Diagnostics	System	
Q Search Expand All   Collapse All   Reload	C	ONTACT REG TABLE	ant Entry Row	_	_	_	_	_
<ul> <li>Call Routing</li> <li>Signaling Groups</li> <li>Networking Interfaces</li> </ul>		Address of Record				Display Registration	Status	
System     Auth and Directory Services	llr	0	Contact Registrant	Registration	n Status - Googl	e Chrome	_	
Protocols           SIP         Image: Simple state         Image: Simple state		A Not secure	gi/phpUI/callTable	Engine.php?p	parentID=1&filte	r=1&parentType=	SIPRegistration&	tty Q
<ul> <li> <sup>1</sup> Local Registrars         <sup>1</sup> Local / Pass-thru Auth Tables         <sup>1</sup> SIP Profiles     </li> </ul>		Contact Registrant Registration S Total 1 SIPRegistrationStatus Row	tatus	_	_	_	July 02, 2021 1	4:10:19 🤤
SIP Server Tables		SIP Server		Sig	naling Group	Registratio	n Status	
Irunk Groups NAT Qualified Prefix Tables	ш	Entry 102 (f-ecs-650.edns.t-ipnet.d		(51	P) telekom	Registered		
Contact Registrant Table     CONTACT REG TABLE								
Message Manipulation								

# **Create TLS Profile**

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

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

- Provide desired description.
  Set TLS protocol as "TLS 1.2 Only".
  Disable "Mutual Authentication".
  Disable "Validate Server FQDN".

- Click "Apply".



### **SIP Server Table**

#### Select Settings > SIP > SIP Server Tables

SIP Server Tables contain 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.

When you configure a SIP server table entry with a DNS SRV record, Ribbon recommends that you do not configure another SIP server table entry with the IPs or FQDNs that the DNS SRV record resolves.

- Create a SIP Server Table with a DNS SRV record.
- Add domain name provided by the Deutsche Telekom.
- Update the Service Name as "sips".
- Use TLS protocol.
- For Remote Authorization Table choose "sipTrunk2" that was created earlier.
- For contact Registration table choose "Telekom contact reg" .
- The FQDN provided from Deutsche Telekom will be resolved under SRV servers.
- Attach the TLS profile created in the previous step.
- Verify the FQDN provided from Deutsche Telekom is resolved under SRV servers with protocol as TLS.



# **Create SRTP Profile**

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

Select Settings > Media > SDES-SRTP Profile. Click the + icon to create a new SRTP profile.

- Provide desired description.
- Set "Operation Option" as Required. This setting permits call connections only if you can use encryption for the call. If the peer device does not support SRTP (Secure Real Time Protocol) for voice encryption over the IP network, the call setup will fail.
- Attach the Crypto suite "AES\_CM\_128\_HMAC\_SHAI\_80" A crypto suite algorithm which uses the 128 bit AES-CM encryption key and a 80 bit HMAC\_SHA1 message authentication tag length.
- Key Identifier Length set to "0" Set this value to 0 to disable the MKI in SDP.
- Click OK.

Q Search	SDES-SRTP Profiles	
Expand All   Collapse All   Reload Call Routing Signaling Groups	Description Crypto Suite	HMAC SHA1 80
<ul> <li>Metworking Interfaces</li> <li>System</li> <li>Auth and Directory Services</li> </ul>	SRTP Config	
<ul> <li>Protocols</li> <li>SIP</li> <li>Security</li> <li>Media</li> </ul>	Description tls           Operation Option         Required           Crypto Suite         AFS CM 128 HMAC SHA1 80	
Media System Configuration  Media Profiles  SDES-SRTP Profiles  Interpretation  Media List  Media List	Key Identifier Length 0 V	

# **Media Profile**

#### Select Settings > Media > Media List.

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

- Create new Media list profile.
- G711 media profiles will be there by default under Media profile list, Additional codecs can be added as per the need.
- Attach the SDES-SRTP profile (Specifies the profile for authentication/encryption protocols applied with this Media List) created in the
- previous step.Click Apply.

Q Search	Description telekom
Expand All   Collapse All   Reload  Call Routing  Signaling Groups  Networking Interfaces  System	Media Profiles List
Auth and Directory Services     Protocols     SIP     Security     Media     Media System Configuration	SDES-SRTP Profile       tls         Media DSCP       46         Dead Call Detection       Disabled         Silence Suppression       Enabled
<ul> <li>Media Profiles</li> <li>SDES-SRTP Profiles</li> <li>Media List</li> <li>Default Media List</li> <li>telekom</li> </ul>	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
<ul> <li>Telephony Mapping Tables</li> <li>SNMP/Alarms</li> </ul>	Passthrough/Tone Detection
<ul> <li>Logging Configuration</li> <li>Emergency Services</li> <li>Configuration</li> <li>Callback Number Pool</li> </ul>	Modem Passthrough Enabled Fax Passthrough Enabled Fax Tone Detection Disabled

#### Select Settings > Media > Media Profiles.

Create a Media profile with G729 codec if needed.

Media Profiles	
Collapse All   Reload Create Media Profile 🗸   🗙 Total 5 Media Profile Rows	
ng Codec C Groups g Interfaces G.711 A-Law C b C G.711 A-Law C	Description Default G711A Default G711u
Directory Services Colec Configuration Usystem Configuration Usystem Configuration Usystem Configuration Usescription Usescription Usescription Usescription Usescription Usescription Usescription Usescription Usescription Description Usescription Description	1729
-SRTP Profiles	

#### (i) Note

As per Deutsche Telekom, T.38 media encryption is not supported. Negotiations within an established connection for T.38 to a UE using encryption are rejected with SIP Error code 488, so that fax transmission will use G.711 with encryption instead.

### **SIP Profile**

#### Select Settings > SIP > SIP Profiles.

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

Create a new SIP profile with the name "Telekom sip profile" with the session timer enabled. The Minimum Acceptable Timer is 600, and the Offered Session Timer is 1800.



# **Signaling Group**

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

#### Select Settings > Signaling Groups

- Create an entry in signaling group named "From/To Telekom".
- Choose "Telekom sip profile " under SIP Profile.
- Choose Call Routing as "From Telekom".

### Note

Initially choose Default call Route. Create the Route, as shown in the call Routing section, and then update the call Route to "From Telekom".

Choose Agent type as "Back-to-Back user agent" and media list as "telekom media list".

- Choose SIP Server Table as "Telekom SIP Server Table".
- Attach the SRTP profile created in the previous steps under "proxy local SRTP crypto profile ID".

ribbon Q. Search Exgand All   Collapse All   Reload	Description telekom Admin State Enabled Service Status Up	
Call Routing     Call Transformation     Description	SIP Channels and Routing	Media Information
Important Control	Action Set 1301e Mone Call Routing Table from telekom No. of Channels 60 SIP Profile TELEKOM SIP PROFILE SIP Mode Basic Call	Supported Audio Proxy * Modes Direct Proxy with Local SRTP V
Default Route Table     from local registrar     from telekom	Agent Type Back-to-Back User Agent SIP Server Table telekom sip server table	Supported Proxy A Video/Application Direct *
FROM CUCM	Channel Hunting Most Idle Notify Lync CAC Profile Disable	Media List ID telekom Proxy Local SRTP Concer profile ID tis
Call Actions     Sionalina Groups     (SIP) telekom     (SIP) telekom	Challenge Request Disable Outbound Proxy IP/FQDN Outbound Proxy Port 5060	Play Ringback Auto on 180 Tone Table Default Tone Table Play Congestion Disable
(SIP) CUCM	Call Proceeding Timer 180 Ulse Reviser as Keen Aliye Enable	Early 183 Disable
System     Auth and Directory Services	Forked Call Answered Too Soon Disable	SDP Enable Music on Hold Disabled
Protocols	SIP Recording SIP Recording Status Disabled	RTCP Disable Multiplexing
<ul> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> </ul>		Mapping Tables
Default SIP Profile     TELEKOM SIP PROFILE     SIP Segue Tables		SIP To Q.850 Override Default (RFC4497) Table
Default SIP Server		Q.850 To SIP Override Default (RFC4497)

Note

If NAT is used, then add the external public IP of the NAT box under static NAT outbound of the Signaling Group that is facing towards the Deutsche Telekom server.

Configure NAT so that the external public IP address does not change frequently. If it does, update the new IP address under "Static NAT Outbound".

- Update the Federated IP/FQDN, i.e. the IPs of the Deutsche Telekom servers and gateway, as provided by Deutsche Telekom.
- Add a listening port for TLS (5061).
- Attach the TLS profile created earlier.

C. Search     Expand All   Collapse All   Reload				Cause Codes	413 - Request Entity Too Large 500 - Server Internal Error	•
Call Routing					SIP IP Details	
(SIP) telekom (SIP) rigistrar				Teams	s Local Media Disable Optimization	
(SIP) CUCM				Signaling/Med	dia Source IP (1	
Vetworking Interfaces				Sig	gnaling DSCP 40	
Auth and Directory Services					NAT Traversal	- 1
Protocols					ICE Support Disabled	
🔻 💋 SIP					-Static NAT - Outbound	
Local Registrars				Outbound N	NAT Traversal None	
🥖 Local / Pass-thru Auth Tables						
🔻 🌽 SIP Profiles					Detection Disabled	
Default SIP Profile						
TELEKOM SIP PROFILE						
V SIP Server Tables			Listen Ports		Federated IP/EODN	
Default SIP Server			Listen Forts		redenated in right	
cucm	Total 3 SIP Li	sten Port Rows		Total 1 SIP Federate	ed IP Row	_
💋 Trunk Groups	Port	Protocol	TLS Profile ID	 IP/FQDN	Netmask/Prefix	
NAT Qualified Prefix Tables	5060	UDP	N/A	2	25	
Remote Authorization Tables				-		
	5060	TCP	N/A			
Gontact Registrant Table	5061	TLS	telekom tls profile			
Model evel SIP Settings						

# **Transformation Table**

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

From the Settings > Call Routing > Transformation.

# To Create a Transformation Table

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

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

- 1. Click the **Create** (+) icon.
- 2. Enter a descriptive name in the **Description** text field.
- 3. Click OK.

Q Search	Transformation
Expand   Collapse   Reload	Total 3 Transformation Table Rows
Call Routing	Description
Transformation	Passthrough Untouched
Towards cucm	🔻 📋 🖸 Towards cucm
i Towards Deutsche Telekc	
💋 Time of Day Table	Description Towards guern
Call Routing Table Call Routing Table	
i from local registrar	
i from telekom	Apply
E FROM CUCM	
in from telekom to cucm	

Similarly create transformation table towards Deutsche Telekom.

Q Search	Transformation
Expand   Collapse   Reload	Total 3 Transformation Table Rows   Description   Passthrough Untouched   Towards cucm     Towards Deutsche Telekom
<ul> <li>Time of Day Table</li> <li>Call Routing Table</li> <li>Default Route Table</li> <li>from local registrar</li> <li>from telekom</li> <li>FROM CUCM</li> <li>from telekom to cucm</li> <li>Call Actions</li> </ul>	Description Towards Deutsche Telekom

In the lab environment we added +4 to the called number while sending out to Deutsche Telekom. Towards CUCM, we removed + . The followings transformation examples are based on the lab setup. It will differ based on the requirements.

Note

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

**Towards Deutsche Telekom** 

Q Search	Towards Deutsche Telekom		
Expand   Collapse   Reload	VI 🖉 I 🕂 I 🗙 I 🥂 Total 1 Transformation E	Intry Row	
All 'All '	Admin State Input Field Type	Input Field Value	Output Field Type
Transformation	Called Address/Number	(.*)	Called Address/Num
Towards cucm	[		
Towards Deutsche Tele	Description add +4		
Time of Day Table Total Day Table	Admin State Enabled 🗸		
Call Routing Table	Match Type Optional (Match One) 🗸		
i from local registrar			
FROM CUCM	Input Field	Output Field	
i from telekom to cucm			
Call Actions	Type Called Address/Number 🗸	Type Called Address/Number	er 🗸
<ul> <li>Signaling Groups</li> <li>(SIP) telekom</li> </ul>	Value (.*)	Value +4\1	
(SIP) rigistrar			
(SIP) CUCM			
Metworking Interfaces			Apply

#### **Towards CUCM**

Q Search	Towards cucm				
Expand Collapse Reload	<b>√101</b> ∔1X	Total 2 Transformation	Entry Rows		
	Admin State	Input Field Type	_	Input Field Value	Output Field Type
<ul> <li>Call Routing</li> <li>Transformation</li> </ul>	▼ □ □ ₩	Called Address/Number		\+(.*)	Called Address/Num
Passthrough Untouched					
Towards cucm				_	
Towards Deutsche Telekc	Description	remove +			
📁 Time of Day Table	Admin State	Enabled 🗸			
Call Routing Table Default Route Table	Match Type	Optional (Match One)			
i from local registrar					
i from telekom					
FROM CUCM		Input Field		Output Field	
Trom telekom to cucm					
Call Actions	Type C	alled Address/Number 🗸	Type C	alled Address/Number	<u> </u>
Signaling Groups	Value \-	+(.*)	Value \1		
(SIP) rejector					
(SIP) CUCM			_		
Metworking Interfaces					
🕨 💋 System					Apply
Auth and Directory Services					

Q Search	Towards cucm	ntry Rows
Call Routing	Admin State Input Field Type	Input Field Value Output Field Type
V Transformation	🕨 📄 🗆 🍢 🛛 Called Address/Number	\+(.*) Called Address/Numbe
Towards Cucm	🔻 📋 🛛 🍢 Calling Address/Number	\+(.*) Calling Address/Nur
<ul> <li>Time of Day Table</li> <li>Call Routing Table</li> <li>Default Route Table</li> <li>from local registrar</li> <li>from telekom</li> </ul>	Description abc Admin State Enabled Match Type Optional (Match One)	
FROM CUCM	Input Field	Output Field
<ul> <li>Signaling Groups</li> <li>(SIP) telekom</li> <li>(SIP) rigistrar</li> <li>(SIP) CUCM</li> </ul>	Type Calling Address/Number  Value \+(.*)	Type Calling Address/Number  Value \1
Metworking Interfaces     System     Auth and Directory Services     Protocols		Арріу

# **Call Routing Table**

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

#### Select Settings > Call Routing > Call Routing Table.

#### Creating an Entry to a Call Routing Table

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

In the SBC Edge, call routing occurs between Signaling Groups.

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

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

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

To create an entry:

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

#### Admin State:

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

#### Route Priority:

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

#### Number/Name Transformation Table:

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

#### **Destination Signaling Groups:**

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

#### Audio Stream Mode:

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

#### Media Transcoding:

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

3. Click Apply.

Call Routing for Deutsche Telekom signaling group: Any signaling coming from Deutsche Telekom will be routed to CUCM

<b>0</b> 00		
noddin	Route Details	
O Search	Description to registrar	
g Beardic.	Admin State Enabled	
Expand All   Collapse All   Reload	Route Priority 1	
P 🤣 Call Routing	Call Priority Normal	
Transformation	Number/Name Transformation Table Towards cucm	
💣 Time of Day Table	Time of Day Restriction None	
🗢 🧀 Call Routing Table		
efault Route Table	Destination Information	
i from local registrar		
from telekom	Destination Type Normal	
FROM CUCM	Message Translation Table None	
from telekom to cucm	Cause Code Reroutes None	
🕨 🧀 Call Actions	Cancel Others upon Forwarding Disabled	
🕨 📁 Signaling Groups	Fork Call No	
🕨 📁 Networking Interfaces	(SIP) CUCM	
🕨 📁 System		
Auth and Directory Services	Destination Signaling Groups *	
Protocols		
r 🌽 SIP	· · · · · · · · · · · · · · · · · · ·	
Local Registrars	Enable Maximum Call Duration Disabled	
Local / Pass-thru Auth Tables		
V SIP Profiles		
TELEVON SID PROFILE	Media Quality of Service	
	Audia Straam Moda DSR Quality Matrice Number of Calls	10
V SIP Server Tables	Video Application Stream Mode Displand Ounline Metrics Time Refere Reter	10
E telekom sin server	Modia Transcoding Enabled Quality Metrics Time Before Kerry	0
cucm	Media List None Enable Min MOS Threshold	Disabled
Truck Groups	Frable May R/T Delay	Enabled
NAT Qualified Brefix Tables	May, P/T Dalay	65535
Remote Authorization Tables	Eashia May, 10tar	Enabled
TELEKOM-REMOTE-AUTH-TABLE	Enable Max. Sitter May Titter	3000
	Max. Jitter	5000

Call Routing for IP-PBX (CUCM) signaling group : Any signaling coming from CUCM will be routed to Deutsche Telekom



# SWe Lite Configuration Towards IP-PBX CUCM

# **SIP Server Table**

SIP Server Tables contain information about the SIP devices connected to the SBC Edge. Create a new SIP Server Table towards IP-PBX (Cisco CUCM)

#### Select Settings > SIP > SIP Server Tables

- Create a SIP Server Table with IP/FQDN.
- Provide CUCM IP in the Host FQDN/IP.
- Provide Port as 5060.
- · Choose Protocol as TCP.
- Click Apply.

Q Search	cucm		
Expand Collapse   Reload	Create SIP Server 🔻   🗙   🖉	Total 1 SIP Server Row	
	Host / Domain	Server Lookup	Port Protocol
Call Routing		IP/FODN	5060 TCP
Signaling Groups			
Metworking Interfaces			<b>T</b>
System	Server Ho	ost	Iransport
Auth and Directory Services			
Protocols	Server Lookup IP/FQDN		Monitor None 🗸
	Priority 1 🗸		
Local Registrars	Host FODN/IP	*	
Local / Pass-thru Auth Tables			
SIP Profiles	Port 5060	* [165535]	
Default SIP Server	Protocol TCP 🗸	*	
telekom sip server table			
Cucm	Remote Authorization	and Contacts	Connection Reuse
💋 Trunk Groups			
p NAT Qualified Prefix Tables	Remote Authorization Table Nor	e 🗸 +	Reuse True 🗸
Remote Authorization Tables	Contact Registrant Table		Sockets 4 ¥
🕨 🥖 Contact Registrant Table			SUCKELS 4 -
V Message Manipulation	Session URI Validation Libe	ral 🗸	Reuse Timeout Forever 👻
Vessage Rule Tables			
telekom			
SMM FOR INV			
SMM FOR REG			Apply

# **Signaling Group Table**

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

#### Select Settings > Signaling Groups

- Create an entry in signaling group named "CUCM".Choose "Default SIP profile" under SIP Profile.
- Choose Call Routing as "From CUCM".
  Choose Sip Mode as "Basic Call".
- Choose Agent Type "Back to Back user agent".
- Choose Sip Server Table created in the previous step.

	SIP C	hannels and Routing			
\$_}				Media Informati	on
ribbor	Action Set Table	None			
	Call Routing Table	FROM CUCM		DSP	
A	No. of Channels	60	Supported Audio	Proxy	
Q Search	SIP Profile	Default SIP Profile	Modes	Direct	DTD
Expand Collapse   Reload	SIP Mode	Basic Call		Proxy with Local S	KIP Y
	Agent Type	Back-to-Back User Agent	Supported	Proxy	
Call Routing	SIP Server Table	cucm	Video/Application Modes	Direct	*
r 💋 Signaling Groups	Load Balancing	Priority: Register All			1
(SIP) telekom	Channel Hunting	Most Idle	Media List ID	Default Media List	
(SIP) rigistrar	Notify Lync CAC Profile	Disable	Proxy Local SRTP Crypto Profile ID	None	
Networking Interfaces	Challenge Request	Disable	Play Ringback	Auto on 180	
System	Outbound Proxy IP/FQDN		Tone Table	Default Tone Table	
Auth and Directory Services	Outbound Proxy Port	5060	Play Congestion	Disable	
Protocols	Call Setup Response Timer	255	Tone	Disable	
🛙 💋 SIP	Call Proceeding Timer	180	Early 183	Disable	
🕨 🥖 Local Registrars	Use Register as Keep Alive	Enable	Allow Refresh SDP	Enable	
💋 Local / Pass-thru Auth Tables	Forked Call Answered Too Soon	Disable	Music on Hold	Disabled	
SIP Profiles	SID Decen	ullu a	RTCP		
V SIP Server Tables	SIP Recoi	aing	Multiplexing	Disable	
Default SIP Server	STR Recording Status Diral	blad	L		
				Mapping Table	s
Trunk Groups					
NAT Qualified Prefix Tables			SIP To Q.850	0 Override Table 1	Default (RFC4497)
Figure Authorization Tables			Q.850 To SI	P Override Table I	Default (RFC4497)
🕨 📁 Contact Registrant Table			Pass-thru Peer SIP	Response Code E	nable
🕨 📹 Mercane Manipulation	1			Activ	rate Windows

- Update the Federated IP/FQDN , i.e. the IP of the CUCM.
- Add a listening port for TCP.

CSIP telekom				Opt Signaling/Media S Signal ICE ———————————————————————————————————	imization Disable Source IP Ethernet 2 IP Traversal Support Disabled atic NAT - Outbound Traversal None tatic NAT - Inbound Detection Disabled
Vetworking Interfaces			Listen Ports		Federated IP/FQDN
Auth and Directory Services	Total 2 SIP Lis	ten Port Rows		Total 1 SIP Federated IF	Row
Protocols					
🔻 💋 SIP	Port	Protocol	TLS Profile ID	IP/FQDN	Netmask/Prefix
🕨 💋 Local Registrars	5060	UDP	N/A		255.255.255.255
📁 Local / Pass-thru Auth Tables	5060	TCP	N/A		
SIP Profiles	5000	TCF	100		
V SIP Server Tables					
E Default SIP Server					
telekom sip server table	Message Manipu	lation Enabled			
(= cucm					
💋 Trunk Groups		Inbound M	Message Manipulation	Outbou	nd Message Manipulation
💋 NAT Qualified Prefix Tables				_	
Figure Authorization Tables		save history in	fo from cucm 🔺	p-a-	i 🔺
🕨 🥖 Contact Registrant Table					A stiuste Minder
Massage Manipulation					ACTIVATE WINDOWS

# **Message Manipulation**

The Message Manipulation feature comprises two primary components that work in concert to modify SIP messages. Those components are Condition Rules and Rule Tables. Conditional rule and rule table for the TLS registration and call to work are shown below.

# Creating 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 match against a; literal value, token, or REGEX.

Settings > SIP > Message Manipulation > Condition Rule Table. Click the Create (+) icon at the top of the Condition Rule Table page.

#### If Authorization is present in INVITE:

- Provide a suitable description for the rule.
- From the Match type drop-down, select "SG USER VALUE 5" as we are checking if the auth is present in the INVITE.
- We have saved the auth header in variable "SG USER VALUE 5" in one of the following Rule tables.

SIP Local Registrars Local / Pass-thru Auth Tables SIP Profiles	SG User Value 1 SG User Value 5 SG User Value 5	Regex	N/A N/A
<ul> <li>SIP Server Tables</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Description auth is present for inv		
Message Manipulation  Message Rule Tables  Condition Rule Table  f auth is present  auth is present for inv  chk if diversion is present  Node-Level SIP Settings	Match Type SG User Value 5 Operation Regex Match Regex Digest realm.*	×	

#### If Authorization is present in REGISTER:

- Provide a suitable description for the rule.
- From the Match type drop-down, select "SG USER VALUE 1" as we are checking if the auth is present in the REGISTER.
- We have saved the auth header in variable "SG USER VALUE 1" in one of the following Rule tables.

V SIP	🔻 📋 🗋 SG User Value 1	Regex	N/A
<ul> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> <li>SIP Server Tables</li> <li>Trunk Groups</li> </ul>	Description If auth is present		
NAT Qualified Prefix Tables     Armonic Authorization Tables     Contact Registrant Table	Match Type		
Message Manipulation     Message Rule Table      Condition Rule Table      if auth is present      auth is present for inv	Match Type SG User Value 1 Operation Regex Match Regex Digest realm.*	×	
chk if diversion is present			

#### If Diversion header is present in INVITE:

- Provide a suitable description for the rule.
- From the Match type drop-down, select "Diversion" .
- Choose Operation as "Equal".
- Choose Match value type as "Token".
- Choose Match Value as Diversion.

SIP	🔻 📋 🗌 diversion	Equals	Token
Local Registrars			
📁 Local / Pass-thru Auth Tables			
🕨 🥖 SIP Profiles	Description chk if dive	ersion is present	
SIP Server Tables			
📁 Trunk Groups			
💋 NAT Qualified Prefix Tables		Match Type	
Remote Authorization Tables		maton type	
Contact Registrant Table	M.L.T.		N.,
Message Manipulation	Match Type	diversion	*
🕨 💋 Message Rule Tables	Operation	Equals 🗸	
Condition Rule Table	Match Value Type	Token 🗸	
if auth is present			N
auth is present for inv	Match Value	diversion	*
chk if diversion is present			

# Creating a SIP Message Rule Table

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+)icon.

Add FQDN provided by Deutsche Telekom in the URI host of the following headers of the outbound SIP messages.

- To
- From
- Req-URI

Add SIP trunk number in URI user for CONTACT header of all outgoing SIP messages.

Select Settings > SIP > Message Manipulation > Message Rule Table

Click the Create Message Rule Table(+) icon.



Telekom - From, To, Request URI sends FQDN in URI Host:

- Provide a description as "Telekom" for the Rule Table.
- Apply the SMM for All messages.
- Click the expand icon next to the Rule Table entry created.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Modify" and header name as "From".
- Under URI host give modify and click add/edit. Provide the FDQN that will replace the URI host in from header.

Protocols	🧹   ⊘   Create Rule 👻	🛛 🗶 🛛 🥕 🛔 Test Mes	sage	Total 9 Message Manipulation Ru	les Rows
Local Registrars Local / Pass-thru Auth Tables	Admin State	Rule Type		Result Type	Description
SIP Profiles	▼ 🛄 🗋 🔍	Header Rule		Optional	change from host to te
🕨 🏓 SIP Server Tables	Test Rule				
💋 Trunk Groups					
📁 NAT Qualified Prefix Tables					
Remote Authorization Tables	Description	<b>[</b> ]	la estre de		
Contact Registrant Table	Description	change from nost to te	l.t-online.de		
Message Manipulation	Condition Expression	Add/Edit			
Message Rule Tables	Admin State	Enabled	<u>×</u>		
telekom	Header Action	Modify	<del>č</del>		
p-asserted	Header Name	From	÷.		
add allow and supported in reg		11011			
🥬 Condition Rule Table					
Node-Level SIP Settings					
SIP Voice Quality Server					
🕨 🥖 CAS	Display Name	Ignore 🗸			
🔻 💋 Security	e olu				
🔻 💋 Users	UR	I Scheme Ignore	~		
Global Security Options	▶ URI	User Info Ignore	~		
Local User Management		URI Host Modify	× Add/	Edit itel t-online de'	
Active User Sessions		URI Port Remove	~		-
Remote Auth Permissions			Total	CDDII:Daram Down	
AD User Group		+ · ×		, seconearan Rows	
C RADIUS User Class			Name	Value	Action
🔻 💋 Login Messages	URI Pa	arameters			
Pre-Login Message				Table is empty	
Post-Login Message					
🔻 💋 SBC Certificates					
Generate SBC Edge CSR					

Under "Telekom" Repeat the same for the To header.

telekom			
🔷   🖉   Create Rule 👻   🗶   🥂	1   Test Message Total 9 Messag	e Manipulation Rules Rows	
Admin State Rule	е Туре	Result Type	Description
🕨 🔲 🗌 🔍 🛛 Head	ider Rule	Optional	change from host to tel.t-online.de
🔻 🗋 🗋 🗤 🛛 Head	ader Rule	Optional	change to host to tel.t-online.de
Test Rule			
Description change to	host to tel.t-online.de		
Condition Expression Add/Edit			
Admin State Enabled	<b>`</b>		
Header Action Modify	× I		
Header Name To	*		
▼ Header Value			
Display Name Ignore	~		
🐨 URI			
URI Scheme			
URI User Info	Ignore V		
URI Host	Modify Add/Edit Stel t-opline	de'	
URI Port	Remove V	na na	
	Total 0 SPRUriParam R	ows	
	<b>T</b> ' <b>O</b>		
URI Parameters	Name Va	lue Action	
ond relative			
	Tak	le is empty	

Under "Telekom" repeat the same for request URI.

▶ 🥖 Auth and Directory Services	telekom						
Protocols	Create Rule		Test Herees	Total O Message M	animulation Bules Do		
	Create Rule		Test Message	iotal 5 Message M	ampulation Rules Ro	13	
Local Registrars	Admin State	Rule Ty	pe		Result Type		Description
Gen Pass-mid Addi Tables	▶ <u>□</u> □ ₩	Header	Rule		Optional		change from ho
SIP Server Tables		Header	Rule		Optional		change to host
📁 Trunk Groups			t Line Bule		Onlineal		
📁 NAT Qualified Prefix Tables		Reque	st Line Rule		Optional		requestine
🕨 🥖 Remote Authorization Tables	Test Rule						
🕨 🥩 Contact Registrant Table							
<ul> <li>Message Manipulation</li> </ul>							
Message Rule Tables	Descriptio	n requestline					
telekom	Condition Expression	Add/Edit					
add allow and supported in reg	Admin Stat	e Enabled	~				
Condition Rule Table	Result Typ	e Optional	~				
Node Lovel SIP Settings							
SIP Voice Quality Server	Request Line Val	ue					
Security		INT	~				
Vsers							
Global Security Options		URI Scheme	Ignore 🗸	<u>.</u>			
Cocal User Management		VRI User Into	Ignore V	•		_	
Carling Active User Sessions		URI Host	Modify 💊	Add/Edit 'tel.t-online	.de'		
Remote Auth Permissions		UKI POR	Ignore 🗸				
E AD User Group			+ 1 ×	Total O SPRUriParam R	ows		
RADIUS User Class						Antion	
▼		URI Parameters	Name	Val	ue	Action	
Pre-Login Message				Teb	1- i		
Post-Login Message				Tab	ie is empty		
V SBC Certificates							

Telekom - add SIP Trunk number in URI user for contact header:

- Click the expand icon next to the Rule Table entry created previously named "Telekom".
  From the Create Rule drop-down box, select Header Rule.
  Provide the desired description.

- Modify Contact header.
- Add SIP Trunk number under URI User.

V C SF	-			
Local Registrars	▼ 🛄 🗋 🦭	Header Rule	Optional	contact
💋 Local / Pass-thru Auth Tables	Test Rule			
🕨 🥖 SIP Profiles				
🕨 🥖 SIP Server Tables				
📁 Trunk Groups	Description For	ntact		
📁 NAT Qualified Prefix Tables	Condition Expression			
🕨 📁 Remote Authorization Tables	Admin State Eng			
🕨 🃁 Contact Registrant Table	Result Type On	ntional V		
🔻 🌽 Message Manipulation	Header Action Mo	odify V		
Message Rule Tables	Header Name Cor	ontact	ż	
telekom	Header Ordinal Number All	✓		
p-asserted				
add allow and supported in reg				
🥬 Condition Rule Table	▼ Header Value			
Node-Level SIP Settings	T URI			
SIP Voice Quality Server				
🕨 🥖 CAS	URI Scheme Ig	gnore 🗸		
🔻 💋 Security	♥ URI User Into			
🔻 💋 Users	URT	I User Modify	Add/Edit	
Global Security Options	Pass	sword lanare	/	
E Local User Management			Tatal A COUL-IllearDaram Davis	
C Active User Sessions		+ · ×	Iotal O SPROFIOSEPParam Rows	
Remote Auth Permissions		Name	e Value	Action
C AD User Group	URI User Param	neters		
C RADIUS User Class			Table is empty	
🔻 💋 Login Messages				
Pre-Login Message				
Post-Login Message	URI Host Ig	gnore 🗸		
🔻 💋 SBC Certificates	URI Port Ig	gnore 🗸		
Generate SBC Edge CSR				

Telekom - add rport in the Via header:

- Click the expand icon next to the Rule Table entry created previously named "Telekom".
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add header parameter "rport" in the Via header.

Protocols	
🔻 🥟 SIP	
🕨 💋 Local Registrars	Description add rport
📁 Local / Pass-thru Auth Tables	Condition Expression Add/Edit
SIP Profiles	Admin State Enabled
🕨 🃁 SIP Server Tables	
📁 Trunk Groups	Result Type Optional V
📁 NAT Qualified Prefix Tables	Header Action Modify 🗸
Remote Authorization Tables	Header Name Via
🕨 📁 Contact Registrant Table	· · · · · · · · · · · · · · · · · · ·
🔻 💋 Message Manipulation	
🔻 🧀 Message Rule Tables	🐨 Header Value
telekom	
E SMM FOR INV	SIP Version Ignore 🗸
E SMM FOR REG	Transport Ignore 🗸
i p-a-i	Host Ignore Y
info from cucm	Dark June 1
i relay-history	
irelay history 2	
🕨 📁 Condition Rule Table	
Node-Level SIP Settings	Header Parameters
📁 SIP Recording	
🕨 💋 Security	+   X Total 1 SPRHeaderParam Row
🕨 📁 Media	Name Value Astion
🕨 📁 Tone Tables	Value Action
Telephony Mapping Tables	🥖 🗌 rport 🤍 Add
🕨 🥟 SNMP/Alarms	

#### Telekom - remove port from request line:

- Click the expand icon next to the Rule Table entry created previously named "Telekom".
  From the Create Rule drop-down box, select Request line Rule.
- Provide the desired description.
- Remove port from request line.

Auth and Directory Services  Protocols  Could be added by the service of the ser	Description     remove port in req line       Condition Expression     Add/Edit       Admin State     Enabled       Result Type     Optional
Initia Gloups     NAT Qualified Prefix Tables     Remote Authorization Tables     Contact Registrant Table     Message Num For Num     SMM FOR INV     SMM FOR REG     p-a-i     save history info from curm     relay-history	
<ul> <li>relay history 2</li> <li>Condition Rule Table</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> <li>Security</li> <li>Media</li> <li>Tone Tables</li> <li>Telephony Mapping Tables</li> </ul>	URI Parameters URI Parameters URI Parameters SIP Version Ignore

Create a new rule table for INVITE messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+)icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK.

NAT Qualified Prefix Tables	s	IP Message Rule Ta	ble	
Remote Authorization Tables	4	📙   🗙   Test Selected T	ables Total 7 SIP Message Manipulation Tabl	e Rows
Message Manipulation	Б	Description		Result Type
Message Rule Tables	E	i telekom		Optional
		🕫 📄 SMM FOR IN	v	Optional
SMM FOR REG				
p-a-i			[	
💼 add + in diversion haeder		Description	SMM FOR INV	
inv - modify history info		Applicable Messages	Selected Messages	
imit hardcoded history info				
info from cucm			Invite 🔺	
relay-history			Add/Edit	
relay history 2	Ē	Message Selection	Remove *	
🔻 💋 Condition Rule Table				
if auth is present			· · · · · · · · · · · · · · · · · · ·	
auth is present for inv		Table Result Type	Ontional ×	
chk if diversion is present		Table Result Type	Optional	

#### SMM for INVITE - save Proxy-Authorization header:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Save the Proxy-Authorization header in variable "SG User Value 5".

£	Note
9	This is used in the Condition Rule Table.

▼ 💋 SIP ▶ 💋 Local Registrars	
💋 Local / Pass-thru Auth Tables	Description save auth
SIP Profiles	Condition Expression Add/Edit
SIP Server Tables	Admin State Enabled 🗸
Trunk Groups NAT Qualified Prefix Tables	Result Type Optional
Ø Remote Authorization Tables	Header Action Modify 🗸
🕨 🥟 Contact Registrant Table	Header Name Proxy-Authorization
V Message Manipulation	
V Wessage Rule Tables	Header Value Copy Value to 🗸 Add/Edit) SG User Value 5

#### SMM for INVITE - If Authorization is present in INVITE delete route:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.

Trunk Groups	*		Header Ru	lle Option	nal sa	ve auth	
NAT Qualified Prefix Tables		V DO V	Header R	ule Optio	onal if	auth is present-dele	te route
Gemote Authorization Tables     Gontact Registrant Table				Message Rule Condition			
Message Manipulation				Match All Conditions			
Message Rule Tables Image: Table telekom		Description	if auth is	auth is present for inv		+×4	
		Condition Expression	Add/Ed				
p-a-i		Admin State	Enabled			oply Cancel	
(and a							

• Remove all Route header from INVITE.

Protocols	▼ □□ ♥	Header Rule	Optional
SIP     Local Registrars     Local / Pass-thru Auth Tables	Test Rule		_
<ul> <li>SIP Profiles</li> <li>SIP Server Tables</li> <li>Trunk Groups</li> </ul>	Description	f auth is present-delete route	
<ul> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Admin State Result Type	Enabled V Optional V	
Message Manipulation     Message Rule Tables     Telekom	Header Action Header Name	Remove ×	
SMM FOR INV	Header Ordinal Number	All 🗸	

### Note

To avoid multiple instances of the same header in INVITE message, All the instances of the header are first removed and then the single instance is added again. Condition Rule is added to achieve it for the following SMM's.

#### SMM for INVITE - add route:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add Route header with the Deutsche Telekom resolved IP.

Protocols				
V SIP	Test Rule			
▶ 🥖 Local Registrars				
💋 Local / Pass-thru Auth Tables		-		
🕨 🃁 SIP Profiles	Description ADD ROLITE			
🕨 📁 SIP Server Tables				
💋 Trunk Groups	Condition Expression Add/Edit			
📁 NAT Qualified Prefix Tables	Admin State Enabled 🗸			
🕨 🥖 Remote Authorization Tables	Result Type Optional 🗸			
🕨 📁 Contact Registrant Table	Header Action Add 🗸			
🔻 🥟 Message Manipulation	Header Name Pouto			
🔻 💋 Message Rule Tables				
telekom [				
SMM FOR INV	N Header Value			
C SMM FOR REG				
e p-a-i				

#### SMM for INVITE - If Authorization is present in INVITE delete P-Early-Media:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove all P-Early-Media header from INVITE.

SIP	Test Rule	
<ul> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> </ul>		
🕨 🍺 SIP Server Tables	Description	f auth is presnt-delete P-Early-Media
💋 Trunk Groups	Condition Expression	Add/Edit) '\${2}'
MAT Qualified Prefix Tables	Admin State	Enabled V
<ul> <li>France Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Result Type	Optional V
🔻 💋 Message Manipulation	Header Action	Remove V
🔻 💋 Message Rule Tables	Header Name	P-Early-Media
telekom		
SMM FOR INV		

#### SMM for INVITE - Add P-Early-Media:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add P-Early-Media header.

Protocols	Test Rule
<ul> <li>Local Registrars</li> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> <li>SIP Server Tables</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> <li>Message Manipulation</li> <li>Message Rule Tables</li> </ul>	Description       Add P-Early-Media         Condition Expression       Add/Edit         Admin State       Enabled         Result Type       Optional         Header Action       Add         Header Name       P-Early-Media
etekom SMM FOR INV SMM FOR REG p-a-i	Header Value Add  Add/Edit Supported'

#### SMM for INVITE - If Authorization is present in INVITE delete Allow-Events:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove all Allow-Events header from INVITE.

V SIP	▼ 🗀 🗆 🖖	Header Rule	Optional
🕨 📁 Local Registrars	Test Dula		
📁 Local / Pass-thru Auth Tables	Test Rule		
🕨 🥖 SIP Profiles			
🕨 📁 SIP Server Tables			
📁 Trunk Groups	Description	if auth is present- delete allow events	
📁 NAT Qualified Prefix Tables	Condition Expression	Add/Edit	
🕨 💋 Remote Authorization Tables	Condition Expression		
🕨 📁 Contact Registrant Table	Admin State	Enabled V	
🔻 💋 Message Manipulation	Result Type	Optional 🗸	
🔻 💋 Message Rule Tables	Header Action	Remove 🗸	
telekom	Header Name	Allow-Events	
E SMM FOR INV		Allow-Events	
SMM FOR REG			

### SMM for INVITE - Add Allow-Events

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.Provide the desired description.
- Add Allow-Events header.

▶ 📁 Protocols	🔻 🛄 🗋 🗛	Header Rule Optional	add a
V SIP	Test Rule		
Local Registrars			
📁 Local / Pass-thru Auth Tables			_
🕨 🃁 SIP Profiles		[]	
🕨 🏓 SIP Server Tables	Description	add allow events	
📁 Trunk Groups	Condition Expression	Add/Edit	
p NAT Qualified Prefix Tables	Admin State	Enabled V	
Remote Authorization Tables	Result Type	Ontional	
🕨 🥖 Contact Registrant Table			
Message Manipulation	Header Action	Add	
🔻 🥟 Message Rule Tables	Header Name	Allow-Events	
telekom			_
SMM FOR INV			
SMM FOR REG	Header Value Add	✓ Add/Edit /refer, messaae-summary, dialo	
e p-a-i			
isave history info from cucm			

#### SMM for INVITE - Remove user and Add transport parameter in request line URI:

- Click the expand icon next to the Rule Table entry created above.From the Create Rule drop-down box, select Request Line Rule.
- Provide the desired description.
- Remove user and Add transport parameter in request line URI.

Auth and Directory Services	
🕨 🥖 Protocols	Description add tis
FlotOcols     Formula     Local Registrars     Local / Pass-thru Auth Tables     SIP Profiles     SIP Server Tables     Trunk Groups     NAT Qualified Prefix Tables     Remote Authorization Tables	Condition Expression Add/Edit Admin State Enabled Result Type Optional   Request Line Value  Method Ignore
Contact Registrant Table     Message Manipulation     Message Mule Tables     Elekom     SMM FOR INV     SMM FOR REG     P-a-I     save history info from cucm     relay-history	VRI     URI Scheme Ignore      VRI User Info Ignore      VRI User Info Ignore      URI Host Ignore      URI Port Ignore      VRI Port Ignore      Total 2 SPRUriParam Rows
<ul> <li>leave history 2</li> <li>leave bit on Rule Table</li> <li>Node-Level SIP Settings</li> <li>SIP Recording</li> </ul>	URI Parameters URI Parameters URI Parameters URI Parameters URI Parameters URI Parameters Action Act
Security     Media     Jone Tables     Jelephony Mapping Tables	SIP Version Ignore

 Note For TLS calls to work INVITE messages sent to Deutsche Telekom should have the following headers.
 The initial INVITE includes the SIP header fields:

 Proxy-Require: mediasec
 Require: mediasec
 Security-Verify: msrp-tls;mediasec
 Security-Verify: sdes-srtp;mediasec
 Security-Verify: dtls-srtp;mediasec

 Additionally, the SDP includes the attribute:

 a=3ge2ae:requested

### SMM for INVITE - If Authorization is present in INVITE delete Proxy-Require:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove all Proxy-Require header from INVITE.

V DIP	▼ 📮 🗆 💐	Header Rule	Optional
🕨 💋 Local Registrars	Test Rule		
📁 Local / Pass-thru Auth Tables			
🕨 💋 SIP Profiles			
SIP Server Tables	Description	If auth is present-delete Proxy-Require	
MAT Qualified Prefix Tables	Condition Expression	Add/Edit '\${2}'	
🕨 📁 Remote Authorization Tables	Admin State	Enabled 🗸	
▶ 💋 Contact Registrant Table	Result Type	Optional 🗸	
<ul> <li>Message Manipulation</li> <li>Message Rule Tables</li> </ul>	Header Action	Remove V	
telekom	Header Name	Proxy-Require	
SMM FOR INV			

### SMM for INVITE - Add Proxy-Require

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add Proxy-Require header with value "mediasec".

V DI SIP	Test Rule
🕨 🥖 Local Registrars	
📁 Local / Pass-thru Auth Tables	
🕨 💋 SIP Profiles	
🕨 📁 SIP Server Tables	Description add Proxy-Require
💋 Trunk Groups	Condition Expression Add/Edit
📁 NAT Qualified Prefix Tables	Admin State Enabled 🗸
🕨 🥖 Remote Authorization Tables	Result Type Ontional
🕨 📁 Contact Registrant Table	
Message Manipulation	Header Action Add
🔻 💋 Message Rule Tables	Header Name Proxy-Require
telekom	
SMM FOR INV	
SMM FOR REG	Header Value Add V Add/Edit 'mediasec'
i p-a-i	
e save history info from cucm	

#### SMM for INVITE - If Authorization is present in INVITE delete Require:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove all Require header from INVITE.

V 🖉 SIP	🔻 🔲 🗌 🐺 🛛 Head	der Rule Optional
🕨 🍺 Local Registrars	Test Rule	
📁 Local / Pass-thru Auth Tables		
🕨 🥩 SIP Profiles		
🕨 📁 SIP Server Tables		
💋 Trunk Groups	Description If auth is p	resent -delete Require
📁 NAT Qualified Prefix Tables	Condition Expression Add/Edit	'\${2}'
🕨 📁 Remote Authorization Tables	Admin State Enabled	~
🕨 🥖 Contact Registrant Table	Result Type Optional	
Vessage Manipulation	Header Action Pomovo	
🔻 🧀 Message Rule Tables	Reader Action Remove	
telekom	Header Name Require	*
E SMM FOR INV		
SMM FOR REG		

#### SMM for INVITE - Add Require:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add Require header with value "mediasec".

SIP	Test Rule
Local Registrars	
📁 Local / Pass-thru Auth Tables	
SIP Profiles	
SIP Server Tables	Description add Require
💋 Trunk Groups	Condition Expression Add/Edit
💋 NAT Qualified Prefix Tables	Admin State Enabled 🗸
Remote Authorization Tables	Result Type Optional V
🕨 📁 Contact Registrant Table	Header Action Add
🔻 🥟 Message Manipulation	
🔻 💋 Message Rule Tables	Header Name Require
telekom	
SMM FOR INV	
SMM FOR REG	Header Value Add V Add/Edit 'mediasec'
p-a-i	

#### SMM for INVITE - If Authorization is present in INVITE delete Security-Verify:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove all Security-Verify header from INVITE.

V DIP	V DO V	Header Rule	Optional
🕨 🥟 Local Registrars			_
📁 Local / Pass-thru Auth Tables	Test Rule		
🕨 🥖 SIP Profiles			
🕨 📁 SIP Server Tables			
📁 Trunk Groups	Description	if suth is present -delete Security-Verify	
📁 NAT Qualified Prefix Tables	Condition Expression		
🕨 📁 Remote Authorization Tables	Condition Expression		
🕨 📁 Contact Registrant Table	Admin State	Enabled 🗸	
🔻 🥟 Message Manipulation	Result Type	Optional 🗸	
🔻 💋 Message Rule Tables	Header Action	Remove 🗸	
telekom	Header Name	Security-Verify	
E SMM FOR INV		Security verify	
E SMM FOR REG			

#### SMM for INVITE - Add Security-Verify:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- AddSecurity-Verify header with value "msrp-tls;mediasec, sdes-srtp;mediasec, dtls-srtp;mediasec".

V 💋 SIP	Test Rule
🕨 💋 Local Registrars	
📁 Local / Pass-thru Auth Tables	
🕨 🃁 SIP Profiles	Description Indel Convite Visite
SIP Server Tables	beschpton add security-verify
💋 Trunk Groups	Condition Expression Add/Edit
💋 NAT Qualified Prefix Tables	Admin State Enabled 🗸
🕨 💋 Remote Authorization Tables	Result Type Optional 🗸
🕨 📁 Contact Registrant Table	Header Action Add
🔻 🤌 Message Manipulation	
🔻 🧀 Message Rule Tables	Header Name Security-Verity
telekom	
SMM FOR INV	
SMM FOR REG	Header Value Add V Add/Edit 'msrp-tls;mediasec, sdes-srtp;m
p-a-i	

#### SMM for INVITE - If Authorization is present in INVITE delete SDP info a=3ge2ae:requested:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Raw Message Rule.
- Provide the desired description.
- Attach Condition Rule "If Auth is present in INVITE" in condition Expression.
- Remove "a=3ge2ae:requested" from INVITE SDP.

Protocols	Test Rule
<ul> <li>SIP</li> <li>Local Registrars</li> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> <li>SIP Server Tables</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Description     If auth is present-delete Sdp val       Condition Expression     Add/Edit       S(2)'       Admin State     Enabled       Result Type     Optional
Message Manipulation  Message Rule Tables  telekom  SMM FOR INV  SMM FOR REG	Match Regex a=3ge2ae:requested * Replace Regex "

#### SMM for INVITE - Add a=3ge2ae:requested in INVITE SDP:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Raw Message Rule.
- Provide the desired description.
- Add "a=3ge2ae:requested" from INVITE SDP.

Protocols	🔻 📋 🗋 👢 Raw Message Rule	Optional
<ul> <li>SIP</li> <li>Local Registrars</li> <li>Local / Pass-thru Auth Tables</li> <li>SIP Profiles</li> <li>SIP Server Tables</li> <li>Trunk Groups</li> <li>NAT Qualified Prefix Tables</li> <li>Remote Authorization Tables</li> <li>Contact Registrant Table</li> </ul>	Test Rule           Description         add sdp           Condition Expression         Add/Edit           Admin State         Enabled           Result Type         Optional	
Message Manipulation Message Rule Tables telekom SMM FOR INV SMM FOR REG p-a-i save history info from cucm	Match Regex \$ * Replace Regex a=3ge2ae:requested *	

#### SMM for INVITE - Add P-Asserted-Identity:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Modify P-Asserted-Identity header, the host IP should have Deutsche Telekom domain.

V SIP	🕨 📴 🗆 🗤 Header Rule	Optional
🥖 Local / Pass-thru Auth Tables	🕨 🛅 🔲 🗤 Header Rule	Optional
SIP Profiles	Raw Message Rule	Optional
Trunk Groups	Raw Message Rule	Optional
NAT Qualified Prefix Tables		-
Remote Authorization Tables	Header Rule	Optional
Contact Registrant Table	Description p asserted identity - add domain	
🔻 🥔 Message Manipulation	Condition Expression Add/Edit	
🔻 💋 Message Rule Tables	Admin State Enabled 💙	
C telekom	Result Type Optional	
SMM FOR INV	Header Action Modify	
SMM FOR REG	Header Name P-Asserted-Identity	
p-a-i	Header Ordinal Number 1++	
save history info from cucm		
elay-history		
relay history 2	w Header Value	
Condition Rule Table		
Node-Level SIP Settings	Display Name Ignore 🗸	
SIP Recording	♥ URI	
Security	UDI Scheme Jacon Ad	
🕨 🏓 Media		
🕨 📁 Tone Tables		
Telephony Mapping Tables	URI Host Modify V Add/Edit 'tel.t-online.de'	
SNMP/Alarms	URI Port Ignore V	
Logging Configuration	L IX Total 0 SPRUriParam Rows	
<ul> <li>Emergency Services</li> </ul>		
	Name Value Action	
	Table is empty	
	Header Parameters	
	+ I X Total 0 SPRHeaderParam Rows	
	Name Value Action	
	Table is empty	

Create a new rule table for REGISTER messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+) icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Register from the Message Selection list.
- Click OK

📁 Trunk Groups	SIP Message Rule Table	
Remote Authorization Table:	+   🗙   Test Selected Tables Total 7 SIP Message Manipulation Table R	ows
Contact Registrant Table	Description	Result Type
Message Rule Tables	▶ 📴 🗋 telekom	Optional
Elekom	▶ 📴 🗋 SMM FOR INV	Optional
SMM FOR REG	🔻 📋 🗋 SMM FOR REG	Optional
p-a-i		
inv - modify history inf	Description SMM FOR REG	
hardcoded history info	Applicable Messages	
relay-history	Register	
relay history 2	Message Selection *	
Condition Rule Table	Remove	
auth is present for inv		
chk if diversion is pres	Table Result Type Optional 🗸	
Node-Level SIP Settings		

SMM for REG - Add Allow in REGISTER:

- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Add" and header name as "Allow".
- Under header value give "Add" and click on add/edit and provide 'ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, MESSAGE, SUBSCRIBE, UPDATE, PRACK, REFER'.
- Click Apply.

NAT Qualified Prefix Tables	a	dd allow	and suppor	ted in	n reg				
Gentact Registrant Table		/ 101 0	Create Rule 🔻	I X	🥖   Test Mess	age Total 2 M	lessage Manipulatio	on Rules Rows	
Message Manipulation	E		Admin State		Rule Type		Result Type		Descri
Message Rule Tables	Ŀ	r 🗀 🗆	₽⁄		Header Rule		Optional		add
p-asserted add allow and supported in rec	F	Fest Rule	-	-	-	_	_	_	
💋 Condition Rule Table	1.								
Node-Level SIP Settings			Description	add					
SIP Voice Quality Server		Conditio	n Expression	Add	[				- 11
🕨 🃁 CAS		Condition	on Expression	Add/i					- 1
▼ 龙 Security	:		Admin State	Enable	ed	<b>×</b>			- 1
🔻 💋 Users	1	_	Result Type	Optio	nal	<u>~</u>			- 1
Global Security Options	Ш	E E	Header Action	Add		~			- 1
Local User Management	Ш		Header Name	Allow		<b>▼</b> *			- 1
Active User Sessions									
Remote Auth Permissions	15								
🔻 🟳 Login Messages		Header	Value Add		✓ Add/Edi	ACK BYE CANCEL INFO	O INVI		
Pre-Login Message			Add		Add/Edi	Men, Bre, CATCEL, INT	C, III III		
Post-Login Message									

#### SMM for REG - Add Supported in REGISTER:

- Under the same Message Rule Table, choose Create Rule from the drop-down box, select Header Rule.
- Provide the desired description.
  Provide Header action as "Add" and header name as "Supported".
- Under header value, give "Add" and click on add/edit and provide '100rel, replaces'.
- Click Apply.

NAT Qualified Prefix Tables	add allow and supp	orted in reg		
Genetic Authorization Tables     Contact Registrant Table	🧹   ⊘   Create Rule	▼   🗙   🥖 12   Test Message	Total 2 Message Manipulation I	Rules Rows
Message Manipulation	Admin State	Rule Type	Result Type	Description
telekom	Þ 🔲 🗆 🦭	Header Rule	Optional	add
p-asserted	▼ 🗀 🗆 🦭	Header Rule	Optional	add supported
Condition Rule Table	Test Rule	_		
Node-Level SIP Settings				
SIP Voice Quality Server CAS	Description	add supported		
🔻 💋 Security	Condition Expression	Add/Edit		
Visers	Admin State	e Enabled 🗸		
Global Security Options	Result Type	e Optional 🗸		
Active User Sessions	Header Action	n Add 🗸		
Remote Auth Permissions	Header Nam	e Supported 🔻	k	
Login Messages      Dra Login Messages				
Post-Login Message	Header Value Add	✓ Add/Edit	rel, replaces	
▼				

Note

For successful registration of trunk to Deutsche Telekom, the following header must be in REGISTER header.

For an initial REGISTER without Authentication Challenge, include the SIP header fields:

• Security-Client: sdes-srtp;mediasec

- Proxy-Require: mediasec
- Require: mediasec

For the following REGISTER with Authentication Challenge, in addition to the originally included SIP header fields it should also contain the following headers:

Security-Verify: msrp-tls;mediasec

- Security-Verify: sdes-srtp;mediasec
- Security-Verify: dtls-srtp;mediasec

#### SMM for REG - Add Security-Client:

- Click the expand icon next to the Rule Table entry created above.
- · From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- Add Security-Client with value "sdes-srtp;mediasec".

V 💋 SIP	Test Rule
🕨 📁 Local Registrars	
📁 Local / Pass-thru Auth Tables	
SIP Profiles	Description add Security-Client
SIP Server Tables Trunk Groups	Condition Expression Add/Edit
MAT Qualified Prefix Tables	Admin State Enabled 🗸
Ø Remote Authorization Tables	Result Type Optional 🗸
Contact Registrant Table	Header Action Add
Vessage Manipulation	Header Name Security-Client
V province and the second seco	
e telekom	
SMM FOR INV	
SMM FOR REG	Header Value Add  Add/Edit 'sdes-srtp;mediasec'
e p-a-i	

#### SMM for REG - Add Proxy-Require:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- Add Proxy-Require with value "mediasec".

Protocols	▼ □□ ₩	Header Rule Optional	add Pr
V 💋 SIP	Test Rule		
🕨 🏓 Local Registrars	Test Rule		_
📁 Local / Pass-thru Auth Tables			
🕨 🥖 SIP Profiles			
🕨 📁 SIP Server Tables	Description	add Proxy-Require	
📁 Trunk Groups	Condition Expression	Add/Edit	
📁 NAT Qualified Prefix Tables			
🕨 📁 Remote Authorization Tables	Admin State	Enabled	
🕨 📁 Contact Registrant Table	Result Type	Optional 🗸	
🔻 🧼 Message Manipulation 🕴	Header Action	Add 🗸	
🔻 💋 Message Rule Tables	Header Name	Proxy-Require *	
telekom			
SMM FOR INV			
SMM FOR REG	Header Value Add	Add/Edit	
e p-a-i	Add Add		
save history info from cucm			

#### SMM for REG - Add Require:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- Add Require with value "mediasec".

▶ 💋 Protocols	🔻 🗀 🗋 🕸	Header Rule	Optional	add Re
	Test Rule			
Local / Pass-thru Auth Tables				
SIP Profiles           Ø SIP Server Tables	Description	add Require		
💋 Trunk Groups	Condition Expression	Add/Edit		
NAT Qualified Prefix Tables           Image: state of the state	Admin State	Enabled 🗸		
Figure Contact Registrant Table	Result Type Header Action	Optional		
<ul> <li>Message Manipulation</li> <li>Message Rule Tables</li> </ul>	Header Name	Require	*	
telekom			-	
SMM FOR INV	Header Value Add	✓ Add/Edit	nediasec'	
save history info from cucm				

#### SMM for REG - save Authorization:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- Save Authorization under variable "SG User Value 1".

▶ protocols	🔻 🔲 🗋 🦞	Header Rule	Optional save a
V 💋 SIP	Test Rule		
🕨 💋 Local Registrars			
📁 Local / Pass-thru Auth Tables			
SIP Profiles		c	
🕨 🃁 SIP Server Tables	Description	save auth	
💋 Trunk Groups	Condition Expression	Add/Edit	
📁 NAT Qualified Prefix Tables	Admin State	Enabled V	
🕨 🥖 Remote Authorization Tables	Beault Ture		
🕨 📁 Contact Registrant Table	Result Type	Optional 🗸	
🔻 🥟 Message Manipulation	Header Action	Modify 🗸	
🔻 💋 Message Rule Tables	Header Name	Authorization *	
telekom			
C SMM FOR INV			
SMM FOR REG	Header Value Conv.	Value to X Add/Edit	alue 1
📄 p-a-i	copy		
is save history info from cucm	L		

Note

This will be used for condition rule table.

#### SMM for REG - add Security-Verify:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- Add condition and check if Authorization header is present.

🕨 📁 Contact Registrant Table 🔒	Test Rule					
The Message Manipulation						
Message Rule Tables     Telekom			Message Rule Cor	ndition		
SMM FOR INV	Description	add securit	Match All Conditions			
SMM FOR REG	Condition Expression	Add/Edit	Match All Conditions	<u> </u>		
p-a-i	Admin State	Enabled	if auth is present	~	+ × 4	
add + in diversion hae	Result Type	Optional				
inv - modify history inf	Header Action	Add			Apply Cancel	
esve history info	Header Name	Security-Ve				

• Add Security-Verify header with value "msrp-tls;mediasec, sdes-srtp;mediasec, dtls-srtp;mediasec".

▶ 💋 Protocols	▼ 🛄 🛛 🗤	Header Rule	Optional	add securit
V DI SIP	Test Rule			
🕨 🥖 Local Registrars				
📁 Local / Pass-thru Auth Tables				
SIP Profiles				
SIP Server Tables	Description	add security-verify if auth is	s present	]
💋 Trunk Groups	Condition Expression	Add/Edit '\${1}'		
📁 NAT Qualified Prefix Tables	Admin State	Enabled		
Remote Authorization Tables	Admin State	Linabled 🔹		
Contact Registrant Table	Result Type	Optional 🗸		
🔻 🥟 Message Manipulation	Header Action	Add 🗸		
🔻 🤣 Message Rule Tables	Header Name	Security-Verify	*	
telekom			=	
SMM FOR INV				
SMM FOR REG	Header Value Add	Add/Edit	mern_tls:mediasec_sdes_srtn:m	
e p-a-i	Add	- Add/Edit	msip-us,meatasec, sues-srip,m	
info from cucm				

Create a new rule table for INVITE messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+) icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK

Message Manipulation     Message Rule Tables	SMM FOR REG	Optional
telekom	🐨 📋 🗌 p-a-i	Optional
SMM FOR REG	Description p-a-i	
inv - modify history inf	Applicable Messages	
is ave history info from interval and interv	Message Selection Add/Edit Remove *	
Condition Rule Table	Table Result Type Optional	

SMM for PAI - remove + from the number sent out to PBX/PSNT end:

#### Note

This SMM depends on the number transformation that is chosen in SWe Lite. For example, in our lab setup the phones registered to the PBX has phone number as 4xxxxxxxx. Any request from Deutsche Telekom will have number +4xxxxxxxxx. These changes are handled by transformation tables in SWe Lite. This will update only 'To', 'From' headers, the changes in P-Asserted-Identity header for the number needs to be done using this SMM. Add regex based on the requirements.

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Modify P-Asserted-Identity header.
- SMM removes + from the number present in the uri user of P-Asserted-Identity header.

Q Search	Admin State		Rule Type			Result Type
Expand All   Collapse All   Reload	▼ 🛄 🗋 🦭		Header R	ule		Optional
	Description pai					
Signaling Groups	Condition Expression Add/E	dit				
Metworking Interfaces	Admin State Enable	d	~			
🕨 🃁 System	Result Type Option	nal	~			
Auth and Directory Services	Header Action Modifi	/	~			
Protocols	Header Name P-Asse	rted-Identit	v	*		
SIP	Header Ordinal Number		~			
Local / Pass-thru Auth Tables			•			
SIP Profiles						
SIP Server Tables	▼ Header Value					
📁 Trunk Groups						
💋 NAT Qualified Prefix Tables	Display Name Ignore	~				
Remote Authorization Tables	VRI URI					
Contact Registrant Table	LIPI Scheme	Ignore	~			
Message Manipulation	v URI User Info	Ignore	•			
telekom						
SMM FOR INV		URI User	Modify	✓ Add/Edit) Mate	:h:\+(.*)	Replace: \1
SMM FOR REG		Password	lanore	~		
<b>p-a-i</b>			+ I X	Total O SPRUriU	serParam Rows	
Telay-history				Name	Value	Action
i relay history 2	URI User Pa	rameters			vuide .	Action
🕨 🥖 Condition Rule Table					Table is empty	
Node-Level SIP Settings						
SIP Recording						
🕨 🃁 Security	URI Host	Ignore	~			
🕨 📁 Media	URI Port	Ignore	~			
Figure Tables				T - 1		
Telephony Mapping Tables		+ 1×		Iotal O SPRUriParam Rows		
SNMP/Alarms			Name	Va	ue	Action
Eugying Configuration	URI Parameters					
				Té	ble is empty	

Create a new rule table for INVITE messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+) icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK

Message Manipulation	🕨 📋 🗋 p-a-i		Optional
telekom	🔻 🔲 🗌 save history	info from cucm	Optional
SMM FOR INV			
SMM FOR REG	Description	save history info from cucm	
💼 p-a-i			
add + in diversion hae	Applicable Messages	Selected Messages 🗸 🗸	
inv - modify history inf		Invite 🔺	
hardcoded history info	Message Selection	Add/Edit Remove *	
Condition Rule Table	Table Result Type	Optional V	

### Note

SWe Lite does not support History Info header. SWe Lite will convert History Info header into Diversion header while relaying it out to Deutsche Telekom. As Deutsche Telekom expects History Info, we are storing the header that we receive from PBX in a local variable. This header will be used later.

#### Save History info - save History Info in a local variable:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description
- · Choose header action as 'Modify' and Header name as 'History Info'.
- Choose "Copy Value to" option to store History Info received from PBX in a local variable "SG User Value 2".

▶ 💋 Protocols	▼ 🛄 🗋 🕸	Header Rule Optional
V SIP	Test Rule	
🕨 📁 Local Registrars		
📁 Local / Pass-thru Auth Tables		
SIP Profiles		
🕨 📁 SIP Server Tables	Description	save all history info
📁 Trunk Groups	Condition Expression	Add/Edit
💋 NAT Qualified Prefix Tables	Admin State	Enabled V
Remote Authorization Tables	Result Type	
🕨 📁 Contact Registrant Table	itebuic type	
🔻 🧀 Message Manipulation	Header Action	Modify
🔻 💋 Message Rule Tables	Header Name	History-Info
telekom	Header Ordinal Number	
C SMM FOR INV		
E SMM FOR REG		
p-a-i	Header Value	Value to X Add/Edit SG User Value 2
💼 save history info from cucm	copy	
relay-history		
ielay history 2		

Create a new rule table for INVITE messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+) icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK

V Message Manipulation	SMM FOR INV	Optional
🔻 💋 Message Rule Tables	SMM FOR REG	Optional
telekom	▶ 📴 🗆 µ-d-i	Optional
SMM FOR REG	▶ 📴 🗋 save history info from cucm	Optional
💼 p-a-i	🔻 📋 🗌 relay-history	Optional
inv - modify history inf		
hardcoded history info	Description relay-history	
save history info from	Applicable Messages V	
relay-history	Invite	
🔻 💋 Condition Rule Table	Add/Edit	
if auth is present	Message Selection Remove	
auth is present for inv		
chk if diversion is pres	· · · · · · · · · · · · · · · · · · ·	
Node-Level SIP Settings	Table Result Type Optional	
💋 SIP Recording	L	
-		

Note

Add the history-info header that was stored in the previous step to the INVITE sent to Deutsche Telekom.

#### Save History info - save History Info in a local variable:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Choose header action as 'Add' and header name as 'History-Info'.
- Add value from variable "SG User Value 2".

🔻 💋 SIP	🔻 🛄 🗌 県	Header Rule Optional	add histo
Local Registrars Local / Pass-thru Auth Tables	Test Rule		
SIP Profiles			
SIP Server Tables	Description	Indel history info	
NAT Qualified Prefix Tables	Condition Expression	Add/Edit ['\$(3)'	
Genetic Authorization Tables      Genetic Registrant Table      Genetic Registratter      Genetic Registrant Table      G	Admin State Result Type	Enabled  V Optional V	
Vessage Rule Tables	Header Action	Add	
telekom .	Header Name	History-Info	
SMM FOR INV			
SMM FOR REG			
p-a-i	Header Value Ad	d  Add/Edit SG User Value 2	
ave history info from cucm			
relay-history			
relay history 2			

### Note

This SMM depends on the number transformation that is chosen in Swe Lite. For example, in our lab setup the phones registered to the PBX has phone number as 4xxxxxxxxx. Any request to Deutsche Telekom will have number +4xxxxxxxxxx. To accommodate this in Diversion header we need to add SMM. This SMM will add + before the number.

Note

To avoid duplicate + on the diversion header during re-Invite we need to remove all the + and then add only one +.

#### Relay History - remove + from diversion header:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Choose header action as 'Modify' and header name as 'Diversion'.
- Remove + using regex.

▶ 💋 Protocols			
V SIP			_
🕨 💋 Local Registrars	Description	emove additional + from diversion	
💋 Local / Pass-thru Auth Tables	Condition Expression	Add/Edit ]	
SIP Profiles	Admin State	Enabled ×	
SIP Server Tables	Pocult Type		
Trunk Groups	Result Type		
NAT Qualified Prefix Tables	Header Action	Modify ~	
Remote Authorization Tables	Header Name	Diversion *	
🕨 📁 Contact Registrant Table	Header Ordinal Number	1st 🗸	
Message Manipulation			
V Message Rule Tables			
Telekom	Header Value Modif	✓ Add/Edit Match: \+(\d{9,15})	Replace: \1
save history info from cucm		Header Parameters	
relay-history			
relay history 2	🕂 l 🗙 Tota	0 SPRHeaderParam Rows	
Condition Rule Table	Name	Value Act	tion
Node-Level SIP Settings			
💋 SIP Recording		Table is empty	
▶ 🍺 Security			

#### Relay History - add + from diversion header:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Choose header action as 'Modify' and header name as 'Diversion'.
- add + using regex.

🕨 🃁 Protocols				
SIP	Description	add + in diversion		
Local / Pass-thru Auth Tables	Condition Expression	Add/Edit		
SIP Profiles	Admin State	Enabled	~	
🕨 📁 SIP Server Tables	Result Type	Optional	~	
💋 Trunk Groups	Header Action	Modify	~	
💋 NAT Qualified Prefix Tables	Header Name	Diversion	*	
Remote Authorization Tables	Header Maine	Diversion		
🕨 💋 Contact Registrant Table	Header Ordinal Number	1st	~	
V Message Manipulation				
🔻 🥻 Message Rule Tables				
i telekom	Header Value Modif	y 🗸 Add/Ed	it Match: (\d{10,15})	Replace: +\1
SMM FOR INV				
SMM FOR REG				
p-a-i		F	leader Parameters	
e save history info from cucm				
🔚 relay-history	🕂 l 🗙 Tota	I <b>O SPRHeaderParam</b> R	ows	
elay history 2			M-lu-	A stran
🕨 📁 Condition Rule Table	- Name		value	Action
Node-Level SIP Settings				
💋 SIP Recording			Table is empty	
🕨 🏓 Security				
🕨 🃁 Media				

# Note

P-Preferred-Identity header is an important header for Deutsche Telekom during forward cases. The P-Preferred-Identity header should carry the details of the instance that forwarded the call. This is same as that of the diversion header value. Hence P-Preferred-Identity header value will be picked from diversion header.

# Relay History - add P-Preferred-Identity:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add condition and check if Diversion header is present (this SMM will be applicable only for forward scenario).

V Messade Manipulation		Header Ri	lle	Optional	add history info
V Message Rule Tables	• 💷 👽	Header Ru	ule	Optional	remove additional + from div
telekom		Header	Message Rule Co	ondition	
SMM FOR REG	- DO V	Header	Match All Conditions	~	
p-a-i			chk if diversion is	present 🗸	+ × 4
inv - modify history inf					
hardcoded history info	Descriptio	n add ppi			Apply Cancel
relay-history	Condition Expressio	n Add/Edit			
relay history 2	Admin Stat	e Enabled			
Condition Rule Table	Result Typ	e Optional			
if auth is present	Header Actio	n Add			
chk if diversion is pres	Header Nam	e P-Preferrec			

- Choose header action as 'Add' and header name as 'P-Preferred-Identity'.
- Get user info from diversion header, Uri host as Deutsche Telekom domain (tel.t-online.de) and additional parameter 'user'.

O Search	V 📄 🗌 🐶		Header Rule	Opti	ional	add ppi
Evened All L Collanse All L Belead	Description	add ppi				
Expand All   Collapse All   Keload	Condition Expression	Add/Edit	·¢/31			
Call Routing	Admin Chate	Enabled	\$(0)			
Signaling Groups	Admin State	Enabled	<b>•</b>			
Sustem	Result Type	Optional	<b>~</b>			
Auth and Directory Services	Header Action	Add	<u> </u>			
Protocols	Header Name	P-Preferre	d-Identity			
V 🖉 SIP						
Local Registrars						
📁 Local / Pass-thru Auth Tables						
🕨 📁 SIP Profiles	Display Name	Ignore	×			
SIP Server Tables	v URI	ignore	-			
Trunk Groups						
NAT Qualified Prefix Tables	U	RI Scheme	lgnore 🗸	]		
Contact Registrant Table	▶ UR	I User Info	Add 🗸	Add/Edit diversion.uri.userinfo.u	user	
Message Manipulation						
V Wessage Rule Tables		URI Host	Add 🗸	Add/Edit 'tel.t-online.de'		
itelekom		URI Port	lgnore 🗸	J		
C SMM FOR INV			+ 1 ×	Total 1 SPRUriParam Row		
SMM FOR REG						
i p-a-i	URI	'arameters	Name	Value		Action
save history info from cucm			🥖 🔲 user	'phone'		Add
relay-history						
Condition Bule Table						
Node Level SIR Settings						
SIP Recording				Header Parameters		
Security						
🕨 🥖 Media	+ 1 X	Total O SPRH	leaderParam Rows			
🕨 🏓 Tone Tables						
Telephony Mapping Tables	Nar	ne		Value	Action	
SNMP/Alarms				Table is success		
Logging Configuration				Tablé is empty		

Create a new rule table for INVITE messages.

Settings > SIP > Message Manipulation > Message Rule Table. Click the Create Message Rule Table(+) icon.

- Provide a description for the Rule Table.
- Apply the SMM only for the Selected messages and choose Invite from the Message Selection list.
- Click OK

V Dessage Manipulation	relay-history				Optional
Message Rule Tables	💌 📋 📄 relay history	2			Optional
SMM FOR INV					
SMM FOR REG	Description	relay history 2			
💼 p-a-i	Applicable Messages	Selected Messages	~		
add + in diversion hae					
inv - modify history inf		Invite	<b>^</b>		
hardcoded history info				Add/Edit	
isave history info from	Message Selection			Remove *	
relay-history					
relay history 2					
♥	Table Result Type	Optional	~		
if auth is present		<u> </u>			

# Note

1st instance of History info relayed to Deutsche Telekom needs to be in the Specific format. Else forwarding wont be successful. The SMM shown below will modify the History info to the following format.

History-Info: <sip:+4XXXXXXXX@tel.t-online.de;cause=302>;index=1

Once that is achieved we delete the Diversion header.

### Relay History 2 - Modify History-info:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add condition and check if Diversion header is present (this SMM will be applicable only for forward scenario).

- · Choose header action as 'Modify' ,header name as 'History-Info' and Header Ordinal Number to 1st.
- Get Uri from P-Preferred-Identity.

Auth and Directory Services				
<ul> <li>Protocols</li> </ul>	Admin State	Rule Type	Result Type	Descript
💌 🌽 SIP	V D V	Header Rule	Optional	modify
Local Registrars				_
📁 Local / Pass-thru Auth Tables	lest Rule			_
SIP Profiles				
SIP Server Tables				
📁 Trunk Groups	Description	modify history		
Difference of the second secon	Condition Expression	Add/Edit '\${3}'		
Remote Authorization Tables	Admin State	Enabled 🗸		
Contact Registrant Table	Result Type			
Message Manipulation	lies des Arties			
Vessage Rule Tables	Header Action	Modiry		
ielekom	Header Name	History-Info		
SMM FOR INV	Header Ordinal Number	1st 🗸		
SMM FOR REG				
i p-a-i				
isave history info from cucm	Header Value Modi	fy  Add/Edit p-preferred-	identity.uri	
irelay-history			~	
relay history 2				

#### Relay History 2 - Modify History-info:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add condition and check if Diversion header is present (this SMM will be applicable only for forward scenario).
- Choose header action as 'Modify' ,header name as 'History-Info' and Header Ordinal Number to 1st.
- Replace 'user=phone' to 'cause=302'.

Protocols	🕨 🗋 🗆 🗤	Header Rule	Optional	modify
V 💋 SIP		Header Rule	Optional	modify
Local Registrars				
🥖 Local / Pass-thru Auth Tables	Test Rule			
SIP Profiles				
SIP Server Tables				
💋 Trunk Groups	Description	modify boodor		ן ו
NAT Qualified Prefix Tables	Description			1
Remote Authorization Tables	Condition Expression	Add/Edit '\${3}'		
Contact Registrant Table	Admin State	Enabled 🗸		
Message Manipulation	Result Type	Optional 🗸		
Vessage Rule Tables	Header Action	Modify 🗸		
i telekom	Header Name	History-Info	~ *	
E SMM FOR INV	Header Ordinal Number	1 ct y	E.	
E SMM FOR REG	fieader Ordinar Number	TSL •		
i p-a-i				
isave history info from cucm				
relay-history	Modif	y Y Add/Edit	Match: user=phone	Replace: cause=302
relay history 2				

#### Relay History 2 - Modify History-info:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add condition and check if Diversion header is present (this SMM will be applicable only for forward scenario).
- Choose header action as 'Modify' ,header name as 'History-Info' and Header Ordinal Number to 1st.
- Add 'Index=1' at the end.

SIP		Hondor Dulo	Ontional	modify
Local Registrars		Header Rule	Optional	mouny
📁 Local / Pass-thru Auth Tables	Test Rule			
SIP Profiles				
🕨 📁 SIP Server Tables				
💋 Trunk Groups	Description	I was to a		
💋 NAT Qualified Prefix Tables	Description	modify history		
Æ det se	Condition Expression	Add/Edit \${3}'		
Contact Registrant Table	Admin State	Enabled 🗸		
Message Manipulation	Result Type	Optional 🗸		
Vessage Rule Tables	Header Action	Modify 🗸		
itelekom	Header Name	History-Info	k .	
E SMM FOR INV		Enstery me		
E SMM FOR REG	Header Ordinal Number	1st 🗸		
e p-a-i				
info from cucm				
ielay-history	Header Value Modif	y Add/Edit Mate	h: \$ Replace: ;index=	1
ili relay history 2				

#### Relay History 2 - Modify History-info:

- Click the expand icon next to the Rule Table entry created above.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Add condition and check if Diversion header is present (this SMM will be applicable only for forward scenario).
- Choose header action as 'Remove' ,header name as 'Diversion' and Header Ordinal Number to 1st.

▶ <i>i</i> Remote Authorization Tables		
Contact Registrant Table		
Message Manipulation	Description	remove diversion
Vessage Rule Tables	Condition Expression	Add/Edit \${3}
SMM FOR INV	Admin State	Enabled V
SMM FOR REG	Result Type	Optional 🗸
i p-a-i	Header Action	Remove 🗸
isave history info from cucm	Header Name	Diversion *
i relay-history	Header Ordinal Number	1st 🗸
relay history 2		

### Updating Signaling Group with Message Manipulation

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

Expand the signaling group towards Deutsche Telekom.

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

- Enable Message Manipulation.
- Choose "Outbound Message Manipulation".
- Add the following SMM's in the same order.

Q Search	S	Signaling Group Table June						June 30, 2021	12:35:38 🗘	
Expand Collapse Reload		/   📙   ⊘   Add SII	📙   🔗   Add SIP SC   🗙 Total 3 Signaling Group Rows							
	15	Туре	Description	Admin State	Service Status		Display			Primary Key
Call Routing		🔻 📋 🗌 SIP	telekom	V	Up		Counters   Chan	nels   Sessions		1
(SIP) telekom	IT.	/ 5061	TLS	telekom tis profile						
(SIP) rigistrar						_				
(SIP) CUCM										
Metworking Interfaces		Message Manipulati	on Enabled V							
dystem     Auth and Directory Services								0.4.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.1.	-	
Protocols	Ш		Indound Wess	age Manipulation				Outbound Message Manipulati	ion	
🔻 💋 SIP	:							telekom	<b>^</b>	
Local Registrars				Up				SMM FOR INV	Up	
Local / Pass-thru Auth Tables		Message Table List		Down		Mess	sage Table List	SMM FOR REG	Down	
V SIP Server Tables				Add/Edit				relay-nistory relay history 2	Add/Edit	
E Default SIP Server				- Remove					- Remove	
itelekom sip server table										
Cucm										
💋 Trunk Groups										Apply
NAT Qualified Prefix Tables										мрргу

Expand the signaling group towards IP-PBX Cisco UCM.

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

- Enable Message Manipulation.
- Choose "Outbound Message Manipulation".
- Add the following SMM's in the same order.

		· · · · · · · · · · · · · · · · · · ·
Q Search	Signaling Group Table	
Expand All   Collapse All   Reload	V   📙   🚫   Add SIP SG   🗙 Total 3 Signaling Group Rows	
🕫 📁 Call Routing	Type Description Admin State S	Service Status Display
🕨 💋 Transformation	🕨 🛄 🗆 SIP telekom 🔍	Up Counters   Channels   Sessions
Time of Day Table	h Ch O STD minister	Dawn Counters I Channels I Sessions
Call Routing Table		
E from local registrar	v 🗋 SIP CUCM 🗤 U	Up Counters   Channels   Sessions
i from telekom	Port Protocol TLS Profile ID	IP/FQDN Netmask/Prefix
EROM CUCM		
i from telekom to cucm	V JUBU UUP INA	
🕨 🥩 Call Actions	/ D 5060 TCP N/A	
🛡 💋 Signaling Groups		
(SIP) telekom		
(SIP) ripistrar	Message Manipulation Enabled V	
Metworking Interfaces	Inbound Message Manipulation	Outbound Message Manipulation
🕨 🧀 System		
Auth and Directory Services	save history info from cucm 🔺 Up	p•a•i ▲ Up
Protocols	Down	Down
V SIP	Message Table List	Message Table List
<ul> <li>Local Registrars</li> <li>Local / Pass thru Auth Tables</li> </ul>	- Remove	Remove
SIP Profiles		

# Section B: CUCM (IP-PBX) Configuration

### Accessing CUCM (Cisco Unified CM Administration)

- 1. Open browse and enter the CUCM IP Address.
- 2. Select Cisco Unified CM Administration from the Navigation drop-down.
- 3. Provide the credentials and click Login.



### **Configure SIP Trunk Security Profile**

Unified Communications Manager Administration groups security-related settings for the SIP trunk to allow you to assign a single security profile to multiple SIP trunks. Security-related settings include device security mode, digest authentication, and incoming/outgoing transport type settings.

- From Cisco Unified CM Administration, navigate to System > Security > SIP Trunk Security Profile.
- Click Add New.

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	ħ
System 👻	Call Routing - Media Resources - Advanced Features - Device - Application - Us	ser Management 👻 Bulk Administration 👻 Help 👻
Find and	List SIP Trunk Security Profiles	
Add N	ew 🔛 Select All 🔛 Clear All 🙀 Delete Selected	
Status -		
(i) 8 re	ords found	
SIP Tru	nk Security Profile (1 - 8 of 8)	
Find SIP T	runk Security Profile where Name 🗸 begins with 🖌	Find Clear Filter
	Name 📥	Description
	DT SBC CORE	DT_SBC_CORE
	Non Secure SIP Conference Bridge	Non Secure SIP Conference Bridge
	Non Secure SIP Trunk Profile	Non Secure SIP Trunk Profile authenticated by null String
	Non Secure SIP Trunk Profile- aish	Non Secure SIP Trunk Profile authenticated by null String
	Non Secure SIP Trunk Profile Pooja UDP	Non Secure SIP Trunk Profile authenticated by null String
	Non Secure SIP Trunk Profile UDP	Non Secure SIP Trunk Profile_UDP
	Secure Profile	TLS Profile
	SfBVideoInterop SecurityProfile	SFB-VideoInterop
Add Nev	/ Select All Clear All Delete Selected	

- Provide the desired Name and Description.
- Choose Non Secure from Device Security Mode.
- No security features except image authentication apply. A TCP or UDP connection opens to Unified Communications Manager.
  From Incoming Transport Type, select TCP+UDP.
- When Device Security Mode is Non Secure, TCP+UDP specifies the transport type.
- Select Outgoing Transport Type as TCP.
- Click Save.

System	ced Features ▼ Device ▼ Application ▼ User Management ▼ Bulk Adr
SIP Trunk Security Profile Configuration	
🔚 Save 🗙 Delete 🗋 Copy 資 Reset 🧷	Apply Config 🕂 Add New
┌ Status	
i Status: Ready	
SIP Trunk Security Profile Information	
Name*	Non Secure SIP Trunk Profile- aish
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure 🗸
Incoming Transport Type*	TCP+UDP 💙
Outgoing Transport Type	TCP 🗸
Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	customers.interopdomain.com
Incoming Port*	5060
Enable Application level authorization	

# **Configure SIP Profiles**

A SIP profile comprises the set of SIP attributes that are associated with SIP trunks and SIP endpoints. SIP profiles include information such as name, description, timing, retry, call pickup URI, and so on. The profiles contain some standard entries that you cannot delete or change.

• From Cisco Unified CM Administration, navigate to Device > Device Settings > SIP Profile.

• Click Add New.

System 👻	Call Routing 👻	Media Resources 👻 Advanced Features 👻	Device 👻	Application 👻	User Managemer	nt 👻 Bulk Administration 👻	Help 👻
Find and	List SIP Profil	es					
Add N	lew Select	All 🔛 Clear All 🙀 Delete Selected					
-Status-							
(i) 10 m	ecords found						
CTD Due	(il. (il. 10	-(10)					
SIP Pro	me (1-10	or 10)					
Find SIP P	rofile where Na	me 🗸 begins with 🖌		Find C	lear Filter 🛛 💠		
		Name *					Description
	SIP Profile					SIP Profile	
	Secure SIP F	Profile				Secure_SIP_Profile	
	<u>SfBVideoInter</u>	op SIPProfile					
	Standard SIP	Profile				Default SIP Profile	
	Standard SIP	Profile - Pooja				Default SIP Profile - Pooj	a
	Standard SIP	Profile -aish				Default SIP Profile	
	Standard SIP	Profile For Cisco VCS				Default SIP Profile For Cis	co Video Communication Server
	Standard SIP	Profile For TelePresence Conferencing				Default SIP Profile For Cis	co TelePresence Conferencing
	Standard SIP	Profile For TelePresence Endpoint				Default SIP Profile For Cis	co TelePresence Endpoint
	Standard SIP	Profile for Mobile Device				Default SIP Profile for Mo	bile Device
Add Nev	v Select All	Clear All Delete Selected					

- Enter a name to identify the SIP profile.Provide description to identify the purpose of the SIP profile.

System - Call Routing - Media Resources -	- Advanced Features 👻 🛛	Device 🔻 Application 👻 U	Jser Management 👻	Bulk Administration 👻 🛛 H	lelp 👻		
SIP Profile Configuration							
🕞 Save 🗙 Delete 🗋 Copy 資 Res	set 🥖 Apply Config 🔓	Add New					
SIP Profile Information							
Name*	Standard SIP Profile -aish	1					
Description	Default SIP Profile						
Default MTP Telephony Event Payload Type*	101						
Early Offer for G.Clear Calls*	Disabled		~				
User-Agent and Server header information*	Send Unified CM Version	Information as User-Agent	~				
Version in User Agent and Server Header*	Major And Minor		~				
Dial String Interpretation*	Phone number consists o	f characters 0-9, *, #, and	~				
Confidential Access Level Headers*	Disabled		~				
Redirect by Application							
Disable Early Media on 180							
Outgoing T.38 INVITE include audio mline	2						
Offer valid IP and Send/Receive mode on	ly for T.38 Fax Relay						
Use Fully Qualified Domain Name in SIP I	Requests						
Assured Services SIP conformance							
Enable External QoS**							
⊂SDP Information							
SDP Session-level Bandwidth Modifier for E	arly Offer and Re-invites*	TIAS and AS		~			
SDP Transparency Profile		Pass all unknown SDP attr	ributes	~			
Accent Audio Codec Preferences in Receive	Arcent Audio Order Preferences in Received Offer*						

- From Early Offer support for voice and video calls drop-down, choose Mandatory (insert MTP if needed).
- ٠ Enable SIP OPTIONS Ping.
  - SIP OPTIONS are requests to the configured destination address on the SIP trunk.
- Click Save.

Reroute Incoming Request to new Trunk based on * Never   Resource Priority Namespace List   Never   Resource Priority Namespace List   Never   Resource Priority Namespace List   Never   SIP Rel1XX Options*  Disabled   V  Disabled  V  Calling Line Identification Presentation*  Default  V  Session Refresh Method*  Invite  V  Early Offer support for voice and video calls*  Mandatory (insert MTP if needed)  V  Early Offer support for voice and video calls*  Mandatory (insert MTP if needed)  V  Early Offer support for voice and video calls*  Mandatory (insert MTP if needed)  V  Early Offer support for voice and video calls  Mandatory (insert MTP if needed)  V  Early Offer support for voice and Number  Reject Anonymous Incoming Calls  Send ILS Learned Destination Route String  Connect Inbound Call before Playing Queuing Announcement  SIP OPTIONS Ping  Ping Interval for Un-service Trunks (seconds)*  Ping Interval for Out-of-service Trunks (seconds)*  Ping Interval for Out-of-service Trunks (seconds)*  Ping Retry Timer (milliseconds)*  Sup Output Ping Retry Count*  Sup Output Support  Sup Output Support  Sup Output Support  Sup Output Support  S	-Trunk Specific Configuration					
Resource Priority Namespace List       < None >       ✓         SIP Rel1XX Options*       Disabled       ✓         Video Call Traffic Class*       Mixed       ✓         Calling Line Identification Presentation*       Default       ✓         Calling Line Identification Presentation*       Default       ✓         Session Refresh Method*       Invite       ✓         Early Offer support for voice and video calls*       Mandatory (insert MTP if needed)       ✓         Early Offer support for voice and video calls       Mandatory (insert MTP if needed)       ✓         Early Offer support for voice and video calls       Mandatory (insert MTP if needed)       ✓         Enable ANAT	Reroute Incoming Request to new Trunk based on ${}^{m{\star}}$	Never	~	)		
SIP Rel1XX Options* Disabled   Video Call Traffic Class* Mixed   Calling Line Identification Presentation* Default   Calling Line Identification Presentation* Default   Calling Line Identification Presentation* Default   Session Refresh Method* Invite   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Calling Line Identification Presentation Name and Number Invite   Calling External Presentation Name and Number Reject Anonymous Incoming Calls   Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String   Connect Inbound Call before Playing Queuing Announcement     SIP OPTIONS Ping   Image Interval for In-service and Partially In-service Trunks with Service Type "None (Default)"   Ping Interval for Out-of-service Trunks (seconds)*   Ping Retry Timer (milliseconds)*   Stop   Ping Retry Timer (milliseconds)*   Stop	Resource Priority Namespace List	< None > V				
Video Call Traffic Class* Mixed   Calling Line Identification Presentation* Default   Calling Line Identification Presentation* Default   Session Refresh Method* Invite   Session Refresh Method* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer support for voice and video calls* Mandatory (insert MTP if needed)   Early Offer Support Early Offer support for voice and video calls*   Enable ANAT Send LS canned Number   Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String   Connect Inbound Call before Playing Queuing Announcement StP OPTIONS Ping   Image: Step OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"   Ping Interval for Ont-of-service Trunks (seconds)* 60   Ping Interval for Out-of-service Trunks (seconds)* 120   Ping Retry Timer (milliseconds)* 500   Ping Retry Count* 500	SIP Rel1XX Options*	Disabled 🗸				
Calling Line Identification Presentation * Default  Session Refresh Method * Invite  Calling Conserved Frage and Video calls * Invite  Calling Conserved Frage and Video calls * Invite  Calling Conserved Frage and Video calls * Invite  Calling Conserved Frage and Number  Reject Anonymous Incoming Calls  Send ILS Learned Destination Route String  Connect Inbound Call before Playing Queuing Announcement  SIP OPTIONS Ping  Calling Interval for In-service and Partially In-service Trunks (seconds) * 60 Ping Interval for Out-of-service Trunks (seconds) * 120 Ping Retry Timer (milliseconds) * 500 Ping Retry Count * 5	Video Call Traffic Class*	Mixed	~	]		
Session Refresh Method * Invite   Early Offer support for voice and video calls * Mandatory (insert MTP if needed)  Early Offer support for voice and video calls * Mandatory (insert MTP if needed)  Enable ANAT  Deliver Conference Bridge Identifier Enable External Presentation Name and Number Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String Connect Inbound Call before Playing Queuing Announcement  SIP OPTIONS Ping  Finable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)" Ping Interval for In-service and Partially In-service Trunks (seconds)* 60 Ping Interval for Out-of-service Trunks (seconds)* 120 Ping Retry Timer (milliseconds)* 500 Ping Retry Count* 5	Calling Line Identification Presentation*	Default	~			
Early Offer support for voice and video calls * Mandatory (insert MTP if needed) Enable ANAT Deliver Conference Bridge Identifier Enable External Presentation Name and Number Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String Connect Inbound Call before Playing Queuing Announcement SIP OPTIONS Ping Enable OPTIONS Ping Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)" Ping Interval for In-service and Partially In-service Trunks (seconds)* Ping Interval for Out-of-service Trunks (seconds)* Ping Retry Timer (milliseconds)* Sign Retry Count*	Session Refresh Method*	Invite	~			
<ul> <li>□ Enable ANAT</li> <li>□ Deliver Conference Bridge Identifier</li> <li>□ Enable External Presentation Name and Number</li> <li>□ Reject Anonymous Incoming Calls</li> <li>□ Reject Anonymous Outgoing Calls</li> <li>□ Send ILS Learned Destination Route String</li> <li>□ Connect Inbound Call before Playing Queuing Announcement</li> <li>SIP OPTIONS Ping</li> <li>■ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"</li> <li>Ping Interval for In-service and Partially In-service Trunks (seconds)*</li> <li>Ping Retry Timer (milliseconds)*</li> <li>■ 500</li> </ul>	Early Offer support for voice and video calls $\!\!\!\!*$	Mandatory (insert MTP if needed)				
Deliver Conference Bridge Identifier Enable External Presentation Name and Number Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String Connect Inbound Call before Playing Queuing Announcement SIP OPTIONS Ping Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)" Ping Interval for In-service and Partially In-service Trunks (seconds)* 60 Ping Interval for Out-of-service Trunks (seconds)* 120 900 900 900 900 900 900 900	Enable ANAT					
	Deliver Conference Bridge Identifier					
□ Reject Anonymous Incoming Calls         □ Reject Anonymous Outgoing Calls         □ Send ILS Learned Destination Route String         □ Connect Inbound Call before Playing Queuing Announcement         SIP OPTIONS Ping         Image: Inable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"         Ping Interval for In-service and Partially In-service Trunks (seconds)*         Ping Interval for Out-of-service Trunks (seconds)*         Ping Retry Timer (milliseconds)*         S00         Ping Retry Count*	Enable External Presentation Name and Number					
□ Reject Anonymous Outgoing Calls         □ Send ILS Learned Destination Route String         □ Connect Inbound Call before Playing Queuing Announcement         SIP OPTIONS Ping         □ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"         Ping Interval for In-service and Partially In-service Trunks (seconds)*         0         Ping Interval for Out-of-service Trunks (seconds)*         120         Ping Retry Timer (milliseconds)*         500	Reject Anonymous Incoming Calls					
□ Send ILS Learned Destination Route String         □ Connect Inbound Call before Playing Queuing Announcement         SIP OPTIONS Ping         Image: Connect Inbound Call before Playing Queuing Announcement         Image: Connect Inbound Call	Reject Anonymous Outgoing Calls					
Connect Inbound Call before Playing Queuing Announcement      SIP OPTIONS Ping      Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"      Ping Interval for In-service and Partially In-service Trunks (seconds)*     ing Interval for Out-of-service Trunks (seconds)*     ing Retry Timer (milliseconds)*      ping Retry Count*	□ Send ILS Learned Destination Route String					
SIP OPTIONS Ping Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)" Ping Interval for In-service and Partially In-service Trunks (seconds)* Ping Interval for Out-of-service Trunks (seconds)* Ping Retry Timer (milliseconds)* Ping Retry Count* 5	Connect Inbound Call before Playing Queuing An	nouncement				
Image: Content of the service is a content of the servi	- SIP OPTIONS Ping					
Ping Interval for In-service and Partially In-service Trunks (seconds)*       60         Ping Interval for Out-of-service Trunks (seconds)*       120         Ping Retry Timer (milliseconds)*       500         Ping Retry Count*       6	Enable OPTIONS Ping to monitor destination st	atus for Trunks with	Service Type "None (Default)"			
Ping Interval for Out-of-service Trunks (seconds)*     120       Ping Retry Timer (milliseconds)*     500       Ping Retry Count*     5	Ping Interval for In-service and Partially In-service	Trunks (seconds)*	60			
Ping Retry Timer (milliseconds)* 500 Ping Retry Count* 5	Ping Interval for Out-of-service Trunks (seconds)*		120			
Ping Retry Count*	Ping Retry Timer (milliseconds)*		500			
	Ping Retry Count*		6			

# **Configure Normalization Script**

SIP trunks can connect to a variety of endpoints, including PBXs, gateways, and service providers. Each of these endpoints implements the SIP protocol a bit differently, causing a unique set of interoperability issues. To normalize messages per trunk, Cisco Unified Communications Manager all ows you to add or update scripts to the system and then associate them with one or more SIP trunks.

- From Cisco Unified CM Administration, choose Device > Device Settings > SIP Normalization Script
- Click Add New.

System	✓ Call Routing ✓ Media Resources ✓ Advanced	Features 👻 Device 👻	Application +	User Management 👻	Bulk Administration 👻	Help 👻	
Find a	nd List SIP Normalization Scripts						
- <b></b> A	dd New 🔛 Select All 🔛 Clear All 💥 Delete	Selected					
Statu	- Status-						
SIP	Normalization Script (1 - 9 of 9)						
Find S	IP Normalization Script where Name 💙 beg	ns with 🗙		Find Clear Filte	r 🕂 📼		
	Name <sup>▲</sup>					Description	
	HCS-PCV-PAI-passthrough	Cisco HCS platform int	egration with E	nterprise IMS			
	aish	modify diversion to his	tory info				
	att-header-passthrough	Provides passthrough (	of header x-att-	loop			
	cisco-meeting-server-interop	Provides interoperabilit	ty between Unif	ied Communication M	lanager (UCM) and Cis	sco Meeting Server	
	cisco-telepresence-conductor-interop	Provides interoperabilit	ty for endpoints	registered to the Tel	ePresence Conductor		
	cisco-telepresence-mcu-ts-direct-interop	Provides interoperabilit	ty between Unif	ied Communications	Manager (UCM) and C	isco TelePresence MCL	
	diversion-counter	Provide capability to ac	djust the divers	ion counter			
	refer-passthrough	Remove Unified CM fro	m the call due	to a blind transfer be	tween SIP trunks		
	<u>vcs-interop</u>	Provides interoperabilit	ty for endpoints	registered to the Vid	leo Communications S	erver (VCS)	
Add	New Select All Clear All Delete Selected						

• Provide the desired Name and Description.

• Add the script under content to convert diversion header to history info.

SIP Normalization Script Configuration					
🔚 Save 🗙 Delete 🎱 Reset 🖧 Add New 🔤 Import File					
i Status: Ready					
SIP Normalization Script Info					
Name*	modify_diversion_to_history_info				
Description	modify diversion to history info	]			
Content*	<pre>M = {} function M.outbound_INVITE(msg) if msg:getHeader("Diversion") then msg:convertDiversionToHI() msg:removeHeader("Diversion") end end return M</pre>				
Script Execution Error Recovery Action*	Message Rollback Only				
System Resource Error Recovery Action*	Disable Script				
Memory Threshold*	50	kilobytes			
Lua Instruction Threshold*	1000	instructions			

# **Trunk Configuration**

Use a trunk device to configure a logical route to a SIP network.

- From Cisco Unified CM Administration, choose Device > Trunk.
- Click Add New.

ahaha cisco	Cisco Unified CM	Administration		
System 👻	Call Routing 👻 Media Resources	✓ Advanced Features ▼ Device	e 🔻 Application 👻 User Management	Bulk Administration
Find and L	.ist Trunks			
🕂 Add Ne	ew			
Trunks				
Find Trunks	s where Device Name	✓ begins with ✓ Select ite	Find Clear Filte	r the contraction of the contrac
			No active query. Please enter y	our search criteria using the options above.
Add New	]			

- From the Trunk Type drop-down list, choose SIP Trunk.
  Choose SIP from Device Protocol drop-down.
  From Trunk Service Type, select the default value (None).

- Click Next.

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions						
System 👻	all Routing 👻 Media Resources 👻 Advanced Features 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻						
Trunk Cor	guration						
Next							
Status Statu	-Status Status: Ready						
Trunk In	mation						
Trunk Typ	SIP Trunk						
Device Pro	col* SIP 🗸						
Trunk Ser	e Type* None(Default)						
Next							

- Enter a unique identifier for the trunk.
- Enter a descriptive name for the trunk.
- Choose the Default Device Pool.

System      Call Routing      Media Resources      Advanced Features	Device
Trunk Configuration	
🔚 Save 🗙 Delete 省 Reset 🕂 Add New	
Trunk Service Type	None(Default)
Device Name*	trunkToDT
Description	trunk to DT
Device Pool*	Default 🗸
Common Device Configuration	< None >
Call Classification *	Use System Default
Media Resource Group List	< None >
Location*	Hub_None V
AAR Group	< None > V
Tunneled Protocol*	None 🗸
QSIG Variant*	No Changes 🗸
ASN.1 ROSE OID Encoding*	No Changes 🗸
Packet Capture Mode*	None 🗸
Packet Capture Duration	0
Media Termination Point Required	
✓ Retry Video Call as Audio	
Path Replacement Support	
Transmit UTF-8 for Calling Party Name	
Transmit UTF-8 Names in QSIG APDU	
Unattended Port	

- Provide the destination address.

  - The Destination Address represents the remote SIP peer with which this trunk will communicate.
     SIP trunks only accept incoming requests from the configured Destination Address and the specified incoming port that is specified in the SIP Trunk Security Profile that is associated with this trunk.
- Choose the SIP Trunk Security Profile created to apply to the SIP trunk.
  Select the SIP Profile created from the list.
- Choose the Normalization Script created previously from the list.
- Click Save.

Destination Destination Address is an SRV					
Destination Ad	dress	Destination	Address IPv6	Destination Port	
1**				5060	
MTP Preferred Originating Codec*	711ulaw	~			
BLF Presence Group*	Standard Presence group	~			
SIP Trunk Security Profile*	Non Secure SIP Trunk Pro	ofile-aish 🗸 🗸			
Rerouting Calling Search Space	< None >	~			
Out-Of-Dialog Refer Calling Search Space	< None >	~			
SUBSCRIBE Calling Search Space	< None >	~			
SIP Profile*	Standard SIP Profile -aish	ו 🗸	View Details		
DTMF Signaling Method*	No Preference	*			
Normalization Script					
Normalization Script aish		×			
Enable Trace					
Parameter Nat	ne	Paramete	er Value		
1 Diversion					
Recording Information					
None					
O This trunk connects to a recording-e	nabled gateway				

• Reset, Restart and Close the window. Refresh the SIP trunk page and wait until the Server status changes from Unknown to Full Service.

Device Reset
Reset 🚱 Restart
Status: Ready
Reset Information
Selected Device: trunkToDT (trunk to DT; SIP Trunk) If a device is not registered with Cisco Unified Communications Manager, you cannot reset or restart it. If a device is registered, to restart a device without shutting it down, click the <b>Restart</b> button. To shut down a device and bring it back up, click the <b>Reset</b> button. To return to the previous window without resetting/restarting the device, click <b>Close</b> .
<b>Note:</b> Resetting a gateway/trunk/media devices <b>drops</b> any calls in progress that are using that gateway/trunk/media devices. Restarting a gateway/media devices tries to preserve the calls in progress that are using that gateway/media devices, if possible. Other devices wait until calls are complete before restarting or resetting. Resetting/restarting a H323 device does not physically reset/restart the hardware; it only reinitializes the configuration loaded by Cisco Unified Communications Manager.
Reset Restart Close
(i) Note

Resetting/restarting a SIP device does not physically reset/restart the hardware; it only reinitializes the configuration that is loaded by Cisco Unified Communications Manager.

For SIP trunks, Restart and Reset behave the same way, so all active calls will disconnect when either choice is pressed.

# **Configure Call Routing**

A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that route calls 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.

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

System 👻	Call Routing 👻	Media Resources 🔻	Advanced Features 👻	Device 👻	Application 👻	User Management 👻	Bulk Administration 👻	Help 👻
Find and	List Route Pa	tterns						
Add N	Add New Elect All 🔛 Clear All 💥 Delete Selected							
-Status —								
(i) 18 records found								
Route Patterns (1 - 18 of 18)								
Find Route	e Patterns wher	e Pattern	<b>∨</b> b	egins with	•	Fin	d Clear Filter 🕂	

- Enter the route pattern, including numbers and wildcards (do not use spaces); for example, for NANP, enter 9.@ for typical local access or 8XXX for a typical private network numbering plan. Valid characters include the uppercase characters A, B, C, and D and \+, which represents the international escape character +.
- Configure the Route Pattern as shown below.
- Choose SIP Trunk created from the gateway or route list drop-down to add the route pattern.

System 👻 Call Routing 👻 Media Resources 👻	Advanced Features - Device - Application	✓ User Managemen	t 👻 Bulk Administration 👻	Help 👻
Route Pattern Configuration				
🔚 Save 🗙 Delete 🗋 Copy 🕂 Add N	ew			
_ Status				
i Status: Ready				
Pattern Definition				
Route Pattern*	9XXXXXXXXXX			
Route Partition	< None >	~		
Description			]	
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*	trunkToDT	~	(Edit)	
Route Option	Route this pattern			
	O Block this pattern No Error	~		
Call Classification* OffNet	~			
External Call Control Profile < None >	~			
□ Allow Device Override ✔ Provide Outside [	Dial Tone 🗌 Allow Overlap Sending 🗍 Urgen	t Priority		

# **Configure End Users**

The End User Configuration window allows you to add, search, display, and maintain information about Unified Communications Manager end users. End users can control phones after you associate a phone in the End User Configuration window.

- In Cisco Unified CM Administration, use the User Management > End User menu path to configure end users.
- Click Add New.

cisco	Cisco Ul For Cisco Un	nified CM A	dministration					
System 👻	Call Routing 👻	Media Resources	<ul> <li>Advanced Features</li> </ul>	Device 👻	Application 👻	User Management 👻	Bulk Administration 👻	Help 👻
Find and I	List Users							
🕂 Add N	lew							
User								
Find User	where First nan	ne	✓ begins with ✓			Find Clear Filter	+ <b>-</b>	
					No active (	query. Please enter your	search criteria using the o	ptions above.
Add New	•							

- Enter the unique end user identification name.
  Enter alphanumeric or special characters for the end user password and confirm the same.
  Enter numeric characters for the end user PIN and confirm.
  Enter the end user last name.

cisco	Cisco UI For Cisco Un	nified CM Ad	ministr ns Solutions	ation					
System 👻	Call Routing 👻	Media Resources 👻	Advanced F	eatures 👻	Device 👻	Application	<ul> <li>User Management</li> </ul>	→ Bulk Administration          →	Help
End User	Configuration								
Save	X Delete	Add New							
-Status-									
i Statu	us: Ready								
- User Info	ormation ——								
User Stat	us	Enabled Local User							
User ID*		496					]		
Password		•••••	•••••	•••••	•••		Edit Credential		
Confirm P	assword	•••••	•••••	•••••	•••		]		
Self-Servi	ice User ID						]		
PIN		•••••	•••••	•••••	•••		Edit Credential		
Confirm P	NIN	•••••	•••••	•••••	•••		]		
Last name	e*	aish1					]		
Middle na	me						]		
First nam	e						]		
Display na	ame						)		
Title							)		
Directory	URI						)		
Telephone	e Number						)		

# Phone Setup

- In Cisco Unified Communications Manager Administration, use the **Device > Phone** menu path to configure phones.
  Click Add New.

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions
System 👻	Call Routing • Media Resources • Advanced Features • Device • Application • User Management • Bulk Administration • Help •
Find and	List Phones
👍 Add N	lew 🕂 Add New From Template 🏢 Select All 🔛 Clear All 🔆 Delete Selected 🎱 Reset Selected 🧷 Apply Config to Selected
-Status	
(i) 16 re	ecords found
Phone	(1 - 16 of 16)
Filone	(1-10010)
Find Phone	e where Device Name 🗸 begins with 🖌 Find Clear Filter 🔂 📼 Select item or enter search text 🗸

- From the Phone Type drop-down, choose Third-party SIP Device(Advanced) Endpoint.
- Click Next.

ahaha cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions					
System 👻	Call Routing 👻 Media Resources 👻 Advanced Features 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 He					
Add a Nev	v Phone					
Next	Next					
Status Statu	Status Status: Ready					
Add New Phone Information						
Start by s Phone Ty	electing the type of phone you wish to add, or <u>click here to add a new phone using a Universal Device Template.</u>					
Next						

- Enter the Media Access Control (MAC) address that identifies Cisco Unified IP Phones. Make sure that the value comprises 12 hexadecimal characters.
- Choose Default Device pool.
- A Device pool defines sets of common characteristics for devices, such as region, date/time group, and soft key template.
- Choose Third-party SIP Device(Advanced) from the phone button template drop-down.
  - The phone button template determines the configuration of buttons on a phone and identifies which feature (line, speed dial, and so on) is used for each button.
- Choose the user ID of the assigned phone user.

System - Call Routing - Media Resources - Advance	ed Features - Device - Application - U	Jser Management 👻 Bulk Administration 👻 Help 👻				
Phone Configuration						
🔚 Save 🗙 Delete 🕒 Copy 🎦 Reset 🧷 -	Apply Config 🚽 Add New					
- Association Phone Type						
Modify Button Items Product Type: Third-party SIP Device (Advanced)						
1 •m: Line [1] - 49 (no partition)	Device Protocol: SIP					
2 stat Line [2] - Add a new DN	⊂Real-time Device Status					
-785	Registration: Registered with Cisco	Unified Communications Manager cucm12				
3 Line (3) - Add a new DN	IPv4 Address: 1	-				
4 erns Line [4] - Add a new DN	Active Load ID: None					
5 The Line [5] - Add a new DN	Download Status: None					
6 erms Line [6] - Add a new DN	C Device Information					
	Davice is Active					
7 The Line (7) - Add a new DN						
8 Ine [8] - Add a new DN	MAC Address* ABCD123321A1 (SEPABCD123321A1)					
	Description	SEPABCD123321A1				
	Device Pool*	Default	View Details			
	Common Device Configuration	< None >	View Details			
	Phone Button Template*	Third-party SIP Device (Advanced)				
	Common Phone Profile*	Standard Common Phone Profile 🗸 🗸	View Details			
	Calling Search Space	< None > 🗸 🗸 🗸	J			
	AAR Calling Search Space	< None >	)			
	Media Resource Group List	< None > V	] []			
	Location*	Hub_None V	)			
	AAR Group	< None > V	] []			
	Device Mobility Mode*	Default 🗸 🗸	View Current Device Mobility Settings			
Owner 🖲 User 🖸 Anonymous (Public/Shared Space)						
	Owner User ID *	49	Acti			

- Choose the security profile Third-party AS-SIP Endpoint Standard SIP Non-Secure Profile to apply to the device.
- Choose the standard sip profile.
- Choose an end user that you want to associate with the phone for this setting that is used with digest authentication (SIP security).
  Click Save.

Phone Configuration						
🔚 Save 🗶 Delete 📄 Copy 🎦 Reset 🧷	Apply Config 🖧 Add New					
	🖬 Use Device Pool Calling	Party Transformation CSS (Caller ID For Calls From This Phone	e)			
	Remote Number					
	Calling Party Transformatio	n CSS   < None > V				
	Use Device Pool Calling	Party Transformation CSS (Device Mobility Related Information	n)			
	Protocol Specific Informa	tion				
	BLF Presence Group*	Standard Presence group 🗸				
	MTP Preferred Originating Co	odec* 711ulaw 🗸 🗸				
	Device Security Profile*	Third-party SIP Device Advanced - Standard SIP No 🗸				
	Rerouting Calling Search Spa	ace < None > V				
	SUBSCRIBE Calling Search S	pace   < None > V	10 B 1 I			
	Digest User		view Details			
	Madia Transination Deint (					
	O Media Termination Point P	vequired				
	Onattended Port     Date DTMS Reservice					
	Alley December 2019 Reception					
	Allow Presentation Shahi	ig using BrCP				
	Allow IX Applicable Media					
	MLPP and Confidential Ac	cess Level Information				
	MLPP Domain <	None > 🗸				
	Confidential Access Mode <	None > V				
	Confidential Access Level <	None > V				

• Click this link to add a remote destination to associate with this device. The Remote Destination Configuration window displays, which allows you to add a new remote destination to associate with this device.

System - Call Routing - Media Resources - Advance	ed Features 💌 Device 💌 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻	
Phone Configuration		Relate
🔚 Save 🗙 Delete 🗋 Copy 睯 Reset 🧷	Apply Config 🕂 Add New	
Status Status: Ready		
Association Modify Button Items	Phone Type Product Type: Third-party SIP Device (Advanced) Device Protocol: SIP Registration: Registered with Cisco Unified Communications Manager cucm12 IPv4 Address: Ad	

- Add the Directory number.
- Click Save.

System 👻 Call Routing 👻 I	Media Resources 👻 Advanced Features 👻	Device - Applicatio	on 👻 User Management 👻 Bulk Administration 👻 H
Directory Number Config	uration		
Save 🗙 Delete 🤮	Reset 🖉 Apply Config		
Status			
i Status: Ready			
Directory Number Inform	nation		
Directory Number*	49		Urgent Priority
Route Partition	< None >	~	
Description			]
Alerting Name			]
ASCII Alerting Name			<u>]</u>
External Call Control Profile	< None >	~	·
Associated Devices	SEPABCD123321A1		
			Edit Device
			Edit Line Appearance
		*	
	~~		
Dissociate Devices		-	
		-	

- Click Apply Config followed by the Reset button.
- Reset, Restart and Close the window.

#### **Device Association**

- Navigate back to User Management > End User.
- In the Device Information field, click Device Association. This will display all the available devices.
- Select the device created in the previous step and save.

System - Call Routing - Me	edia Resources 👻 Advanced Features 👻	Device - Application -	User Management 👻 Bulk Adr	ninistration 👻 Help 👻
End User Configuration				
🔚 Save 🗙 Delete 🕂	Add New			
L	· · · ·		-	
Device Information				
Controlled Devices Available Profiles	SEPABCD123321A1		Device Association Line Appearance Assoc	iation for Presence
	**		Ŧ	
CTI Controlled Device Profiles			*	

# Supplementary Services and Features Coverage

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

Sr. No.	Supplementary Services/ Features	Coverage
1	SIP Trunk Registration	$\checkmark$
2	Inbound Call-Mobile PSTN	✓

3	Outbound Call-Mobile PSTN	$\checkmark$
4	Inbound call-Landline PSTN	$\checkmark$
5	Outbound call-Landline PSTN	$\checkmark$
6	Basic Call With Different Codecs	✓
7	Voice Mail	✓
8	Call Forward	$\checkmark$
9	FAX using G711	✓
10	Call Hold and Resume Outbound	$\checkmark$
11	Call Hold and Resume Inbound	$\checkmark$
12	Anonymous Calls Outbound	✓
13	Session Timers	✓
14	FAX - transcoding	✓
15	Call Transfer (Blind)	✓
16	Call Transfer (Attended)	✓
17	Cancel Call	✓
18	Long Duration Calls	$\checkmark$

#### Legend

Supported	$\checkmark$
Not Supported	X

# Note

Observation - Any call to the PSTN mobile display the caller's number with the country code, whereas any call to the PSTN landline excludes the country code.

# Caveats

• NA

# Support

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

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

# References

For detailed information about Ribbon products and solutions, visit: https://ribboncommunications.com/products

# **Conclusion**

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

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

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

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