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

Table of Contents

Interoperable Vendors

Deutsche Telekom

Copyright

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

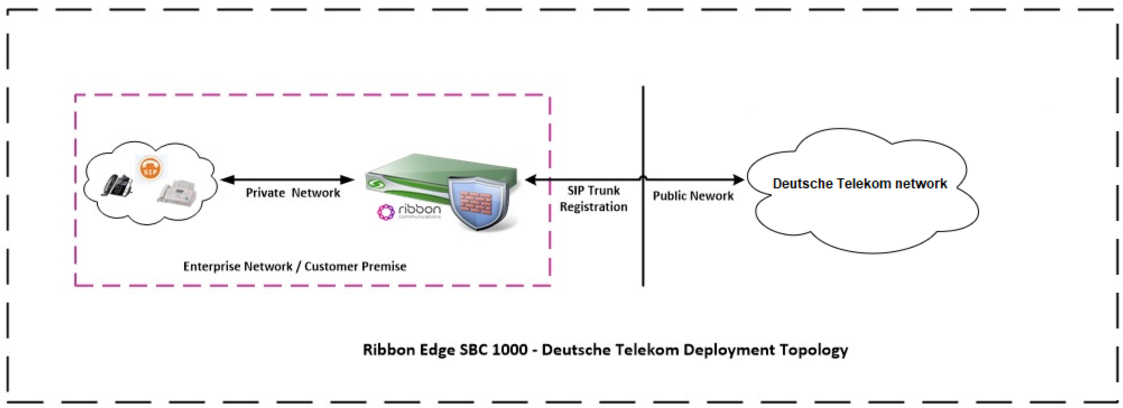
The configuration uses the following equipment and software:

Table 1: Requirements

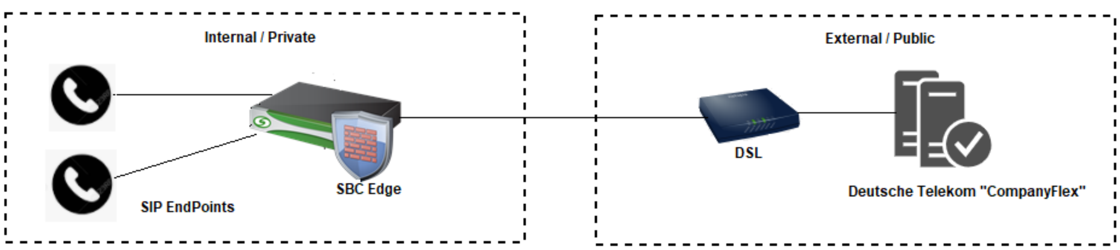
Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC 1000	8.0.1
Third-party Equipment	DSL Line	NA
Deutsche Telekom	Deutsche Telekom "CompanyFlex" SIP trunks	NA
Administration and Debugging Tools	Wireshark LX Tool	3.2.7 2.1.0.6

Network Topology

SBC 1000 - Deutsche Telekom Deployment Topology

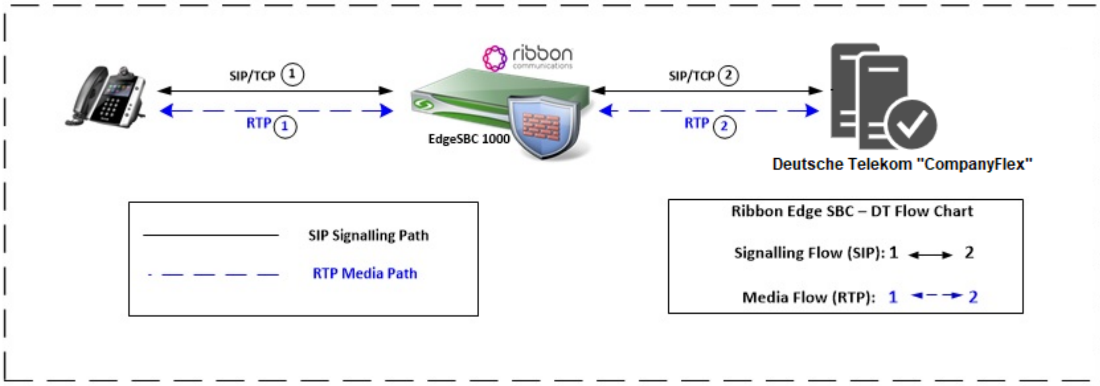


SBC 1000 - Deutsche Telekom Lab Topology



SBC 1000 - Deutsche Telekom Lab Topology

Signaling and Media Flow



Installing SBC 1000/2000

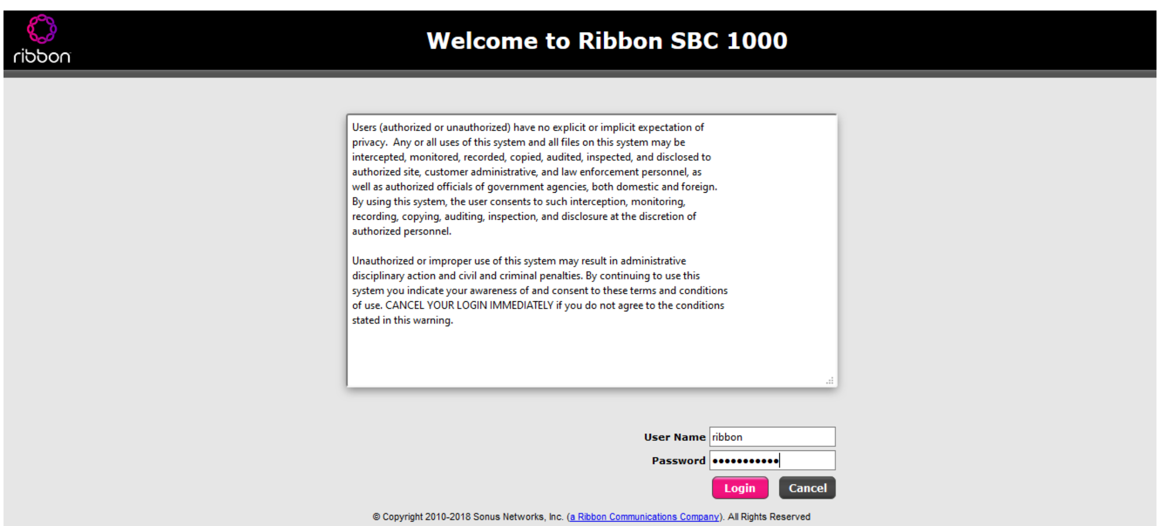
Refer to the following document for installing the SBC 1000: https://doc.rbbn.com/pages/viewpage.action?pageId=229474498

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.

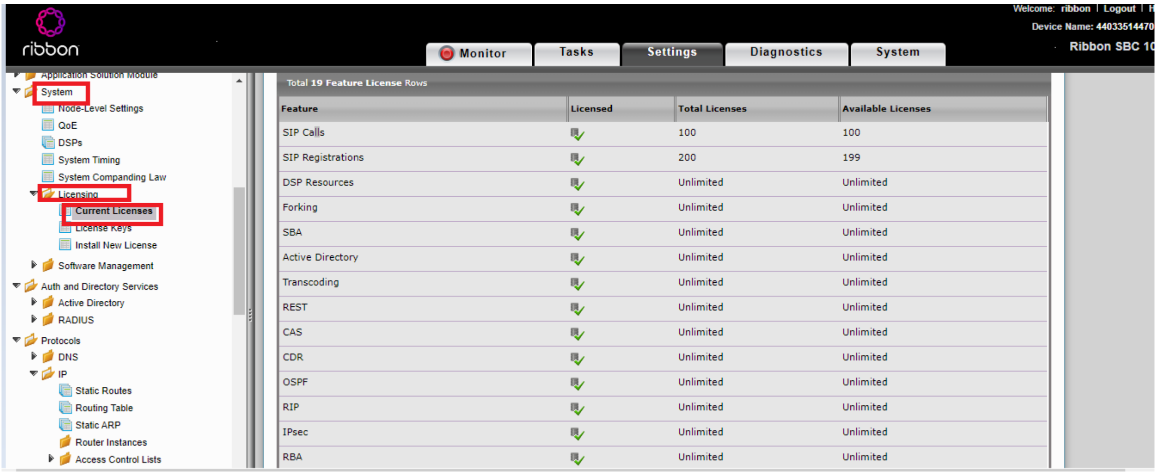


License

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.**



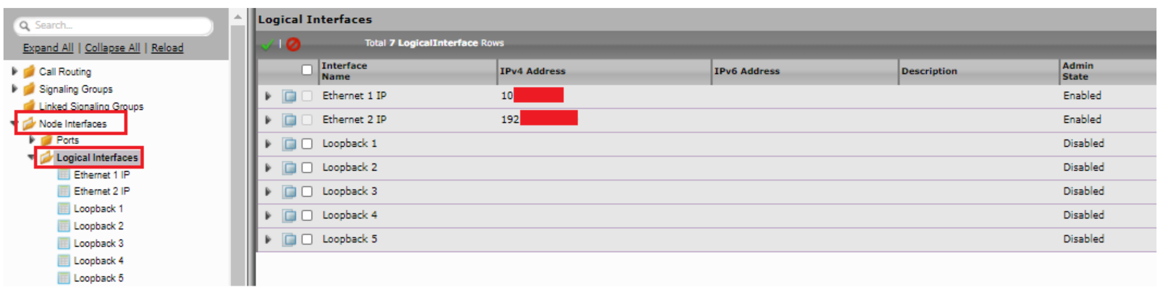
For more details on Licenses, refer to SBC 1000, SBC 2000 Licenses.

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.

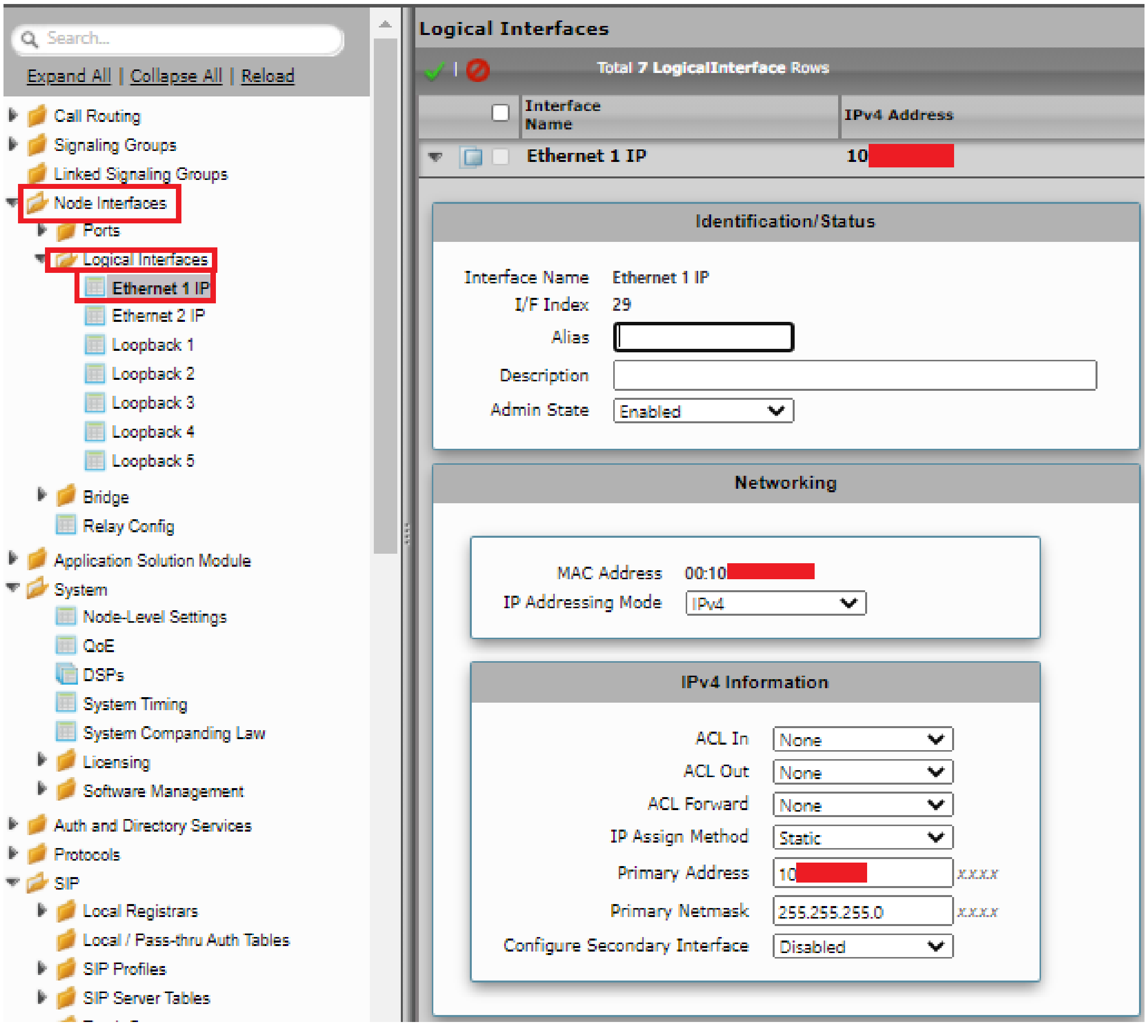
From the **Settings** tab, navigate to **Networking Interfaces > Logical Interfaces.**



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

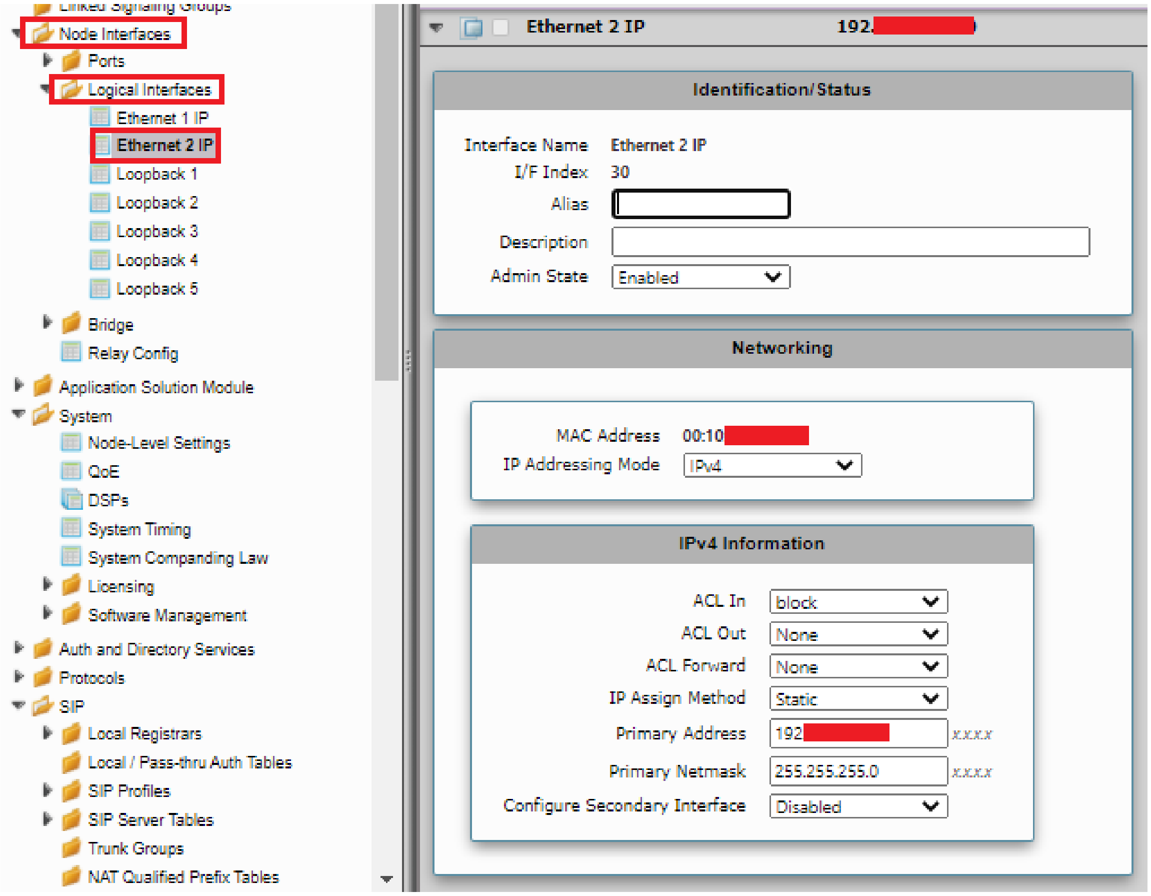
Ethernet 1 IP

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.



Ethernet 2 IP

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



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

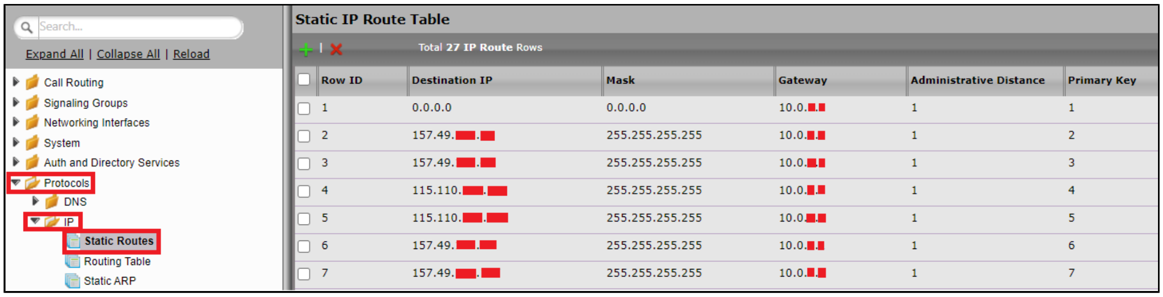
Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network that you can only access through one point or one interface (single path access or default route).

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.



SBC 1000 Configuration for Access End

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

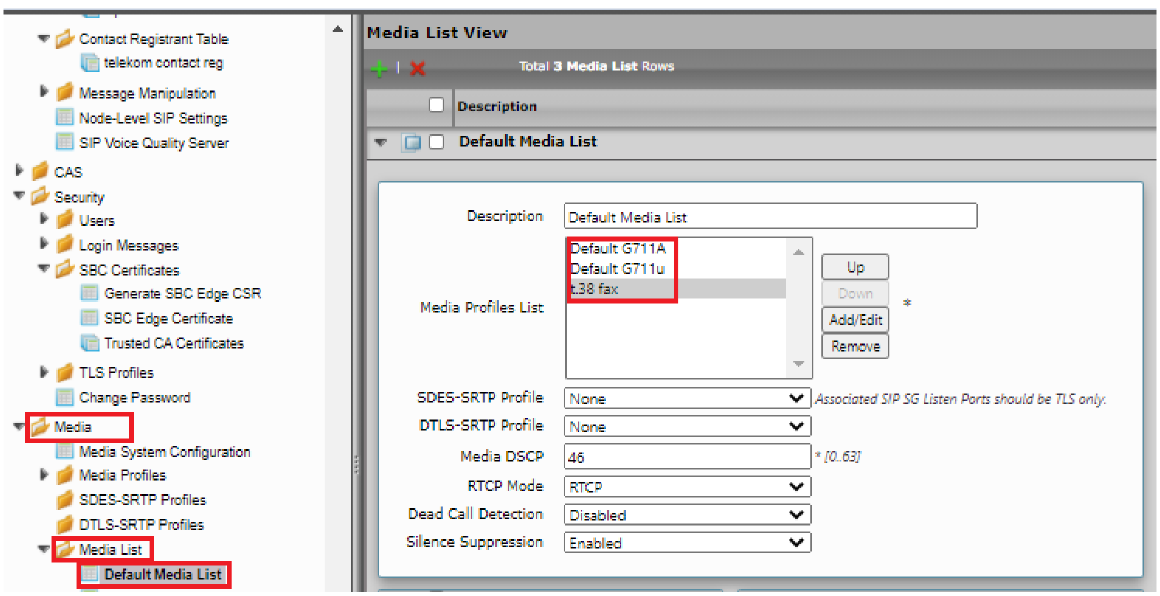
For assistance visit : https://doc.rbbn.com/

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.

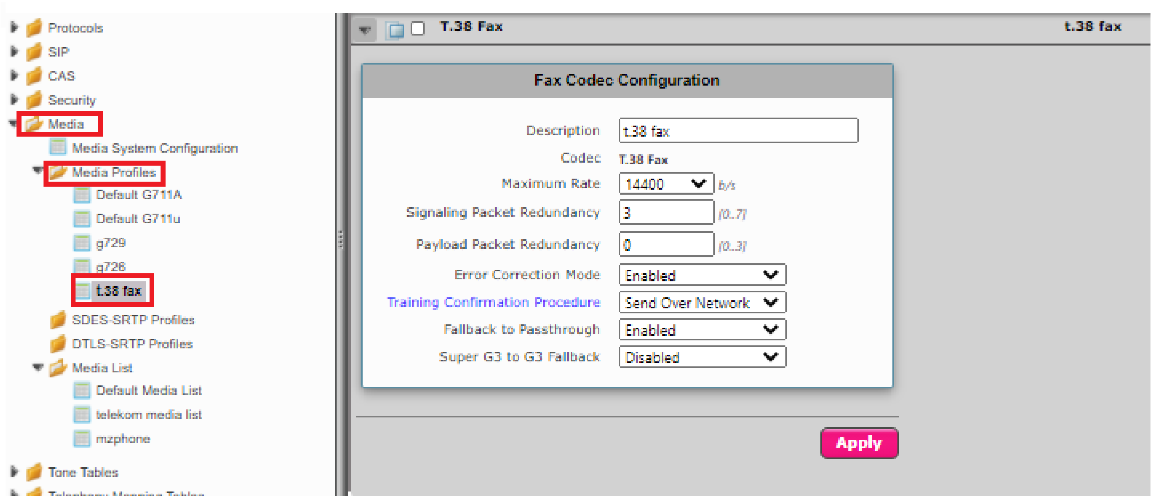
Use default media profile with codec G.711.



ⓘ 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.

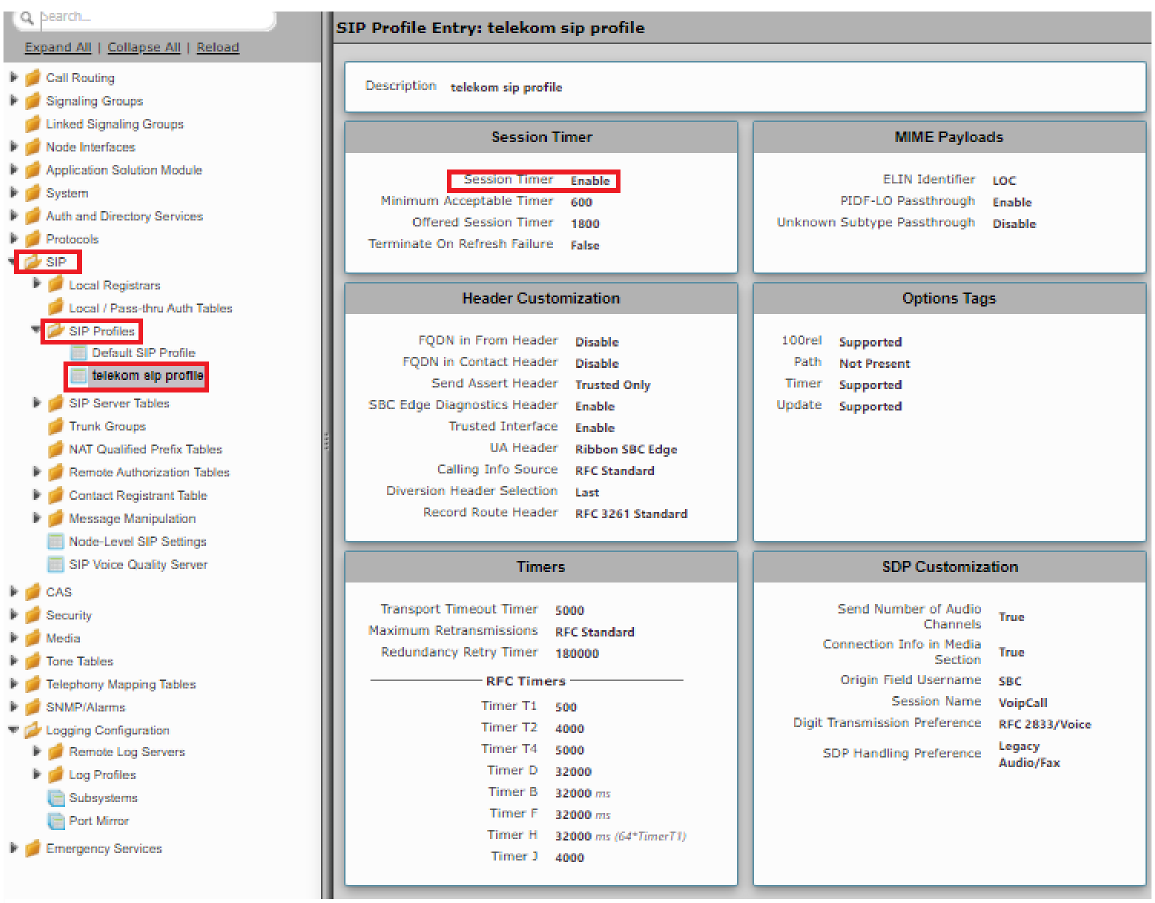


It is recommended to use a maximum packet time (max pTime) of 20ms for all Voice Codecs.

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.

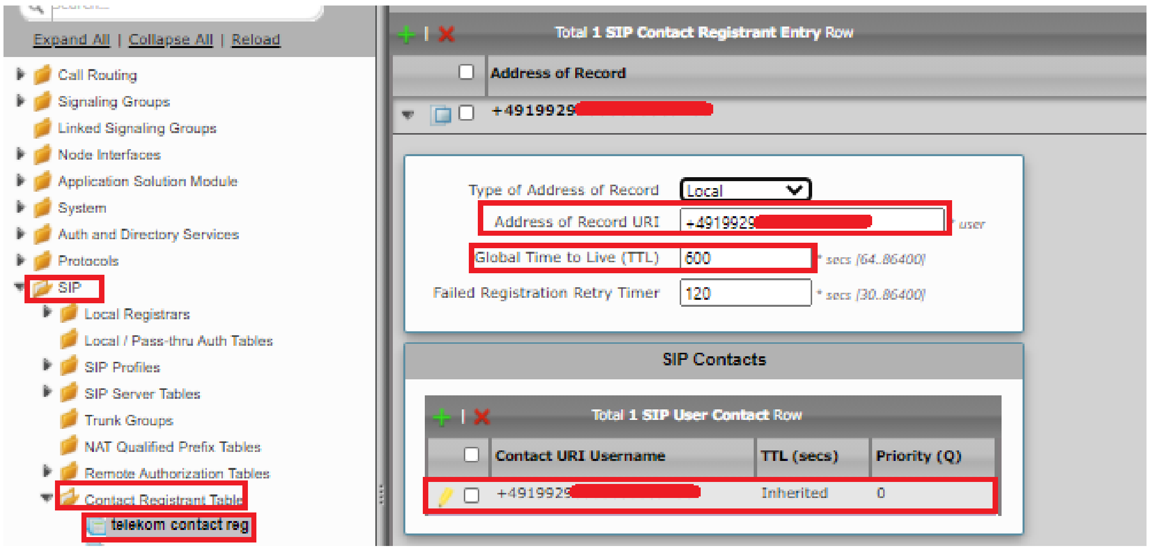


Contact Registration Table

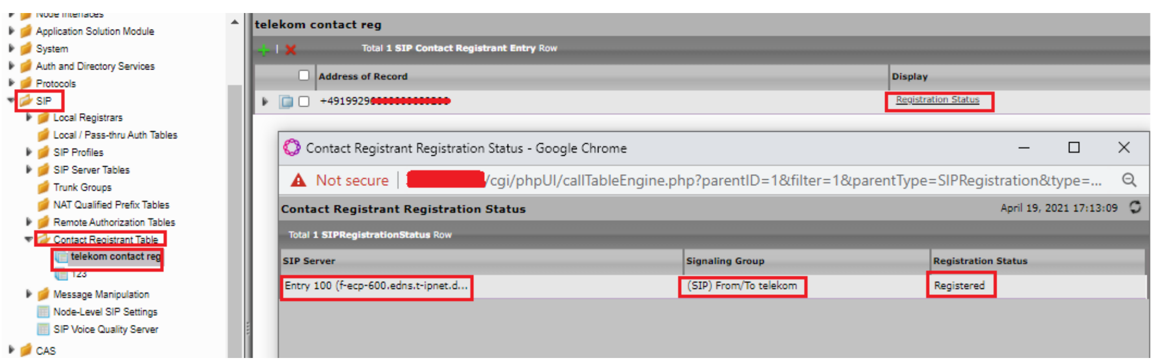
Select **Settings > SIP > Contact Registration Table**.

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".



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

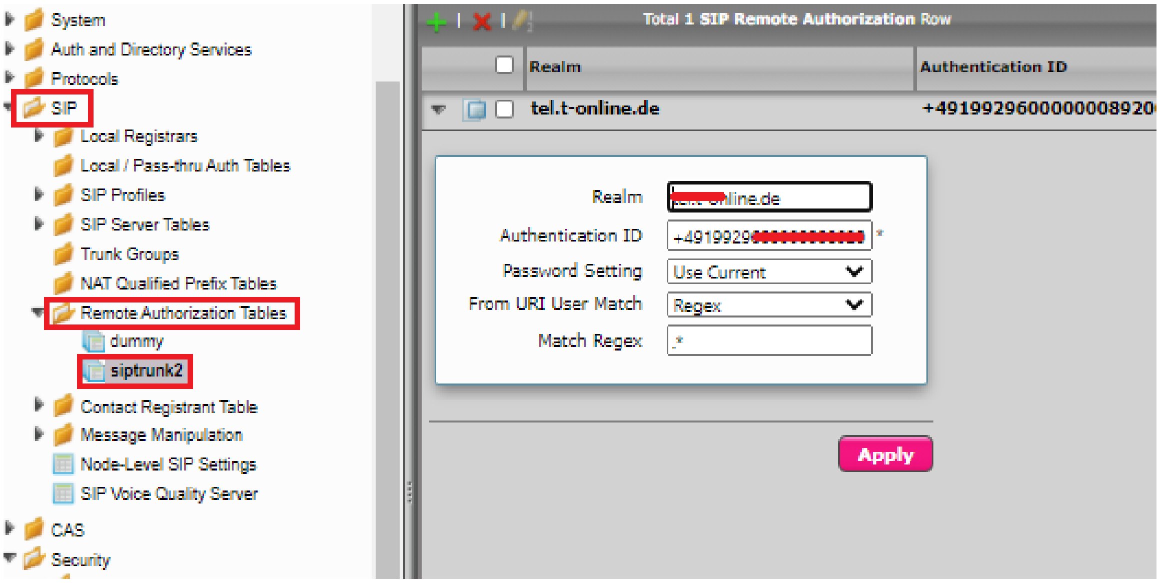


Remote Authorization Table

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".

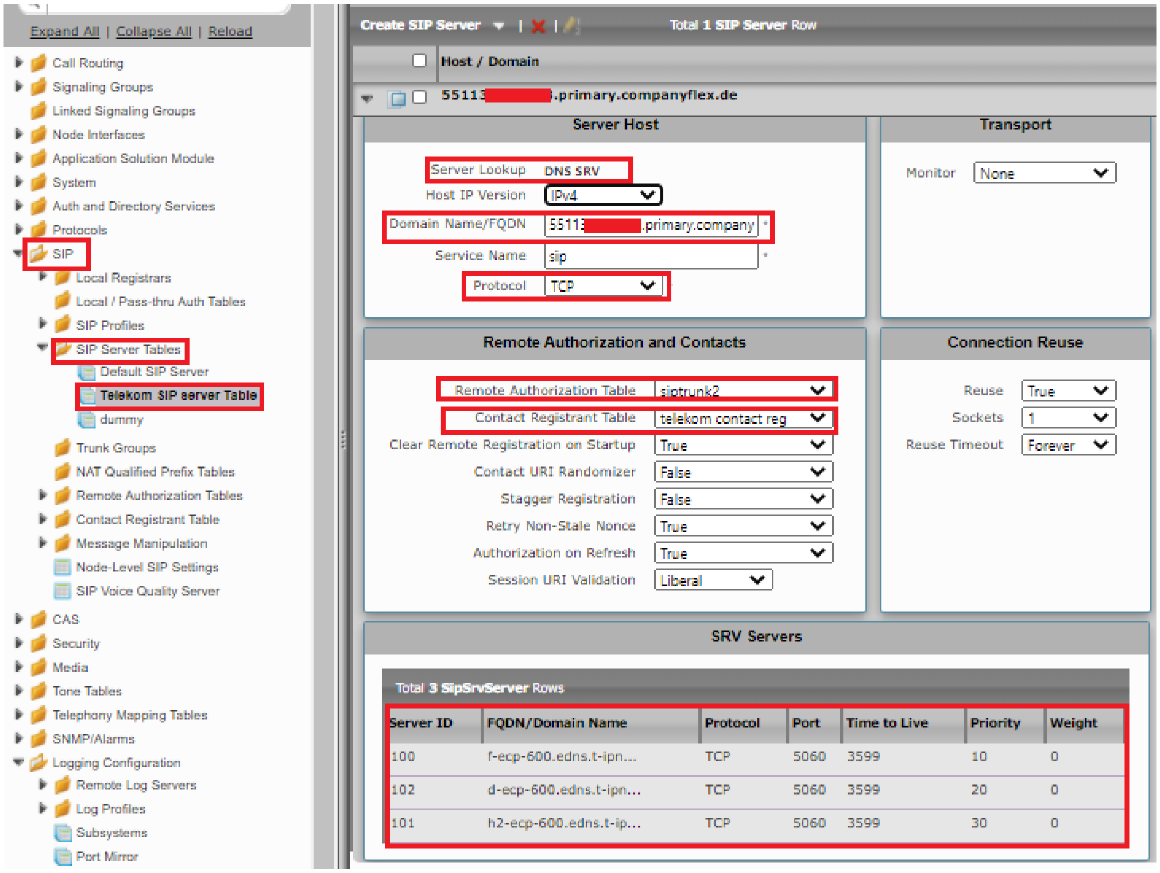


Sip Server Table

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.



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.

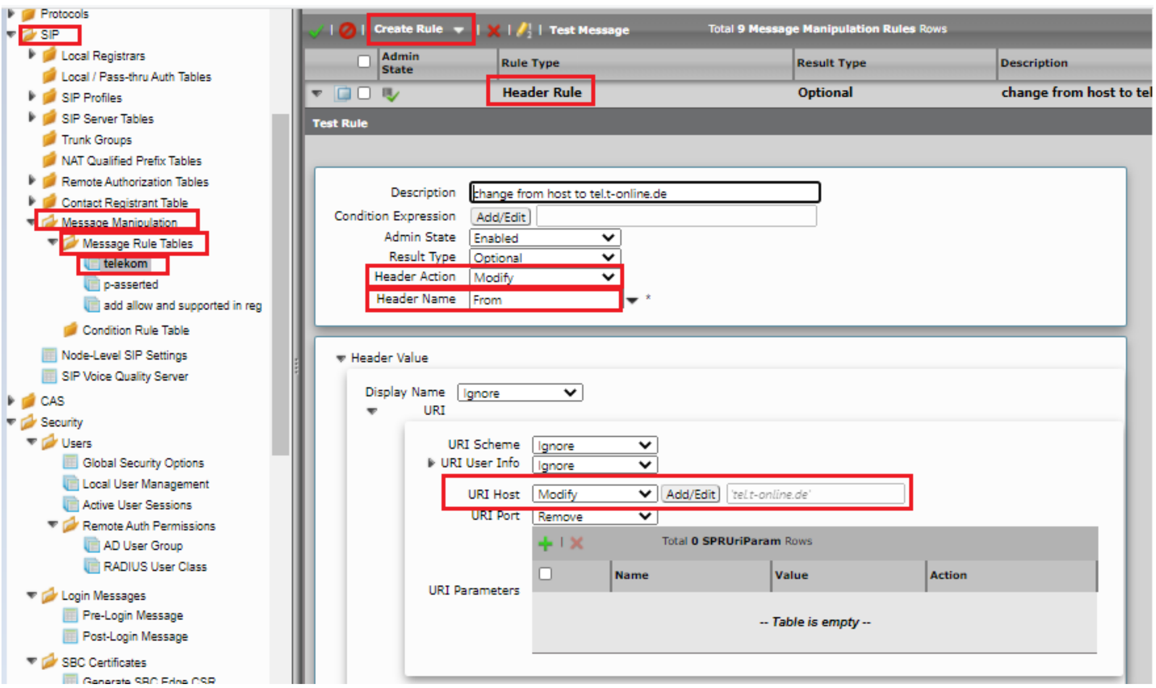
- Supported
- Allow

Select Settings > SIP > Message Manipulation > Message Rule Table

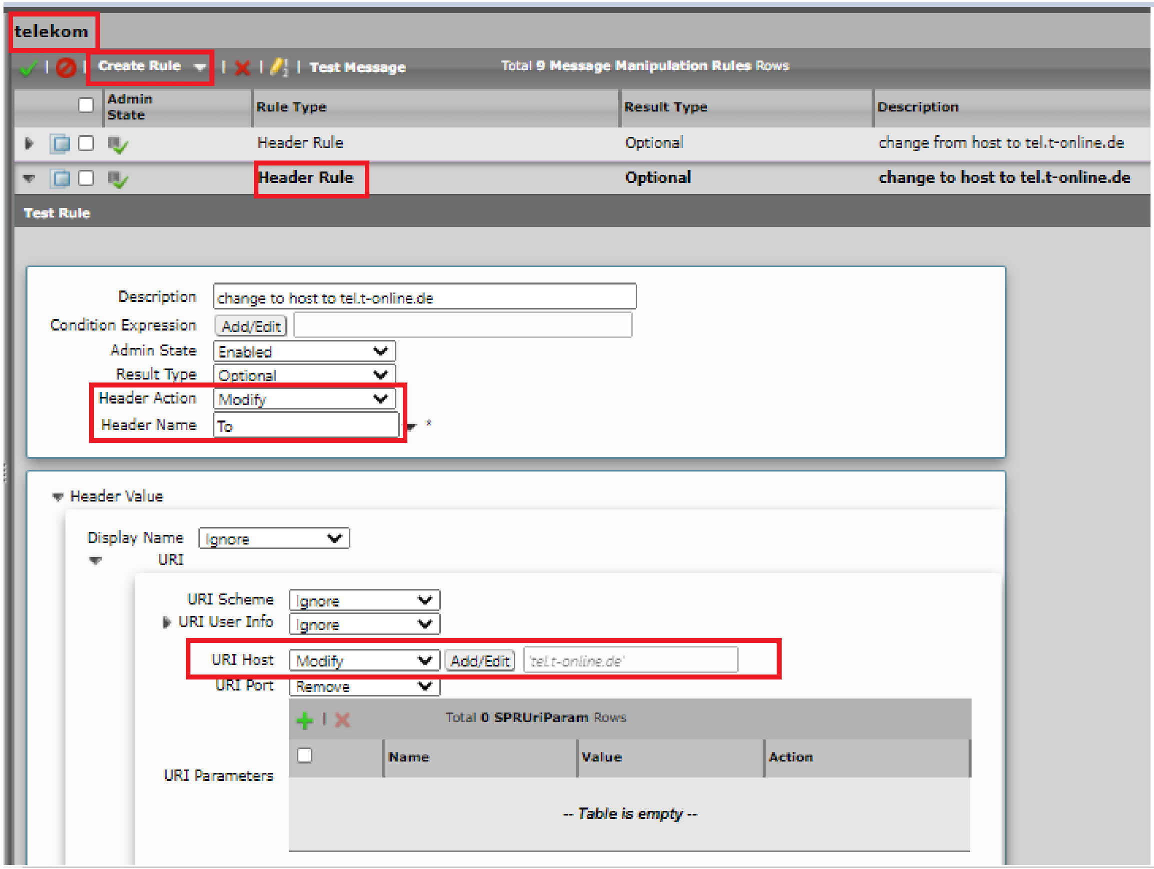
Click the Create Message Rule Table( ) icon.

Message Manipulation - From, To , Request URI sends FQDN in URI host.

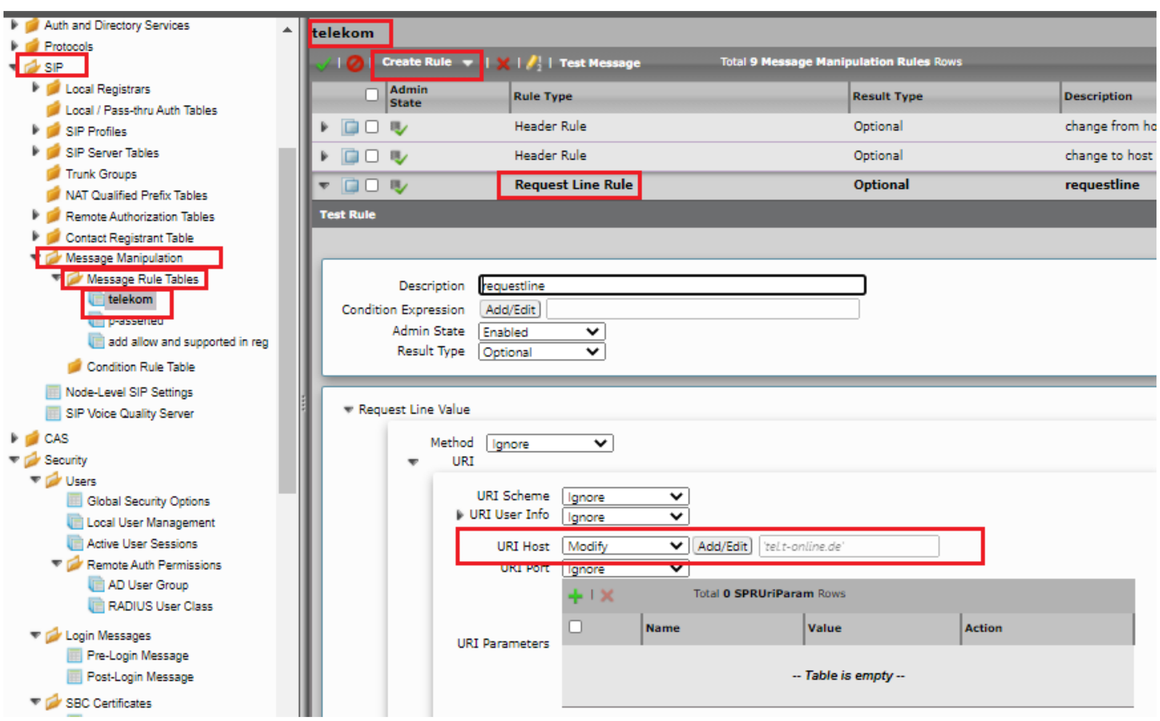
- Provide a description as "Telekom" for the Rule Table.
- Apply the SMM for All messages.
- Click the expand icon next to the Rule Table entry created.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Modify" and header name as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



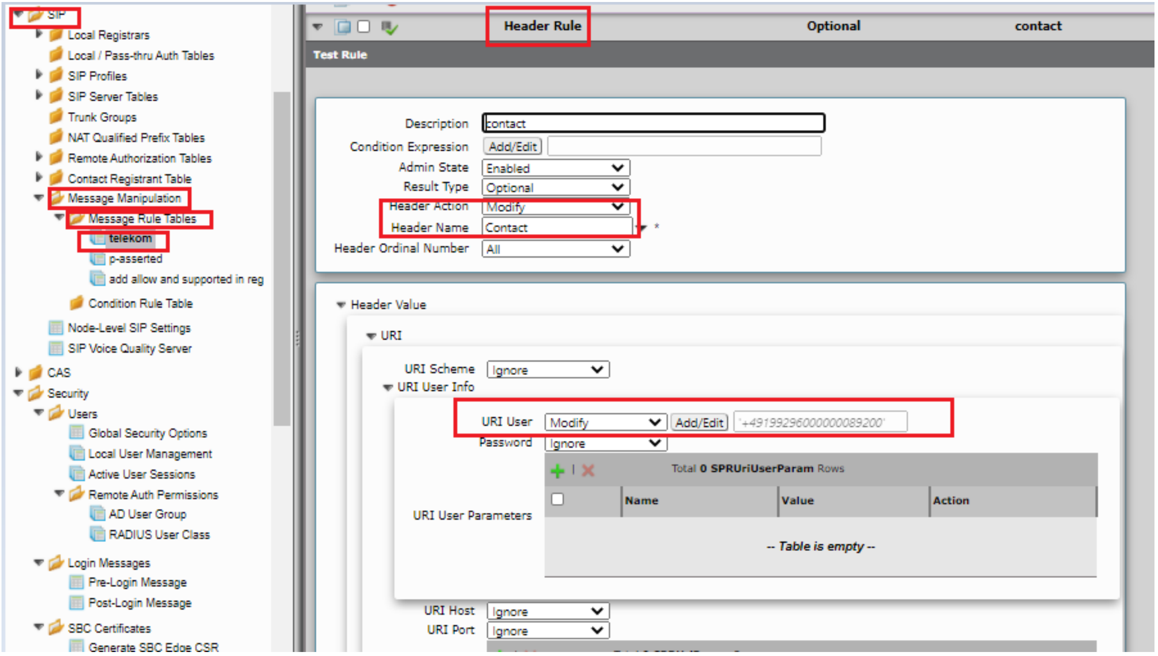
Under "Telekom" Repeat the same for To header.



Under "Telekom" repeat the same for request URI.



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



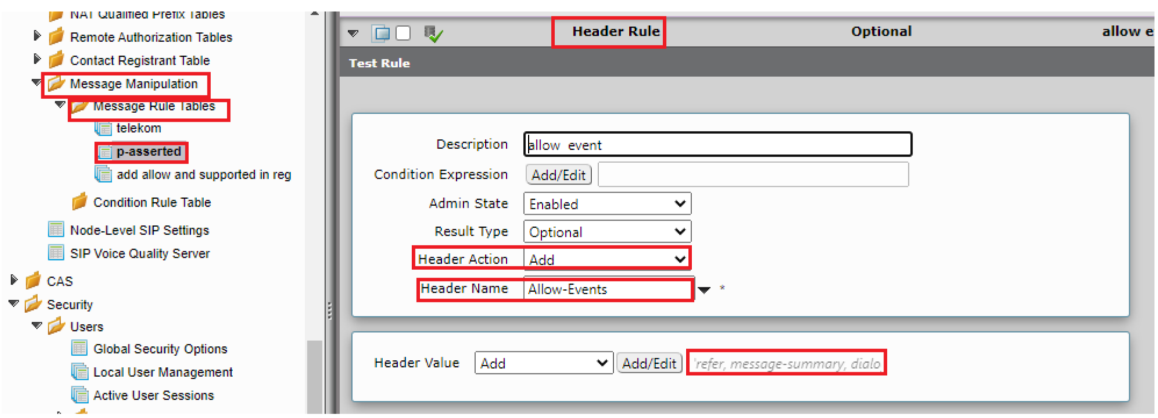
Message Manipulation - Add Allow-Events in INVITE

Click the Create Message Rule Table( ) icon.

Provide a suitable description for the Rule Table.

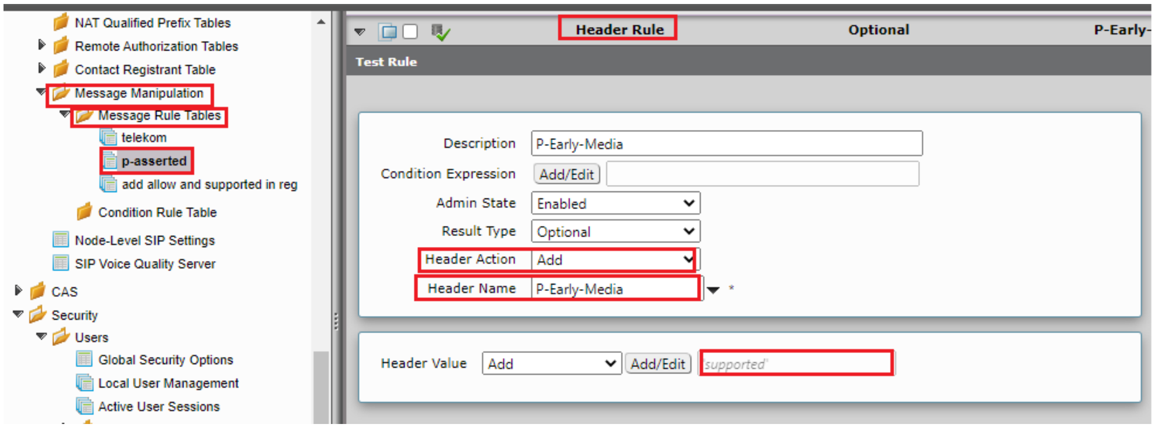
Choose "INVITE" message under Applicable Messages.

- 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**.



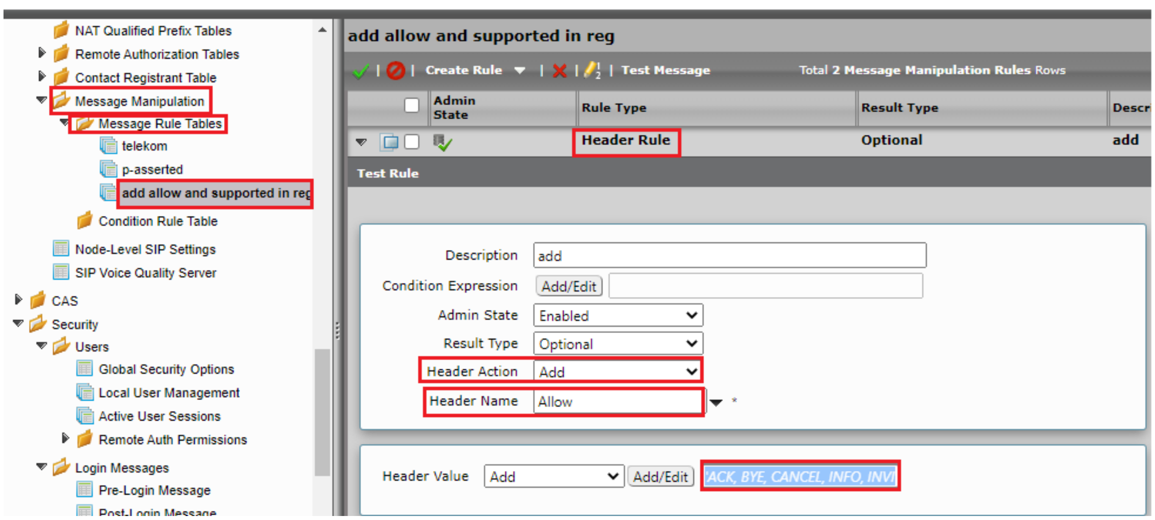
Message Manipulation - Add Allow in REGISTER

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

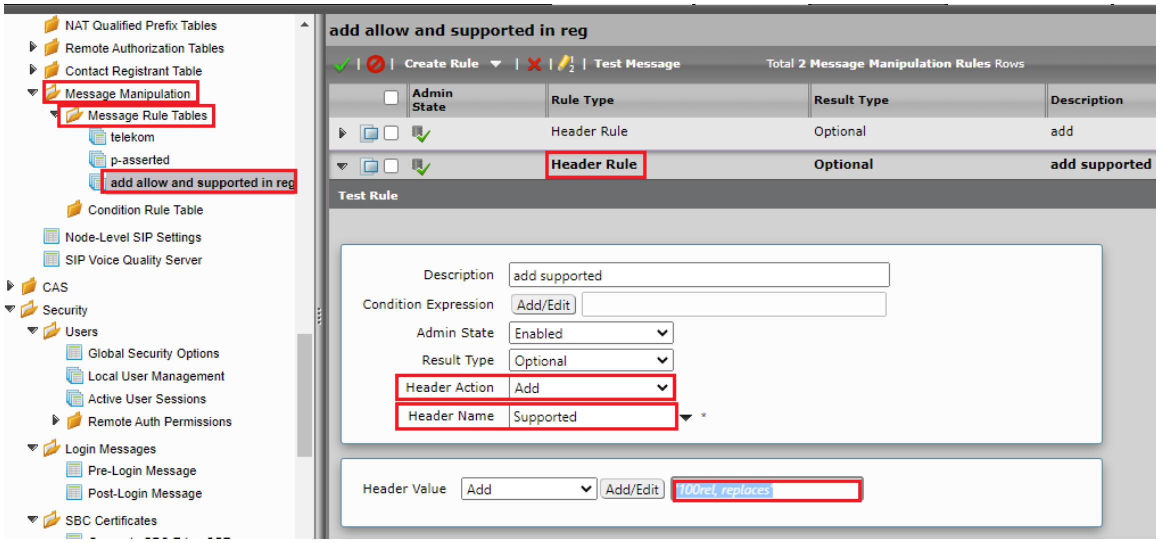
Choose "REGISTER" message under Applicable Messages.

- 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

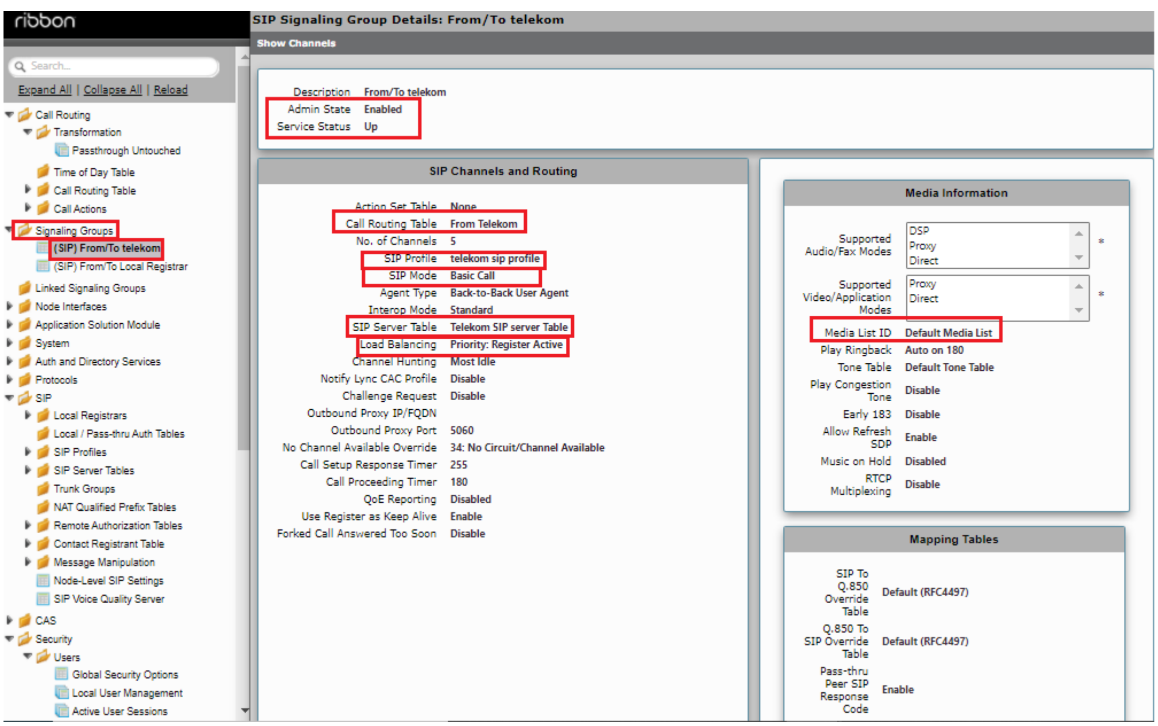
Signaling Groups allow grouping telephony channels together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected.

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".

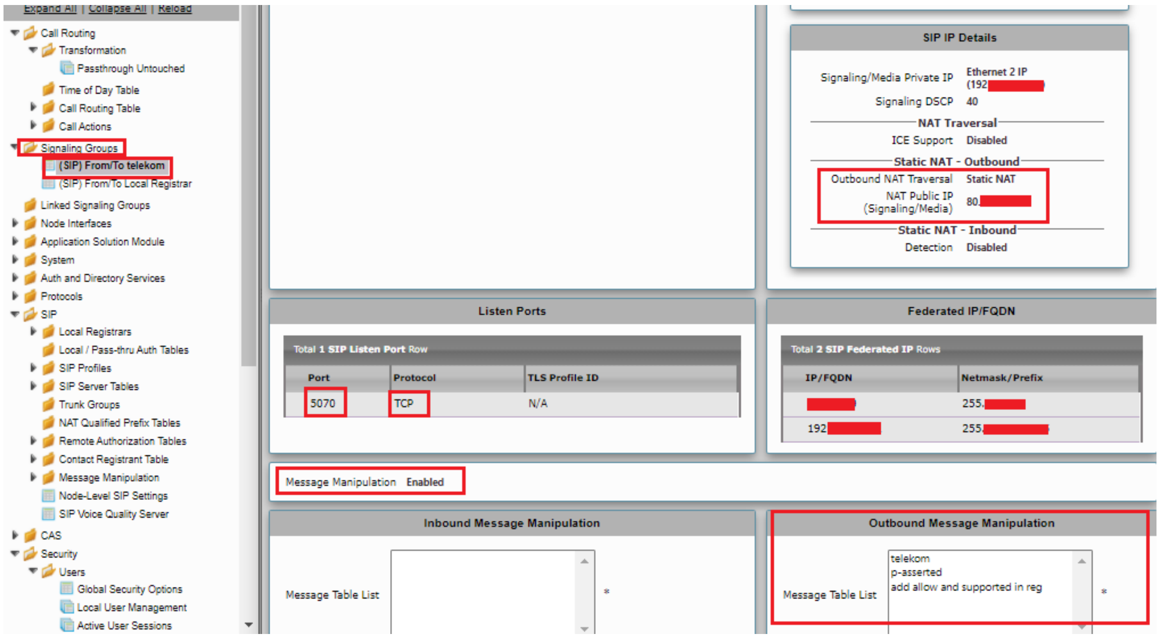
- Choose Agent type as "Back-to-Back user agent" and media list as "default media list".
- Choose SIP Server Table as "Telekom Sip Server Table".



> ⚠ 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**.



> ⚠ 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.

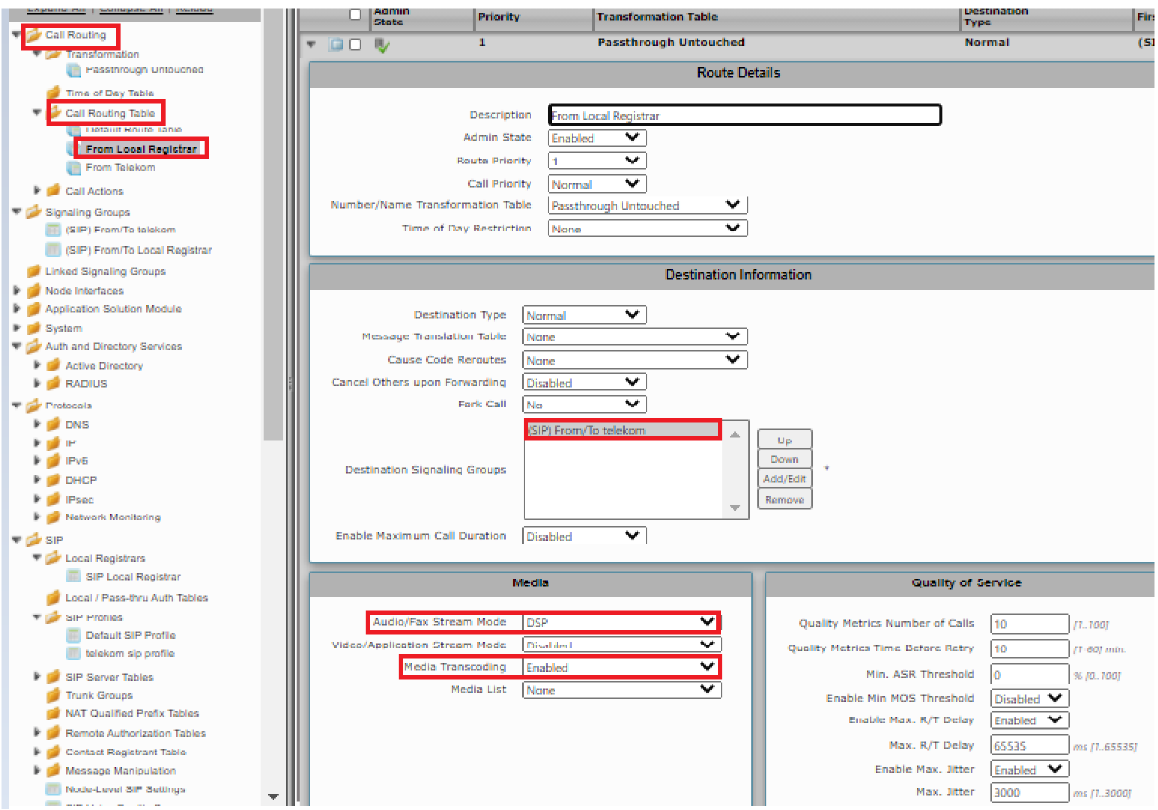
   Number/Name Transformation Table:
   Specifies the Transformation Table to use for this routing entry. This drop-down list is populated from the entries in the Transformation Table.

   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.

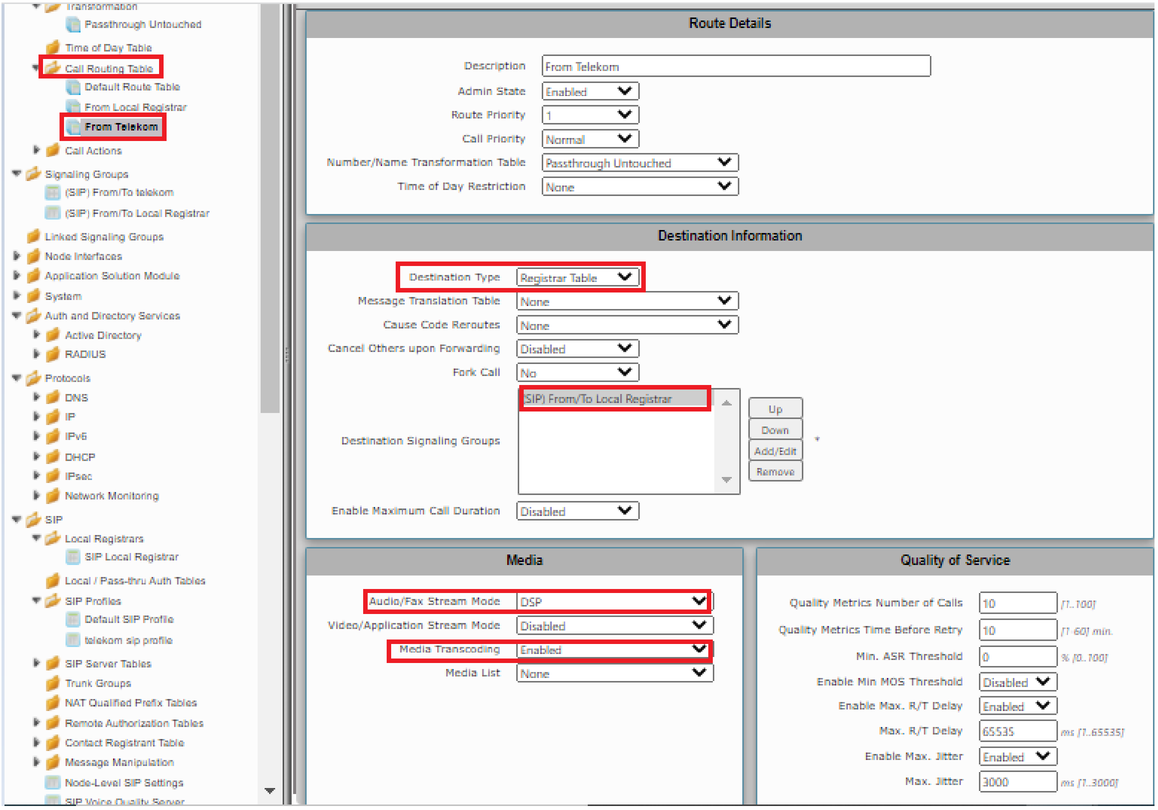
   Audio Stream Mode:
   DSP (default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support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



Call Routing for "From Telekom".



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.

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

Interoperable Vendors

Deutsche Telekom

Copyright

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

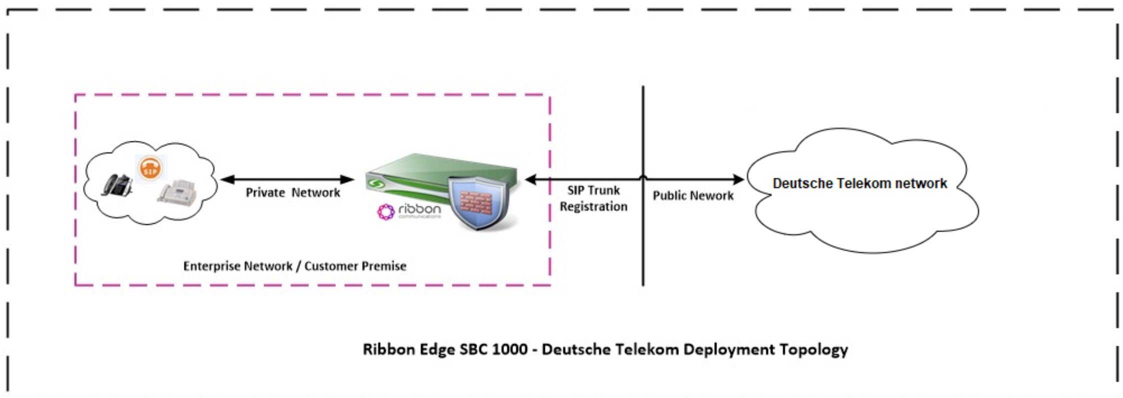
The configuration uses the following equipment and software:

Table 2: Requirements

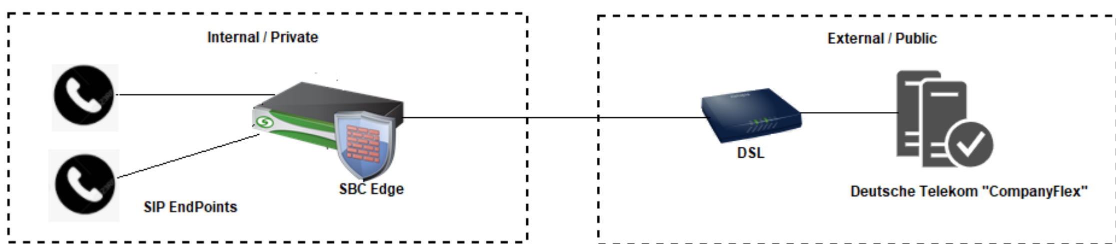
Product	Equipment	Software Version
Ribbon Networks	Ribbon SBC 1000	8.0.1
Third-party Equipment	DSL Line	NA
Deutsche Telekom	Deutsche Telekom "CompanyFlex" SIP trunks	NA
Administration and Debugging Tools	Wireshark LX Tool	3.2.7 2.1.0.6

Network Topology

SBC 1000 - Deutsche Telekom Deployment Topology

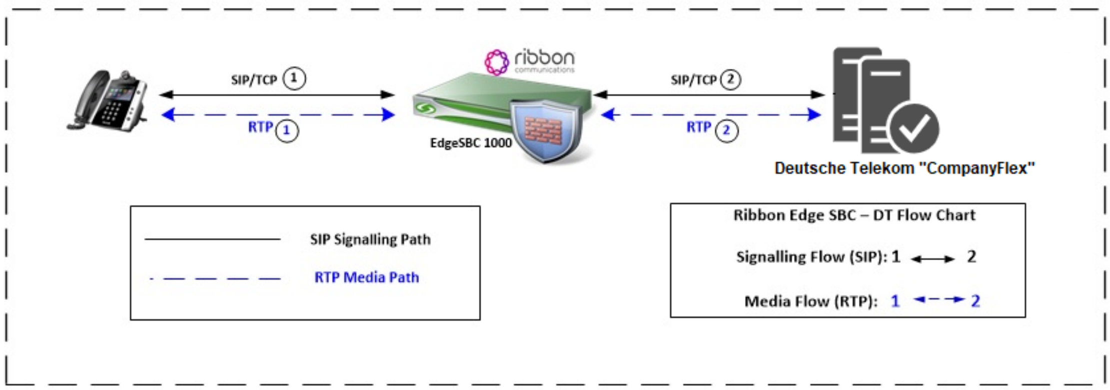


SBC 1000 - Deutsche Telekom Lab Topology



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.

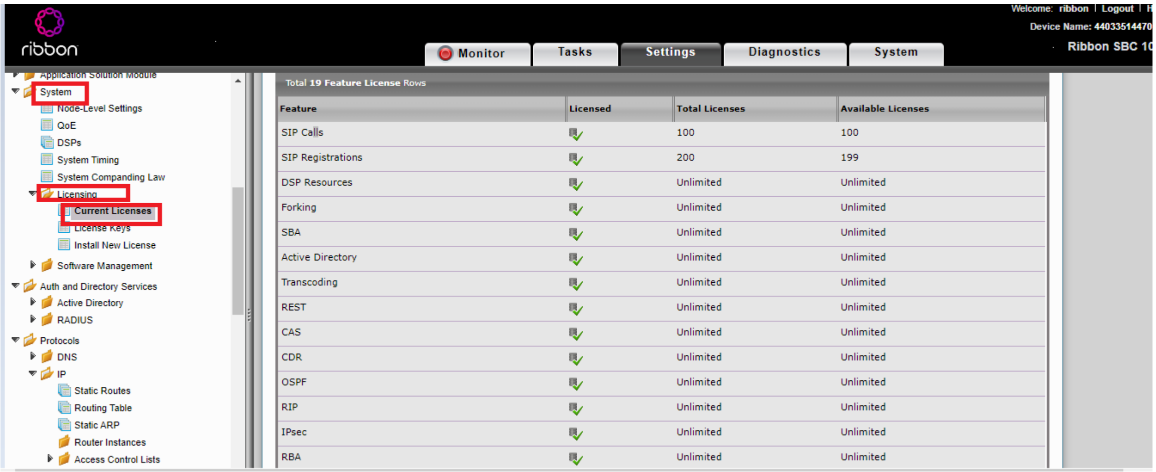
Click **Enter** and log in with a valid User ID and Password.

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.**



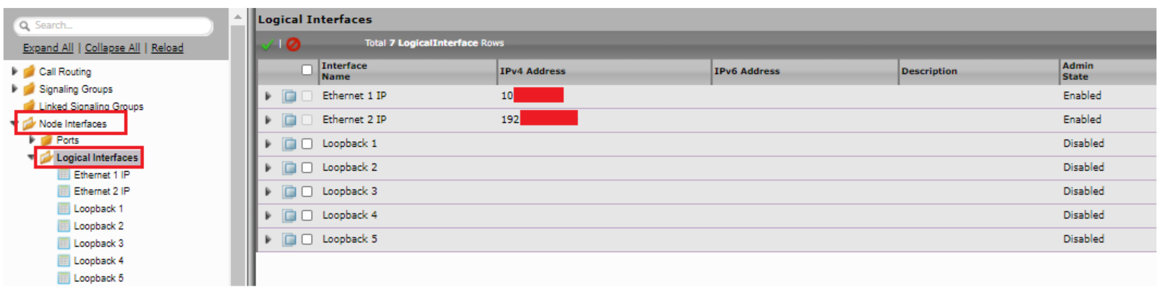
For more details on Licenses, refer to SBC 1000, SBC 2000 Licenses.

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.

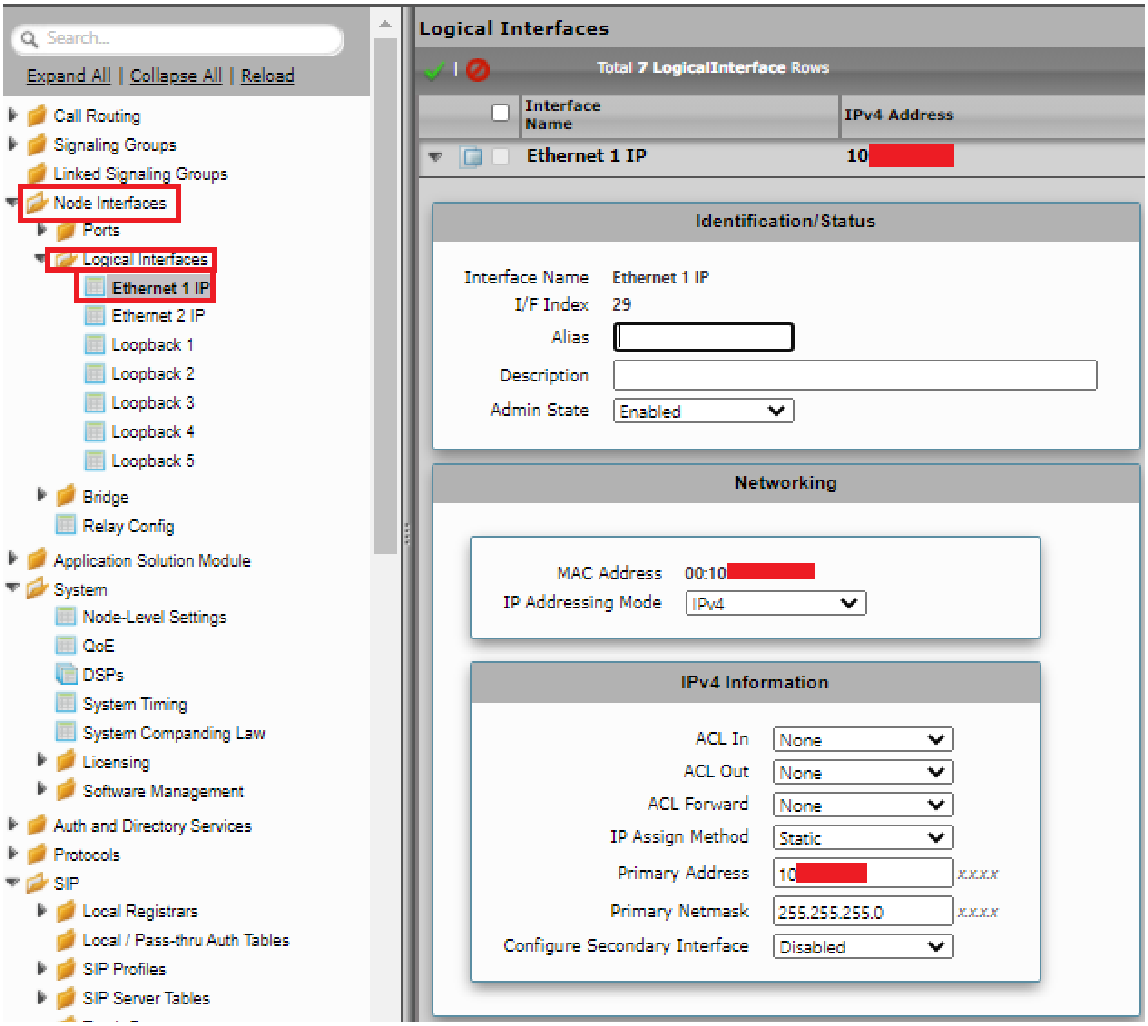
From the **Settings** tab, navigate to **Networking Interfaces > Logical Interfaces.**



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

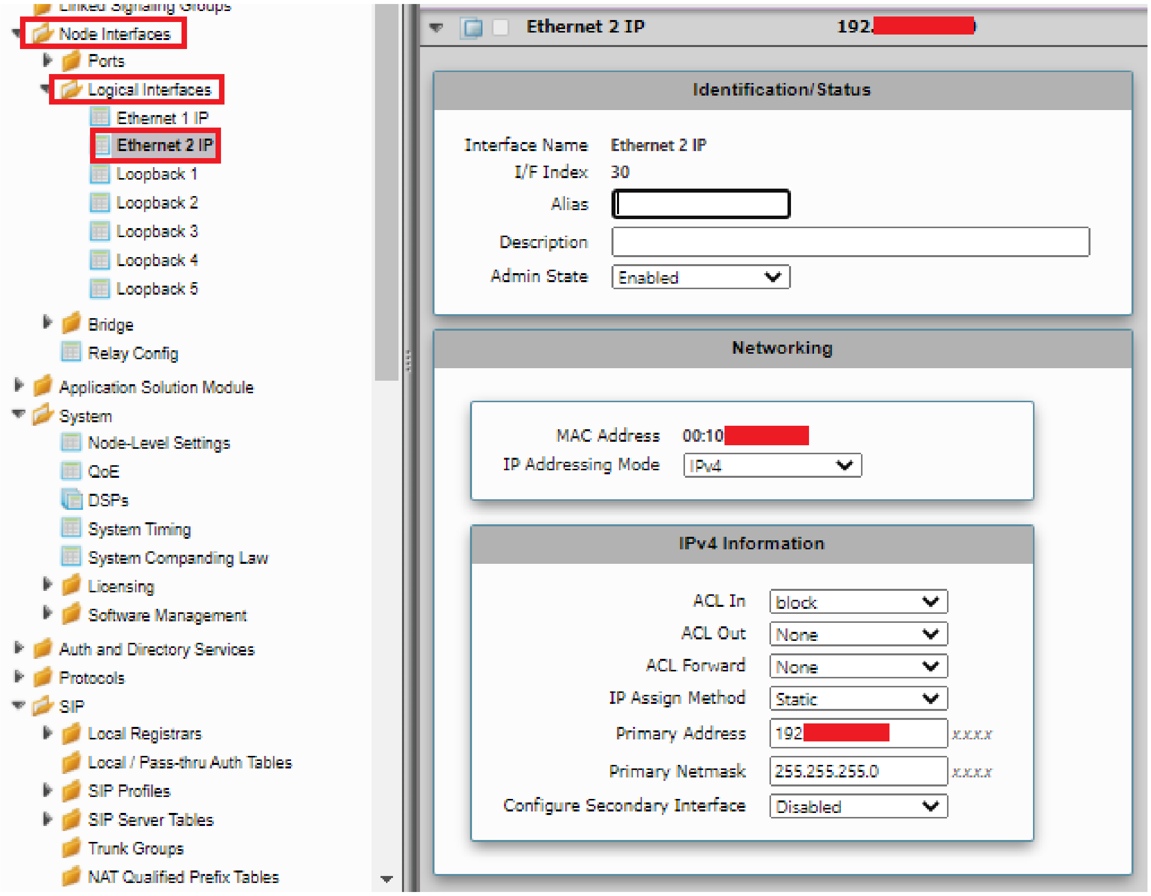
Ethernet 1 IP

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.



Ethernet 2 IP

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



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

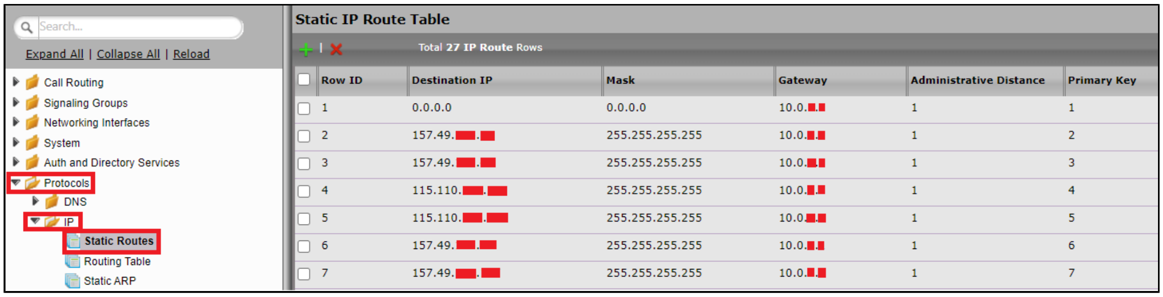
Static routes are used to create communication to remote networks. In a production environment, static routes are mainly configured for routing from a specific network to another network that you can only access through one point or one interface (single path access or default route).

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.



SBC 1000 Configuration for Access End

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

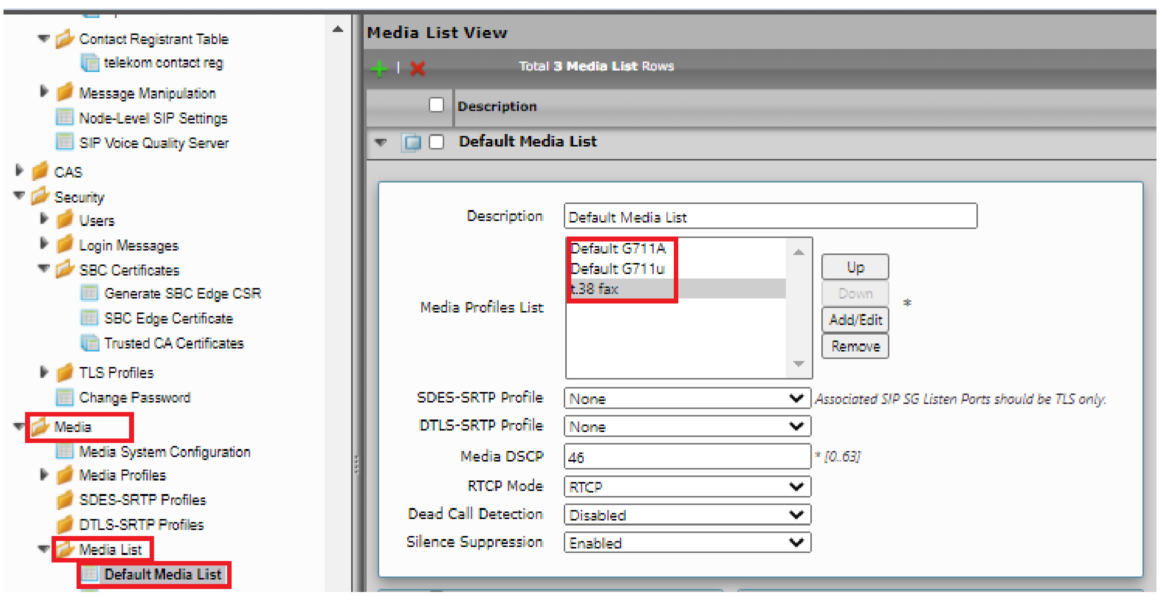
For assistance visit : https://doc.rbbn.com/

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.

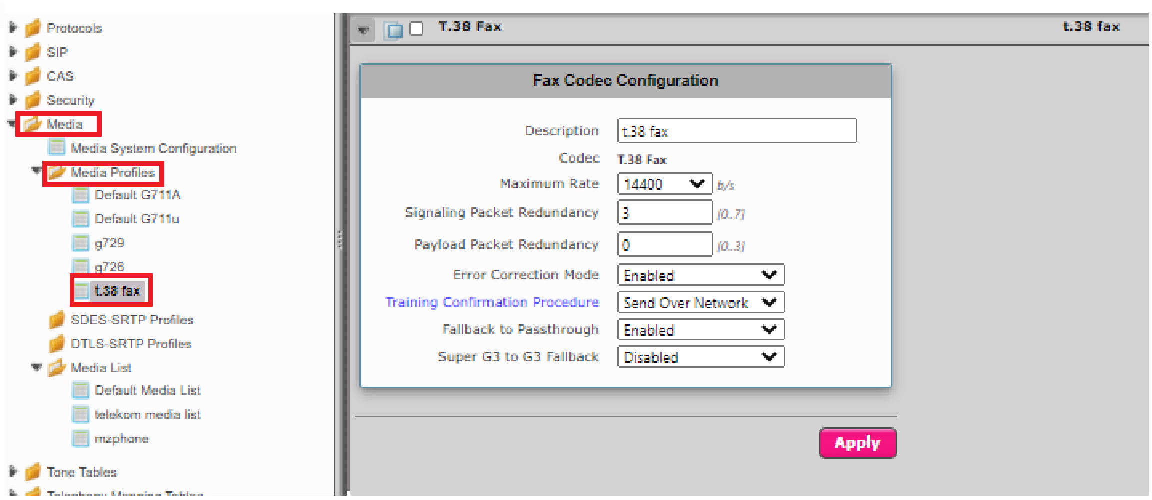
Use default media profile with codec G.711.



> ⓘ 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.

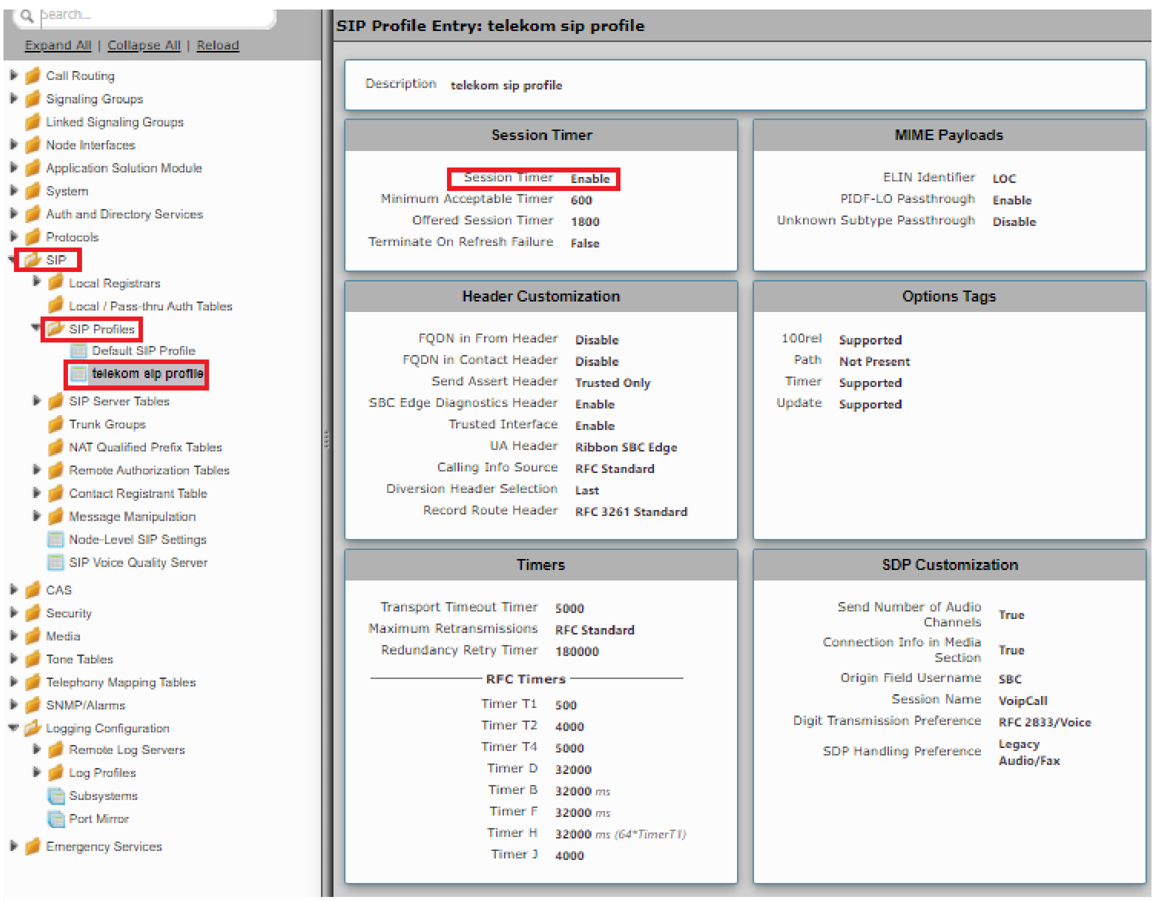


It is recommended to use a maximum packet time (max pTime) of 20ms for all Voice Codecs.

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.

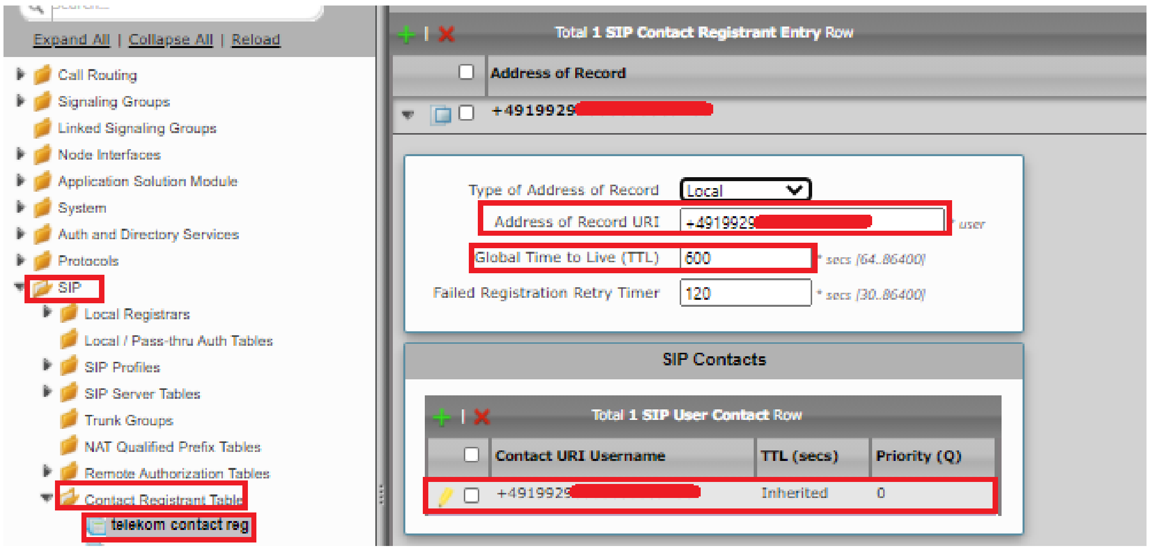


Contact Registration Table

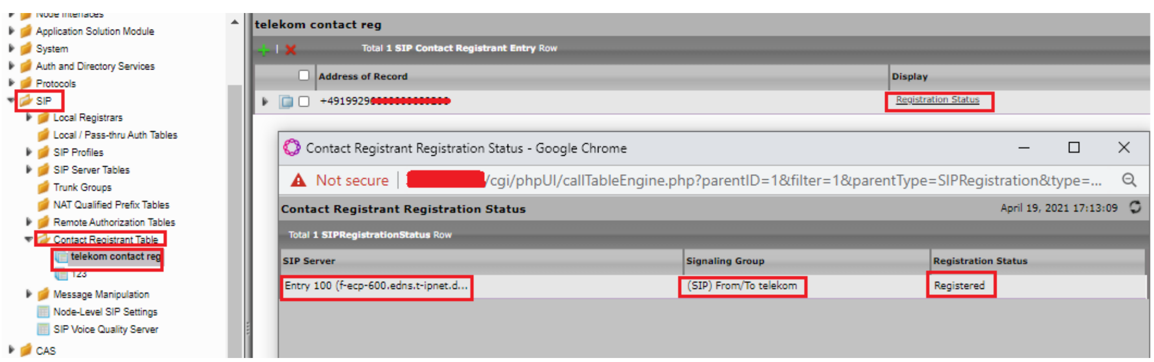
Select **Settings > SIP > Contact Registration Table**.

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".



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

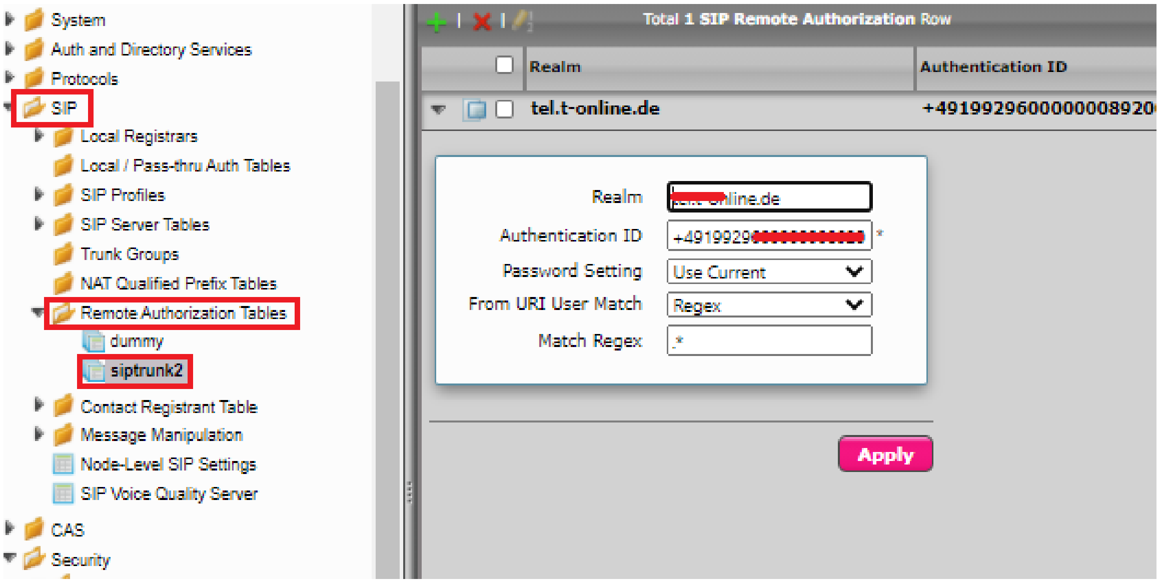


Remote Authorization Table

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".

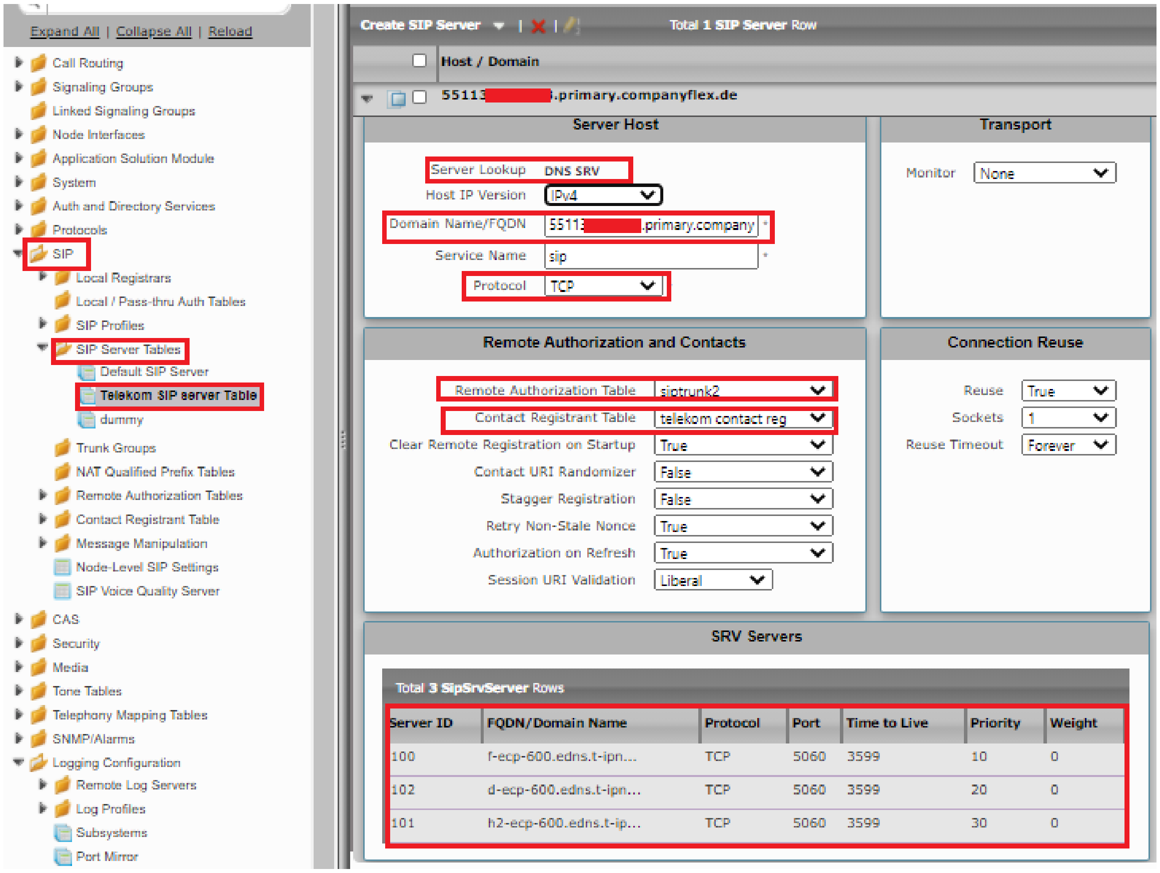


Sip Server Table

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.



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.

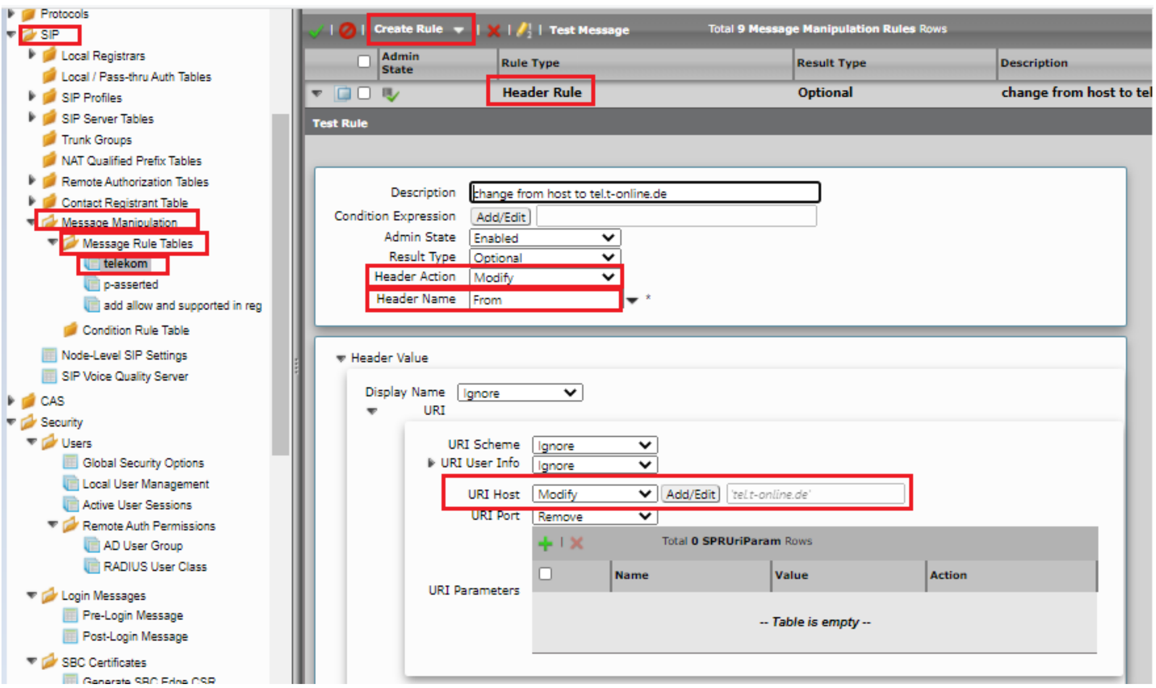
- Supported
- Allow

Select Settings > SIP > Message Manipulation > Message Rule Table

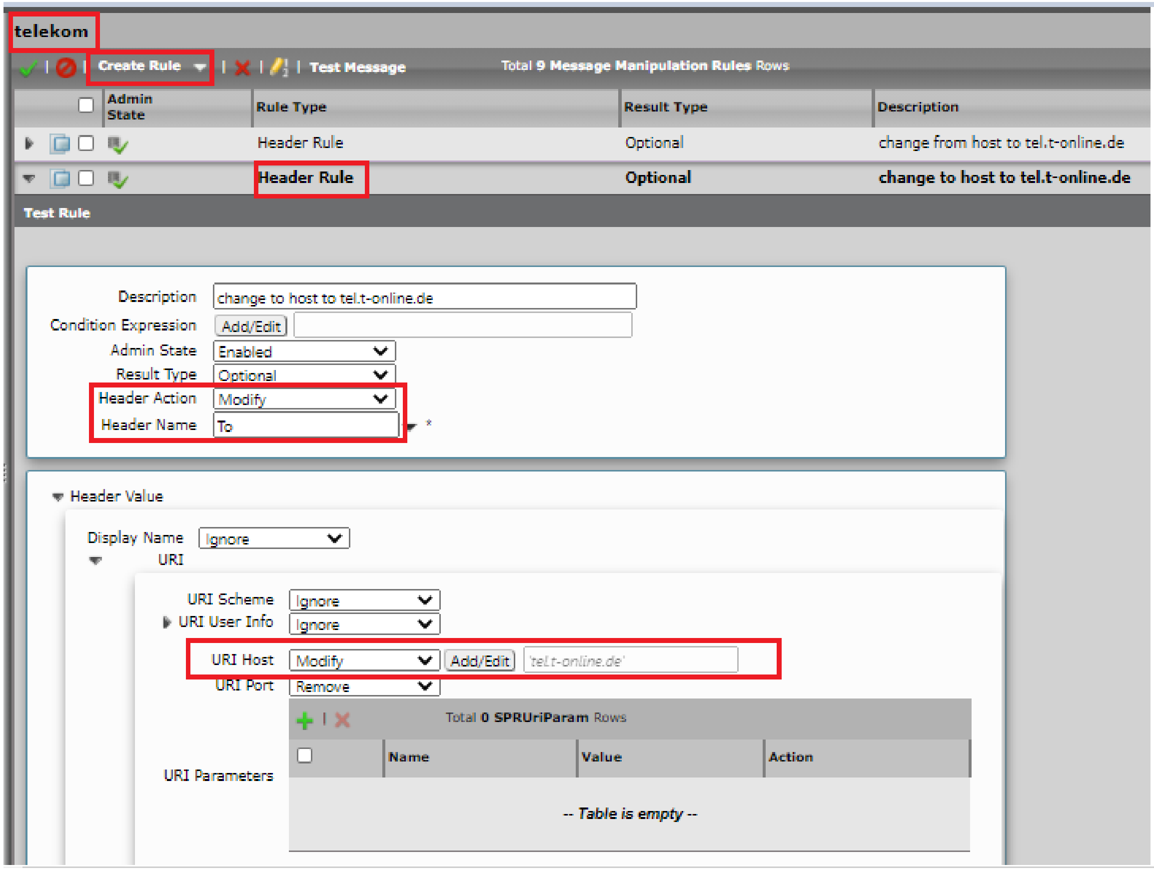
Click the Create Message Rule Table(+) icon.

Message Manipulation - From, To , Request URI sends FQDN in URI host.

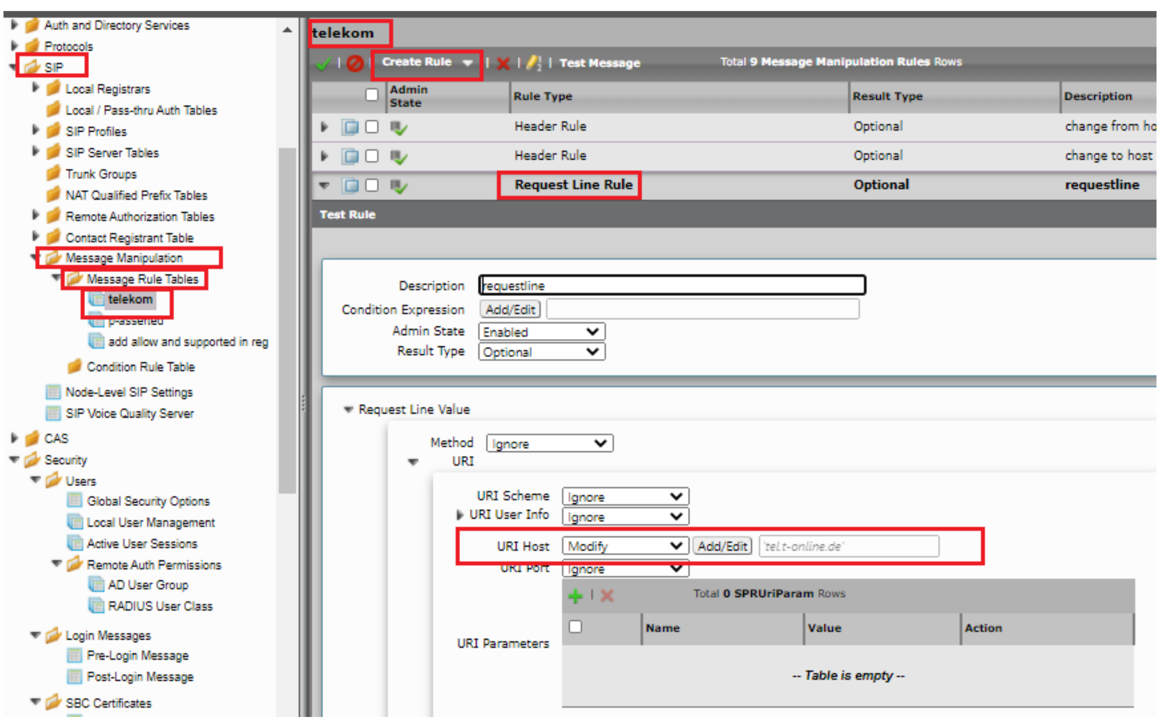
- Provide a description as "Telekom" for the Rule Table.
- Apply the SMM for All messages.
- Click the expand icon next to the Rule Table entry created.
- From the Create Rule drop-down box, select Header Rule.
- Provide the desired description.
- Provide Header action as "Modify" and header name as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



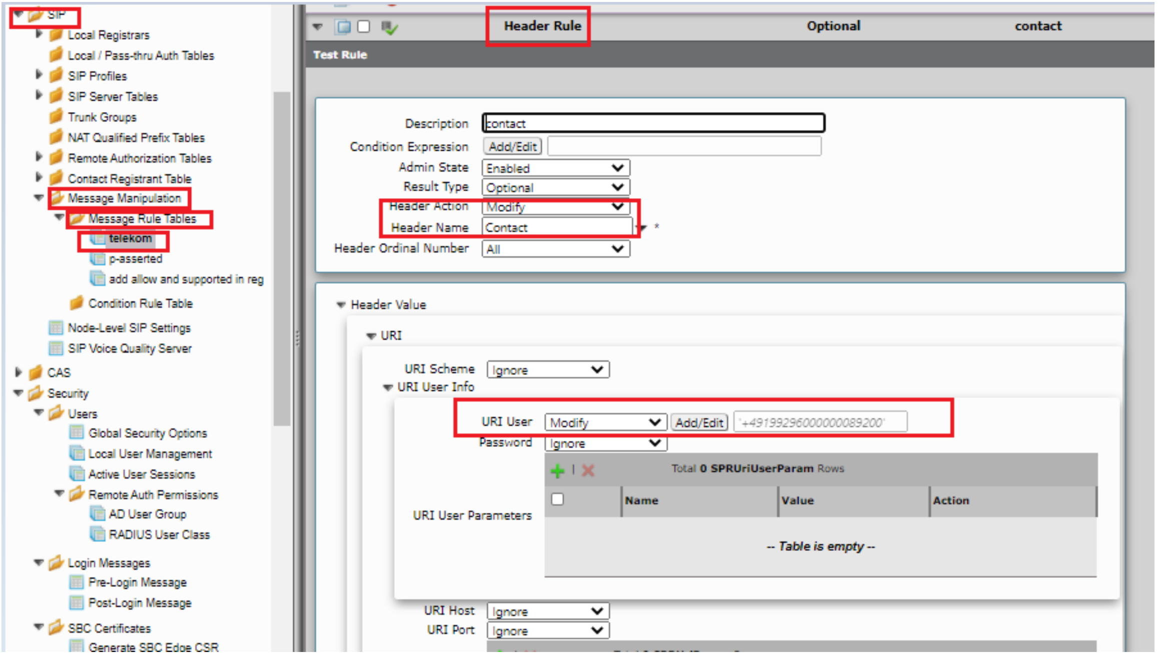
Under "Telekom" Repeat the same for To header.



Under "Telekom" repeat the same for request URI.



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



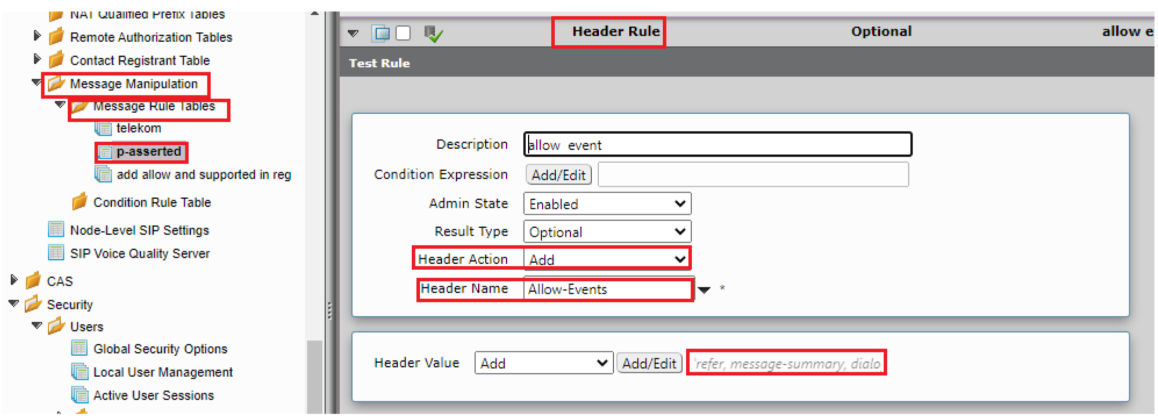
Message Manipulation - Add Allow-Events in INVITE

Click the Create Message Rule Table( ) icon.

Provide a suitable description for the Rule Table.

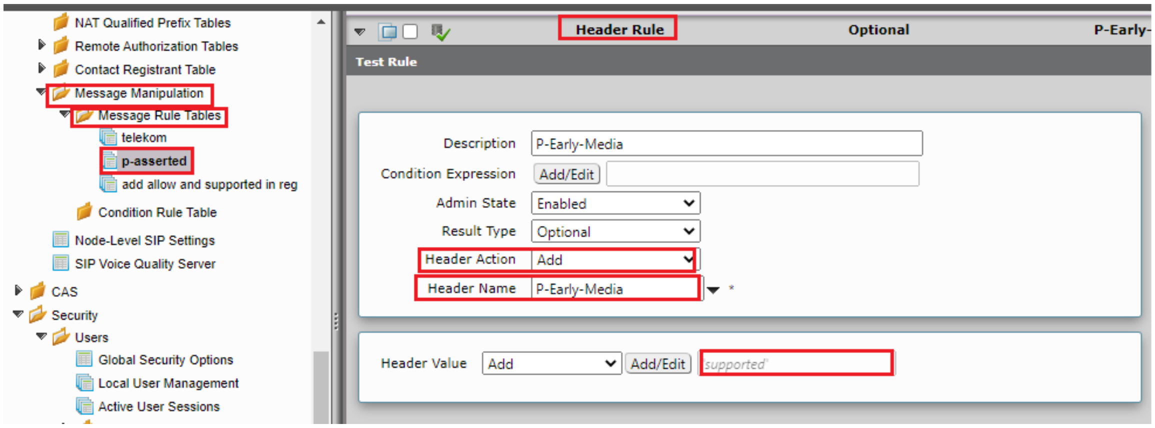
Choose "INVITE" message under Applicable Messages.

- 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**.



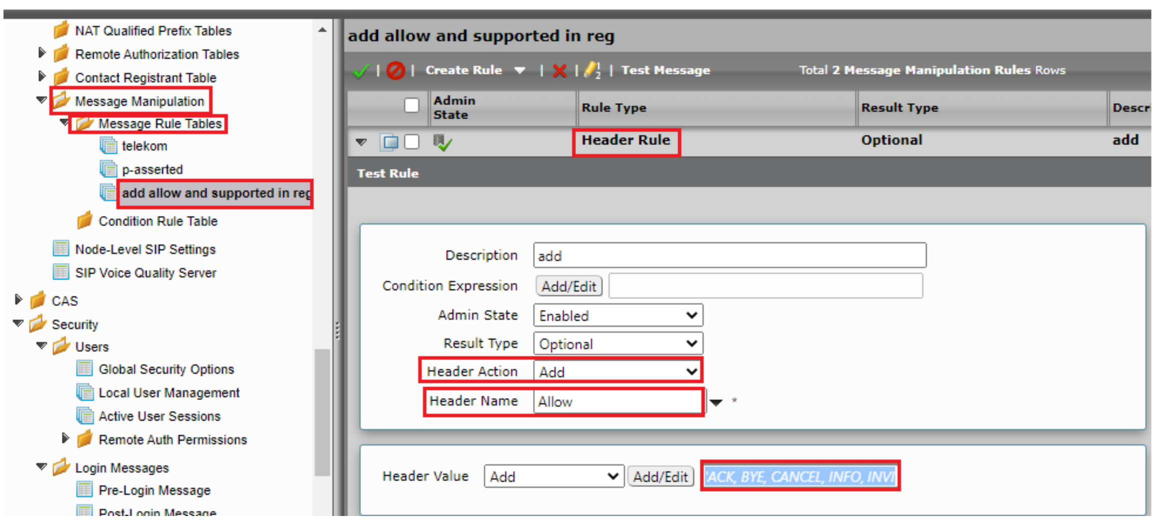
Message Manipulation - Add Allow in REGISTER

Click the Create Message Rule Table(+) icon.

Provide a suitable description for the Rule Table.

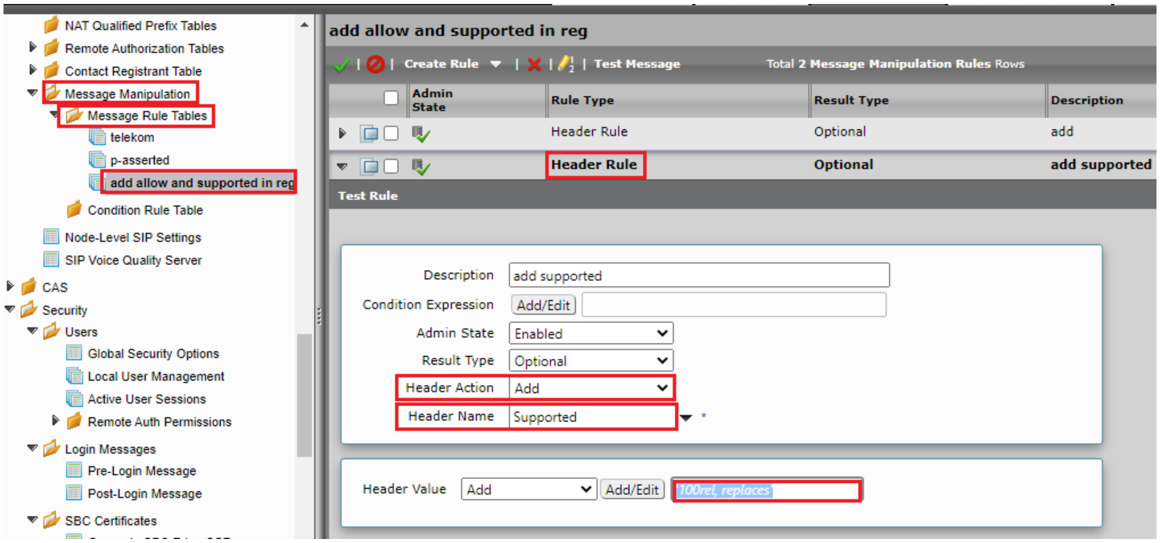
Choose "REGISTER" message under Applicable Messages.

- 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

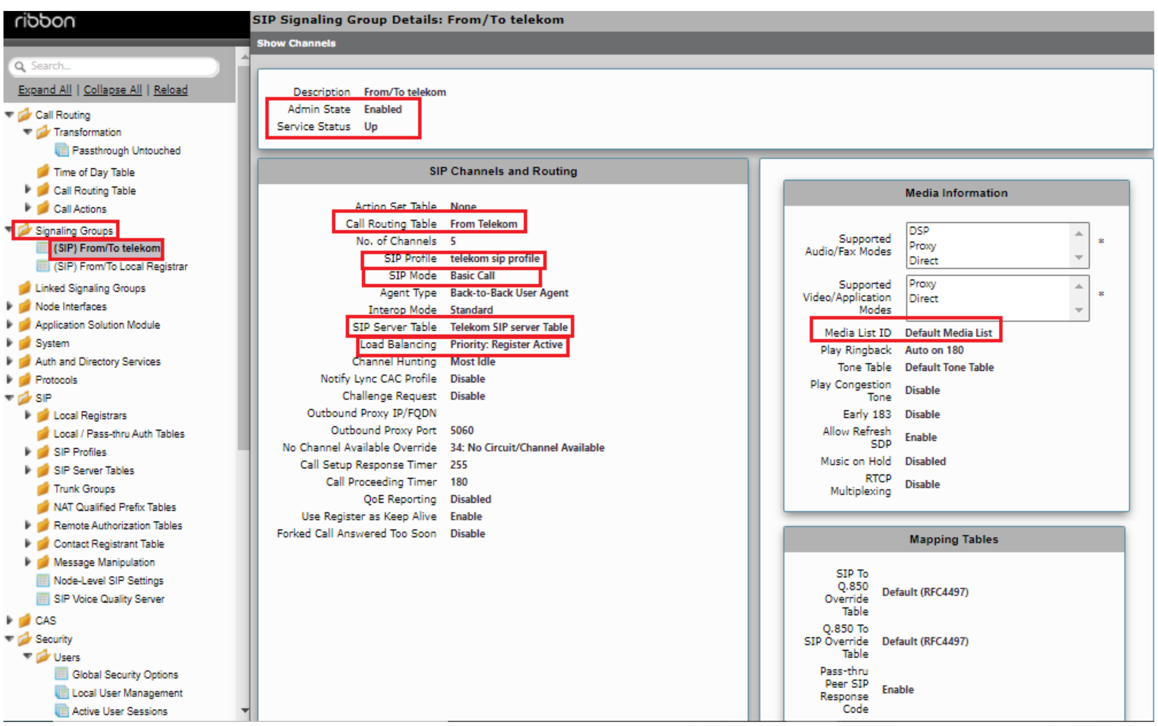
Signaling Groups allow grouping telephony channels together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected.

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".

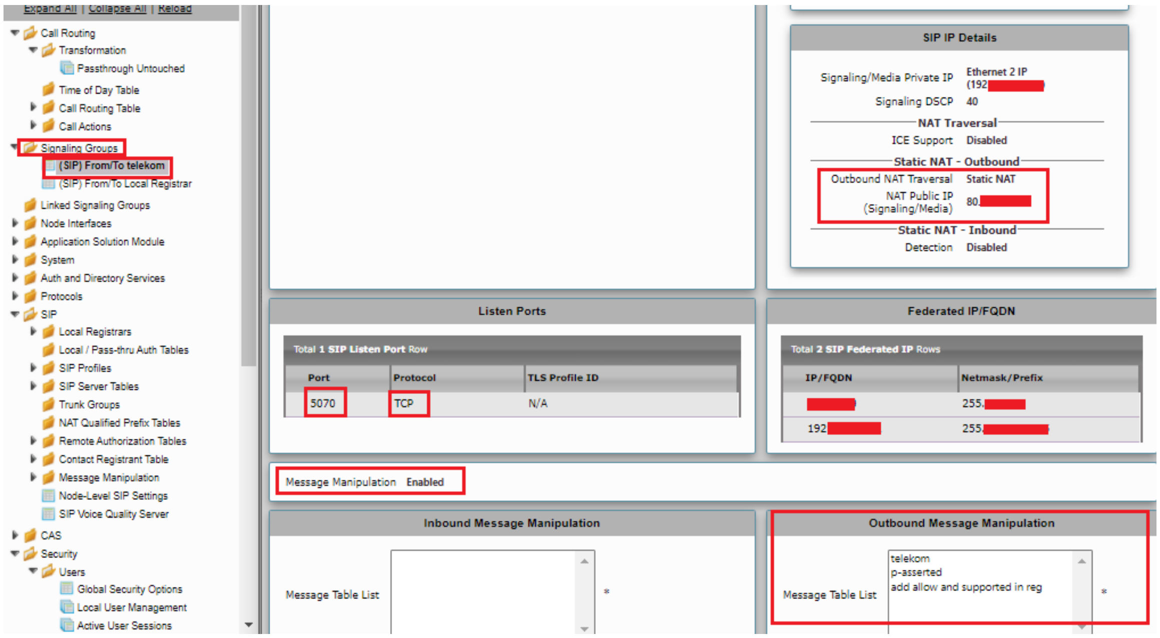
- Choose Agent type as "Back-to-Back user agent" and media list as "default media list".
- Choose SIP Server Table as "Telekom Sip Server Table".



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**.



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.

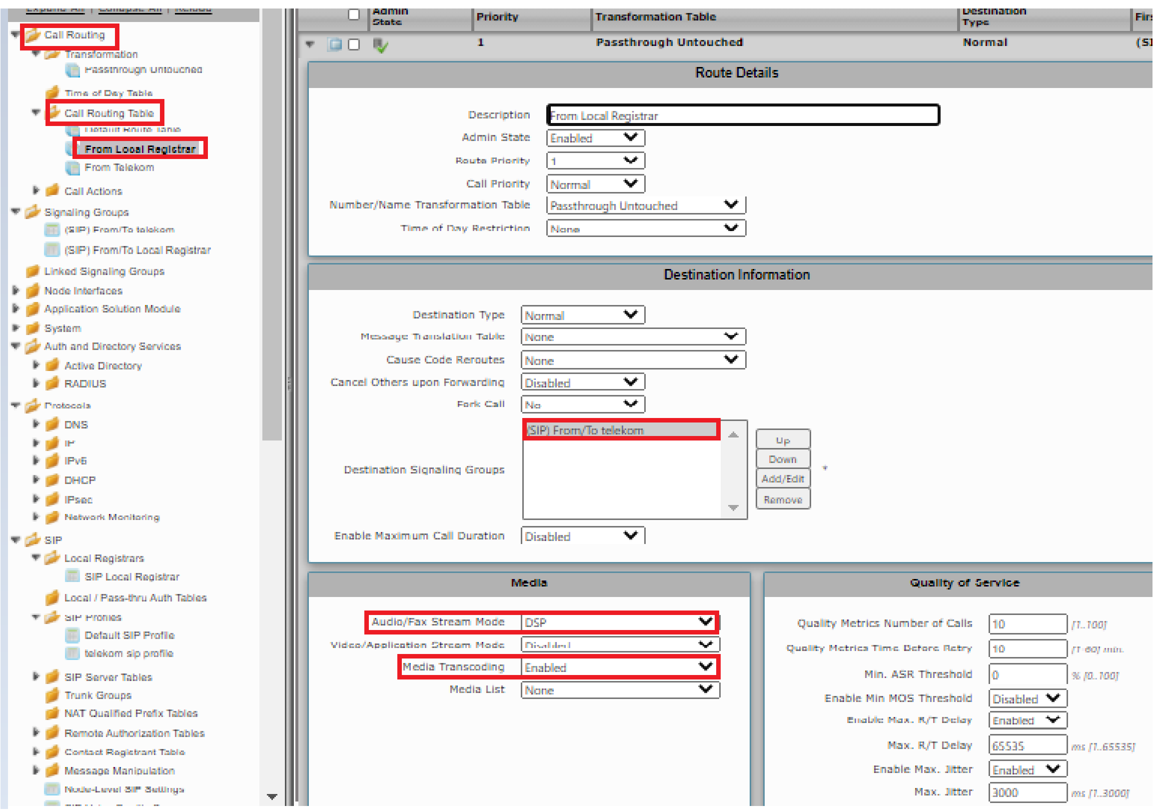
   Number/Name Transformation Table:
   Specifies the Transformation Table to use for this routing entry. This drop-down list is populated from the entries in the Transformation Table.

   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.

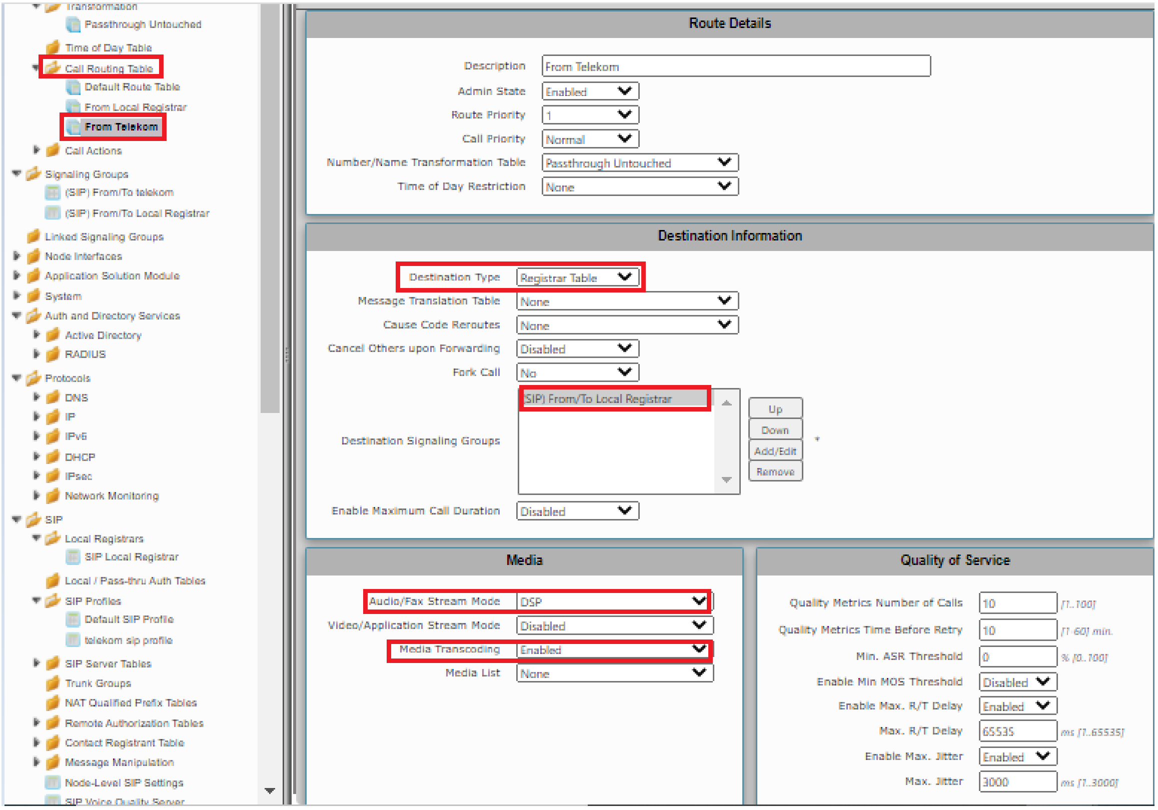
   Audio Stream Mode:
   DSP (default entry): The SBC uses DSP resources for media handling (transcoding), but does not facilitate the capabilities/features between endpoints that are not supported within the SBC (codec/capability mismatch). When the DSP is configured, the Signaling Groups enabled to support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



Call Routing for "From Telekom".



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.