

# Ribbon SBC Edge 1K R8.0 Interop with Deutsche Telekom CompanyFlex SIP Trunks : Interoperability Guide



## Table of Contents

- [Interoperable Vendors](#)
- [Copyright](#)
- [Document Overview](#)
  - [About Ribbon SBC 1k](#)
  - [About Deutsche Telekom](#)
- [Scope](#)
- [Non-Goals](#)
- [Audience](#)
- [Prerequisites](#)
- [Product and Device Details](#)
- [Network Topology](#)
  - [SBC 1000 - Deutsche Telekom Deployment Topology](#)
  - [SBC 1000 - Deutsche Telekom Lab Topology](#)
  - [Signaling and Media Flow](#)
- [Installing SBC 1000/2000](#)
- [SBC 1000 Configuration with TCP](#)
  - [Accessing SBC 1000](#)
  - [License](#)
    - [View License](#)
  - [View Networking Interfaces](#)
  - [Configure Static Routes](#)
  - [SBC 1000 Configuration for Access End](#)
  - [SBC 1000 Configuration for Deutsche Telekom End](#)
    - [Media Profile](#)
    - [Sip Profile](#)
    - [Contact Registration Table](#)
    - [Remote Authorization Table](#)
    - [Sip Server Table](#)
    - [Message Manipulation](#)
    - [Signaling Group](#)
  - [Call Routing table](#)
    - [Creating an Entry to a Call Routing Table](#)
    - [Call Routing table for "From Local Registrar"](#)
    - [Call Routing for "From Telekom".](#)
- [Supplementary Services and Features Coverage](#)
- [Caveats](#)
- [Support](#)
- [References](#)
- [Conclusion](#)
- [Interoperable Vendors](#)
- [Copyright](#)
- [Document Overview](#)
  - [About Ribbon SBC 1k](#)
  - [About Deutsche Telekom](#)
- [Scope](#)
- [Non-Goals](#)
- [Audience](#)
- [Prerequisites](#)
- [Product and Device Details](#)
- [Network Topology](#)
  - [SBC 1000 - Deutsche Telekom Deployment Topology](#)
  - [SBC 1000 - Deutsche Telekom Lab Topology](#)
  - [Signaling and Media Flow](#)
- [Installing SBC 1000/2000](#)

- SBC 1000 Configuration with TCP
  - Accessing SBC 1000
  - License
    - View License
  - View Networking Interfaces
  - Configure Static Routes
  - SBC 1000 Configuration for Access End
  - SBC 1000 Configuration for Deutsche Telekom End
    - Media Profile
    - Sip Profile
    - Contact Registration Table
    - Remote Authorization Table
    - Sip Server Table
    - Message Manipulation
    - Signaling Group
  - Call Routing table
    - Creating an Entry to a Call Routing Table
    - Call Routing table for "From Local Registrar"
    - Call Routing for "From Telekom".
- Supplementary Services and Features Coverage
- Caveats
- Support
- References
- Conclusion

# Interoperable Vendors

---

Deutsche Telekom

## Copyright

---

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved. The compilation (meaning the collection, arrangement and assembly) of all content on this site is protected by U.S. and international copyright laws and treaty provisions and may not be used, copied, reproduced, modified, published, uploaded, posted, transmitted or distributed in any way, without prior written consent of Ribbon Communications Inc.

The trademarks, logos, service marks, trade names, and trade dress ("look and feel") on this website, including without limitation the RIBBON and RIBBON logo marks, are protected by applicable US and foreign trademark rights and other proprietary rights and are the property of Ribbon Communications Operating Company, Inc. or its affiliates. Any third-party trademarks, logos, service marks, trade names and trade dress may be the property of their respective owners. Any uses of the trademarks, logos, service marks, trade names, and trade dress without the prior written consent of Ribbon Communications Operating Company, Inc., its affiliates, or the third parties that own the proprietary rights, are expressly prohibited.

## Document Overview

---

This document depicts the configuration details for Ribbon SBC 1000 interworking & compliance against Deutsche Telekom CompanyFlex SIP Trunking solution.

### About Ribbon SBC 1k

The Ribbon Session Border Controller provides best-in class communications security. The SBC 1000 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 and other hardware as well as 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 1000 GUI is required. Understanding the basic concepts of TCP/UDP, IP /Routing, and SIP/RTP is also necessary to complete the configuration and any required troubleshooting.

## Prerequisites

---

The following aspects are required before proceeding with the interop:

- Ribbon SBC 1000/2000 or SWe Lite series
- SBC License
- 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 details, visit <https://hilfe.companyflex.de/de/einrichtung/einrichtung-sip-trunk>

## Product and Device Details

---

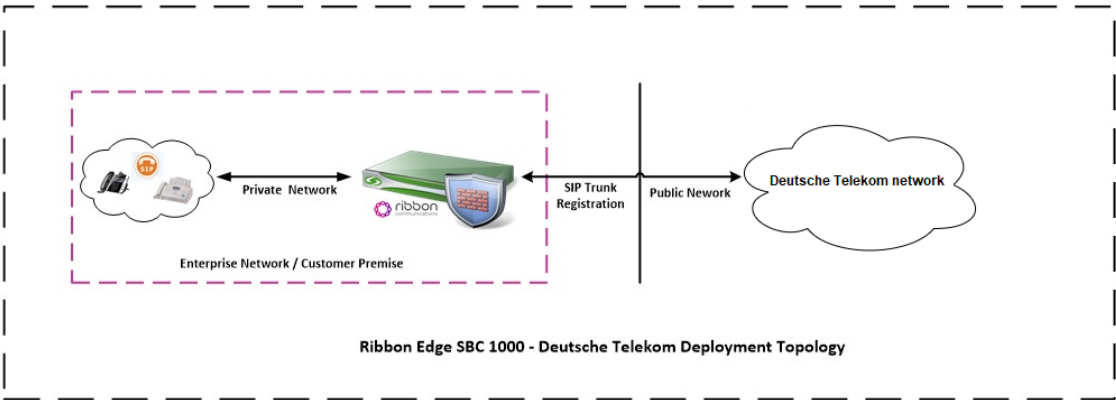
The configuration uses the following equipment and software:

Table 1: Requirements

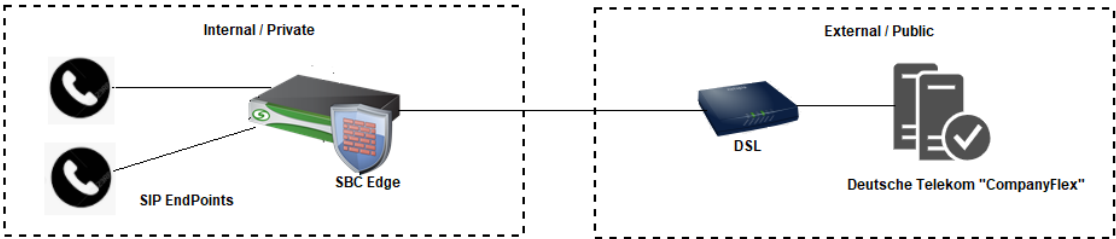
Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC 1000	8.0.1
Third-party Equipment	DSL Line	NA
Deutsche Telekom	Deutsche Telekom "CompanyFlex" SIP trunks	NA
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

## Network Topology

### SBC 1000 - Deutsche Telekom Deployment Topology

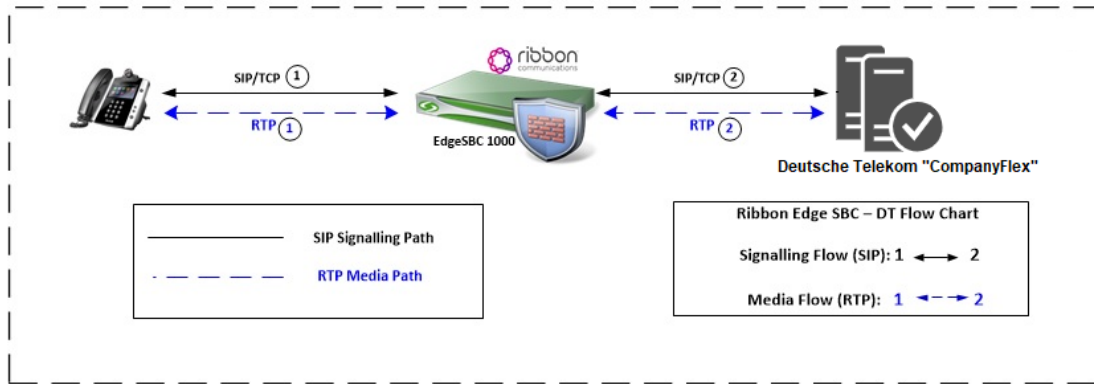


### SBC 1000 - Deutsche Telekom Lab Topology



SBC 1000 - Deutsche Telekom Lab Topology

## Signaling and Media Flow



## Installing SBC 1000/2000


Refer to the following document for installing the SBC 1000: <https://doc.rbbn.com/pages/viewpage.action?pageId=229474498>

## SBC 1000 Configuration with TCP

### Accessing SBC 1000

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 1000**

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

Password

© Copyright 2010-2018 Sonus Networks, Inc. (a Ribbon Communications Company). All Rights Reserved

## License

### View License

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

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

ribbon

Monitor Tasks Settings Diagnostics System

Application Solution Module

System

Node-Level Settings

QoS

DSPs

System Timing

System Companding Law

Licensing

Current Licenses

License Keys

Install New License

Software Management

Auth and Directory Services

Active Directory

RADIUS

Protocols

DNS

IP

Static Routes

Routing Table

Static ARP

Router Instances

Access Control Lists

Total 19 Feature License Rows

Feature	Licensed	Total Licenses	Available Licenses
SIP Calls	✓	100	100
SIP Registrations	✓	200	199
DSP Resources	✓	Unlimited	Unlimited
Forking	✓	Unlimited	Unlimited
SBA	✓	Unlimited	Unlimited
Active Directory	✓	Unlimited	Unlimited
Transcoding	✓	Unlimited	Unlimited
REST	✓	Unlimited	Unlimited
CAS	✓	Unlimited	Unlimited
CDR	✓	Unlimited	Unlimited
OSPF	✓	Unlimited	Unlimited
RIP	✓	Unlimited	Unlimited
IPsec	✓	Unlimited	Unlimited
RBA	✓	Unlimited	Unlimited

Welcome: ribbon | Logout | Device Name: 44033514470 Ribbon SBC 1000

For more details on Licenses, refer to [SBC 1000, SBC 2000 Licenses](#).

## View Networking Interfaces

The SBC 1000 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 1000 supports user-created VLAN logical sub-interfaces.

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

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

Search...

Expand All Collapse All Reload

Call Routing

Signaling Groups

Linked Signaling Groups

Node Interfaces

Ports

Logical Interfaces

Ethernet 1 IP

Ethernet 2 IP

Loopback 1

Loopback 2

Loopback 3

Loopback 4

Loopback 5

Logical Interfaces

Total 7 LogicalInterface Rows

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State
Ethernet 1 IP	10.10.10.10			Enabled
Ethernet 2 IP	192.168.1.1			Enabled
Loopback 1				Disabled
Loopback 2				Disabled
Loopback 3				Disabled
Loopback 4				Disabled
Loopback 5				Disabled



For the interop, this app note uses the same interface for Administrator and Ethernet1.

### 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 or you can assign a static IP as well.

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

- Call Routing
- Signaling Groups
- Linked Signaling Groups
- Node Interfaces**
  - Ports
  - Logical Interfaces**
    - Ethernet 1 IP**
    - Ethernet 2 IP
    - Loopback 1
    - Loopback 2
    - Loopback 3
    - Loopback 4
    - Loopback 5
- Bridge
  - Relay Config
- Application Solution Module
- System
  - Node-Level Settings
  - QoS
  - DSPs
  - System Timing
  - System Companding Law
- Licensing
- Software Management
- Auth and Directory Services
- Protocols
  - SIP
    - Local Registrars
    - Local / Pass-thru Auth Tables
    - SIP Profiles
    - SIP Server Tables

### Logical Interfaces

Total 7 LogicalInterface Rows

Interface Name	IPv4 Address
Ethernet 1 IP	10

Identification/Status

Interface Name

Ethernet 1 IP

I/F Index

29

Alias

Description

Admin State

Enabled

Networking

MAC Address

00:10

IP Addressing Mode

IPv4

IPv4 Information

ACL In

None

ACL Out

None

ACL Forward

None

IP Assign Method

Static

Primary Address

10

Primary Netmask

255.255.255.0

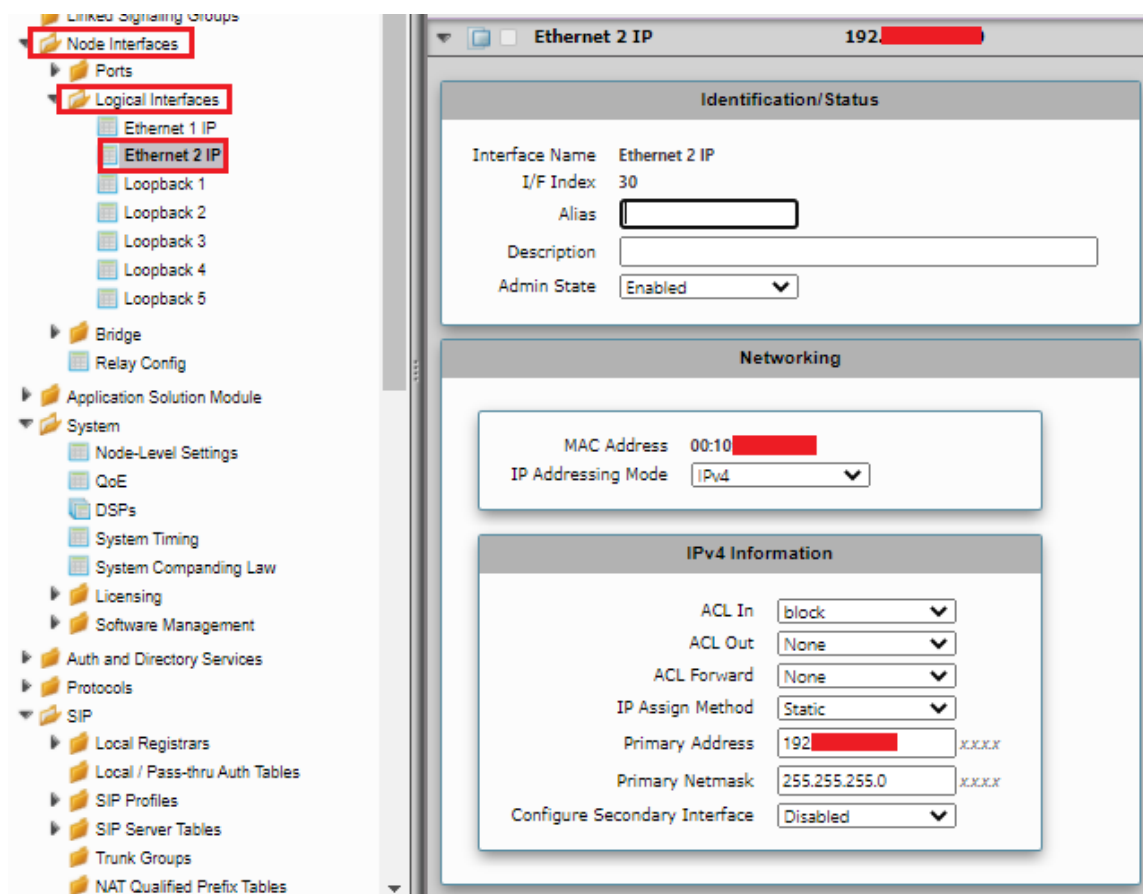
Configure Secondary Interface

Disabled

## Ethernet 2 IP

Configure this Ethernet 2 interface as follows as per the requirement .This interface will face the Deutsche Telekom interface.

Confidential and Proprietary. Copyright © 2020-2023 Ribbon Communications Operating Company, Inc. © 2020-2023 ECI Telecom Ltd.



#### Attention

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

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



Use Static IP address in the interface towards the Deutsche Telekom.

## 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.



Static IP Route Table						
Total 27 IP Route Rows						
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key	
1	0.0.0.0	0.0.0.0	10.0.1.1	1	1	
2	157.49.1.1	255.255.255.255	10.0.1.1	1	2	
3	157.49.1.1	255.255.255.255	10.0.1.1	1	3	
4	115.110.1.1	255.255.255.255	10.0.1.1	1	4	
5	115.110.1.1	255.255.255.255	10.0.1.1	1	5	
6	157.49.1.1	255.255.255.255	10.0.1.1	1	6	
7	157.49.1.1	255.255.255.255	10.0.1.1	1	7	

## SBC 1000 Configuration for Access End

Configure the Signaling profile, Route, Media profile, SIP profile, SIP registrar, etc. based on the requirement.

For assistance visit : <https://doc.rbbn.com/>

## SBC 1000 Configuration for Deutsche Telekom End

### 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.

Use default media profile with codec G.711.

**Media List View**

Total 3 Media List Rows

☐ Description

☐ Default Media List

Description: Default Media List

Media Profiles List:

- Default G711A
- Default G711u
- T.38 fax

Up Down Add/Edit Remove

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

DTLS-SRTP Profile: None

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

RTCP Mode: RTCP

Dead Call Detection: Disabled

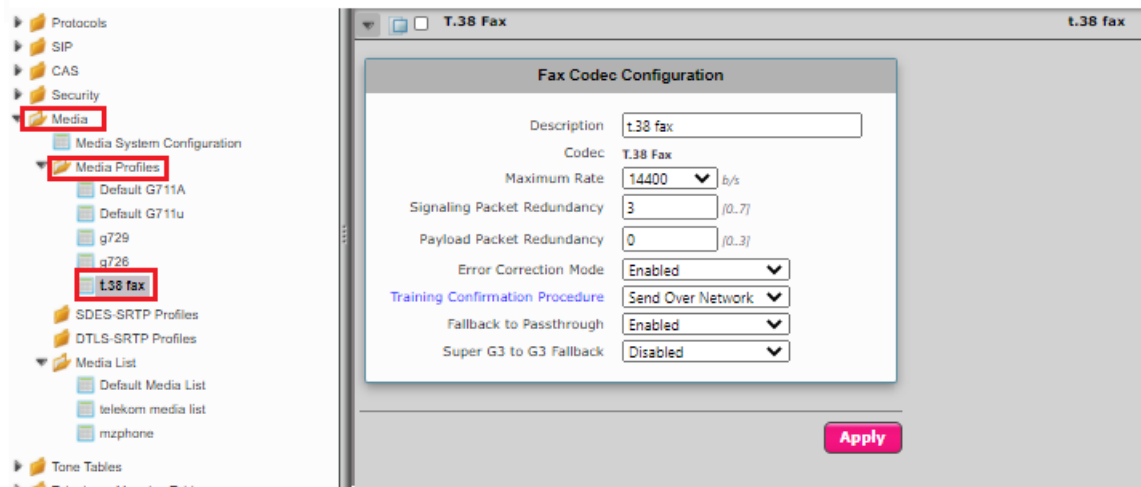
Silence Suppression: Enabled



Add T.38 in the Default Media list only if fax is involved.

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

Create a Media profile with T.38 codec.



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

## 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.

Search...

Expand All | Collapse All | Reload

Call Routing

Signaling Groups

Linked Signaling Groups

Node Interfaces

Application Solution Module

System

Auth and Directory Services

Protocols

**SIP**

Local Registrars

Local / Pass-thru Auth Tables

**SIP Profiles**

Default SIP Profile

**telekom sip profile**

SIP Server Tables

Trunk Groups

NAT Qualified Prefix Tables

Remote Authorization Tables

Contact Registrant Table

Message Manipulation

Node-Level SIP Settings

SIP Voice Quality Server

CAS

Security

Media

Tone Tables

Telephony Mapping Tables

SNMP/Alarms

Logging Configuration

Remote Log Servers

Log Profiles

Subsystems

Port Mirror

Emergency Services

### SIP Profile Entry: telekom sip profile

Description: telekom sip profile

Session Timer		MIME Payloads	
Session Timer	Enable	ELIN Identifier	LOC
Minimum Acceptable Timer	600	PIDF-LO Passthrough	Enable
Offered Session Timer	1800	Unknown Subtype Passthrough	Disable
Terminate On Refresh Failure	False		

Header Customization		Options Tags	
FQDN in From Header	Disable	100rel	Supported
FQDN in Contact Header	Disable	Path	Not Present
Send Assert Header	Trusted Only	Timer	Supported
SBC Edge Diagnostics Header	Enable	Update	Supported
Trusted Interface	Enable		
UA Header	Ribbon SBC Edge		
Calling Info Source	RFC Standard		
Diversion Header Selection	Last		
Record Route Header	RFC 3261 Standard		

Timers		SDP Customization	
Transport Timeout Timer	5000	Send Number of Audio Channels	True
Maximum Retransmissions	RFC Standard	Connection Info in Media Section	True
Redundancy Retry Timer	180000	Origin Field Username	SBC
<b>RFC Timers</b>		Session Name	VoipCall
Timer T1	500	Digit Transmission Preference	RFC 2833/Voice
Timer T2	4000	SDP Handling Preference	Legacy Audio/Fax
Timer T4	5000		
Timer D	32000		
Timer B	32000 ms		
Timer F	32000 ms		
Timer H	32000 ms (64*Timer T1)		
Timer J	4000		

## 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 "Type of address of record" as local.
- Provide the SIP Trunk number provided by Deutsche Telekom under the "Address of record URI".
- Provide 600 sec for Global Timer to Live and 120 sec 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".

Expand All | Collapse All | Reload

Call Routing  
Signaling Groups  
Linked Signaling Groups  
Node Interfaces  
Application Solution Module  
System  
Auth and Directory Services  
Protocols  
SIP  
Local Registrars  
Local / Pass-thru Auth Tables  
SIP Profiles  
SIP Server Tables  
Trunk Groups  
NAT Qualified Prefix Tables  
Remote Authorization Tables  
Contact Registrant Table  
telekom contact reg

Total 1 SIP Contact Registrant Entry Row

Address of Record

+4919929

Type of Address of Record: Local

Address of Record URI: +4919929 user

Global Time to Live (TTL): 600 secs [64..86400]

Failed Registration Retry Timer: 120 \* secs [30..86400]

SIP Contacts

Total 1 SIP User Contact Row

Contact URI Username	TTL (secs)	Priority (Q)
+4919929	Inherited	0

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

telekom contact reg

Total 1 SIP Contact Registrant Entry Row

Address of Record

+4919929

Display

Registration Status

Contact Registrant Registration Status - Google Chrome

Not secure | /cgi/phpUI/callTableEngine.php?parentID=1&filter=1&parentType=SIPRegistration&type=...

Contact Registrant Registration Status

April 19, 2021 17:13:09

Total 1 SIPRegistrationStatus Row

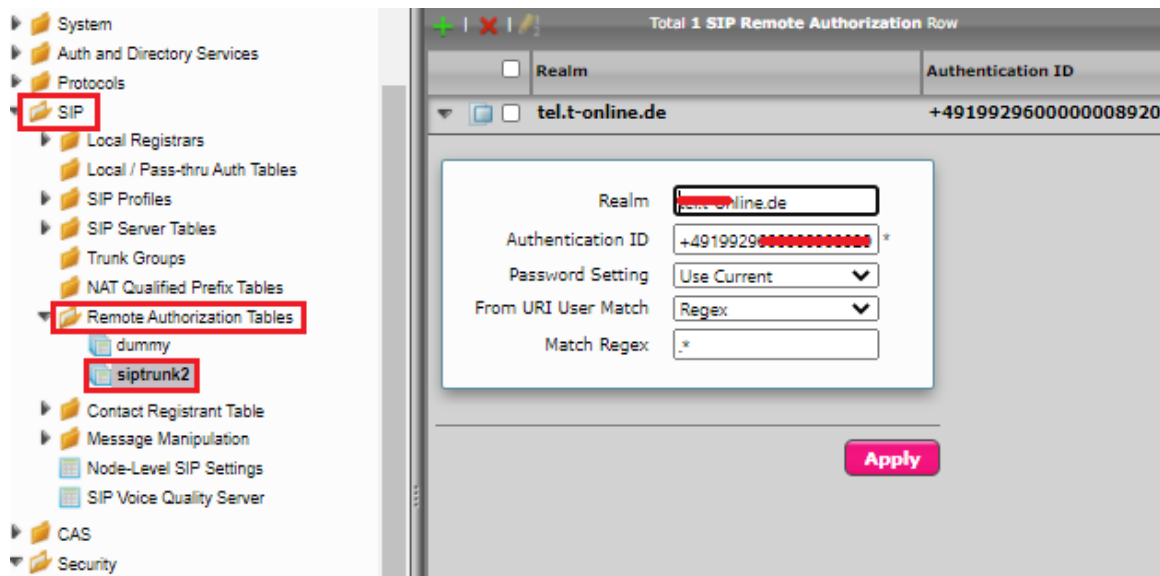
SIP Server	Signaling Group	Registration Status
Entry 100 (f-ecp-600.edns.t-iptel.d...)	(SIP) From/To telekom	Registered

## 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".



## 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.
- Use TCP 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.

**SRV Servers**

Server ID	FQDN/Domain Name	Protocol	Port	Time to Live	Priority	Weight
100	f-ecp-600.edns.t-ipn...	TCP	5060	3599	10	0
102	d-ecp-600.edns.t-ipn...	TCP	5060	3599	20	0
101	h2-ecp-600.edns.t-ip...	TCP	5060	3599	30	0

## Message Manipulation

The Message Manipulation feature work in concert to modify SIP messages. Below Message Manipulation are used to avoid registration and call failures.

### The SMM performs the following actions:

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

- To
- From
- Req-URI

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

Add new headers for all outbound INVITE messages.

- P-Early-Media
- Allow-Events

Add new header for all outbound REGISTER messages.

- Supported
- Allow

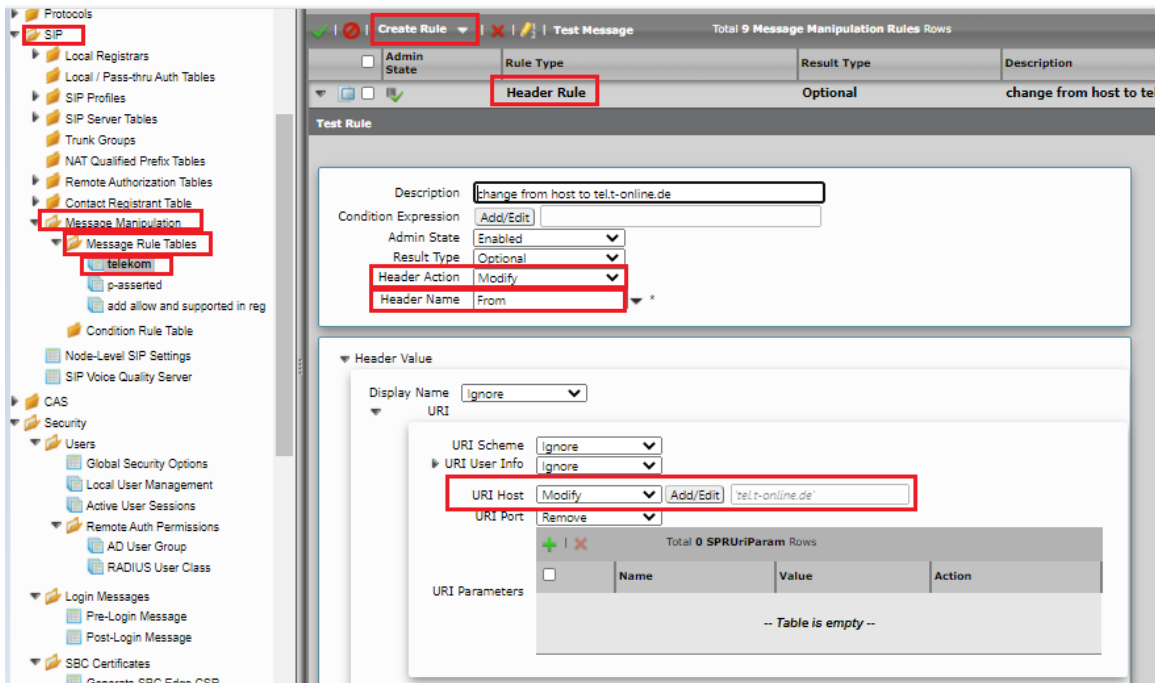
Select Settings > SIP > Message Manipulation > Message Rule Table

Click the Create Message Rule Table(+) icon.



#### Message Manipulation - 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 on add/edit and provide the fqdn that will replace the URI host in from header.



Under "Telekom" Repeat the same for To header.

**telekom**

✓ | ✗ | ⚙ | Test Message Total 9 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	change from host to tel.t-online.de
<input type="checkbox"/>	<b>Header Rule</b>	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

Result Type: Optional

Header Action: **Modify**

Header Name: **To** \*

Header Value

Display Name: Ignore

URI

URI Scheme: Ignore

URI User Info: Ignore

**URI Host: Modify Add/Edit 'tel.t-online.de'**

URI Port: Remove

URI Parameters

Name	Value	Action
-- Table is empty --		

Under "Telekom" repeat the same for request URI.

**telekom**

✓ | ✗ | ⚙ | Test Message Total 9 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	change from host to tel.t-online.de
<input type="checkbox"/>	Header Rule	Optional	change to host to tel.t-online.de
<input type="checkbox"/>	<b>Request Line Rule</b>	Optional	requestline

**Test Rule**

Description: requestline

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Request Line Value

Method: Ignore

URI

URI Scheme: Ignore

URI User Info: Ignore

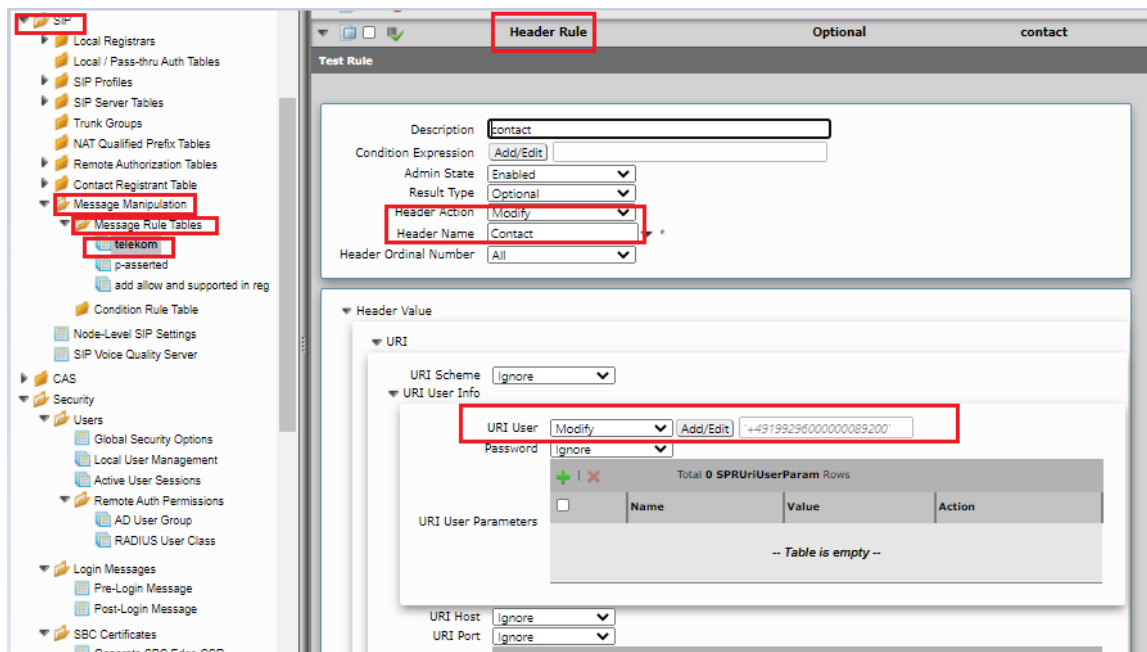
**URI Host: Modify Add/Edit 'tel.t-online.de'**

URI Port: Ignore

URI Parameters

Name	Value	Action
-- Table is empty --		

Create message manipulation under "telekom" so that the contact header has SIP trunk number in URI user for all the sip messages .



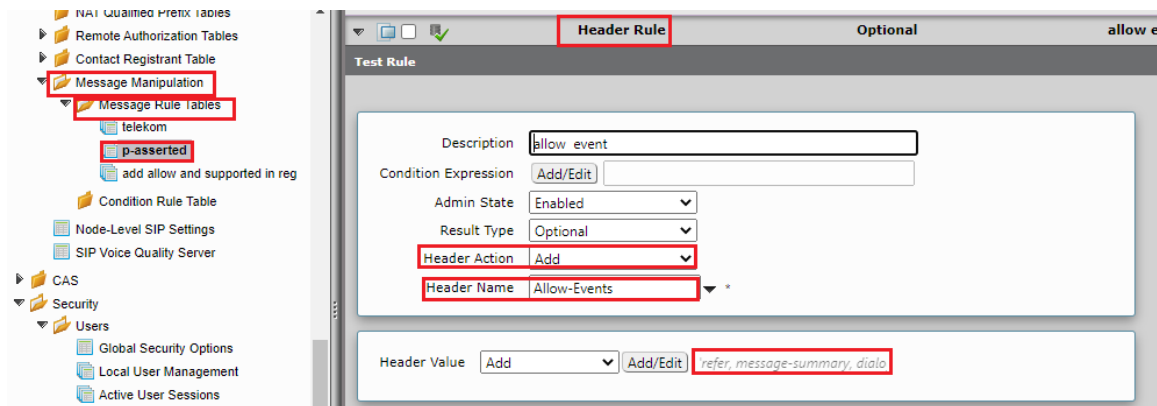
### Message Manipulation - Add Allow-Events in INVITE

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

Choose "INVITE" message under Applicable Messages.

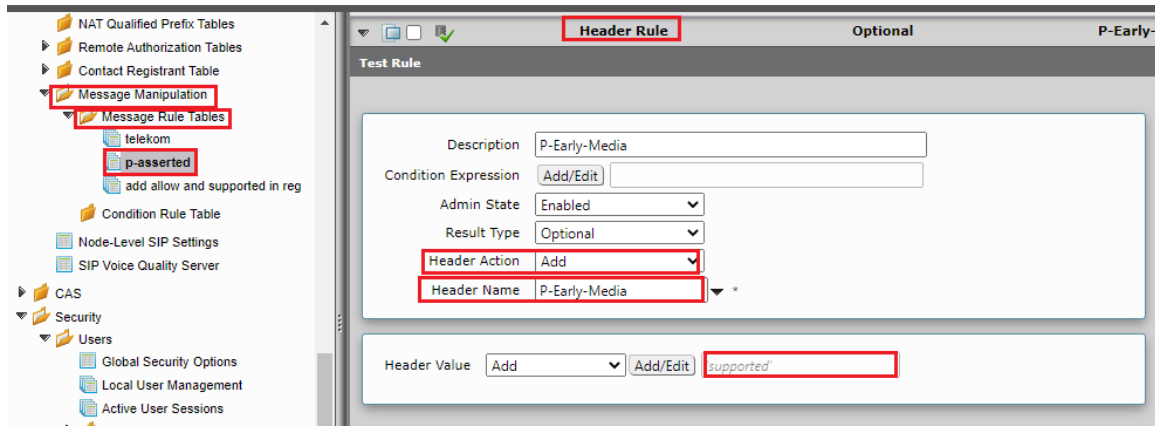
- From the Create Rule drop-down-box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Add" and header name as "Allow-Events".
- Under header value give "Add" and click on add/edit and provide 'refer, message-summary, dialog'.
- Click on **Apply**.



### Message Manipulation - Add P-Early-Media in INVITE

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





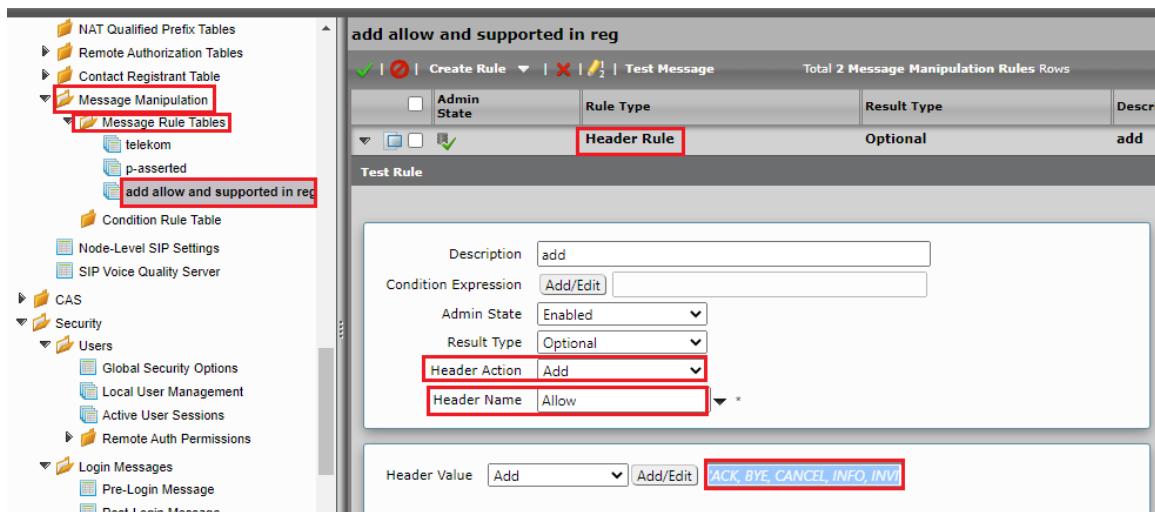
### Message Manipulation - Add Allow in REGISTER

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

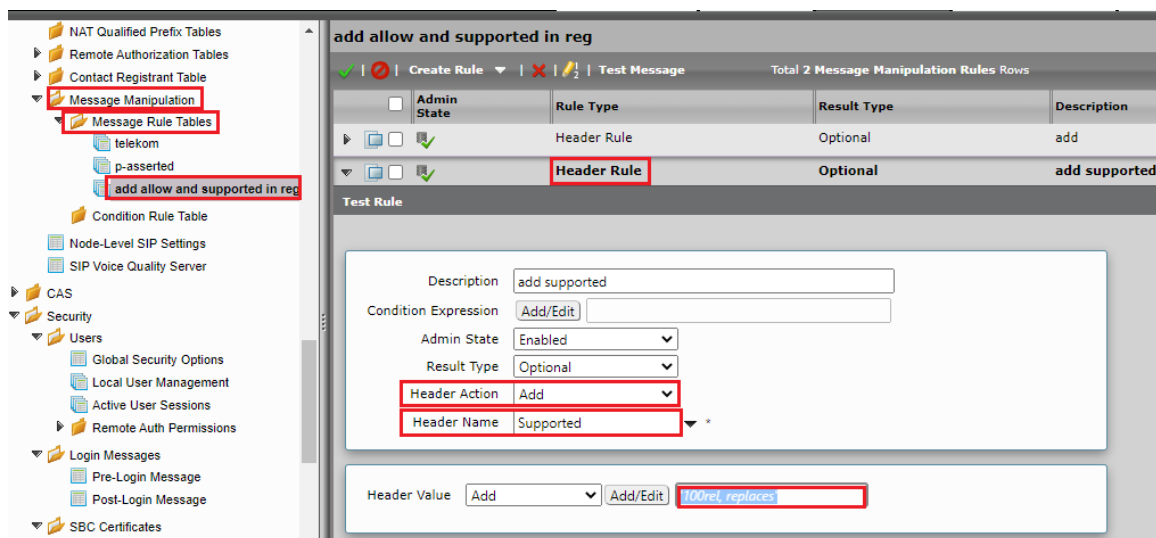
Choose "REGISTER" message under Applicable Messages.

- 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 on **Apply**.



### Message Manipulation - 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 on **Apply**.



## 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.

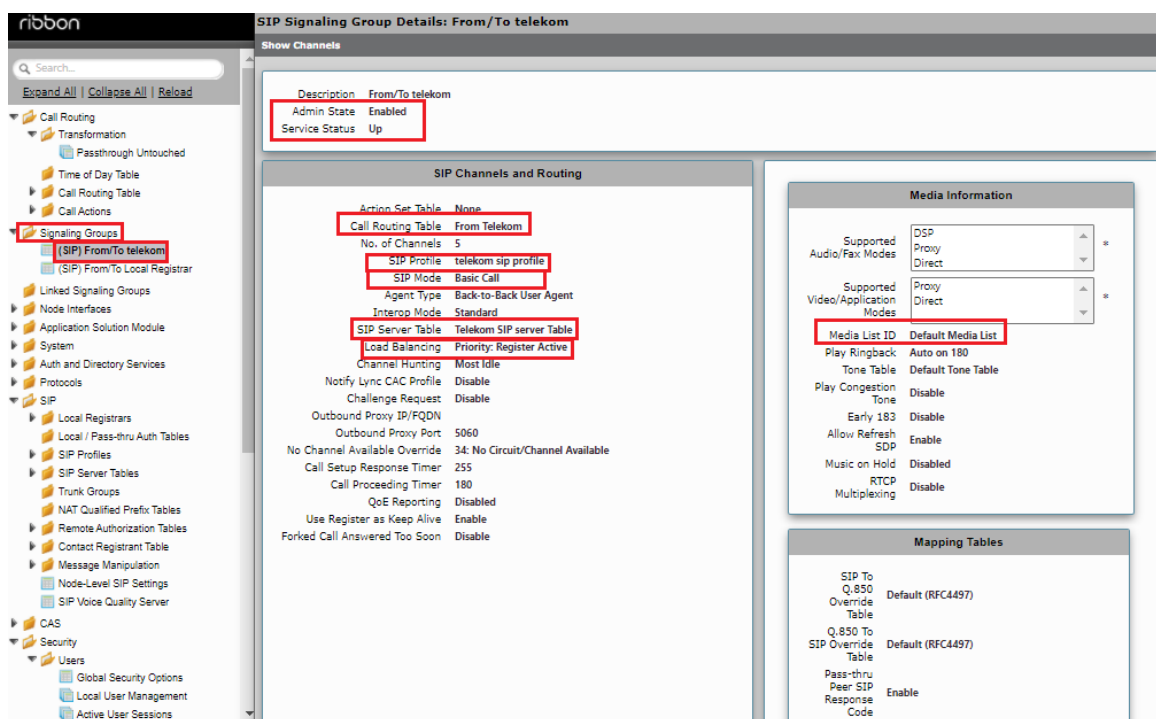
### Select Settings > Signaling Groups

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



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 "default media list".
- Choose SIP Server Table as "Telekom Sip Server Table".





Because a NAT is used in the test environment, add the external public IP of the NAT box under static NAT outbound of the Sig Group that is facing towards the Deutsche Telekom server.

- 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 TCP.
- Add message manipulation under the outbound section that we created earlier to add a domain instead of IP, for a successful call.
  - Enable Message Manipulation.
  - Click **Add/Edit** on Outbound Message Manipulation.
  - This displays a drop-down list of available message tables. Select an entry and click **Apply**.

The screenshot shows the SBC configuration interface with the following sections:

- SIP IP Details:**
  - Signaling/Media Private IP: 192.168.1.1
  - Ethernet 2 IP: 192.168.1.1
  - Signaling DSCP: 40
  - NAT Traversal: Enabled
  - ICE Support: Disabled
  - Static NAT - Outbound:
    - Outbound NAT Traversal: Static NAT
    - NAT Public IP (Signaling/Media): 80.1.1.1
  - Static NAT - Inbound: Disabled
- Listen Ports:**

Port	Protocol	TLS Profile ID
5070	TCP	N/A
- Federated IP/FQDN:**

IP/FQDN	Netmask/Prefix
192.168.1.1	255.255.255.255
- Message Manipulation:** Enabled
- Outbound Message Manipulation:**
  - Message Table List: telekom, p-asserted, add allow and supported in reg



Configure NAT box so that the external public IP doesn't change frequently. Incase if there is a change, update the Static NAT outbound section with the new allocated public IP address.

## 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:

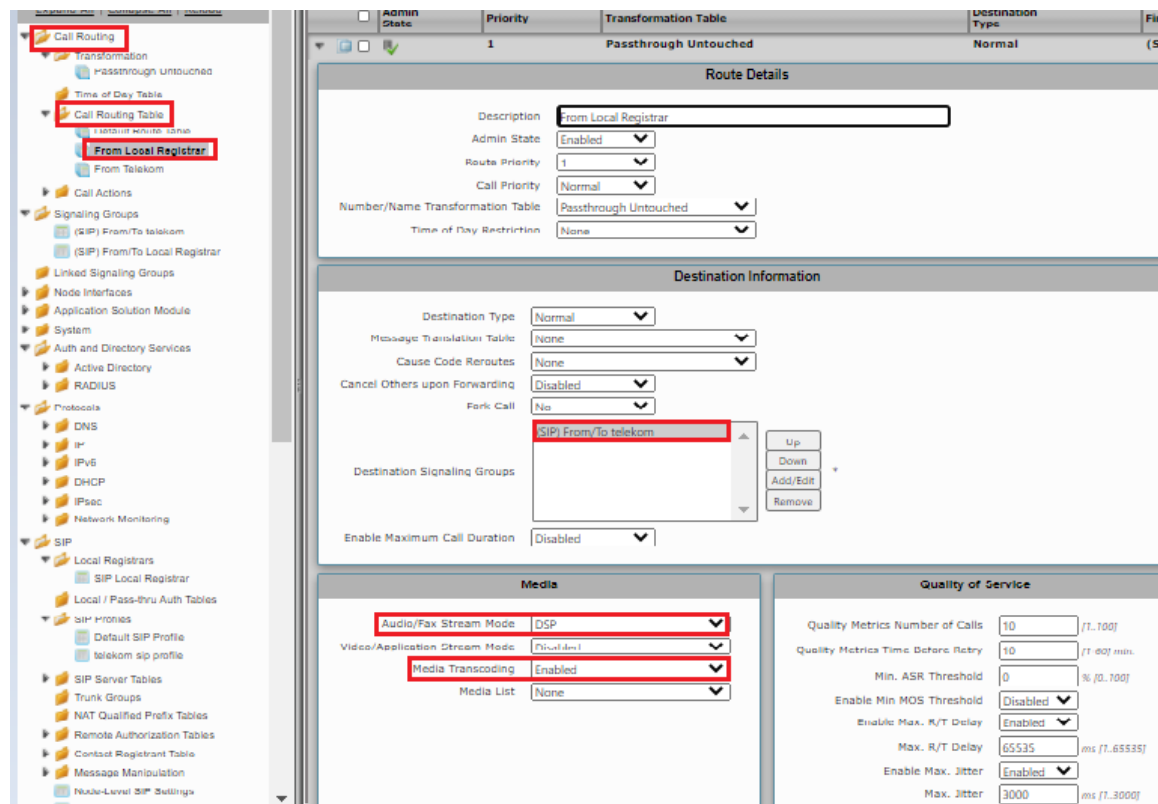
DSP (default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support DSP are attempted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

#### Media Transcoding:

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

3. Click **Apply**.

## Call Routing table for "From Local Registrar"



The screenshot shows the configuration interface for a Call Routing entry named "From Local Registrar". The left sidebar contains a tree view with categories like Call Routing, Signaling Groups, and SIP. The main area is divided into several sections:

- Route Details:** Includes fields for Description (From Local Registrar), Admin State (Enabled), Route Priority (1), Call Priority (Normal), Number/Name Transformation Table (Passthrough Untouched), and Time of Day Restriction (None).
- Destination Information:** Includes Destination Type (Normal), Message Translation Table (None), Cause Code Reroutes (None), Cancel Others upon Forwarding (Disabled), Fork Call (No), and Destination Signaling Groups (SIP From/To telekom).
- Media:** Includes Audio/Fax Stream Mode (DSP), Video/Application Stream Mode (Passthrough), Media Transcoding (Enabled), and Media List (None).
- Quality of Service:** Includes Quality Metrics Number of Calls (10), Quality Metrics Time Before Retry (10), Min. ASR Threshold (0), Enable Min MOS Threshold (Disabled), Enable Max. R/T Delay (Enabled), Max. R/T Delay (65535), Enable Max. Jitter (Enabled), and Max. Jitter (3000).

## Call Routing for "From Telekom".

The screenshot displays the configuration interface for a SIP route. The left sidebar shows a tree view of configuration options, with 'From Telekom' highlighted. The main area is divided into four sections:

- Route Details:** Includes fields for Description (From Telekom), Admin State (Enabled), Route Priority (1), Call Priority (Normal), Number/Name Transformation Table (Passthrough Untouched), and Time of Day Restriction (None).
- Destination Information:** Includes Destination Type (Registrar Table), Message Translation Table (None), Cause Code Reroutes (None), Cancel Others upon Forwarding (Disabled), Fork Call (No), Destination Signaling Groups (SIP From/To Local Registrar), and Enable Maximum Call Duration (Disabled).
- Media:** Includes Audio/Fax Stream Mode (DSP), Video/Application Stream Mode (Disabled), Media Transcoding (Enabled), and Media List (None).
- Quality of Service:** Includes Quality Metrics Number of Calls (10), Quality Metrics Time Before Retry (10), Min. ASR Threshold (0), Enable Min MOS Threshold (Disabled), Enable Max. R/T Delay (Enabled), Max. R/T Delay (65535), Enable Max. Jitter (Enabled), and Max. Jitter (3000).

## 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	✓
2	Inbound Call-Mobile PSTN	✓
3	Outbound Call-Mobile PSTN	✓
4	Inbound call-Landline PSTN	✓
5	Outbound call-Landline PSTN	✓
6	Basic Call With Different Codecs	✓
7	Voice Mail	✓
8	FAX using T.38	✓
9	Call Forward	✗
10	FAX using G711 Fallback	✓
11	Call Hold and Resume Outbound	✓
12	Call Hold and Resume Inbound	✓
13	Anonymous Calls Outbound	✓
14	Session Timers	✓

15	FAX - transcoding	✓
16	Call Transfer (Blind)	✗
17	Call Transfer (Attended)	✗
18	486 Busy	✓
19	487 Request Terminated	✓
20	Long Duration Calls	✓

#### Legend

Tested	✓
Not Tested	✗

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

## Caveats

- NA

## Support

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

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

## References

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

## Conclusion

This Interoperability Guide describe the configuration steps required for **Ribbon SBC 1000 / 2000** 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.

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved.



## Table of Contents

- [Interoperable Vendors](#)
- [Copyright](#)
- [Document Overview](#)
  - [About Ribbon SBC 1k](#)
  - [About Deutsche Telekom](#)
- [Scope](#)
- [Non-Goals](#)
- [Audience](#)
- [Prerequisites](#)
- [Product and Device Details](#)
- [Network Topology](#)
  - [SBC 1000 - Deutsche Telekom Deployment Topology](#)
  - [SBC 1000 - Deutsche Telekom Lab Topology](#)
  - [Signaling and Media Flow](#)
- [Installing SBC 1000/2000](#)
- [SBC 1000 Configuration with TCP](#)
  - [Accessing SBC 1000](#)
  - [License](#)
    - [View License](#)
  - [View Networking Interfaces](#)
  - [Configure Static Routes](#)
  - [SBC 1000 Configuration for Access End](#)
  - [SBC 1000 Configuration for Deutsche Telekom End](#)
    - [Media Profile](#)
    - [Sip Profile](#)
    - [Contact Registration Table](#)
    - [Remote Authorization Table](#)
    - [Sip Server Table](#)
    - [Message Manipulation](#)
    - [Signaling Group](#)
  - [Call Routing table](#)
    - [Creating an Entry to a Call Routing Table](#)
    - [Call Routing table for "From Local Registrar"](#)
    - [Call Routing for "From Telekom"](#)
- [Supplementary Services and Features Coverage](#)
- [Caveats](#)
- [Support](#)
- [References](#)
- [Conclusion](#)
- [Interoperable Vendors](#)
- [Copyright](#)
- [Document Overview](#)
  - [About Ribbon SBC 1k](#)
  - [About Deutsche Telekom](#)
- [Scope](#)
- [Non-Goals](#)
- [Audience](#)
- [Prerequisites](#)
- [Product and Device Details](#)
- [Network Topology](#)
  - [SBC 1000 - Deutsche Telekom Deployment Topology](#)
  - [SBC 1000 - Deutsche Telekom Lab Topology](#)
  - [Signaling and Media Flow](#)
- [Installing SBC 1000/2000](#)

- SBC 1000 Configuration with TCP
  - Accessing SBC 1000
  - License
    - View License
  - View Networking Interfaces
  - Configure Static Routes
  - SBC 1000 Configuration for Access End
  - SBC 1000 Configuration for Deutsche Telekom End
    - Media Profile
    - Sip Profile
    - Contact Registration Table
    - Remote Authorization Table
    - Sip Server Table
    - Message Manipulation
    - Signaling Group
  - Call Routing table
    - Creating an Entry to a Call Routing Table
    - Call Routing table for "From Local Registrar"
    - Call Routing for "From Telekom".
- Supplementary Services and Features Coverage
- Caveats
- Support
- References
- Conclusion



# Interoperable Vendors

---

Deutsche Telekom

## Copyright

---

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved. The compilation (meaning the collection, arrangement and assembly) of all content on this site is protected by U.S. and international copyright laws and treaty provisions and may not be used, copied, reproduced, modified, published, uploaded, posted, transmitted or distributed in any way, without prior written consent of Ribbon Communications Inc.

The trademarks, logos, service marks, trade names, and trade dress ("look and feel") on this website, including without limitation the RIBBON and RIBBON logo marks, are protected by applicable US and foreign trademark rights and other proprietary rights and are the property of Ribbon Communications Operating Company, Inc. or its affiliates. Any third-party trademarks, logos, service marks, trade names and trade dress may be the property of their respective owners. Any uses of the trademarks, logos, service marks, trade names, and trade dress without the prior written consent of Ribbon Communications Operating Company, Inc., its affiliates, or the third parties that own the proprietary rights, are expressly prohibited.

## Document Overview

---

This document depicts the configuration details for Ribbon SBC 1000 interworking & compliance against Deutsche Telekom CompanyFlex SIP Trunking solution.

### About Ribbon SBC 1k

The Ribbon Session Border Controller provides best-in class communications security. The SBC 1000 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 and other hardware as well as 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 1000 GUI is required. Understanding the basic concepts of TCP/UDP, IP /Routing, and SIP/RTP is also necessary to complete the configuration and any required troubleshooting.

## Prerequisites

---

The following aspects are required before proceeding with the interop:

- Ribbon SBC 1000/2000 or SWe Lite series
- SBC License
- 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 details, visit <https://hilfe.companyflex.de/de/einrichtung/einrichtung-sip-trunk>

## Product and Device Details

---

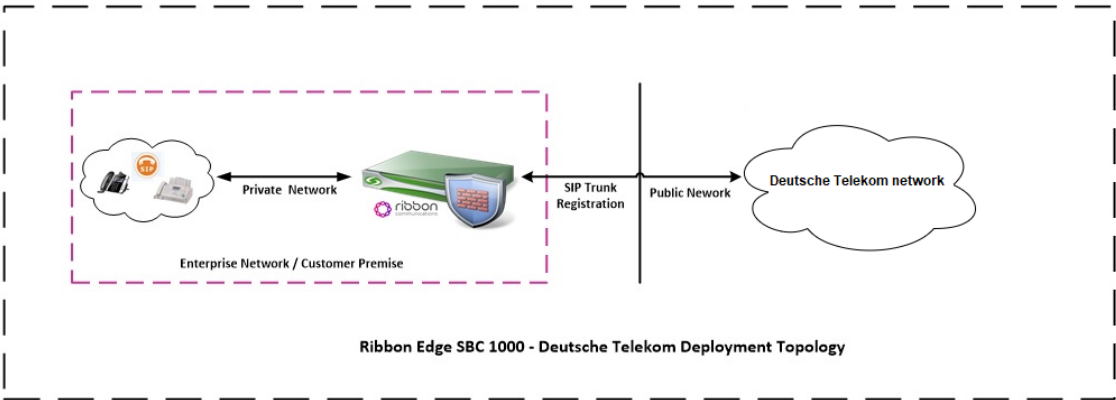
The configuration uses the following equipment and software:

Table 2: Requirements

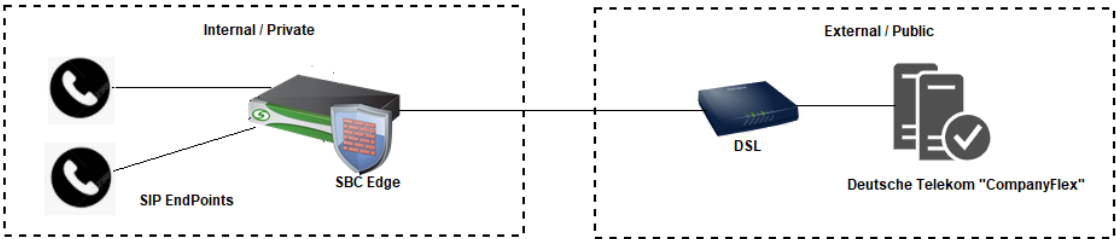
Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC 1000	8.0.1
Third-party Equipment	DSL Line	NA
Deutsche Telekom	Deutsche Telekom "CompanyFlex" SIP trunks	NA
Administration and Debugging Tools	Wireshark	3.2.7
	LX Tool	2.1.0.6

## Network Topology

### SBC 1000 - Deutsche Telekom Deployment Topology

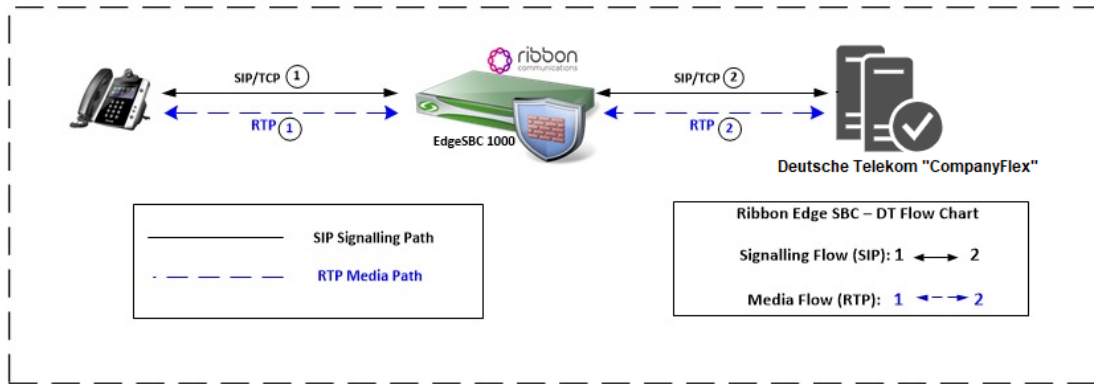


### SBC 1000 - Deutsche Telekom Lab Topology



SBC 1000 - Deutsche Telekom Lab Topology

## Signaling and Media Flow



## Installing SBC 1000/2000


Refer to the following document for installing the SBC 1000: <https://doc.rbbn.com/pages/viewpage.action?pageId=229474498>

## SBC 1000 Configuration with TCP

### Accessing SBC 1000

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 1000**

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

Password

© Copyright 2010-2018 Sonus Networks, Inc. (a Ribbon Communications Company). All Rights Reserved

## License

### View License

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

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

Application Solution Module

System

Node-Level Settings

QoS

DSPs

System Timing

System Companding Law

Licensing

Current Licenses

License Keys

Install New License

Software Management

Auth and Directory Services

Active Directory

RADIUS

Protocols

DNS

IP

Static Routes

Routing Table

Static ARP

Router Instances

Access Control Lists

Monitor Tasks Settings Diagnostics System

Welcome: ribbon | Logout | Device Name: 44033514470 Ribbon SBC 1000

Total 19 Feature License Rows

Feature	Licensed	Total Licenses	Available Licenses
SIP Calls	100	100	100
SIP Registrations	200	199	199
DSP Resources	Unlimited	Unlimited	Unlimited
Forking	Unlimited	Unlimited	Unlimited
SBA	Unlimited	Unlimited	Unlimited
Active Directory	Unlimited	Unlimited	Unlimited
Transcoding	Unlimited	Unlimited	Unlimited
REST	Unlimited	Unlimited	Unlimited
CAS	Unlimited	Unlimited	Unlimited
CDR	Unlimited	Unlimited	Unlimited
OSPF	Unlimited	Unlimited	Unlimited
RIP	Unlimited	Unlimited	Unlimited
IPsec	Unlimited	Unlimited	Unlimited
RBA	Unlimited	Unlimited	Unlimited

For more details on Licenses, refer to [SBC 1000, SBC 2000 Licenses](#).

## View Networking Interfaces

The SBC 1000 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 1000 supports user-created VLAN logical sub-interfaces.

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

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

Search...

Expand All Collapse All Reload

Call Routing

Signaling Groups

Linked Signaling Groups

Node Interfaces

Ports

Logical Interfaces

Ethernet 1 IP

Ethernet 2 IP

Loopback 1

Loopback 2

Loopback 3

Loopback 4

Loopback 5

Logical Interfaces

Total 7 LogicalInterface Rows

Interface Name	IPv4 Address	IPv6 Address	Description	Admin State
Ethernet 1 IP	10			Enabled
Ethernet 2 IP	192			Enabled
Loopback 1				Disabled
Loopback 2				Disabled
Loopback 3				Disabled
Loopback 4				Disabled
Loopback 5				Disabled

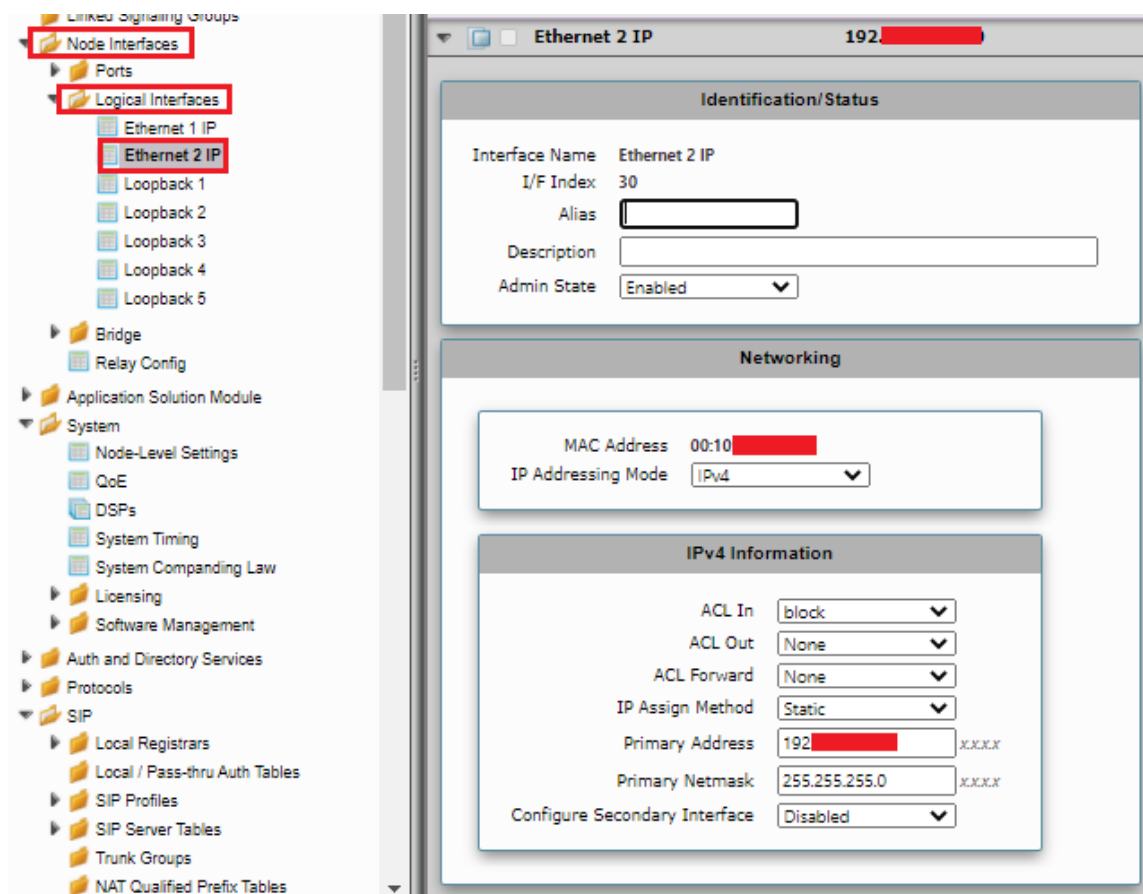


For the interop, this app note uses the same interface for Administrator and Ethernet1.

## 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 or you can assign a static IP as well.





#### Attention

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

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



Use Static IP address in the interface towards the Deutsche Telekom.

## 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.

Static IP Route Table						
Total 27 IP Route Rows						
Row ID	Destination IP	Mask	Gateway	Administrative Distance	Primary Key	
1	0.0.0.0	0.0.0.0	10.0.0.1	1	1	
2	157.49.0.0	255.255.255.255	10.0.0.1	1	2	
3	157.49.0.0	255.255.255.255	10.0.0.1	1	3	
4	115.110.0.0	255.255.255.255	10.0.0.1	1	4	
5	115.110.0.0	255.255.255.255	10.0.0.1	1	5	
6	157.49.0.0	255.255.255.255	10.0.0.1	1	6	
7	157.49.0.0	255.255.255.255	10.0.0.1	1	7	

## SBC 1000 Configuration for Access End

Configure the Signaling profile, Route, Media profile, SIP profile, SIP registrar, etc. based on the requirement.

For assistance visit : <https://doc.rbbn.com/>

## SBC 1000 Configuration for Deutsche Telekom End

### 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.

Use default media profile with codec G.711.

**Media List View**

Total 3 Media List Rows

☐ Description

☐ Default Media List

Description: Default Media List

Media Profiles List:

- Default G711A
- Default G711u
- T.38 fax

Up Down Add/Edit Remove

SDP-SRTP Profile: None

DTLS-SRTP Profile: None

Media DSCP: 46

RTCP Mode: RTCP

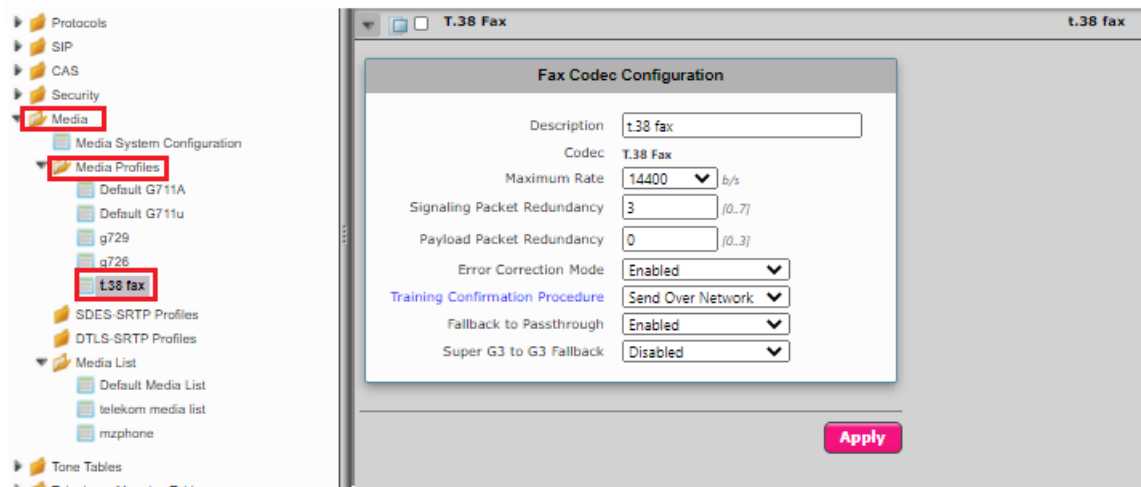
Dead Call Detection: Disabled

Silence Suppression: Enabled

**i** Add T.38 in the Default Media list only if fax is involved.

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

Create a Media profile with T.38 codec.



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

## 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.

**SIP Profile Entry: telekom sip profile**

Description: telekom sip profile

Session Timer		MIME Payloads	
Session Timer	Enable	ELIN Identifier	LOC
Minimum Acceptable Timer	600	PIDF-LO Passthrough	Enable
Offered Session Timer	1800	Unknown Subtype Passthrough	Disable
Terminate On Refresh Failure	False		

Header Customization		Options Tags	
FQDN in From Header	Disable	100rel	Supported
FQDN in Contact Header	Disable	Path	Not Present
Send Assert Header	Trusted Only	Timer	Supported
SBC Edge Diagnostics Header	Enable	Update	Supported
Trusted Interface	Enable		
UA Header	Ribbon SBC Edge		
Calling Info Source	RFC Standard		
Diversion Header Selection	Last		
Record Route Header	RFC 3261 Standard		

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



## 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 "Type of address of record" as local.
- Provide the SIP Trunk number provided by Deutsche Telekom under the "Address of record URI".
- Provide 600 sec for Global Timer to Live and 120 sec 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".

Expand All | Collapse All | Reload

Call Routing  
Signaling Groups  
Linked Signaling Groups  
Node Interfaces  
Application Solution Module  
System  
Auth and Directory Services  
Protocols  
SIP  
Local Registrars  
Local / Pass-thru Auth Tables  
SIP Profiles  
SIP Server Tables  
Trunk Groups  
NAT Qualified Prefix Tables  
Remote Authorization Tables  
Contact Registrant Table  
telekom contact reg

Total 1 SIP Contact Registrant Entry Row

Address of Record

+4919929

Type of Address of Record: Local

Address of Record URI: +4919929 user

Global Time to Live (TTL): 600 secs [64..86400]

Failed Registration Retry Timer: 120 \* secs [30..86400]

SIP Contacts

Total 1 SIP User Contact Row

Contact URI Username	TTL (secs)	Priority (Q)
+4919929	Inherited	0

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

telekom contact reg

Total 1 SIP Contact Registrant Entry Row

Address of Record

+4919929

Display

Registration Status

Contact Registrant Registration Status - Google Chrome

Not secure | http://localhost/cgi/phpUI/callTableEngine.php?parentID=1&filter=1&parentType=SIPRegistration&type=...

Contact Registrant Registration Status

April 19, 2021 17:13:09

Total 1 SIPRegistrationStatus Row

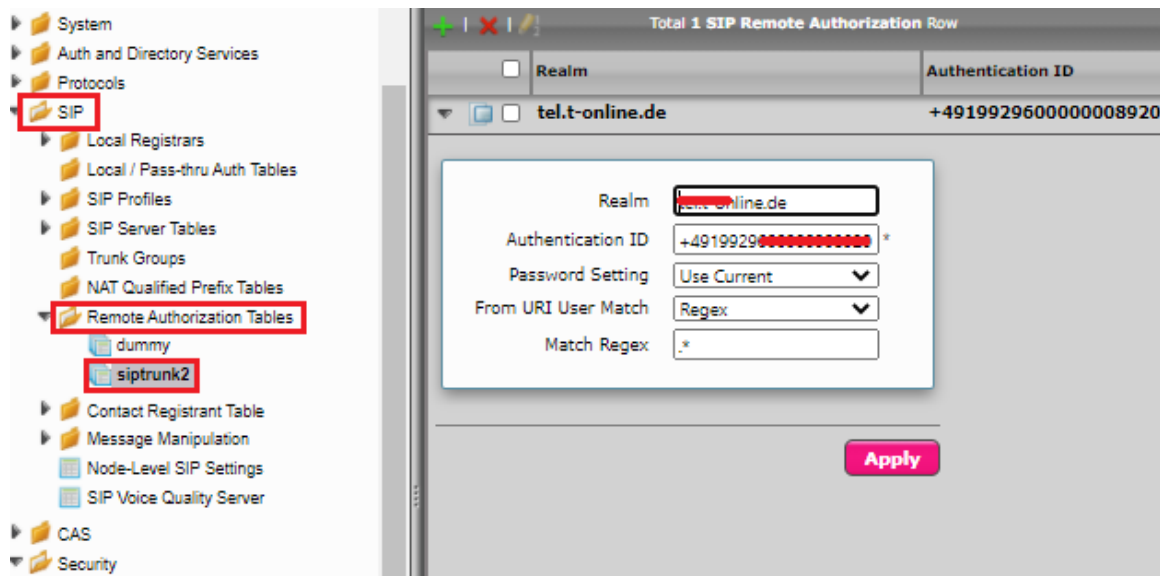
SIP Server	Signaling Group	Registration Status
Entry 100 (f-ecp-600.edns.t-iptel.d...)	(SIP) From/To telekom	Registered

## 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".



## 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.
- Use TCP 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.

Expand All | Collapse All | Reload

Call Routing  
 Signaling Groups  
 Linked Signaling Groups  
 Node Interfaces  
 Application Solution Module  
 System  
 Auth and Directory Services  
 Protocols  
**SIP**  
 Local Registrars  
 Local / Pass-thru Auth Tables  
 SIP Profiles  
**SIP Server Tables**  
 Default SIP Server  
**Telekom SIP server Table**  
 dummy  
 Trunk Groups  
 NAT Qualified Prefix Tables  
 Remote Authorization Tables  
 Contact Registrant Table  
 Message Manipulation  
 Node-Level SIP Settings  
 SIP Voice Quality Server  
 CAS  
 Security  
 Media  
 Tone Tables  
 Telephony Mapping Tables  
 SNMP/Alarms  
 Logging Configuration  
 Remote Log Servers  
 Log Profiles  
 Subsystems  
 Port Mirror

Create SIP Server | X | Total 1 SIP Server Row

Host / Domain  
 55113. primary.companyflex.de

Server Host

Server Lookup DNS SRV  
 Host IP Version IPv4  
 Domain Name/FQDN 55113. primary.companyflex.de  
 Service Name sip  
 Protocol TCP

Transport  
 Monitor None

Remote Authorization and Contacts

Remote Authorization Table siptrunk2  
 Contact Registrant Table telekom contact reg  
 Clear Remote Registration on Startup True  
 Contact URI Randomizer False  
 Stagger Registration False  
 Retry Non-State Nonce True  
 Authorization on Refresh True  
 Session URI Validation Liberal

Connection Reuse  
 Reuse True  
 Sockets 1  
 Reuse Timeout Forever

SRV Servers

Total 3 SipSrvServer Rows

Server ID	FQDN/Domain Name	Protocol	Port	Time to Live	Priority	Weight
100	f-ecp-600.edns.t-ipn...	TCP	5060	3599	10	0
102	d-ecp-600.edns.t-ipn...	TCP	5060	3599	20	0
101	h2-ecp-600.edns.t-ipn...	TCP	5060	3599	30	0

## Message Manipulation

The Message Manipulation feature work in concert to modify SIP messages. Below Message Manipulation are used to avoid registration and call failures.

### The SMM performs the following actions:

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

- To
- From
- Req-URI

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

Add new headers for all outbound INVITE messages.

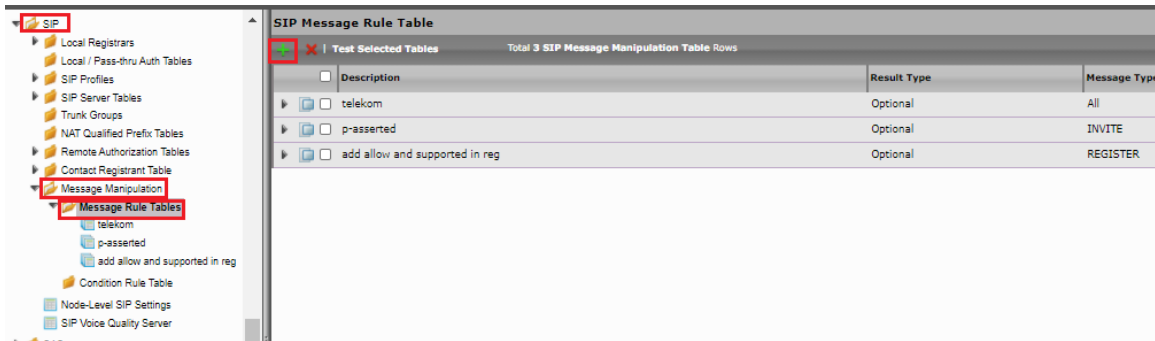
- P-Early-Media
- Allow-Events

Add new header for all outbound REGISTER messages.

- Supported
- Allow

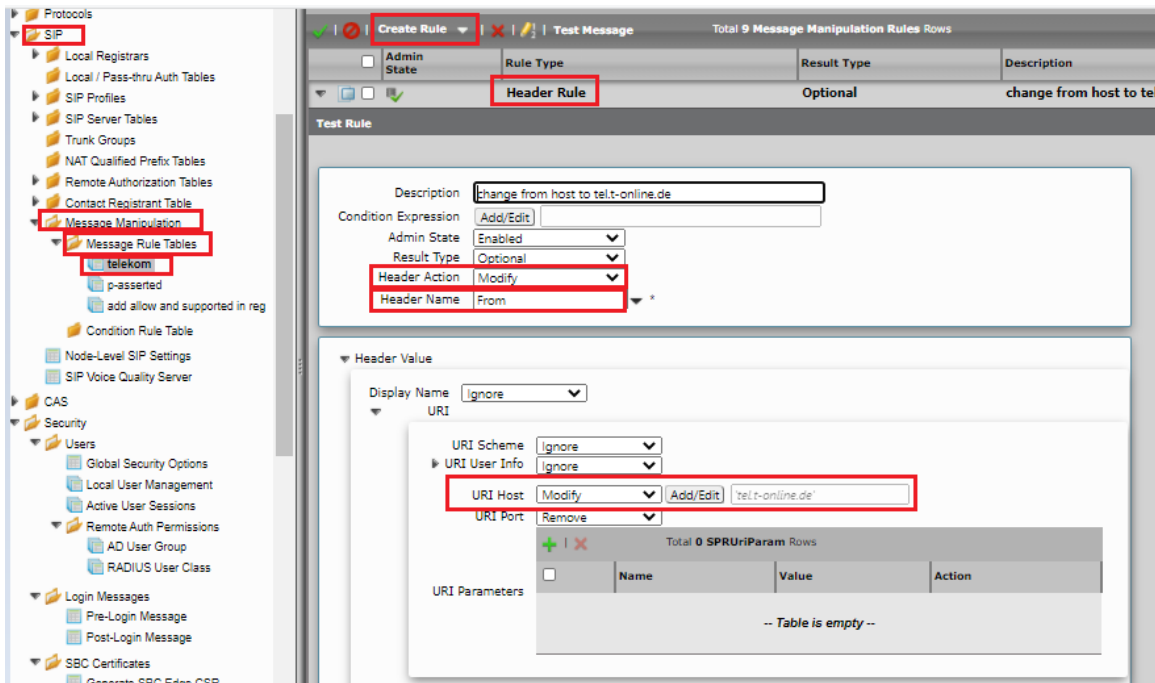
Select Settings > SIP > Message Manipulation > Message Rule Table

Click the Create Message Rule Table(+) icon.



#### Message Manipulation - 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 on add/edit and provide the fqdn that will replace the URI host in from header.



Under "Telekom" Repeat the same for To header.

**telekom**

✓ | ✗ | ⚙ | Test Message Total 9 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	change from host to tel.t-online.de
<input type="checkbox"/>	<b>Header Rule</b>	Optional	change to host to tel.t-online.de

**Test Rule**

Description: change to host to tel.t-online.de

Condition Expression:

Admin State:

Admin State: Enabled

Result Type: Optional

Header Action: **Modify**

Header Name: **To** \*

Header Value

Display Name:

URI

URI Scheme:

URI User Info:

**URI Host:   'tel.t-online.de'**

URI Port:

URI Parameters

Name	Value	Action
-- Table is empty --		

Under "Telekom" repeat the same for request URI.

**telekom**

✓ | ✗ | ⚙ | Test Message Total 9 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
<input type="checkbox"/>	Header Rule	Optional	change from host to tel.t-online.de
<input type="checkbox"/>	Header Rule	Optional	change to host to tel.t-online.de
<input type="checkbox"/>	<b>Request Line Rule</b>	Optional	requestline

**Test Rule**

Description: requestline

Condition Expression:

Admin State:

Admin State: Enabled

Result Type: Optional

Request Line Value

Method:

URI

URI Scheme:

URI User Info:

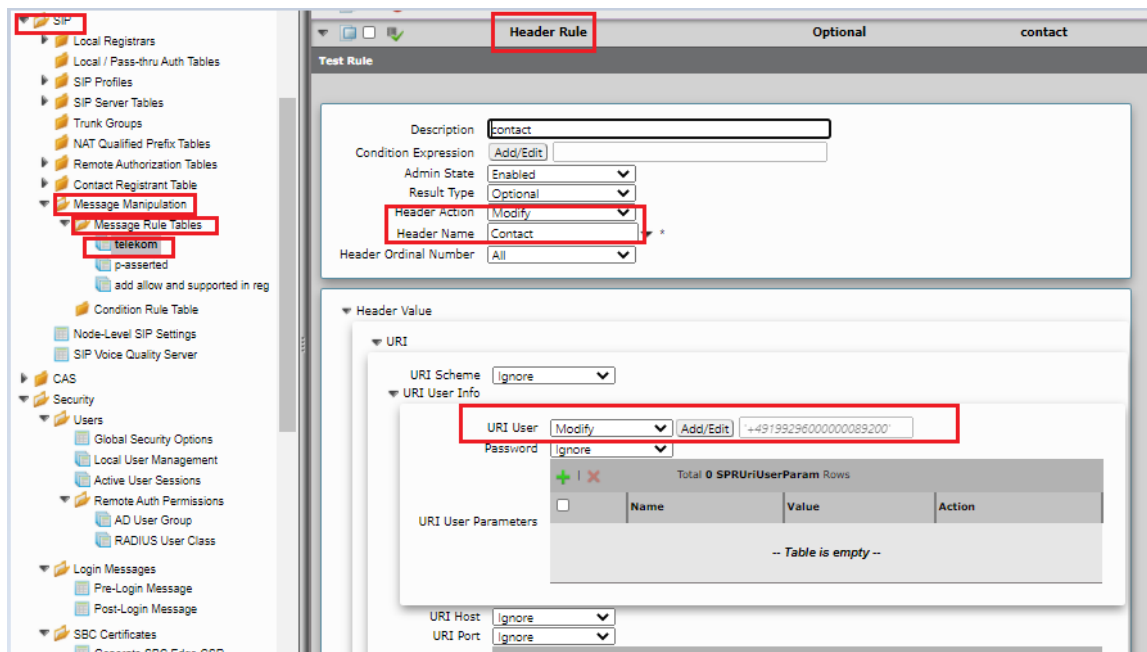
**URI Host:   'tel.t-online.de'**

URI Port:

URI Parameters

Name	Value	Action
-- Table is empty --		

Create message manipulation under "telekom" so that the contact header has SIP trunk number in URI user for all the sip messages .



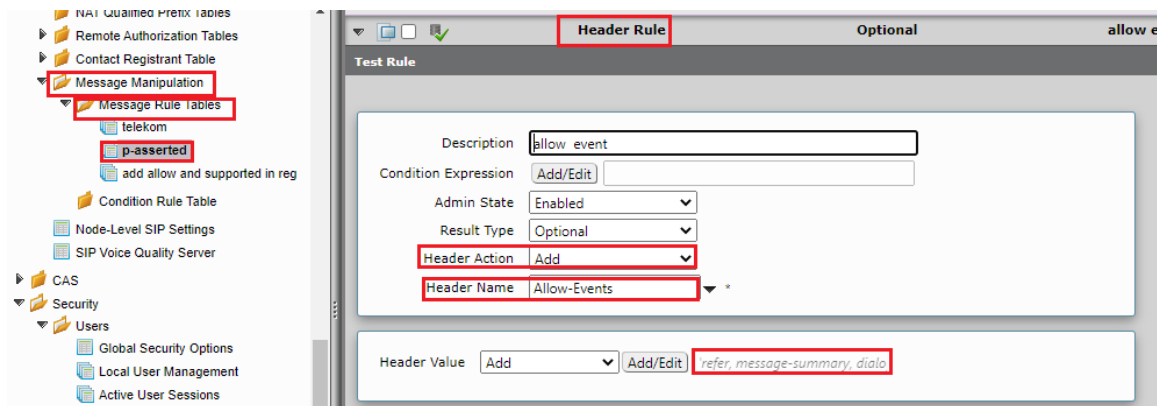
### Message Manipulation - Add Allow-Events in INVITE

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

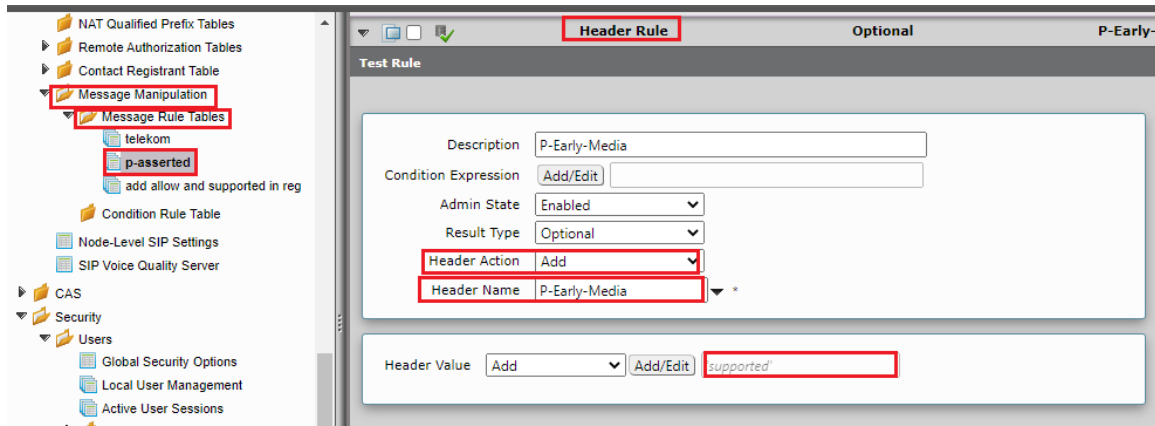
Choose "INVITE" message under Applicable Messages.

- From the Create Rule drop-down-box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Add" and header name as "Allow-Events".
- Under header value give "Add" and click on add/edit and provide 'refer, message-summary, dialog'.
- Click on **Apply**.



### Message Manipulation - Add P-Early-Media in INVITE

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



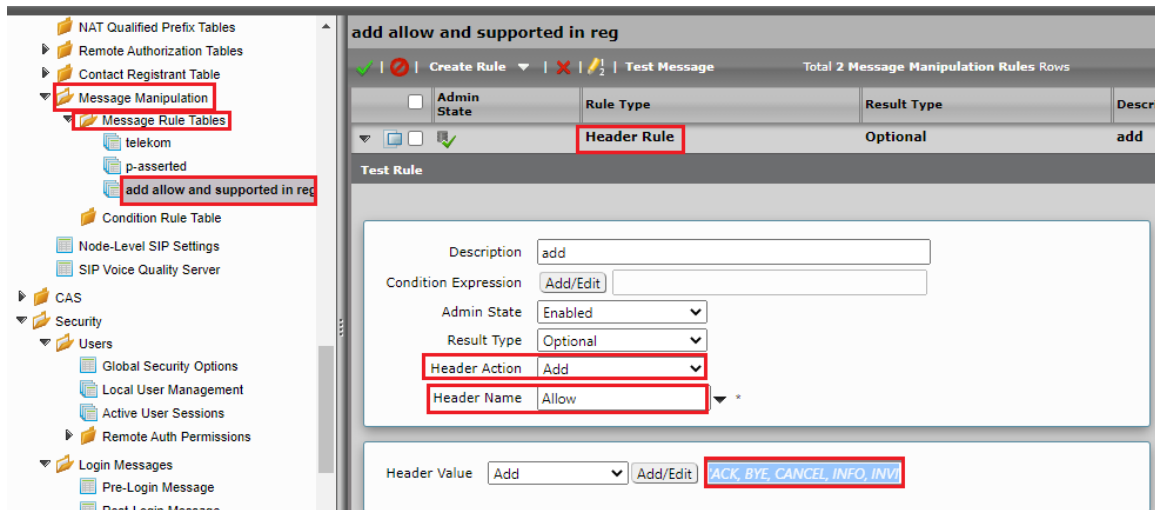
### Message Manipulation - Add Allow in REGISTER

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

Choose "REGISTER" message under Applicable Messages.

- 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 on **Apply**.



### Message Manipulation - 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 on **Apply**.

**add allow and supported in reg**

✓ | ✗ | Create Rule | ✗ | ✗ | Test Message | Total 2 Message Manipulation Rules Rows

Admin State	Rule Type	Result Type	Description
✓	Header Rule	Optional	add
✓	Header Rule	Optional	add supported

**Test Rule**

Description: add supported

Condition Expression: Add/Edit

Admin State: Enabled

Result Type: Optional

Header Action: Add

Header Name: Supported

Header Value: Add

## 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".



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 "default media list".
- Choose SIP Server Table as "Telekom Sip Server Table".

**SIP Signaling Group Details: From/To telekom**

Show Channels

Description: From/To telekom

Admin State: Enabled

Service Status: Up

**SIP Channels and Routing**

Action Set Table: None

Call Routing Table: From Telekom

No. of Channels: 5

SIP Profile: telekom sip profile

SIP Mode: Basic Call

Agent Type: Back-to-Back User Agent

Interop Mode: Standard

SIP Server Table: Telekom SIP server Table

Load Balancing: Priority: Register Active

Channel Hunting: Most Idle

Notify Lync CAC Profile: Disable

Challenge Request: Disable

Outbound Proxy IP/FQDN: 5060

Outbound Proxy Port: 5060

No Channel Available Override: 34: No Circuit/Channel Available

Call Setup Response Timer: 255

Call Proceeding Timer: 180

QoS Reporting: Disabled

Use Register as Keep Alive: Enable

Forked Call Answered Too Soon: Disable

**Media Information**

Supported Audio/Fax Modes: DSP

Supported Video/Application Modes: Proxy Direct

Media List ID: Default Media List

Play Ringback: Auto on 180

Tone Table: Default Tone Table

Play Congestion Tone: Disable

Early 183: Disable

Allow Refresh SDP: Enable

Music on Hold: Disabled

RTCP Multiplexing: Disable

**Mapping Tables**

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

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

Pass-thru Peer SIP Response Code: Enable





Because a NAT is used in the test environment, add the external public IP of the NAT box under static NAT outbound of the Sig Group that is facing towards the Deutsche Telekom server.

- 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 TCP.
- Add message manipulation under the outbound section that we created earlier to add a domain instead of IP, for a successful call.
  - Enable Message Manipulation.
  - Click **Add/Edit** on Outbound Message Manipulation.
  - This displays a drop-down list of available message tables. Select an entry and click **Apply**.

The screenshot shows the SBC configuration interface with the following sections:

- SIP IP Details:**
  - Signaling/Media Private IP: 192.168.1.1
  - Ethernet 2 IP: 192.168.1.1
  - Signaling DSCP: 40
  - NAT Traversal: Enabled
  - ICE Support: Disabled
  - Static NAT - Outbound:
    - Outbound NAT Traversal: Static NAT
    - NAT Public IP (Signaling/Media): 80.1.1.1
  - Static NAT - Inbound:
    - Detection: Disabled
- Listen Ports:**

Port	Protocol	TLS Profile ID
5070	TCP	N/A
- Federated IP/FQDN:**

IP/FQDN	Netmask/Prefix
192.168.1.1	255.255.255.255
192.168.1.1	255.255.255.255
- Message Manipulation:**
  - Message Manipulation: Enabled
  - Inbound Message Manipulation: (Empty list)
  - Outbound Message Manipulation:
    - Message Table List: telekom, p-asserted, add allow and supported in reg



Configure NAT box so that the external public IP doesn't change frequently. Incase if there is a change, update the Static NAT outbound section with the new allocated public IP address.

## 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:

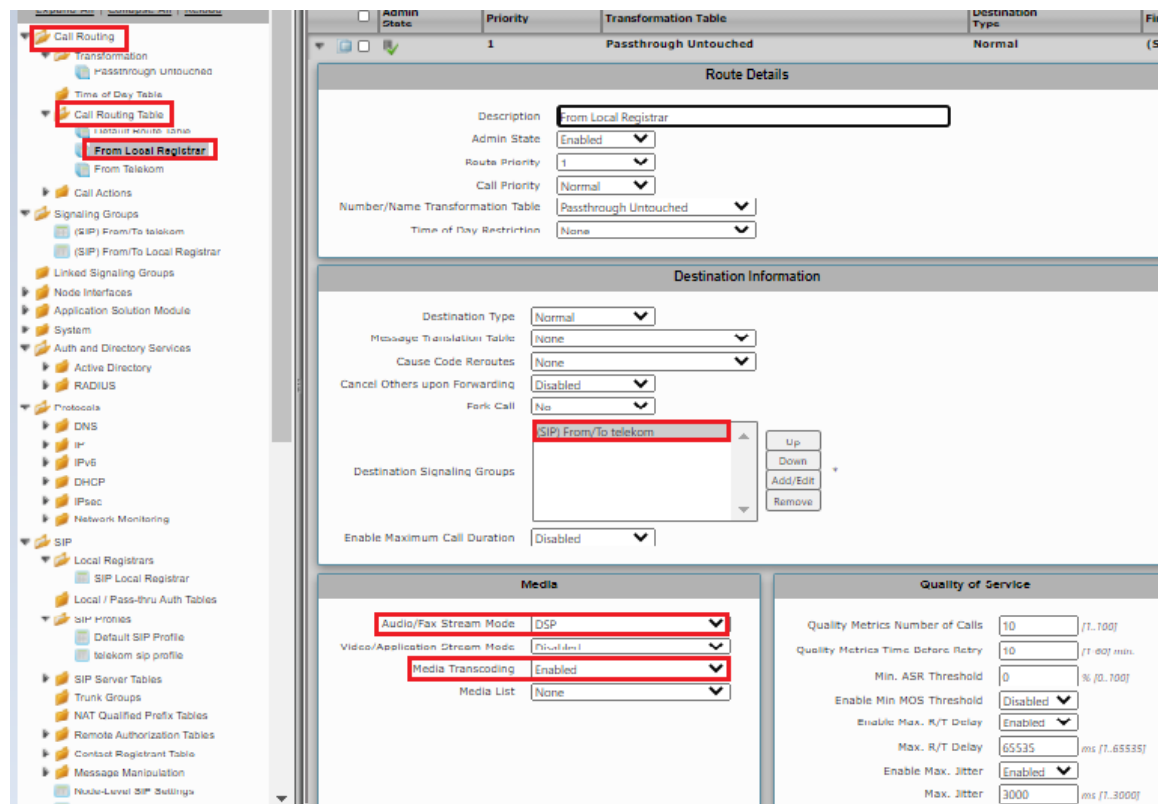
DSP (default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support DSP are attempted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

#### Media Transcoding:

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

3. Click **Apply**.

## Call Routing table for "From Local Registrar"



The screenshot shows the configuration interface for a Call Routing entry named "From Local Registrar". The interface is divided into several sections:

- Route Details:**
  - Description: From Local Registrar
  - Admin State: Enabled (indicated by a green checkmark icon)
  - Route Priority: 1
  - Call Priority: Normal
  - Number/Name Transformation Table: Passthrough Untouched
  - Time of Day Restriction: None
- Destination Information:**
  - Destination Type: Normal
  - Message Translation Table: None
  - Cause Code Reroutes: None
  - Cancel Others upon Forwarding: Disabled
  - Fork Call: No
  - Destination Signaling Groups: SIP From/To telekom (highlighted with a red box)
  - Buttons: Up, Down, Add/Edit, Remove
  - Enable Maximum Call Duration: Disabled
- Media:**
  - Audio/Fax Stream Mode: DSP (highlighted with a red box)
  - Video/Application Stream Mode: (highlighted with a red box)
  - Media Transcoding: Enabled (highlighted with a red box)
  - Media List: None
- Quality of Service:**
  - Quality Metrics Number of Calls: 10 (range [1..100])
  - Quality Metrics Time Before Retry: 10 (range [1..60] min)
  - Min. ASR Threshold: 0 (range [% [0..100])
  - Enable Min MOS Threshold: Disabled
  - Enable Max. R/T Delay: Enabled
  - Max. R/T Delay: 65535 (range [ms [1..65535])
  - Enable Max. Jitter: Enabled
  - Max. Jitter: 3000 (range [ms [1..3000])

The left sidebar shows the navigation tree with "Call Routing" and "From Local Registrar" highlighted.

## Call Routing for "From Telekom".

The screenshot displays the configuration interface for a SIP service. The left sidebar shows a tree view of configuration categories, with 'From Telekom' highlighted under 'Call Routing Tables'. The main panel is divided into four sections:

- Route Details:** Includes fields for Description (From Telekom), Admin State (Enabled), Route Priority (1), Call Priority (Normal), Number/Name Transformation Table (Passthrough Untouched), and Time of Day Restriction (None).
- Destination Information:** Includes Destination Type (Registrar Table), Message Translation Table (None), Cause Code Reroutes (None), Cancel Others upon Forwarding (Disabled), Fork Call (No), Destination Signaling Groups (SIP From/To Local Registrar), and Enable Maximum Call Duration (Disabled).
- Media:** Includes Audio/Fax Stream Mode (DSP), Video/Application Stream Mode (Disabled), Media Transcoding (Enabled), and Media List (None).
- Quality of Service:** Includes Quality Metrics Number of Calls (10), Quality Metrics Time Before Retry (10), Min. ASR Threshold (0), Enable Min MOS Threshold (Disabled), Enable Max. R/T Delay (Enabled), Max. R/T Delay (65535), Enable Max. Jitter (Enabled), and Max. Jitter (3000).

## 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	✓
2	Inbound Call-Mobile PSTN	✓
3	Outbound Call-Mobile PSTN	✓
4	Inbound call-Landline PSTN	✓
5	Outbound call-Landline PSTN	✓
6	Basic Call With Different Codecs	✓
7	Voice Mail	✓
8	FAX using T.38	✓
9	Call Forward	✗
10	FAX using G711 Fallback	✓
11	Call Hold and Resume Outbound	✓
12	Call Hold and Resume Inbound	✓
13	Anonymous Calls Outbound	✓
14	Session Timers	✓

15	FAX - transcoding	✓
16	Call Transfer (Blind)	✗
17	Call Transfer (Attended)	✗
18	486 Busy	✓
19	487 Request Terminated	✓
20	Long Duration Calls	✓

#### Legend

Tested	✓
Not Tested	✗



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

## Caveats

- NA

## Support

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

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

## References

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

<https://ribboncommunications.com/products>

## Conclusion

This Interoperability Guide describe the configuration steps required for **Ribbon SBC 1000 / 2000** 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.

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved.