

NetMatch-S CI User Guide SBC Configuration for WEBEX Calling

Document History

Document released version	Document released date	Notes
1.0	10-01-2023	First Version with NMSCI release 5.8.0-20240111
1.1	19-03-2024	Added chapter 4 (Webex Calling Side configuration) Added chapter 2 (Multi-tenant configuration on SBC side)
1.2	16-05-2024	Added paragraph 3.3 (Caveats)

Abstract

This document is the operator guide relevant to the **SBC Configuration in particular oriented to Webex Calling to PSTN interworking**.

Table of contents

1	INTRODUCTION.....	5
2	MULTI-TENANT CONFIGURATION ON SBC SIDE	5
3	MAIN ASSUMPTIONS.....	6
3.1	Media Bypass option.....	6
3.2	Call Transfer scenarios	6
3.3	Caveats.....	6
3.3.1	OPUS support	6
3.3.2	High Availability	6
3.3.3	Media Optimisation.....	7
3.3.4	Max Concurrent calls.....	7
3.3.5	Media Optimisation.....	7
3.3.6	SBC Maintenance management.....	7
3.3.7	Reference Software release and hardware requirements	7
3.3.8	Options.....	7
3.3.9	US Federal Environment	7
4	WEBEX CALLING SIDE CONFIGURATION.....	8
4.1	Multitenancy configuration	10
5	CONFIGURING ITALTEL'S SBC NETMATCH-S CLOUD INSIDE FOR NO MEDIA BYPASS SCENARIOS.....	12
5.1	Prerequisites	12
5.2	Login to the product	13
5.3	License Management Interface.....	13
5.4	Network configuration	14
5.4.1	Subnet.....	14
5.4.2	IP Interface Addresses	17
5.5	How to configure DNS Service.....	19
5.5.1	How to configure DNS/ENUM Service Manager	20
5.5.2	How to configure DNS/ENUM Interfaces	21
5.5.3	How to configure DNS/ENUM Peers.....	22
5.5.4	How to configure DNS/ENUM Routing Tables	23
5.6	How to manage Certificate	25
5.6.1	Create Certificate Signing Request (CSR).....	25
5.6.2	Update Certificate Signing Request (CSR)	27
5.6.3	Import CA Certificate	29
5.6.4	Create CA Profile.....	31
5.6.5	Create Trustiness Profile	33
5.7	How to import SIP Manipulations	35
5.8	How to create SIP Profiles	37
5.8.1	Create SIP Profiles for No Media Bypass option.....	37
5.9	How to create Media Interfaces.....	39
5.10	How to create Media Domains	42
5.10.1	Create Webex Calling Media Domain for No Media Bypass option.....	42
5.10.2	Create PSTN Media Domain for No Media Bypass option	43
5.11	How to create SIP Interfaces	45
5.11.1	Create SIP Interface for PSTN side	46
5.11.2	Create SIP Interface for Webex Calling side.....	47
5.12	How to create SIP Peers and SIP Peer Group on PSTN side.....	50
5.12.1	Create SIP Peers on PSTN side.....	50

5.12.2 Create SIP Peer Groups on PSTN side	53
5.13 How to create SIP Domains	55
5.13.1 Create SIP Domain for PSTN side for No Media Bypass option	55
5.13.2 Create SIP Domain for Webex Calling side for No Media Bypass option	61
5.14 How to create Transcoding Rules	65
5.14.1 Create Transcoding Rules from Webex Calling to PSTN	66
5.14.2 Create Transcoding Rules from PSTN to Webex Calling	67
5.15 How to create Interconnection.....	68
5.15.1 Create Interconnection from PSTN to Webex Calling for No Media Bypass option	69
5.15.2 Create Interconnection from Webex Calling to PSTN for No Media Bypass option	71

1 Introduction

This document shows how to connect Italtel's SBC named NetMatch-S CI (also referred to as simply "the Product" in the remainder of the document) to Webex Calling and refers to the Italtel SBC configuration only. For configuring **Webex Calling** side, the Cisco's cloud-calling product, please refer to <https://developer.webex.com/docs/webex-calling-overview>.

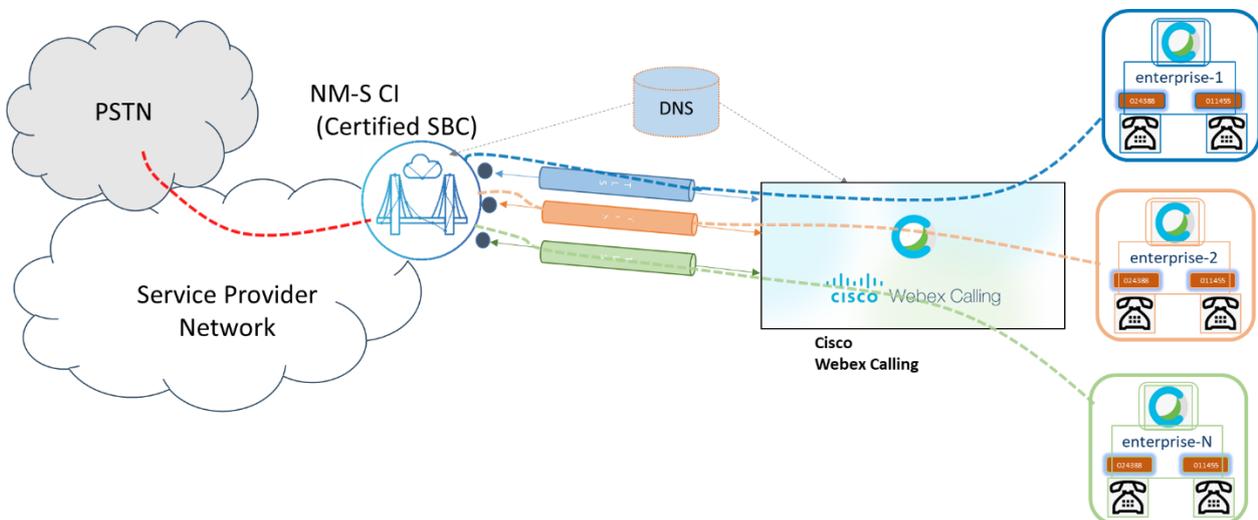
This document is meant for IT or telephony experts.

2 Multi-tenant configuration on SBC side

One of the allowed configurations foreseen by Cisco is to have an SBC certified with Webex Calling interoperability and hosted in the SP network, whilst providing PSTN interworking to the connected Enterprises' Webex Calling tenants.

The Italtel SBC supports this configuration and offers PSTN connectivity through the SIP trunking functionality, enabling Webex Calling to be used as office phone system.

In this scenario the SBC can be hosted in a Service Provider's network, serving in a centralized way the PSTN voice interworking service for the connected Enterprises' Webex tenants. The SBC can virtualized in "slices" (SIP Interfaces) and each one of them, dedicated to one enterprise, has a dedicated IP Address announced on the public network towards Cisco Webex Calling platform: each instance of SBC, at a configuration level, is split into several interfaces (each one of them with its own "IP:Port" socket and its own TLS certificate) shown externally to the public network.



3 Main Assumptions

3.1 Media Bypass option

The product has been certified for the Non-Media Bypass option scenarios “without ICE media path optimization”, refer to Cisco documentation:

<https://www.cisco.com/c/dam/en/us/td/docs/solutions/CVD/Collaboration/hybrid/AltDesigns/PA-WbxCall.pdf>

3.2 Call Transfer scenarios

As per the Call Transfer Scenarios, the product has been certified without supplementary services SIP REFER enable on NetMatch-S CI, refer to Cisco documentation:

https://help.webex.com/en-us/article/jr1i3r/Configure-Local-Gateway-on-Cisco-IOS-XE-for-Webex-Calling#id_100573

The protocol validation option considered for Refer method should not be present in “Allow” SIP header received from the SBC.

3.3 Caveats

3.3.1 OPUS support

NetMatch-S CI supports OPUS with SHA1_80 encryption. GCM_256 encryption is not supported at the moment

3.3.2 High Availability

The product can be deployed either in High Availability (HA) mode or in a Single (or noHA) mode.

From the perspective of the VNFC's (the subcomponent it is made of), when deployed in HA mode, they can be deployed in 1+1 scheme or in 2N scheme or N+1 (N+M) scheme.

1+1 scheme means 1 is working and the other one is in stand-by.

2N scheme means the component are all in Active mode (i.e. working) and there will be double the number of strictly needed components so that:

- in normal mode the components work with half of their capacity
- if a server fails, half of the components will be working still capable of serving the whole traffic.

Let's call it overprovisioning and this case the overprovisioning is $N+N = 2N$. This scheme applies to the deployment over a couple of hardware support (e.g. blades or servers).

In a more complex datacenter where each component can be installed with complex anti-affinity rules and thus over numerous blades / servers, the N+1 mode can be adopted, meaning an overprovisioning of 1 component (all Active). The system will be dimensioned so that the traffic can be supported by N subcomponents and can face a single subcomponent failure, with no service disruption. Also N+M scheme can be applied with M as the overprovisioning level.

3.3.3 Media Optimisation

Media Optimisation is under test and will be available next releases.

3.3.4 Max Concurrent calls

The product can be deployed with different sizes and can scale from 50 sessions to 20k+, depending on the needs.

3.3.5 Media Optimisation

The solution is deployed within the Service Provider's network and be used as a multi-tenant SBC solution where customer is offered with PSTN access through a local gateway.

3.3.6 SBC Maintenance management

If the product is deployed in HA mode (which is the most used model), each functionality (SIP, Media, Operation and Maintenance) is redundant and can survive to the fault of the working element. The software upgrade of such a network element can be done with no service disruption, by orchestrating the change of each redundant functions. This way there's no need of putting the network element in maintenance mode (therefore it is transparent to the Wx Calling platform).

Anyway, if needed, the SBC can be put in maintenance mode for any reason, at SIP Interface level (i.e. trunk). That case from the Wx Calling point of view the IP Address exposed by the product will answer to the Options with 503 response message.

Same apply in case of license expiration (503 to any request).

3.3.7 Reference Software release and hardware requirements

The reference release supporting Wx Calling IW is 5.8 and upper.

If the product is deployed as a virtual application in a DC the minimum requirements are 8vCPU and 8GB RAM. The product is scalable to higher performances with different footprints in terms of vResources.

3.3.8 Options

The SBC marks the Webex Calling node as down solely based on the response to OPTIONS and not for the INVITE messages.

3.3.9 US Federal Environment

The platform is not tested for US Federal Environment.

4 Webex Calling side configuration

Before to configure Netmatch SBC, if is necessary to configure the trunk on Control Hub, according to the guidelines in the following link: [Configure trunks, route groups, and dial plans for Webex Calling](#)
 The required steps are:

- Add a “certificate based” Trunk

Add Trunk

Location
 This location is where the trunk is physically connected. To create a new location, visit the [Locations](#) page.

HQ - Milano Caldera

Name
 TrunkNMS

Trunk Type
 Choose the right trunk type for this local gateway. [Learn more on trunk type](#)

Certificate based

Device Type
 Cisco Unified Border Element

(the Device Type should be “Italtel Netmatch-S”)

Enterprise Session Border Controller (SBC) Address
 Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC.
 You must have the domain for your SBC's FQDN/SRV claimed or verified [before you can use this address.](#) [Manage your domains](#)

FQDN
 SRV

Hostname * Domain * Port *

ItaltelNMS italtel.com 5061

Valid address

FQDN
 ItaltelNMS.italtel.com:5061

Maximum number of concurrent calls *

The trunk is successfully created:

Add Trunk



TrunkNMS Successfully Created.

Visit [Route Group](#) page to add trunk(s) to a route group.
 Visit [Locations](#) page to configure PSTN connection to individual locations.
 Visit [Dial Plans](#) page to use this trunk as the routing choice for a dial plan.

Trunk Info

Status ⓘ

- Unknown

Webex Calling edge proxy address (FQDN)

peering1.eun.sipconnect.bclid.webex.com:5062
 peering2.eun.sipconnect.bclid.webex.com:5062
 peering3.eun.sipconnect.bclid.webex.com:5062
 peering4.eun.sipconnect.bclid.webex.com:5062

Webex Calling edge proxy address (SRV)

eun01.sipconnect.bclid.webex.com

On NMS configuration it will be used the SRV configuration, as Cisco recommends.

The trunk is created and can be associated to a given location on Webex Calling

Calling

Numbers
Virtual Lines
Call Routing
Managed Gateways
Features
PSTN
Service Settings

Trunk
Route Group
Dial Plans
Verify Call Routing
Zone
Trusted Network Edge

Trunk

SIP trunks provide connectivity to a customer-owned PSTN service and to an on-premises IP PBX deployment. These were previously accessed via the Local Gateway configuration page.

Name	Location	Trunk Type	In Use
TrunkNMS	HQ - Milano Caldera	Certificate based	No

4.1 Multitenancy configuration

It is possible to configure multitenancy, on Webex Calling side, in two different ways

- Different IP different FQDN

Ex Customer 1 FQDN : Customer1.italtel.com (IP 138.132.80.80)

Enterprise Session Border Controller (SBC) Address

Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC.

You must have the domain for your SBC's FQDN/SRV [claimed or verified](#) before you can use this address. [Manage your domains](#)

FQDN

SRV

Hostname *

Customer1

Domain *

italtel.com

Port *

5061

Valid address

FQDN

Customer1.italtel.com:5061

Ex Customer 2 FQDN : Customer2.italtel.com (IP 138.132.80.90)

Enterprise Session Border Controller (SBC) Address

Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC.

You must have the domain for your SBC's FQDN/SRV [claimed or verified](#) before you can use this address. [Manage your domains](#)

FQDN

SRV

Hostname *

Customer2

Domain *

italtel.com

Port *

5061

Valid address

FQDN

Customer2.italtel.com:5061

- Same IP (different port), different FQDN

Ex Customer 1 FQDN : Customer1.italtel.com (IP 138.132.80.80:5061)

Enterprise Session Border Controller (SBC) Address

Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC.

You must have the domain for your SBC's FQDN/SRV [claimed or verified](#) before you can use this address. [Manage your domains](#)

FQDN

SRV

Hostname *

Domain *

Port *

Valid address

FQDN

Customer1.italtel.com:5061

Ex Customer 2 FQDN : Customer2.italtel.com (IP 138.132.80.80:5062)

Enterprise Session Border Controller (SBC) Address

Select the type and enter an FQDN or SRV address for Webex Calling to reach out to your Enterprise SBC.

You must have the domain for your SBC's FQDN/SRV [claimed or verified](#) before you can use this address. [Manage your domains](#)

FQDN

SRV

Hostname *

Domain *

Port *

Valid address

FQDN

Customer2.italtel.com:5062

5 Configuring Italtel's SBC Netmatch-S Cloud Inside for No Media Bypass scenarios

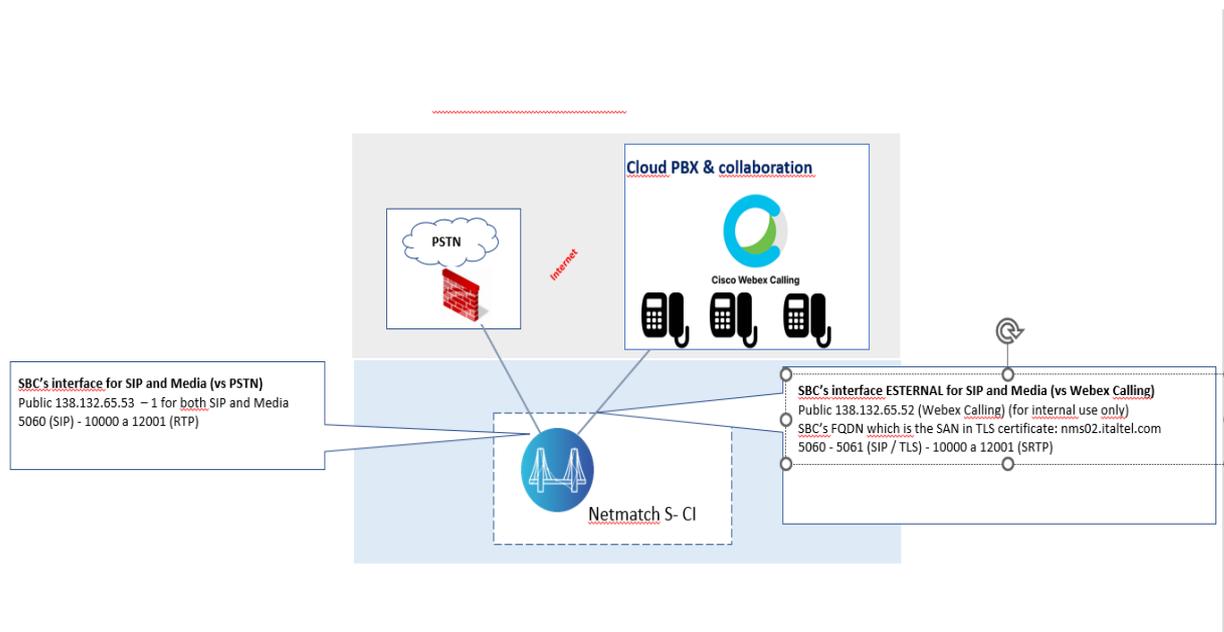
This section shows how to configure Italtel' SBC for interworking with Webex Calling. Some hints and GUI snapshots relevant to a sample case of configuration are inserted in the document to ease the explanation of the configuration and some fields filling: they are highlighted in **yellow** background or shown in tables. They are based on the example scheme shown below in section 5.1.

5.1 Prerequisites

Before you begin the configuration, make sure you have the following information available for the product you want to pair:

- Public IP address
- SRV Proxy name
- Public certificate
- Low Level Design

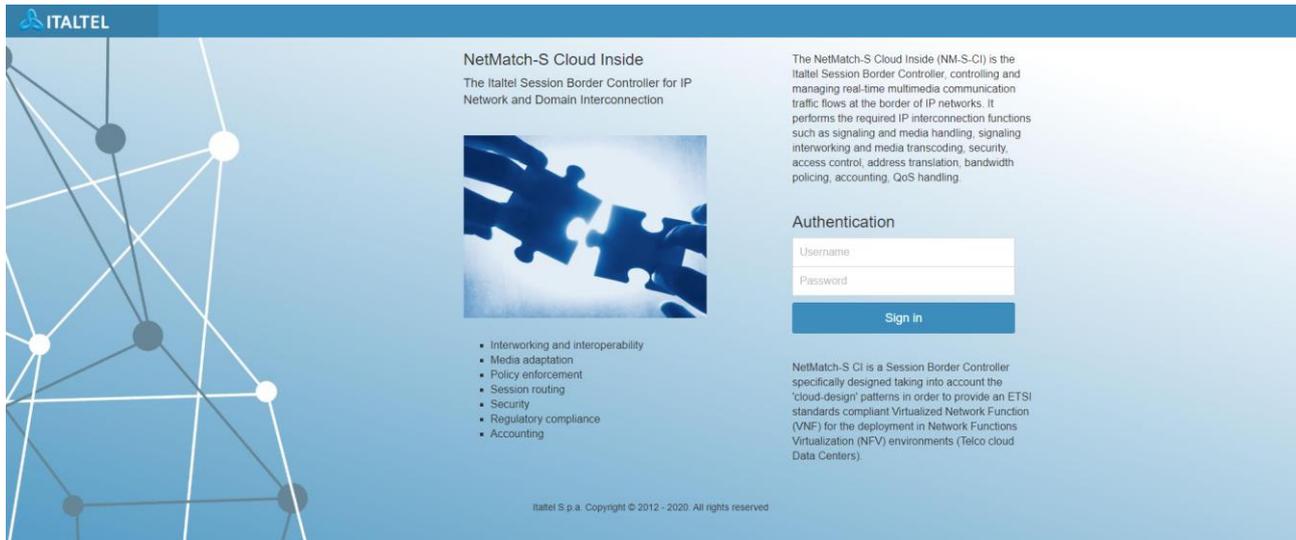
Here below an example of LLD is shown for No Media Bypass.



5.2 Login to the product

NetMatch-S CI provides an Advanced Graphical User Interface (GUI) through HTTP/HTTPS connection using a dedicated management address.

Type the URL of NetMatch-S CI (e.g. <https://138.132.66.69:8443/NMSCI-WebGui/>) in your browser to access the GUI.



Type username and password in the **Authentication** form and click **Sign in** to log into the system.

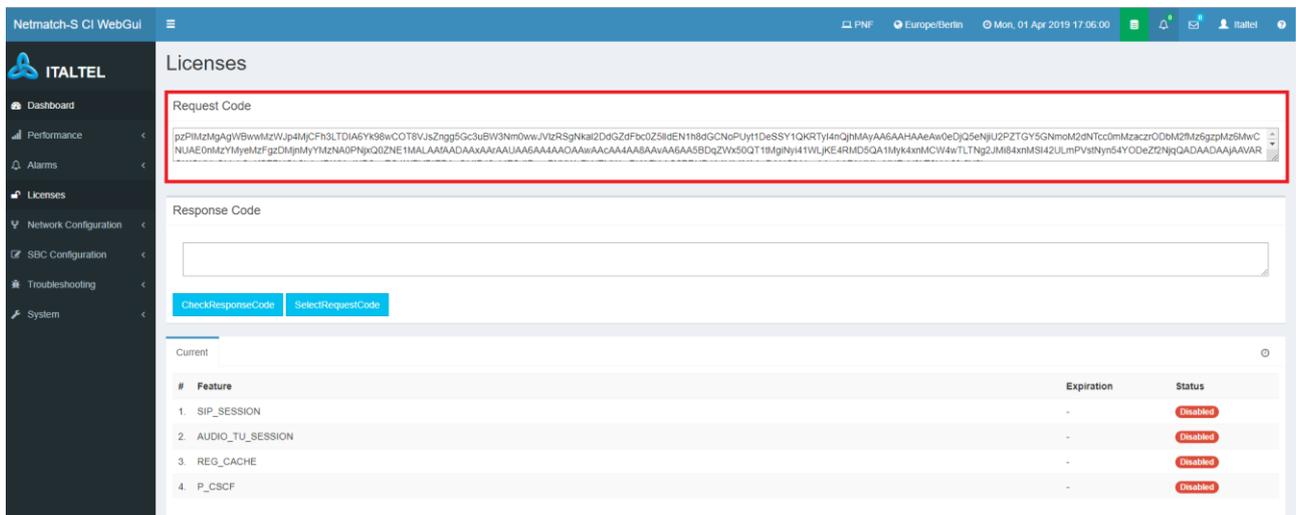
5.3 License Management Interface

The **License Management Service** enables the product to deliver specific features, referred to the purchase made by the customer on the set of features that the product is able to offer.

After the product has been installed, no default license is available; therefore, the product will not offer any functionality.

To obtain a valid license, follow the steps below:

- Click **Licenses** in the left side menu, to open the License information page.



- Copy Request Code and send it to Point of Contact
- Insert the Response Code, obtained by Italtel reference, in the Response Code Box:

Licenses

Request Code

vYpaMzMgAgWFwwMDWJ25Mz5BhCAH7Ns6Qm3tMANnCIKZzJPB0YMDxcz5MnoJ4ha0Dg5CNjXBOZXPXNGYXJRj8aFmMtxQ0ZktVki5GzAAIAAxAAAAoAA0AAhAA4AAwAAwAwMeDcQ5h
 NJVU2oZTDY5PNm5M2SNTDco6NDnly1NTDAwGRj9FBeQkgZD3MUnYzLQUCE37QjChFwMj3M23MDBc4zQjIRC6MTCEKPAAPAApAAHAAcAPxsEAAEAaAKAA6AAvpOAAAsAAVAAgAAtBDFZ

Response Code

rKFKvWMOcrlcJvIIPOc2q2ASIM8H02KFOteHxiGHFSBDTLed4i8HxnKbWUMMMfsWUYUOBxUabOOHHVRI3wYnSDotAVnJhUc8HItPBxluZ7FHDMc2XsD9yGDazWL97IyAfnyJtPpCtDO6bvm
 JSdb3e4c4Y7qaUMNjX7OTu4WMjBtOIHxjtEsulsDggMhIKG7ul65Xx8QA3SuHqmbSNXD3HckH83f3247d

CheckResponseCode
SelectRequestCode

- Click “CheckResponseCode” button and verify the License in now ACTIVE with the following Features MTF, SIP_SESSION, AUDIO_TU_SESSION, TLS, WEBEX_CALLING, SRTP:

#	Feature	Expiration	Status
1	MTF	2000-01-01 00:00:00	Enabled
2	SIP_SESSION	2000-01-01 00:00:00	1000
3	AUDIO_TU_SESSION	2000-01-01 00:00:00	100
4	REG_CACHE	-	Disabled
5	P_CSCF	-	Disabled
6	TTL	-	Disabled
7	CDR	-	Disabled
8	QOS	-	Disabled
9	TLS	2000-01-01 00:00:00	Enabled
10	ATCF	-	Disabled
11	MSTEAMS	-	Disabled
12	ARNT	-	Disabled
13	SRTP	2000-01-01 00:00:00	Enabled
14	SDP_SCREENING	-	Disabled
15	SYS_UM	-	Disabled
16	SIPREC	-	Disabled
17	WEBEX_CALLING	2000-01-01 00:00:00	Enabled

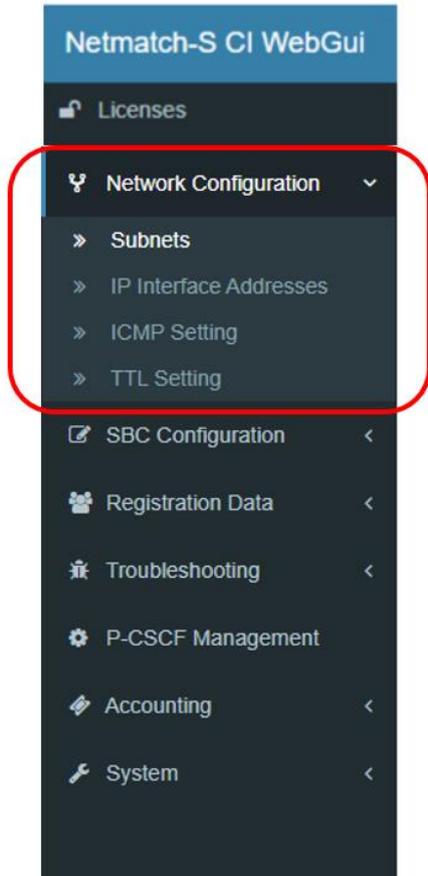
5.4 Network configuration

In the product, you have to configure subnets to be used for SIP signaling and media.

5.4.1 Subnet

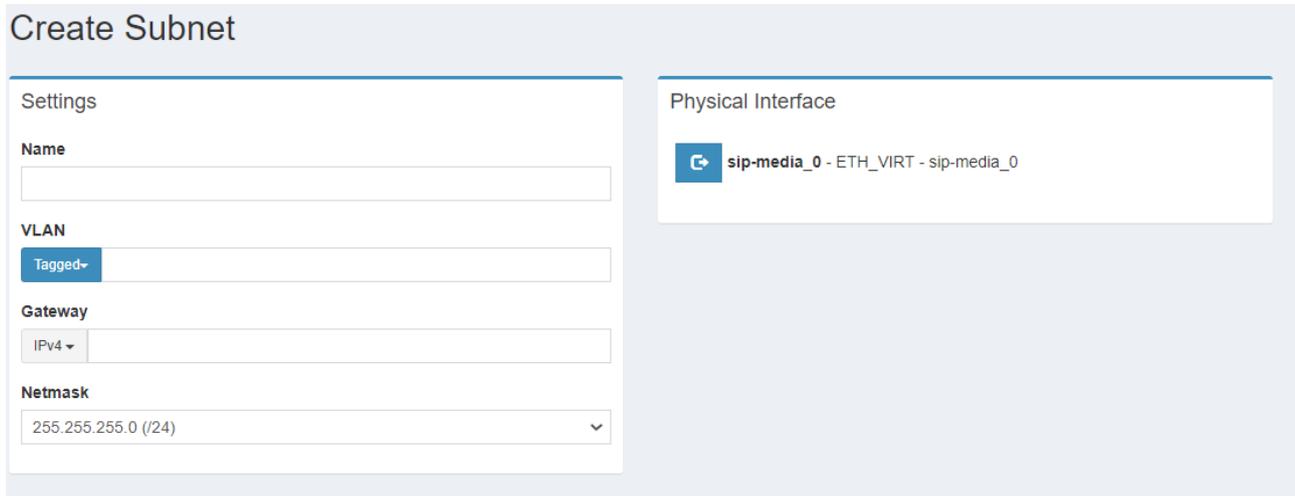
To configure the subnets, follow the steps below:

- Click **Network Configurations/Subnet** in the left side menu, to open the configuration page:

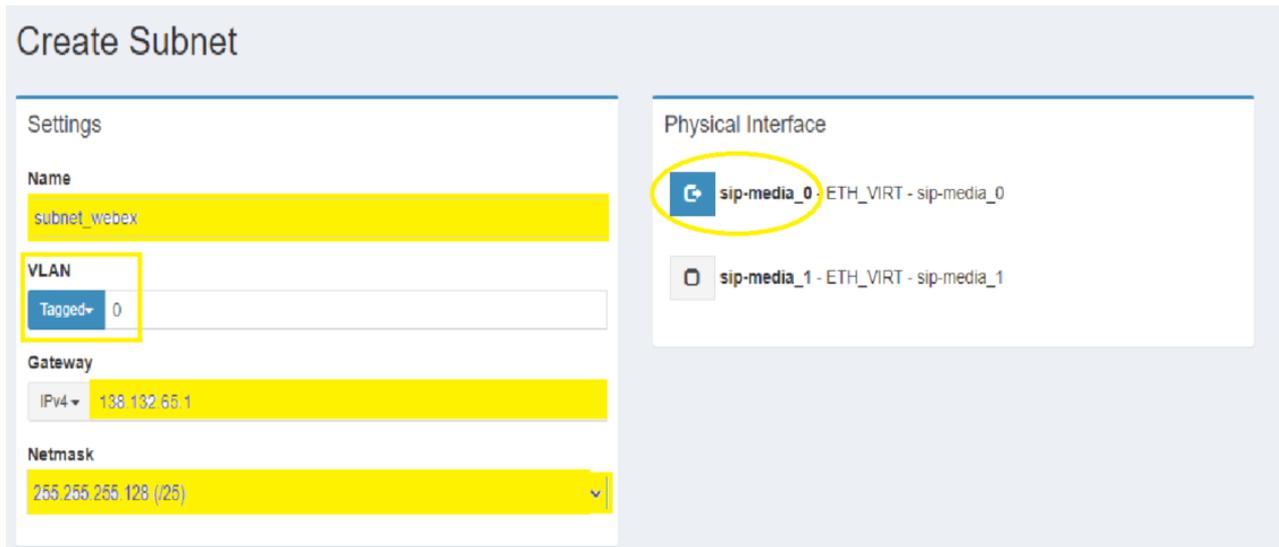


Then to access the **Create Subnet** view, click on [+ New](#) button

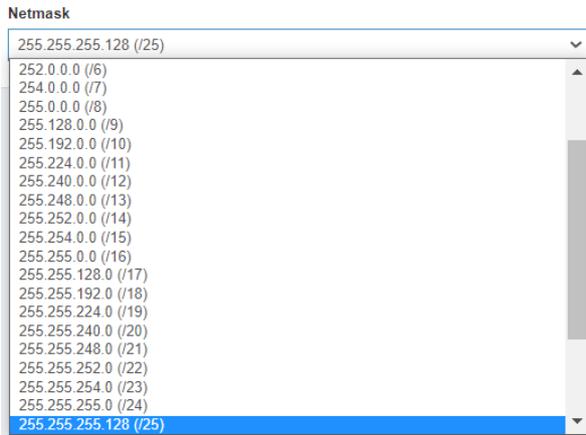
The **Create Subnet** view will be displayed:



We suggest creating one subnet associated to the physical Interface `sip-media_0` according to your network design, like in the following examples:



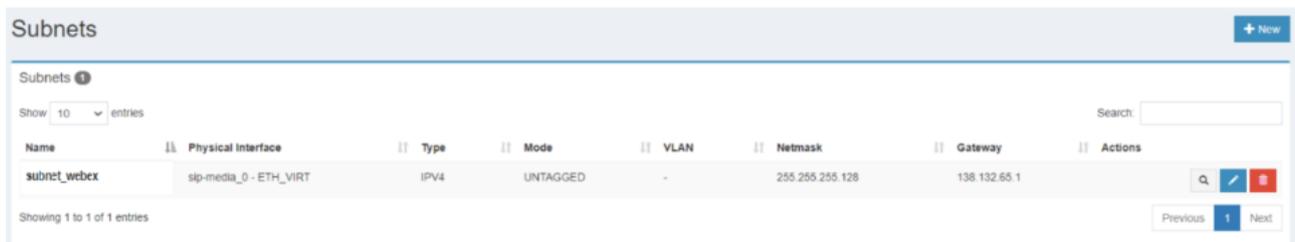
- In the **Name** field, insert a descriptive (logical) name to be used as subnet.
- In the **VLAN** field, the dropdown menu on the left-hand side of the VLAN ID field allows you to specify whether the VLAN should be tagged or untagged. If the VLAN is tagged, the ID is used as an actual tag and will be added to each Ethernet frame that is sent on a trunk. Untagged VLANs have ID equal to -1.
- In the **Gateway** field, Virtual IP address of the gateway to be used to access to the external network. IPv4 or IPV6 addresses are supported.
- In the **Gateway** field, Netmask for the gateway address. You specify the netmask by choosing the relevant integer value In the dropdown menu on the right-hand side. (The system will automatically display the netmask in dotted form in the left-hand field). When editing the form fields the Gateway box you must insert a valid IP address and the Netmask drop-down menu will propose a choice between all possible valid subnet mask values.



- In the **Physical Interface** field, any physical network interface is a named software representation by the operating system of the O&M to the user to enable him to configure the hardware or virtual network device

Name	VLAN	Gateway	Netmask	Physical Interface
subnet_webex	Untagged	138.132.65.1	255.255.255.128 /25	sip-media_0

If everything is correctly configured, upon clicking  the list view with the new subnet will be displayed:

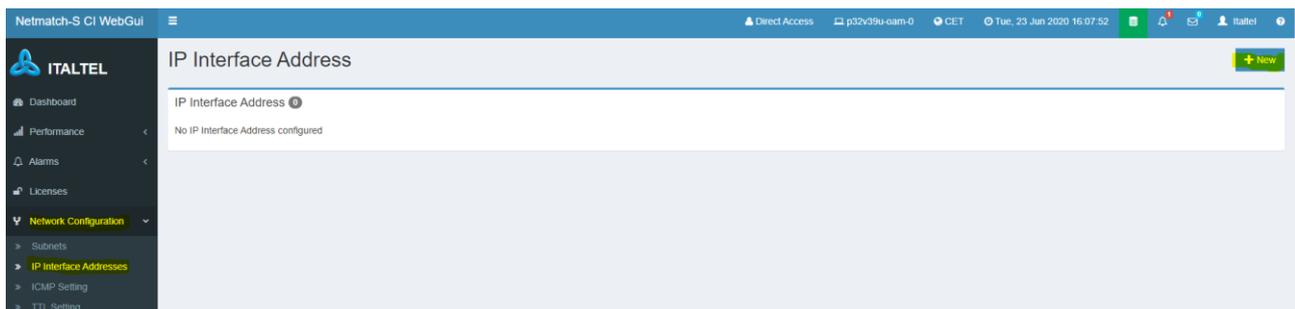


5.4.2 IP Interface Addresses

This section describes how new IP interface addresses can be configured in the product for Media and SIP Interface.

To configure the IP interface Addresses, follow the steps below:

- click **IP Interface Addresses** in the **Network Configuration** menu, and then click on  button:



- For each subnet created you can select to a choice between all possible valid IP, according to you network design, like in this example:

For Webex Calling side **138.132.65.52**:

Create IP Interface Address

Network Interfaces

Subnet

subnet_webex - 138.132.65.1/255.255.255.128 (0@sip-media_0)

Address

138.132.65.52

NAT

DISABLED

For PSTN side **138.132.65.53**:

Create IP Interface Address

Network Interfaces

Subnet

sud_pstn - 138.132.66.1/255.255.255.0 (66@sip-media_1)

Address

138.132.66.53

NAT

DISABLED

A list view with the IP Interface Address will be displayed at the end of the configurations:

IP Interface Address + New

IP Interface Address Search

Show 10 entries

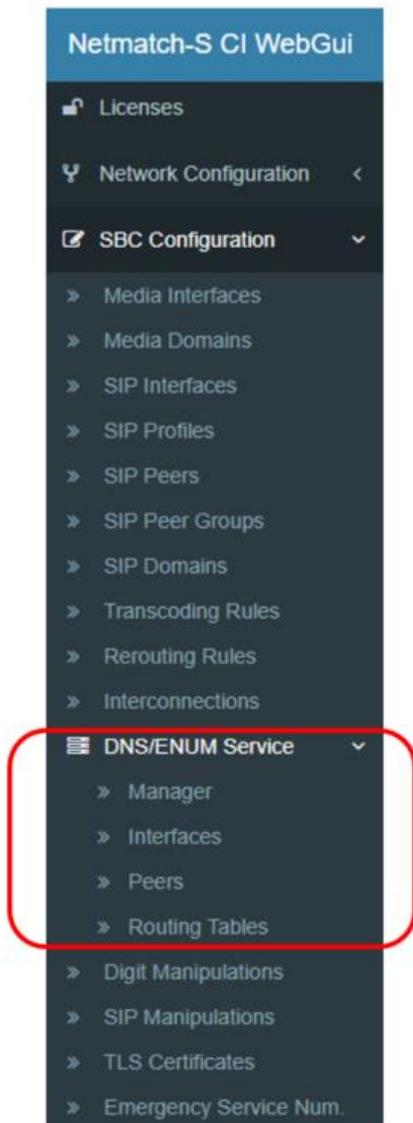
Address	Subnet Name	Netmask	Gateway	Vlan	Actions
138.132.65.52	subnet_webex	255.255.255.128/25	138.132.65.1	UNTAGGED	✖
138.132.65.53	sud_pstn	255.255.255.128/25	138.132.65.1	UNTAGGED	✖

Showing 1 to 2 of 2 entries Previous 1 Next

5.5 How to configure DNS Service

This feature allows operators to configure all the resources needed to perform DNS queries to external servers.

In order to access the configuration sub-menu, choose **SBC Configuration > DNS/ENUM Service** link into the side menu:



5.5.1 How to configure DNS/ENUM Service Manager

This section describes how to globally enable or disable queries to a DNS or ENUM server in order to access the configuration page, choose **Manager** Link into the sub-menu.

DNS Servers

DNS/ENUM Managers Search:

Show 10 entries

Name	DNS Servers Queries	ENUM Servers Queries	Actions
internal	true	true	<input type="button" value="Q"/> <input type="button" value="✎"/>

Showing 1 to 1 of 1 entries

1

By choosing the modify action , the appropriate form will be presented in which it is possible to enable or disable the DNS or ENUM queries separately.

You have to disable **ENUM** feature with **false** and enable **DNS** feature (**default value true**):

Server Settings

Capabilities

DNS Servers Queries

true
▼

ENUM Servers Queries

false
▼

To apply the changes, choose the  button.

5.5.2 How to configure DNS/ENUM Interfaces

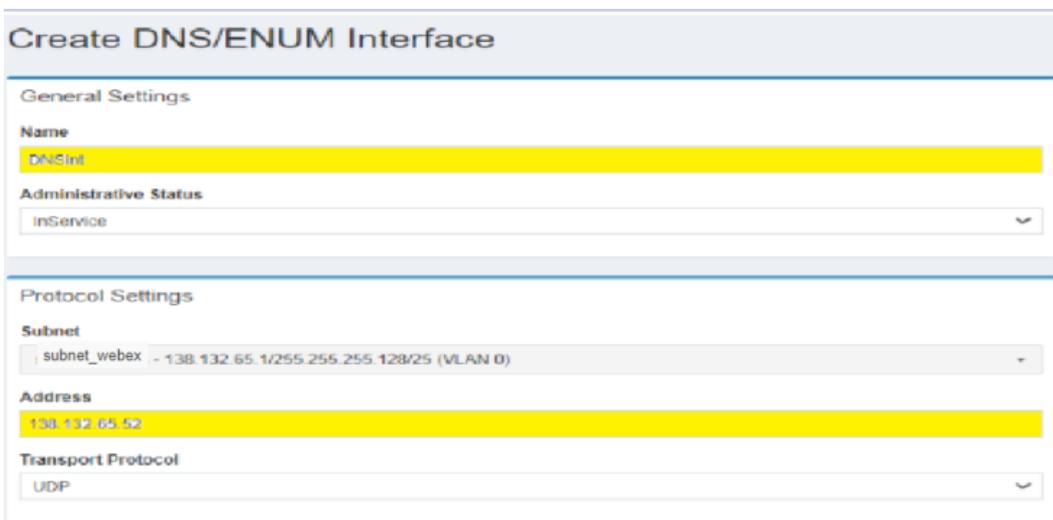
This section describes how to configure the external interface, in terms of IP Address and Port, to be used for DNS queries.

In order to access the configuration page, choose Interfaces Link into the sub-menu.

The **List DNS/ENUM Interfaces**, if any configured, is displayed:



Click  to create a new Interface, the following view is displayed:

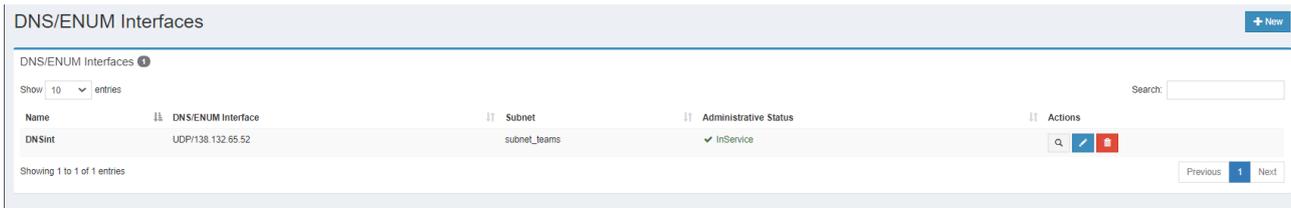


- In the **Name** field, insert a descriptive (logical) name to be used for this Interface. This is meant for the user's convenience only and does not affect the queries.
- The **administrative status** field is used to manage the update operations upon this interface.
- By filling the 'Subnet' and 'Address' fields, select the external address to be assigned to the Interface that can reach a DNS Server (e.g. **138.132.65.52**) according to your network design.
- Begin by searching and selecting one of the configured **Subnet**. The subnet list is available through a live search, which allows the dynamic search of the subnet name
- Then choose in the drop-down menu one of the **IP Interface Addresses** configured for the selected subnet

Parameter	Value
-----------	-------

Name	DNSInt
Administrative status	InService
Subnet	subnet_webex
Address	138.132.65.52

Once the form is completely filled in, click on  button to complete the creation; then the DNS Interfaces list page will be shown.



The screenshot shows a table with one entry:

Name	DNS/ENUM Interface	Subnet	Administrative Status	Actions
DNSInt	UDP/138.132.65.52	subnet_jeams	✓ InService	[Search] [Edit] [Delete]

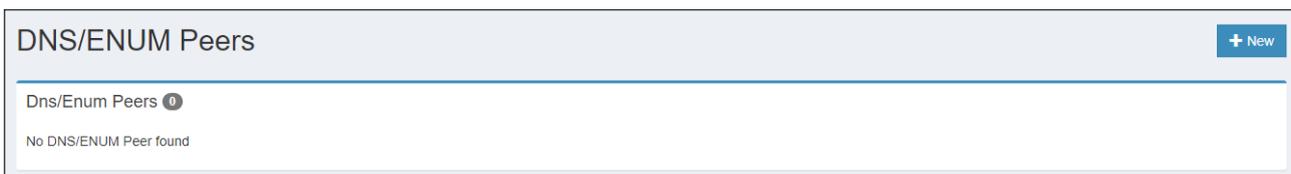
Showing 1 to 1 of 1 entries

5.5.3 How to configure DNS/ENUM Peers

This section describes how to configure the parameters for every external DNS server to be interrogated (e.g. Public DNS Server IP = 8.8.8.8).

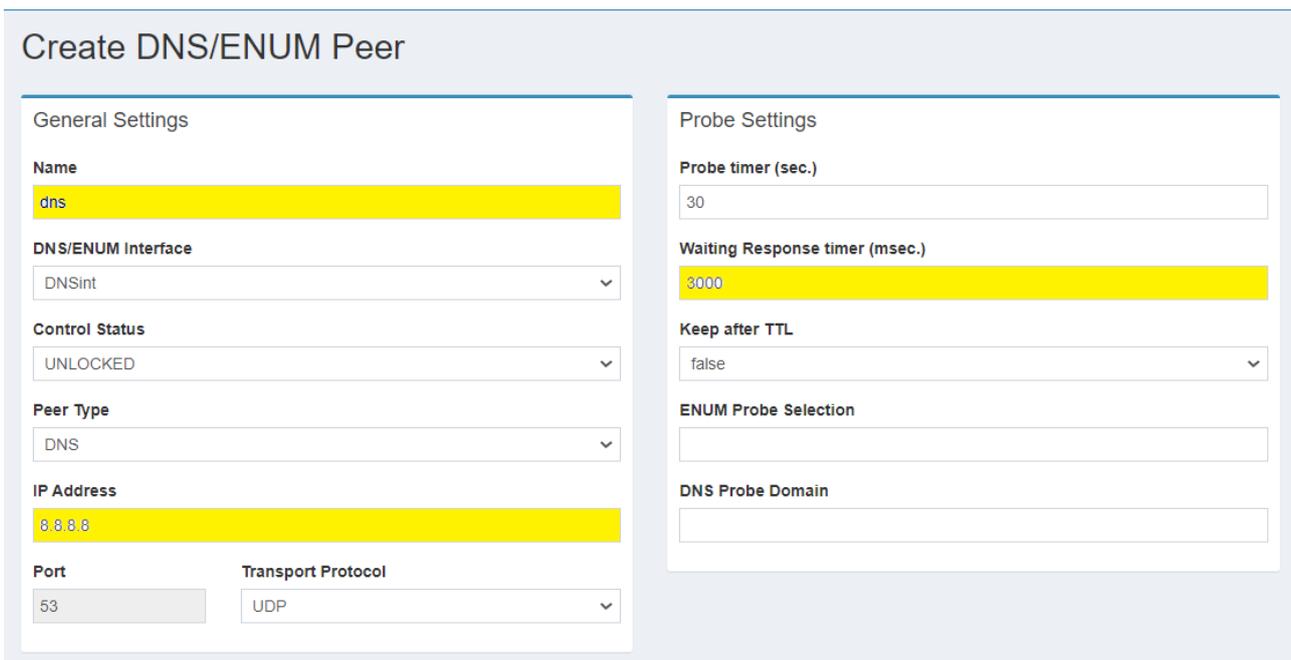
In order to access the configuration page, choose **Peers** Link into the sub-menu.

The **List DNS/ENUM Peers**, if any configured, is displayed:



The screenshot shows a list titled "Dns/Enum Peers" with the message "No DNS/ENUM Peer found".

Click on  button to access the form for configuring the DNS peer parameters.



The form is divided into two main sections: General Settings and Probe Settings.

General Settings:

- Name: dns
- DNS/ENUM Interface: DNSInt
- Control Status: UNLOCKED
- Peer Type: DNS
- IP Address: 8.8.8.8
- Port: 53
- Transport Protocol: UDP

Probe Settings:

- Probe timer (sec.): 30
- Waiting Response timer (msec.): 3000
- Keep after TTL: false
- ENUM Probe Selection: (empty)
- DNS Probe Domain: (empty)

The following describe only the information that you have to change or insert:

- **Name** is a symbolic label to refer to the DNS peer
- **IP Address** and **Port** of the remote DNS
- **Waiting Response timer** is the maximum time waited for a remote server to reply back (expressed in milliseconds)

Parameter	Value
Name	dns
IP Address	8.8.8.8
Waiting Response Timer	3000

Once the form is completed, click on  button to apply the configuration and return to the DNS/ENUM Peer list page.

5.5.4 How to configure DNS/ENUM Routing Tables

This section describes how to configure a group of alternative remote peer referring to the same DNS Server in order to apply load balancing or high availability policies.

In this example only 1 DNS is considered (e.g. 8.8.8.8)

In order to access the configuration page, choose **Routing Tables** Link into the sub-menu.

The **List DNS/ENUM Routing Tables**, if any configured, is displayed.

Click on  button to go to the creation page:

Create DNS/ENUM Routing Table

Settings

Name
dns

Zone Validation Mode
GENERIC

DNS/ENUM Routing Table Type
DNS

Scan Mode
MASTER_SLAVE

DNS/ENUM Peer list

Name	Role
DNS - dns - 8.8.8.8:53 UDP	MASTER
+ Add	

The following describes only the information that you have to change or insert:

- **Name** is a symbolic label to refer to this specific Routing Table (e.g. dns)

Once the form is completed, click on  button to apply the configuration and return to the DNS/ENUM Routing Tables list page:

DNS/ENUM Routing Tables + New

DNS/ENUM Routing Tables 1

Show 10 entries Search:

Name	Type	Scan mode	Dns/Enum Peers	Actions
dns	DNS	FAILOVER_MECHANISM	[M] dns - 8.8.8.8:53	🔍 ✎ 🗑️

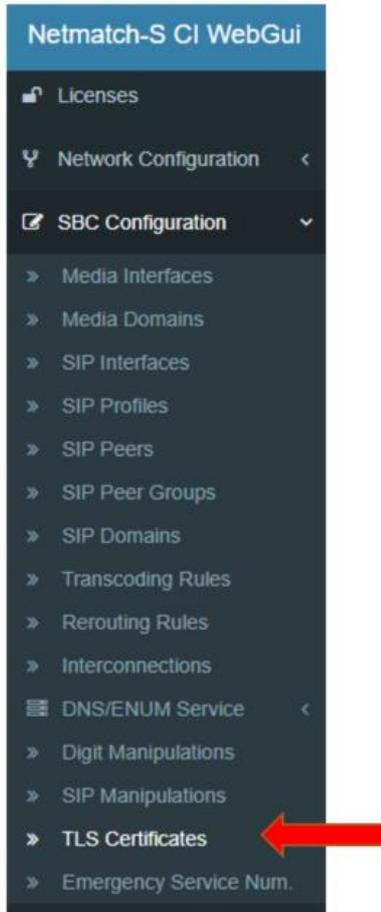
Showing 1 to 1 of 1 entries Previous **1** Next

5.6 How to manage Certificate

The section below shows how to manage a certificate. The certificate is used by the SBC to authenticate the connection with Webex Calling.

5.6.1 Create Certificate Signing Request (CSR)

This mode provides the generation of Certificate Signing Request (CSR) on NetMatch-S CI.



Manage TLS Certificates

+ Import Certificate
+ Import CA Certificate
+ Create Self-Signed
+ Request CSR

< Cancel
✔ Save

Certificate Signing Request

<p>General</p> <p>Name (Identifier) cert-webex-ok</p> <hr/> <p>Subject Information</p> <p>Common Name (CN) nms02.italtel.com</p> <p>Organizational Unit (OU) Research</p> <p>Organization (O) Italtel</p> <p>Locality (L) Milano</p> <p>State (S) Italia</p> <p>Country (C) IT</p> <p>Email Address (E) mario.rossi@italtel.com</p> <p>Password *****</p>	<p>General Extensions</p> <p>Authority Info Access</p> <hr/> <p>SIP Extensions</p> <p>Subject Alternative Names nms02.italtel.com</p> <p style="text-align: right;">+ Add</p>	<p>Security</p> <p>Key Encryption Algorithm PBE SHA1 3DES</p> <hr/> <p>Key Size 2048</p> <p>Signature Algorithm SHA1withRSA</p>
---	---	--

The fields to fill in:

General

name = the name of the certificate. (e.g., cert-webex-ok)

Subject Information

Common Name (CN) = User Certificates: You should enter the person's full name. Mandatory, it cannot be null. (e.g. nms02.italtel.com)

Organizational Unit (OU) = the Organizational Unit field can be used to differentiate between different divisions within an organization. Mandatory, it cannot be null. (e.g. Research)

Organization (O) = the name you specify for the Organization field should be the legal name for your organization that is registered with the appropriate city, state, or country/region authority. Mandatory, it cannot be null. (e.g. Italtel)

Locality (L) = the Locality field denotes the city that the organization resides in. Mandatory, it cannot be null. (e.g. Milano)

State (S) = the State or Province field specifies where the organization is physically located. Mandatory, it cannot be null. (e.g. Italia)

Country (C) = requires country/region code. Mandatory, 2 or 3 specific characters of a country/region. (e.g. IT)

EmailAddress (E) = the e-mail of the person who generate the request. Mandatory, has to present e-mail specific characters. (e.g. mario.rossi@italtel.com)

Password = password. Mandatory, composed at least of six characters

General Extensions (e.g. Italtel12345)

Authority Info Access (AIA) = the authority information access extension indicates how to access information and services for the issuer of the certificate in which the extension appears. Information and services may include on-line validation services and CA policy data. Optional.

SIP Extensions

Subject Alternative Name (SAN) = the subject alternative name extension allows identities to be bound to the subject of the certificate. These identities may be included in addition to or in place of the identity in the subject field of the certificate. Defined options include an Internet electronic mail address, a DNS name, an IP address, and a Uniform Resource Identifier (URI). Mandatory. (e.g. **nms02.italtel.com**)

This value is the SBC FQDN, the same value will be set into the ' Forced Local FQDN ' in WEBEX CALLING SIP DOMAIN

Security settings

In this panel, the used security settings are shown, as Key **Encryption Algorithm**, **Key Size** and **Signature Algorithm**.

At the End click on Save button .

Click to download the CSR file, as seen in figure below:

Tls Certificates 3

Show entries Search:

Name	Type	Subject	Issuer	Valid From	Valid Until	Actions
cer-webex-ok	Certificate Signing Request	CN=nms02.italtel.com, C=IT, ST=Italtel, L=Milano, E=massimiliano.nucita@italtel.com, OU=Research, O=Italtel	-	-	-	  

Then send the CSR to the Certification Authority

5.6.2 Update Certificate Signing Request (CSR)

Then once received the signed certificate, the user can update the CSR clicking the update-icon



Then import cert-webex-ok.pem (signed certificate received):

Upload Signed Certificate for: cer-webex-ok

Upload Signed Certificate

Certificate file

Scegli file cert-webex-ok.pem

Please use PEM format



At the end, you have a list like this:

Manage TLS Certificates

[+ Import Certificate](#)
[+ Import CA Certificate](#)
[+ Create Self-Signed](#)
[+ Request CSR](#)

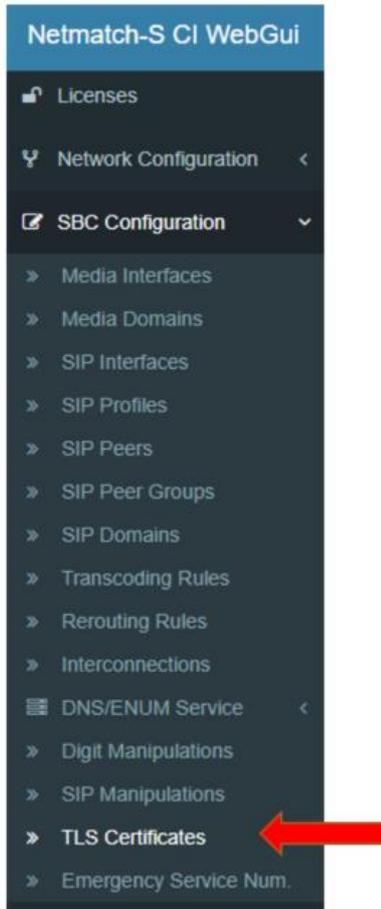
Tls Certificates 7

Show entries Search:

Name	Type	Subject	Issuer	Valid From	Valid Until	Actions
cert-webex-ok	Imported Certificate	CN=nms02.italtel.com, C=, ST=, L=, E=, OU=, O=	CN=R3, C=US, ST=, L=, E=, OU=, O=Let's Encrypt	Fri Oct 27 14:59:23 CEST 2023	Thu Jan 25 13:59:22 CET 2024	↓ ✖

5.6.3 Import CA Certificate

This mode provides importing of a Certification Authority (CA) certificate for NetMatch-S CI and Webex Calling.



Manage TLS Certificates



Upload a new CA certificate

Upload new certificate

Name

ISRG

Certificate file

Scegli file ISRG.pem

Please use PEM format

Upload

The fields to fill in:

Name = the name of the certificate. (e.g. ISRG)

Certificate file = the CA certificate file .pem (e.g. ISRG.pem)

Upload a new CA certificate

Upload new certificate

Name

IdenTrust

Certificate file

Scegli file IdenTrust.pem

Please use PEM format

Upload

The fields to fill in:

Name = the name of the certificate. (e.g. IdenTrust)

Certificate file = the CA certificate file .pem (e.g. IdenTrust.pem)

Upload a new CA certificate

Upload new certificate

Name

letsEncrypt

Certificate file

Scegli file letsEncrypt.pem

Please use PEM format

Upload

The fields to fill in:

Name = the name of the certificate. (e.g. letsEncrypt)

Certificate file = the CA certificate file .pem (e.g. letsEncrypt.pem)

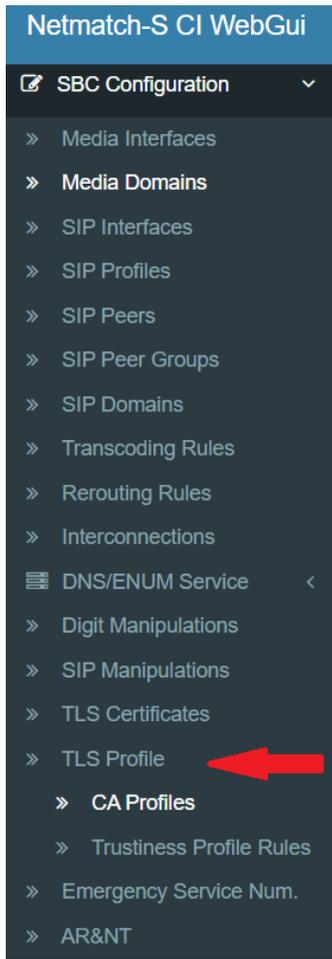
At the end, you have a list like this:

IdenTrust	Certification Authority	CN=IdenTrust Commercial Root CA 1, C=US, ST=, L=, E=, OU=, O=IdenTrust	CN=IdenTrust Commercial Root CA 1, C=US, ST=, L=, E=, OU=, O=IdenTrust	Thu Jan 16 19:12:23 CET 2014	Mon Jan 16 19:12:23 CET 2034	 
ISRG	Certification Authority	CN=ISRG Root X1, C=US, ST=, L=, E=, OU=, O=Internet Security Research Group	CN=ISRG Root X1, C=US, ST=, L=, E=, OU=, O=Internet Security Research Group	Thu Jun 04 13:04:38 CEST 2015	Mon Jun 04 13:04:38 CEST 2035	 
letsEncrypt	Certification Authority	CN=R3, C=US, ST=, L=, E=, OU=, O=Let's Encrypt	CN=ISRG Root X1, C=US, ST=, L=, E=, OU=, O=Internet Security Research Group	Fri Sep 04 02:00:00 CEST 2020	Mon Sep 15 18:00:00 CEST 2025	 

Note: If you have more files of root-ca provided by CA, please import all and then create a CA Profile.

5.6.4 Create CA Profile

This mode allows you to the create a Certification Authority Profile with list of (CA) for NetMatch-S CI and Webex Calling.



The following describes the information that you have to create a new CA Profile:

Click on  to create a new CA Profile; then the following view is displayed:

Create CA Profile

Settings

Name
webEx-CA-list

CA Name List All

rootNov2023

Selected CA Name List All

idenTrust
letsEncrypt
ISRG

>>
<<

And then with  insert in list the CA that you need to insert in you CA Profile

The following describes only the information that you have to change or insert to create a new CA Profile.

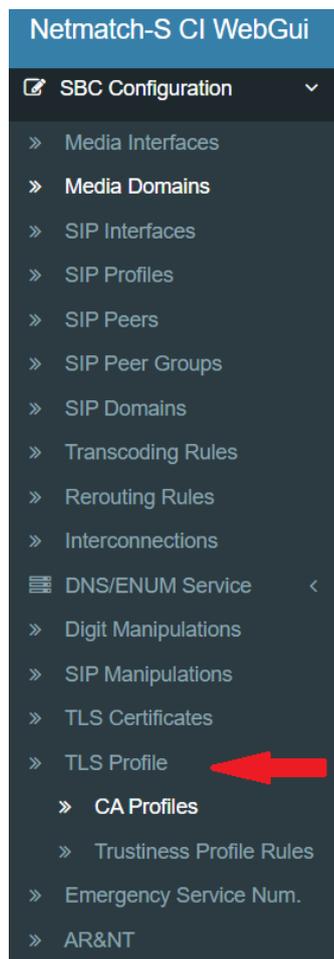
In the **Name** field, insert a descriptive (logical) name to be used for this CA profile. This name will be used to associate the SIP Interface (e.g. webEx-CA-list).

At the end, you have a list like this:



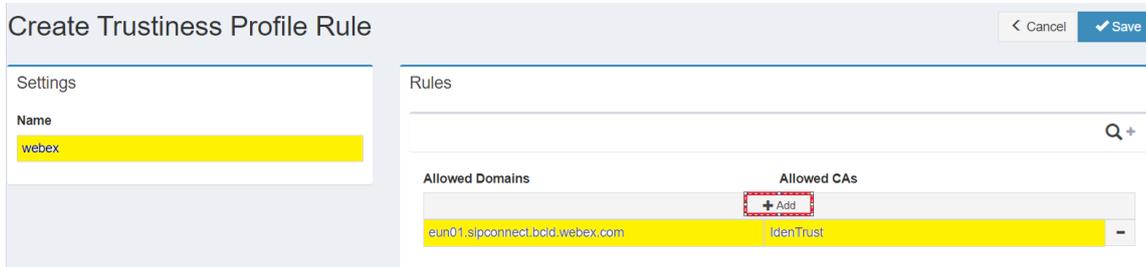
5.6.5 Create Trustiness Profile

This mode allows you to insert a new trustiness rule that can be formed by the admitted domain, the admitted CA or both.



The following describes the information that you have to create a new Trustiness Profile Rule:

Click on **+ New** to create a new Trustiness Profile; then the following view is displayed:



With **+ Add** insert Domain and CA of Webex Calling allowed in your Trustiness Profile

The following describes only the information that you have to change or insert to create a new Trustiness Profile.

In the **Name** field, insert a descriptive (logical) name to be used for this Trustiness profile. This name will be used to associate the SIP Interface (e.g. **webex**).

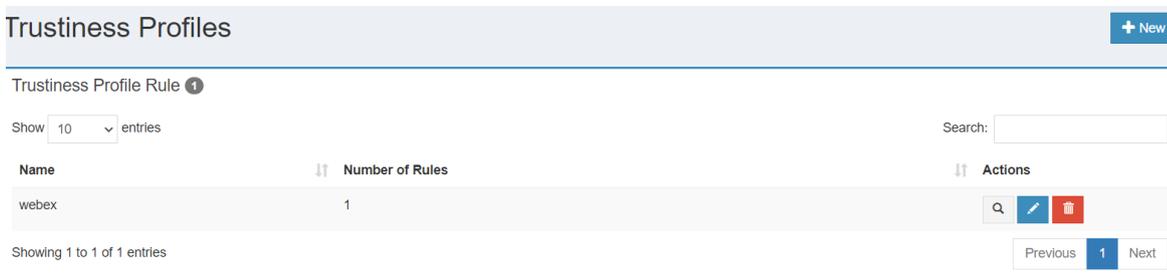
In the **Allowed Domains** field, insert a Name of the allowed domain (e.g. **eun01.sipconnect.bclid.webex.com**)

In the **Allowed CAs** field, insert a Name of the allowed Certification Authority (e.g. **IdenTrust**)

Click **Save** to confirm the creation of Trustiness Profile.

Parameter	Value
Name	webex
Allowed Domains	eun01.sipconnect.bclid.webex.com
Allowed CAs	IdenTrust

At the end for example you have:

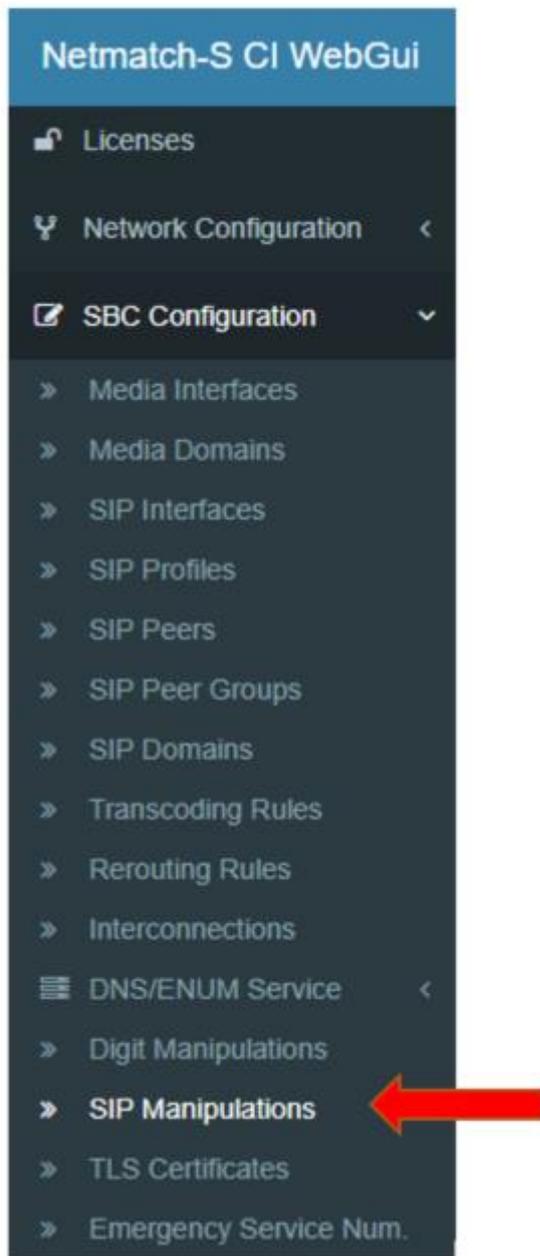


5.7 How to import SIP Manipulations

For those scenarios, where different equipment vendors provide different SIP implementations or where particular SIP profiles are required by the interconnected Service Providers/Enterprises, the product provides the SIP Manipulation feature to ensure the adaptation of SIP signalling interfaces.

The product is able to insert, delete or modify any SIP field in the received SIP messages, before forwarding them.

The access to the SIP Manipulation functionality is available through the **SIP Manipulations** item inside the **SBC Configuration** menu. Choosing this one, the list of the already available Sip Manipulations rules is displayed, if any.



A Sip manipulation rule can also be imported into the SBC.

In this configuration is necessary to import these Sip Manipulation:

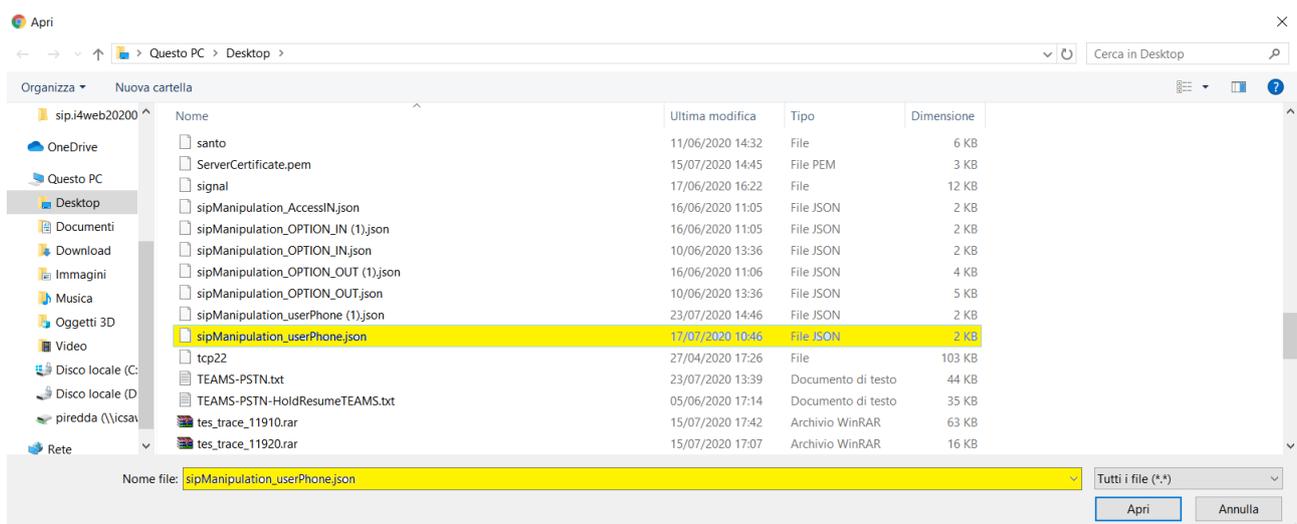
- sipManipulation_userPhoneWebex.json
- sipManipulation_userPhone.json
- sipManipulation_outPstnWebex.json

[sipManipulation_userPhoneWebex.json](#)

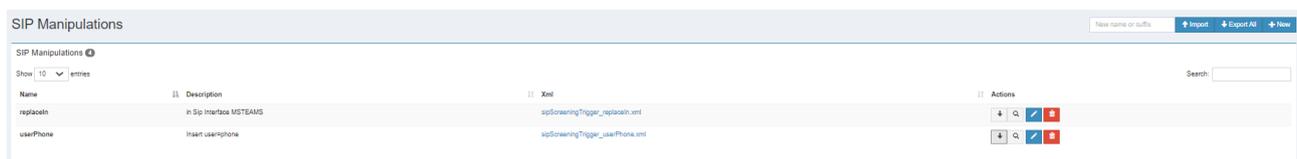
[sipManipulation_userPhone.json](#)

[sipManipulation_userPhoneWebex.json](#)

You can store these files in your computer or other repository and then import it by click on  button.

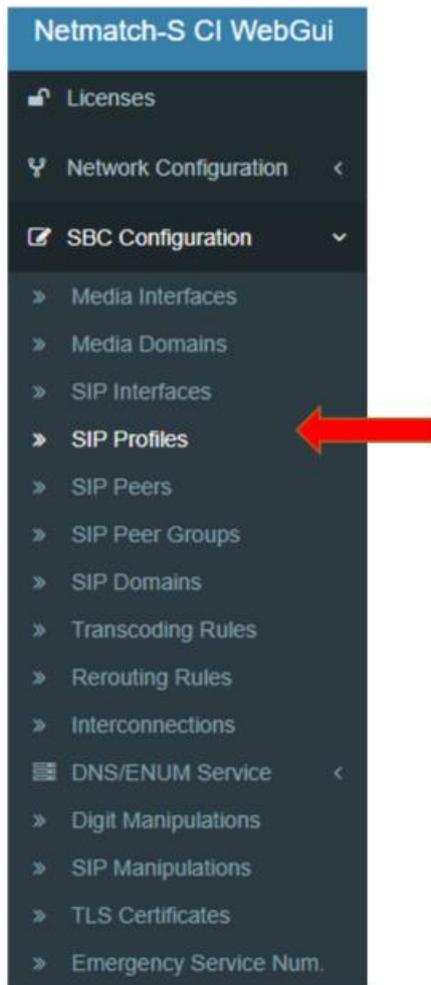


Then, if the operation is successful, the **Sip Manipulation** appears in the list and you can use them subsequently in a **Sip Domain** and **Sip Interface** configuration, if needed.



5.8 How to create SIP Profiles

A SIP Profile is a set of SIP protocol settings that is possible to associate to a SIP domain. These settings include standard SIP timer configuration and White/Blacklist management on both SIP Methods and SIP headers. It is possible to associate the same SIP profile to different SIP domains. In order to create a SIP Profiles, select **SBC Configuration >> SIP Profiles** in the main menu:



5.8.1 Create SIP Profiles for No Media Bypass option

The following describes the information that you have to change or insert to create a Sip Profiles for No Media Bypass options:

Click on  to create a new SIP profile; then the following view is displayed:

The following describes only the information that you have to change or insert to create a new SIP Profile.

In the **Name** field, insert a descriptive (logical) name to be used for this SIP profile. This name will be used to associate the SIP profile during the configuration of a SIP Domain

The **Incoming Managed SIP Methods** and **Outgoing Managed SIP Methods** sections allow configuring separately two list of methods to be accepted or rejected in the corresponding directions of SIP messages. A **Filter Modality** (Disabled / Blacklist / Whitelist) is associated to each methods list to define the application criteria.

To add SIP methods to the list, click **+Add** and select the chosen method from the drop-down menu.

For each list is possible to set the **Filter Modality**.

SIP Headers in a Blacklist, as well as those outside a Whitelist, will be removed from SIP message if not mandatory.

In **VGW options Tab** set **200 msec** in **Timer wait response from vGW** and in **Timer wait response from protected vGW** (this value is used for the duration of SIP T1 when Peer is resolved by DNS), the **Rerouting Match Type** in set to **DIRECT** for rerouting on value of these **Rerouting trigger responses**: 408, 3xx, 5xx if add

Parameter	Value
Name	noreferWebex

Incoming Methods	Managed	SIP	REFER
Filter modality			Blacklist
Outgoing Methods	Managed	SIP	REFER
Filter Modality			Blacklist
Timer waits response from VGW			200
Timer waits response from protected VGW			200
Rerouting Match Type			DIRECT
Rerouting Trigger Responses			408, 3, 5

Click  to confirm the creation of SIP Profile.

At the end the following view is displayed:



The screenshot shows a table of SIP Profiles with the following columns: Name, Timer T1 (msec), Timer T2 (sec), Timer C (sec), Timer D (sec), Timer H (msec), IN Methods, OUT Methods, IN Headers, OUT Headers, Notification Events, VGW options, and Actions. The 'noreferWebex' profile is highlighted in yellow and has 'REFER' in red in the IN and OUT Methods columns, indicating a blacklist modality. Other profiles like 'norefer' and 'refer' have 'ALLOW ALL' in green, indicating a whitelist modality.

Name	Timer T1 (msec)	Timer T2 (sec)	Timer C (sec)	Timer D (sec)	Timer H (msec)	IN Methods	OUT Methods	IN Headers	OUT Headers	Notification Events	VGW options	Actions
norefer	500	40	180	32	32000	REFER	REFER	ALLOW ALL	ALLOW ALL	SET		
noreferWebex	500	40	180	32	32000	REFER	REFER	ALLOW ALL	ALLOW ALL	SET		
refer	500	40	180	32	32000	ALLOW ALL	INFO	ALLOW ALL	ALLOW ALL	SET		
SYSTEM_DEFAULTS	500	40	180	32	32000	ALLOW ALL	ALLOW ALL	ALLOW ALL	ALLOW ALL			

For Methods and Headers columns, the green character means “allow” (Whitelist modality) while the red character means “reject” (Blacklist modality).

5.9 How to create Media Interfaces

In order to create a Media Interface, select **SBC Configuration >> Media Interfaces** in the main menu:



You have to configure 2 media interfaces.

Click on **+ New** to create a first new Media Interface the following view is displayed:

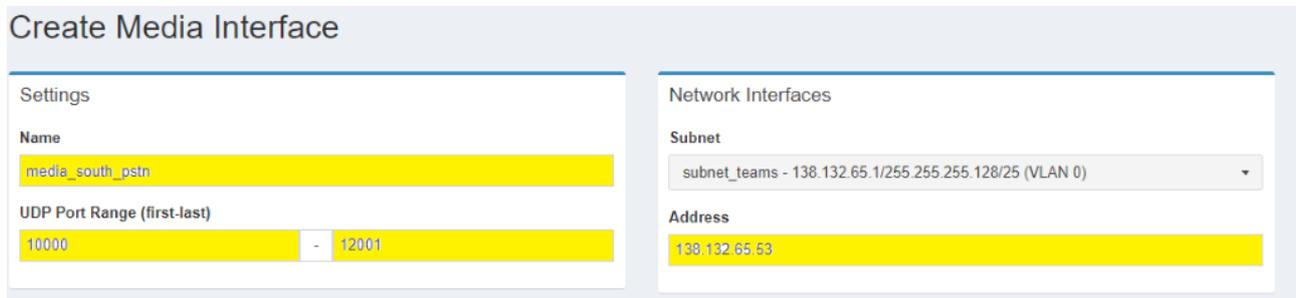
Media Interface for Webex Calling side:

As reported in Cisco Port Reference for Webex calling (<https://help.webex.com/en-us/article/b2exve/Port-Reference-Information-for-Webex-Calling>) the media ports on local gateway must be configured **from port 8000 to 48198**.

Create Media Interface

<div style="border: 1px solid #ccc; padding: 5px; margin-bottom: 5px;"> Settings <p>Name media_north_webex</p> <p>UDP Port Range (first-last) 10000 - 12001</p> </div>	<div style="border: 1px solid #ccc; padding: 5px;"> Network Interfaces <p>Subnet subnet_webex 138.132.65.1/255.255.255.128/25 (VLAN 0)</p> <p>Address 138.132.65.52</p> </div>
---	---

Media Interface for PSTN side:



The **Name** field is a label identifying the Media Interface to recall it during the configuration of the SIP Interface.

The **UDP Port Range** field is used to assign a range of UDP ports to the media interface.

In the **'Network Interfaces'** panel, select the external address to be assigned to the Media Interface between those who are provided according to your network interface configurations.

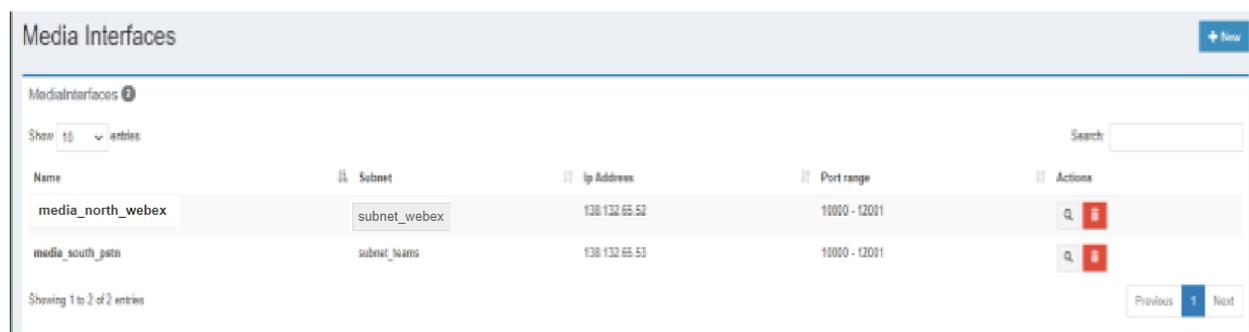
First, you search and select one of the configured **Subnet** as follow; the list is available through a useful live search, which allows the dynamic search of the subnet name.

Then choose in the drop-down menu one of the **IP Interface Addresses** configured for the selected subnet.

Click  to confirm the creation of Media Interface.

Name	First UDP Port	Last UDP Port	Subnet	Address
media_north_webex	10000	12001	subnet_webex	138.132.65.52
media_south_pstn	10000	12001	subnet_pstn	138.132.65.53

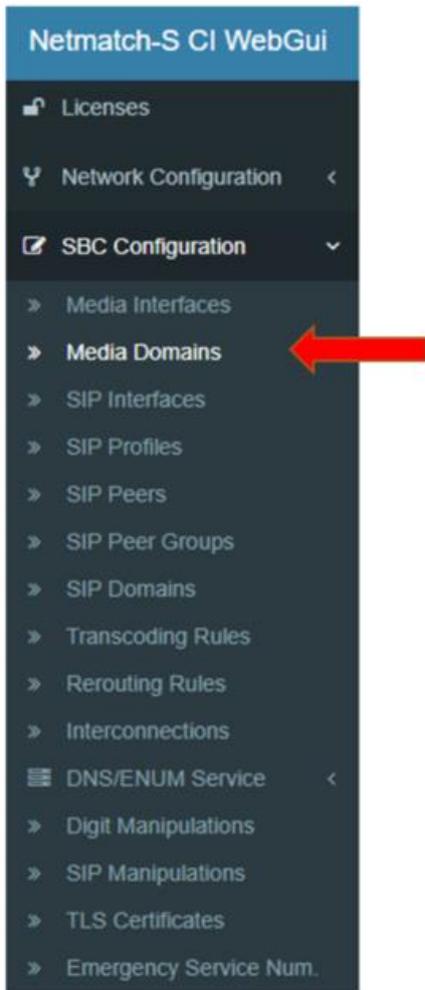
At the end, for example we have:



Name	Subnet	Ip Address	Port range	Actions
media_north_webex	subnet_webex	138.132.65.52	10000 - 12001	 
media_south_pstn	subnet_teams	138.132.65.53	10000 - 12001	 

5.10 How to create Media Domains

In order to create a Media Domain, select **SBC Configuration >> Media Domains** in the main menu:



You have to configure 2 media Domains for Team side and PSTN side.

Click on **+ New** to create a new Media Domain; the following view is displayed:

5.10.1 Create Webex Calling Media Domain for No Media Bypass option

The following describes the information that you have to change or insert to create Webex Calling Media Domain for No Media Bypass options:

Create Media Domain

Settings

Name
dom_webex

Media Release
DISABLED

SRTP
MANDATORY

IceType
DISABLED

This section includes configuration parameters for media domain.

The **Name** field is a label identifying the Media Domain to recall it during the configuration of the SIP Interface.

In case of **SRTP** feature enabled, it is possible to configure the SRTP Type field.

The Direct Routing Interface requires the use of SRTP only, so you need to configure the SBC to operate the same way.

Name	SRTP
dom_webex	MANDATORY

5.10.2 Create PSTN Media Domain for No Media Bypass option

The following describes the information that you have to change or insert to create PSTN Media Domain:

Create Media Domain

Settings

Name
dom_south_pstn

Media Release
DISABLED

SRTP
DISABLED

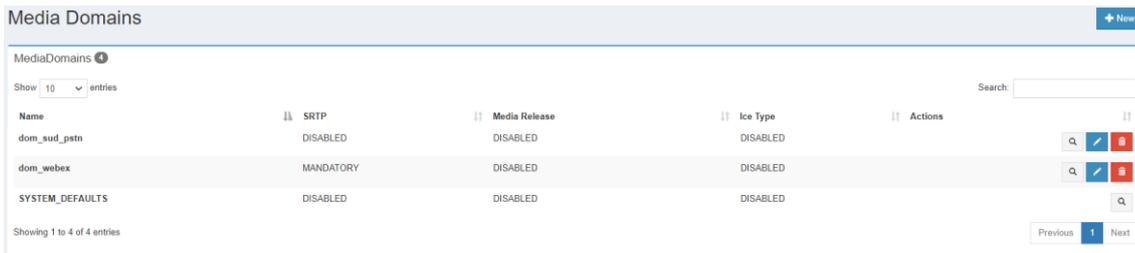
IceType
DISABLED

This section includes configuration parameters for media domain.

The **Name** field is a label identifying the Media Domain to recall it during the configuration of the SIP Interface.

Name
dom_south_pstn

At the end for example you have 2 Media Domains

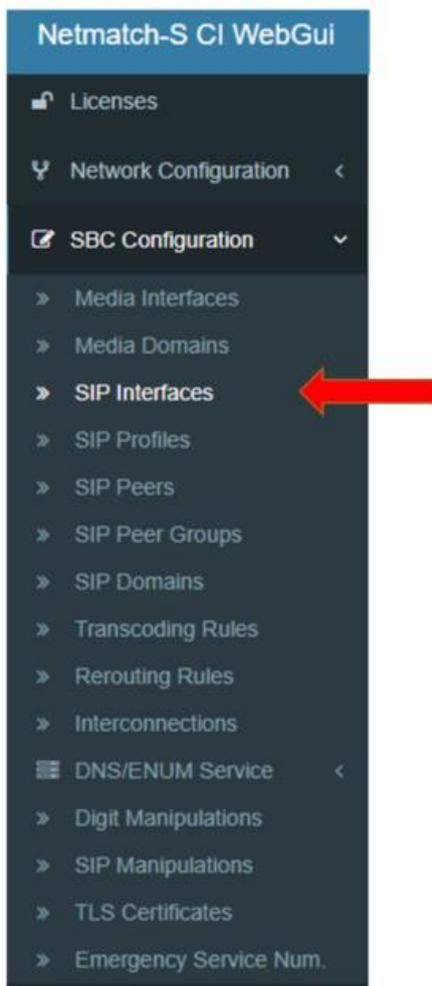


Name	SRTP	Media Release	Ice Type	Actions
dom_sud_pstn	DISABLED	DISABLED	DISABLED	  
dom_webex	MANDATORY	DISABLED	DISABLED	  
SYSTEM_DEFAULTS	DISABLED	DISABLED	DISABLED	

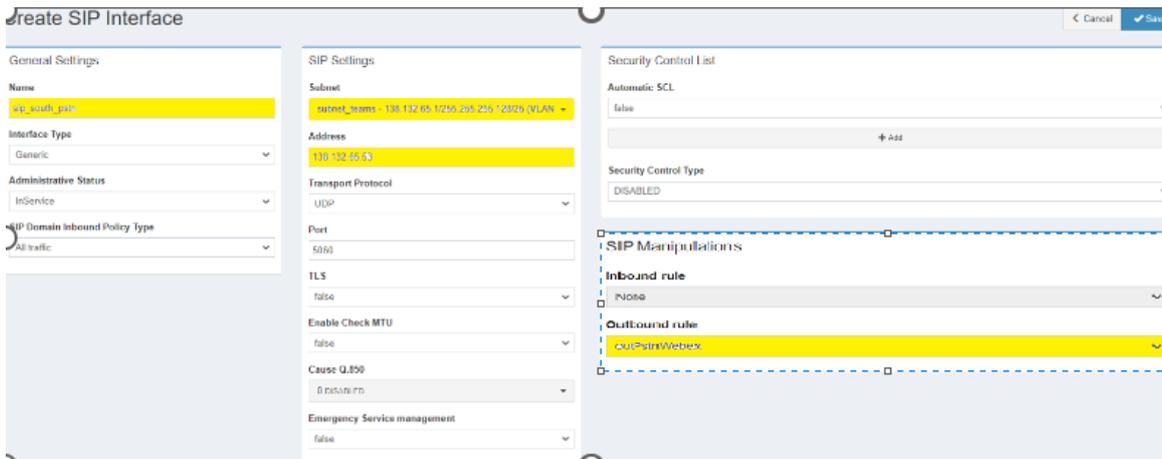
5.11 How to create SIP Interfaces

This section shows how to configure a SIP Interface. A SIP Interface defines a listening port and protocol type (UDP, TCP, or TLS) for SIP signalling traffic on a specific logical IP network interface.

In order to create a SIP Interface, select **SBC Configuration >> SIP Interfaces** from the main menu:



Click  to create a new SIP Interface: the following view is displayed:



5.11.1 Create SIP Interface for PSTN side

The following describes the information that you have to change or insert to create a sip Interface on PSTN side:

In the **Name** field, insert a descriptive (logical) name to be used for this SIP Interface.

In the 'SipSetting' panel, select the external address to be assigned to the Sip Interface, between those provided according to your network interface configurations.

First, search and select one of the configured **Subnet** as follow; the list is available through a useful live search, which allows the dynamic search of the subnet name.

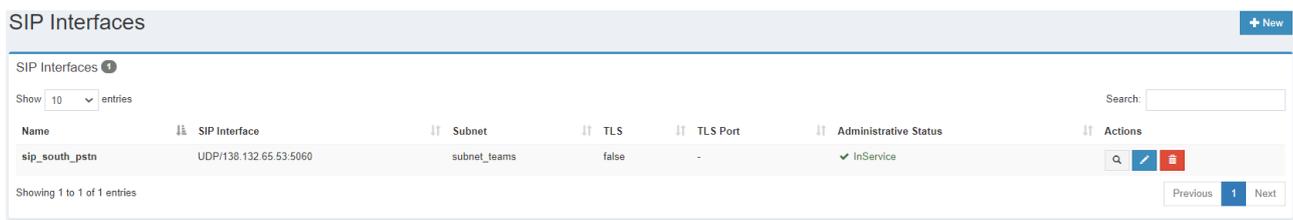
Then choose in the drop-down menu one of the **IP Interface Addresses** configured for the selected subnet.

Then add SIP Manipulation in Outbound Rule, choosing in the drop-down menu.

Click  to confirm the creation of Sip Interface.

Parameter	Value
Name	sip_south_pstn
Subnet	subnet_pstn
Address	138.132.65.53
Transport	UDP
SIP Manipulation Outbound	outPSTNwebex

At the end for example you have:



5.11.2 Create SIP Interface for Webex Calling side

The following describes only the information that you have to change or insert to create a sip Interface on Webex Calling side:

In the **Name** field, insert a descriptive (logical) name to be used for this SIP Interface.

In the 'Sip Settings' panel, select the external address to be assigned to the Sip Interface between those provided according to your network interface configurations.

First, search and select one of the configured **Subnet** as follow; the list is available through a useful live search, which allows the dynamic search of the subnet name.

Then choose in the drop-down menu one of the **IP Interface Addresses** configured for the selected subnet.

In the **Transport Protocol** field, choose the transport protocol to be used for SIP signalling.

In the **TLS** field, you can enable TLS feature

Then add SIP Manipulation in Inbound and Outbound Rule, choosing in the drop-down menu.

Click  to confirm the creation of Sip Interface.

Parameter	Value
Name	sip_webex
Subnet	subnet_webex
Address	138.132.65.52
Transport	TCP
TLS	true
TLS Version	1.2

SIP Settings

Subnet
subnet_webex 138.132.65.1/255.255.255.128/25 (VLAN ▾)

Address
138.132.65.52

Transport Protocol
TCP ▾

FE L4 Termination
false ▾

Port
5060

TLS
true ▾

Enable Check MTU
false ▾

Cause Q.850
0 DISABLED ▾

Emergency Service management
false ▾

When **TLS** field is **true**, a new 'TLS Setting' panel appear:

TLS Settings

TLS Port

TLS Certificate

Trusted TLS CAs

Authentication Type

Trustiness Profile

Add P-Served-User from certificate

TLS Version

Under TLS Settings you have to change or insert the following parameters:

TLS Certificate (select Cert-webex-ok for Webex Calling)

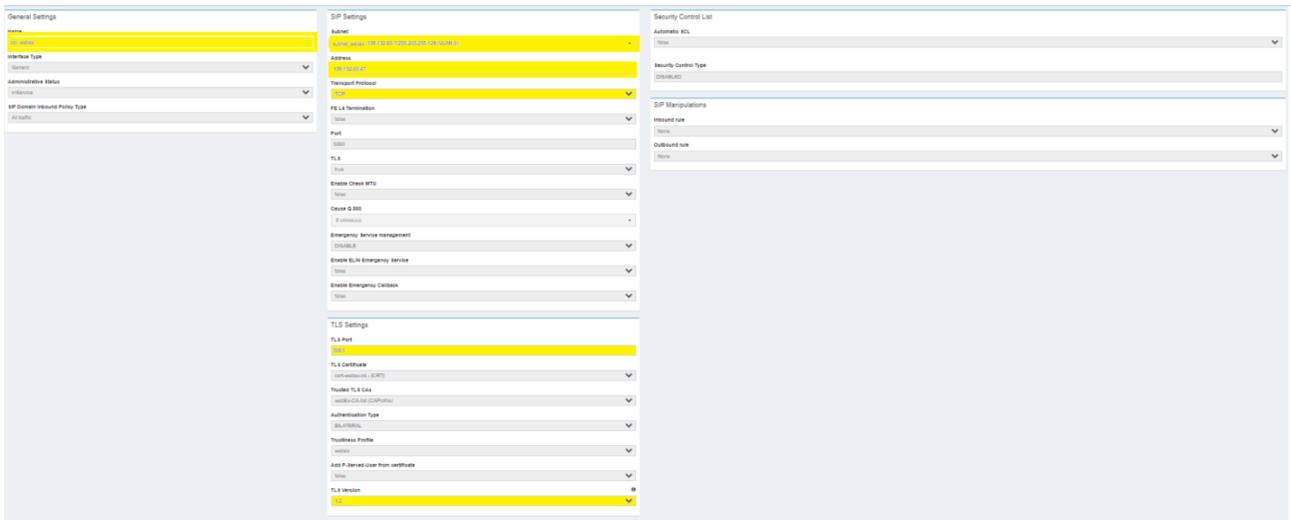
Trusted TLS CAs (select WebEx-CA-List for Webex Calling)

Authentication Type (select BILATERAL for Webex Calling)

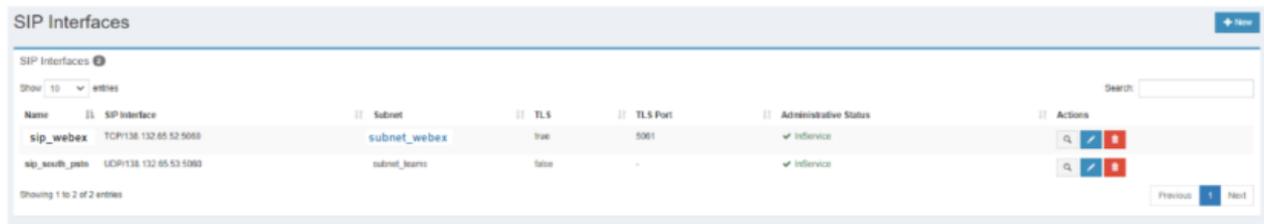
Trustiness Profile (select webex for Webex Calling)

TLS Version TLS version (ALL, 1.0, 1.1 or 1.2) (select 1.2 for Webex Calling)

To create the sip Interface on Webex Calling side you have to change or insert only the highlighted fields:



At the end for example you have:



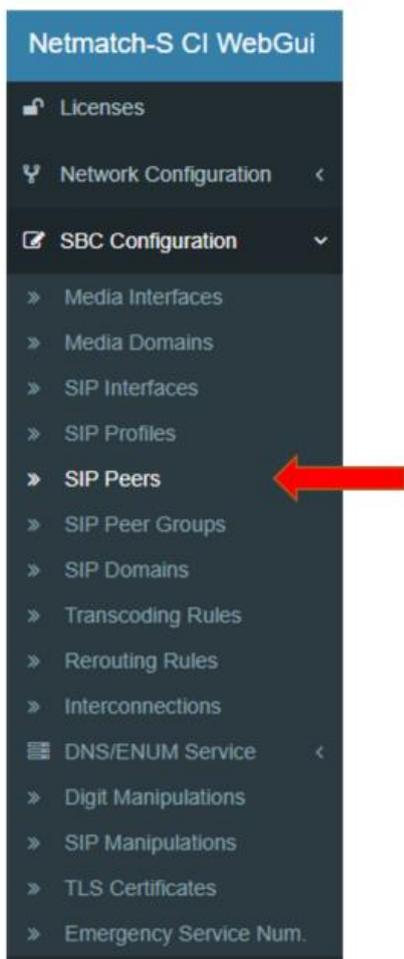
Name	SIP Interface	Subnet	TLS	TLS Port	Administrative Status	Actions
sip_webex	TCP138.132.65.52.5060	subnet_webex	true	5061	✓ In Service	[Search] [Add] [Edit] [Delete]
sip_south_peer	UDP138.132.65.53.5060	subnet_south	false	-	✓ In Service	[Search] [Add] [Edit] [Delete]

Showing 1 to 2 of 2 entries

5.12 How to create SIP Peers and SIP Peer Group on PSTN side

5.12.1 Create SIP Peers on PSTN side

In order to create a SIP Agent (**only for PSTN side**), select **SBC Configuration >> SIP Peers** submenu in the main menu:



Click [+ New](#) to create a new SIP Peer; the following view is displayed:

The following describes the information that you have to change or insert to create a sip Peer on PSTN side:

Create SIP Peer

Settings

Name
pstn

Administrative Status
InService

IP Address
52.178.167.0

Port
5060

Transport Protocol
UDP

Enable TLS
false

TLS Port
5061

Peer Shared
false

Probe Settings

Probe Method
OPTIONS

Probe On Request
Disabled

Probe Domain
First connected

Probe Timer
60

Call Admission Control

Max Act Sess. (CAC)
0

Max In Sess. (CAC IN)
0

Max Out Sess. (CAC OUT)
0

Max CPS
0

Max In CPS
0

Max Out CPS
0

Max In Bandwidth (Kbit/Sec)
0

Max Out Bandwidth (Kbit/Sec)
0

Max Bandwidth (Kbit/Sec)
0

In the **Name** field, insert a descriptive (logical) name to be used for this SIP Agent. This name will be used to associate the SIP Agent during the configuration of the Hunting Group.

In the **IP Address** and **Port** fields enter the IP address and port of the remote SIP agent.

The other fields depend of your configuration in PSTN side, for example if you need probe method or other.

Name	Status	IP Address	Port	Transport	Probe Method
pstn	InService	52.178.167.0	5060	UDP	OPTIONS

At the end for example you have

SIP Peers

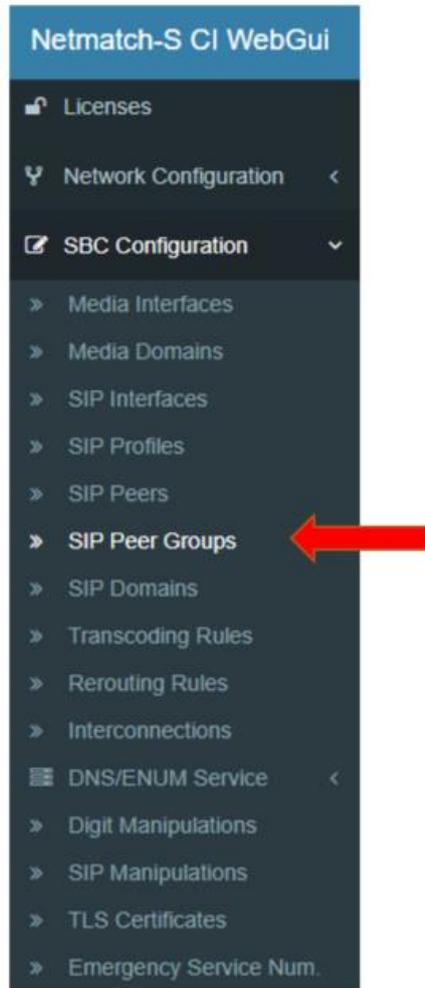
Showing 10 entries

Name	Address	Port	Transport	TLS	TLS Port	Probe Method	Probe On Req	Probe Domain	Shared	Administrative Status	Operational State	Actions
pstn	52.178.167.0	5060	UDP	false	5061	OPTIONS	Disabled	pstn.com	false	✓ InService	✓ InService	[Edit] [Delete]

Showing 1 to 1 of 1 entries

5.12.2 Create SIP Peer Groups on PSTN side

In order to create a SIP Peer Group (**only for PSTN side**), select **SBC Configuration >> SIP Peer Groups** from the main menu:



Click [+ New](#) to create a new SIP Peer Group; the following view is displayed:

The following describes the information that you have to change or insert to create a sip Peer Groups on PSTN side:

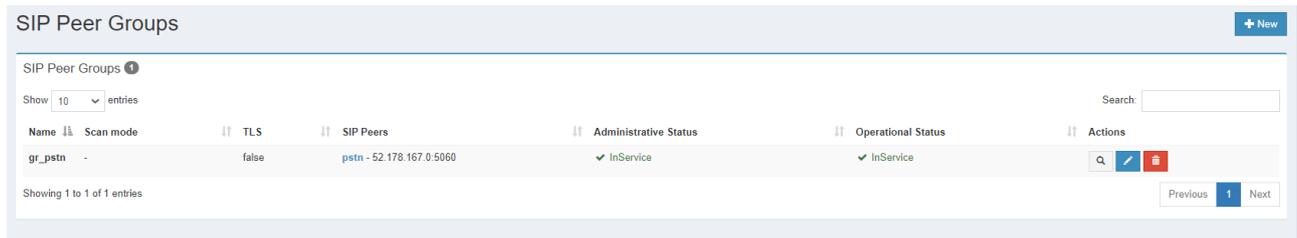


In the **Name** field, insert a descriptive (logical) name to be used for this Hunting Group. This name will be used to associate the Hunting Group in a SIP Domain. (e.g. **gr_pstn**).

In this example, we suppose you have in PSTN side just one Peer in one Peer Group.

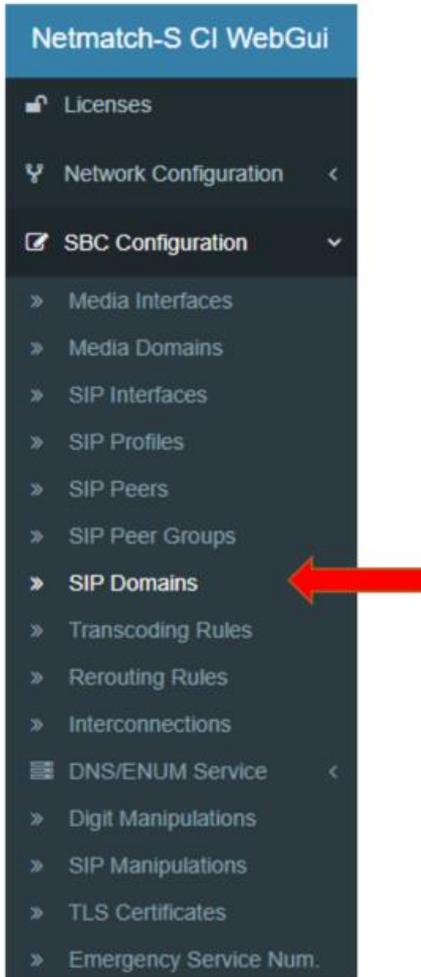
Name	Status	Peer List
gr_pstn	InService	pstn

At the end for example you have:



5.13 How to create SIP Domains

In order to create a SIP Domain, select **SBC Configuration >> SIP Domains** in the main menu:



5.13.1 Create SIP Domain for PSTN side for No Media Bypass option

Click **+ New** to create a new SIP Domain; the following view is displayed:

The following describes the information that you have to change or insert to create SIP Domain PSTN for each section.

In the **Settings Logical** section, you can provide the following information:

Create SIP Domain

 Settings
MTF Interworking **Logical**

Name

pstn.com

SIP Interface

sip_south_pstn

Media Interface

media_south_pstn

Media Domain

dom_south_pstn

SIP Profiles

SYSTEM_DEFAULTS

Type

GENERIC

DNS Query

DISABLED

ENUM Query

DISABLED

Enable TLS

false

The **Name** field, a string identifier.

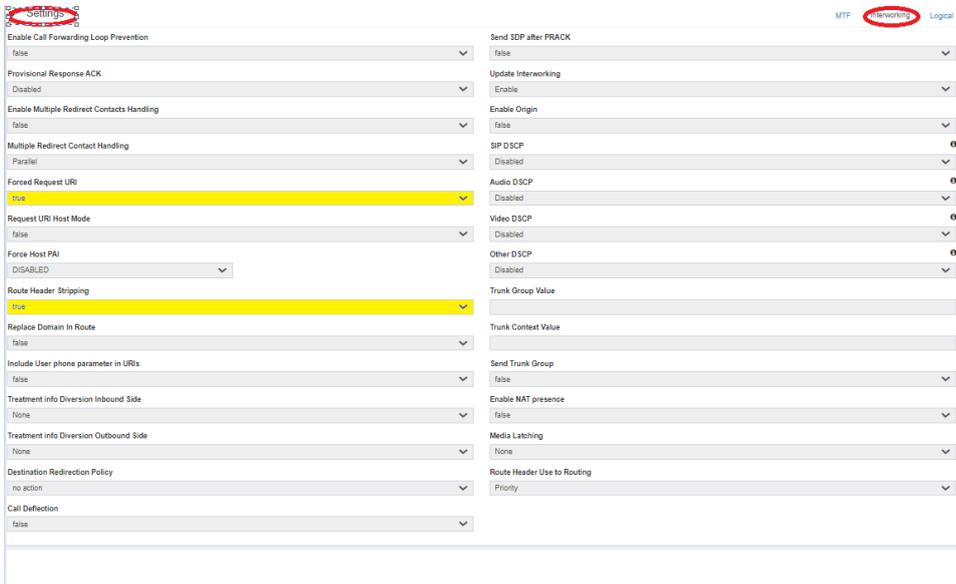
The **Sip Interface** field (drop-down menu) will show all the SIP Interfaces previously configured on the system, and that will be used for signalling flows exchange.

In the **Media Interface** field (drop-down menu) will show all the Media Interface previously configured on the system, and that will be used for media flows exchange.

In the **Media Domain** field (drop-down menu) will show all Media Domain previously configured on the system, and that will be used for media different features configured, like a profile with different configuration.

Parameter	Value
Name	pstn.com
Sip Interface	sip_south_pstn
Media Interface	media_south_pstn
Media Domain	dom_south_pstn

In the **Settings Interworking** section, you can provide the following information:



The screenshot shows the 'Settings Interworking' configuration page. The 'Forced Request URI' and 'Route Header Stripping' fields are highlighted in yellow. The 'Forced Request URI' is set to 'True' and 'Route Header Stripping' is set to 'True'. Other fields include 'Enable Call Forwarding Loop Prevention' (false), 'Provisional Response ACK' (Disabled), 'Enable Multiple Redirect Contacts Handling' (false), 'Multiple Redirect Contact Handling' (Parallel), 'Request URI Host Mode' (false), 'Force Host PAI' (DISABLED), 'Replace Domain In Route' (false), 'Include User phone parameter in URIs' (false), 'Treatment info Diversion Inbound Side' (None), 'Treatment info Diversion Outbound Side' (None), 'Destination Redirection Policy' (no action), 'Call Deflection' (false), 'Send SDP after FRACK' (false), 'Update Interworking' (Enable), 'Enable Origin' (false), 'SIP DSCP' (Disabled), 'Audio DSCP' (Disabled), 'Video DSCP' (Disabled), 'Other DSCP' (Disabled), 'Trunk Group Value', 'Trunk Context Value', 'Send Trunk Group' (false), 'Enable NAT presence' (false), 'Media Latching' (None), and 'Route Header Use to Routing' (Priority).

The **Forced Request URI** field sets the Forced replacement of the Request-URI with the IP address and destination port.

The **Route Header Stripping** field Enabling to strip the top most route typically inserted by NM-S-CI on outgoing side.

Parameter	Value
Forced Request URI	True
Route Header Stripping	True

5.13.1.1 Adding SIP, Digit and SDP Manipulations

In addition, it is possible to associate to a SIP Domain one or more manipulation rules. You can associate SIP and Digit Manipulations previously configured and one or more SDP Manipulations. All these settings can be defined for Inbound and Outbound direction.

In the following example, for all calls incoming into the selected SIP domain, the SIP Manipulation “**userPhone**” (previously imported) is applied in the Inbound Side:



In the following example, for all Domain Role and call side, insert these SDP Manipulations

Inbound/Incoming

Inbound/Incoming	
Action	Param
Remove Media Stream	m=video
Remove Line	m=application

Outbound **Inbound**

SDP Manipulations

Incoming

Action

Remove Media Stream

Media to remove

m=video

Incoming

Action

Remove Media Stream

Media to remove

m=application

Outbound/Incoming

Outbound/Incoming	
Action	Param
Remove Media Stream	m=video
Remove Line	m=application

Outbound Inbound |

SDP Manipulations

Incoming

Action

Remove Media Stream ▼

Media to remove

m=video

Incoming

Action

Remove Media Stream ▼

Media to remove

m=application

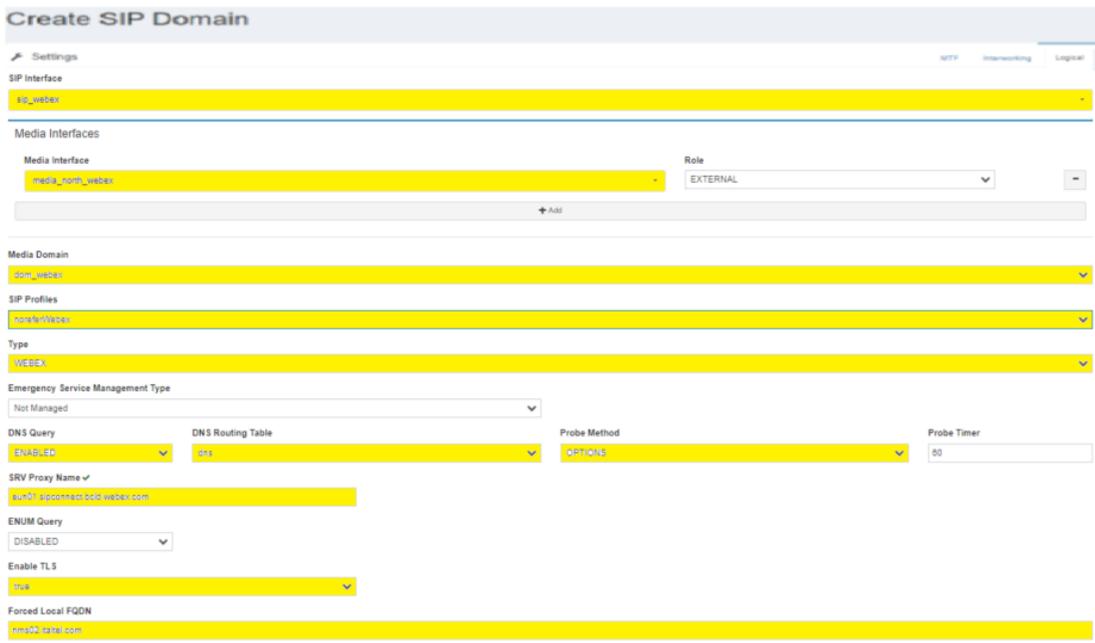
Note: In the SDP Manipulation section, you can also introduce, for example, an action to remove SDP codec that SBC must not forward to PSTN domain.

5.13.2 Create SIP Domain for Webex Calling side for No Media Bypass option

Click **+ New** to create a new SIP Domain the following view is displayed:

The following describes the information that you have to change or insert to create SIP Domain WEBEX for each section.

In the **Settings Logical** section, you can provide the following information:



The screenshot shows the 'Create SIP Domain' configuration page. The fields are as follows:

- SIP Interface:** A drop-down menu with 'sbc_webex' selected.
- Media Interfaces:** A table with one entry: 'media_sbc_webex' and 'EXTERNAL'.
- Media Domain:** A drop-down menu with 'sbc_webex' selected.
- SIP Profiles:** A drop-down menu with 'norefWebex' selected.
- Type:** A drop-down menu with 'WEBEX' selected.
- Emergency Service Management Type:** A dropdown menu with 'Not Managed' selected.
- DNS Query:** A dropdown menu with 'ENABLED' selected.
- DNS Routing Table:** A dropdown menu with 'sbc' selected.
- Probe Method:** A dropdown menu with 'ACTIVE' selected.
- Probe Timer:** A text input field with '50'.
- SRV Proxy Name:** A text input field with 'sbc.sbc.com'.
- ENUM Query:** A dropdown menu with 'DISABLED' selected.
- Enable TLS:** A dropdown menu with 'true' selected.
- Forced Local FQDN:** A text input field with 'sbc.sbc.com'.

The **Name** field, a string identifier.

The **Sip Interface** field (drop-down menu) will show all the SIP Interfaces previously configured on the system, and that will be used for signalling flows exchange.

In the **Media Interface** field (drop-down menu) will show all the Media Interface previously configured on the system, and that will be used for media flows exchange.

In the **Media Domain** field (drop-down menu) will show all Media Domain previously configured on the system, and that will be used for media different features configured, like a profile with different configuration.

The **SIP Profiles** (drop-down menu) will show all profiles previously configured on the system in order to choose one of them to apply a particular set of values for the typical SIP parameters (Timers, Allowed Methods, Allowed Headers, etc.).

In case of No Media Bypass option select **norefWebex**

The **DNS Query** select box provides the opportunity to enable the respective queries.

The **DNS routing Table** provides the list of Routing Tables configured for DNS service.

The **Probe Method** provides the different modalities to probe.

In the **Enable TLS** field, **select true**.

In the **Type** field, select a SIP domain type.

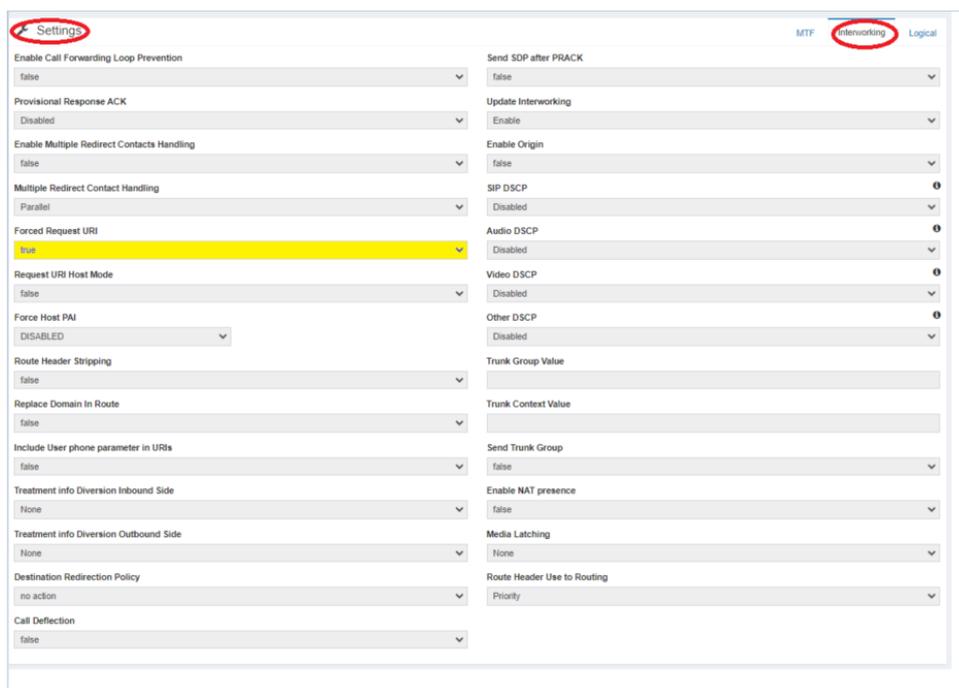
The **SRV Proxy name** field, insert the domain name to discover the access edge service and performing a DNS SRV lookup

In the **Forced Local FQDN** field, the FQDN (Fully Qualified Domain Name) specified in this field overwrites the domain part of the Contact and Record Route Headers.

This value is the SBC FQDN, the same value set into the 'Subject Alternative Names [SAN]' field in the TLS certificate.

Parameter	Value
Name	sip_webex.com
SIP Interface	sip_webex
Media Interface	media_north_webex
Media Domain	dom_north_webex
SIP Profiles	noreferWebex
Type	WEBEX
DNS Query	ENABLED
DNS Routing Tabled	dns
Probe Method	OPTIONS
SRV Proxy name	eun01.sipconnect.bcl.d.webex.com
Enable TLS	True
Forced Local FQDN	nms02.italtel.com

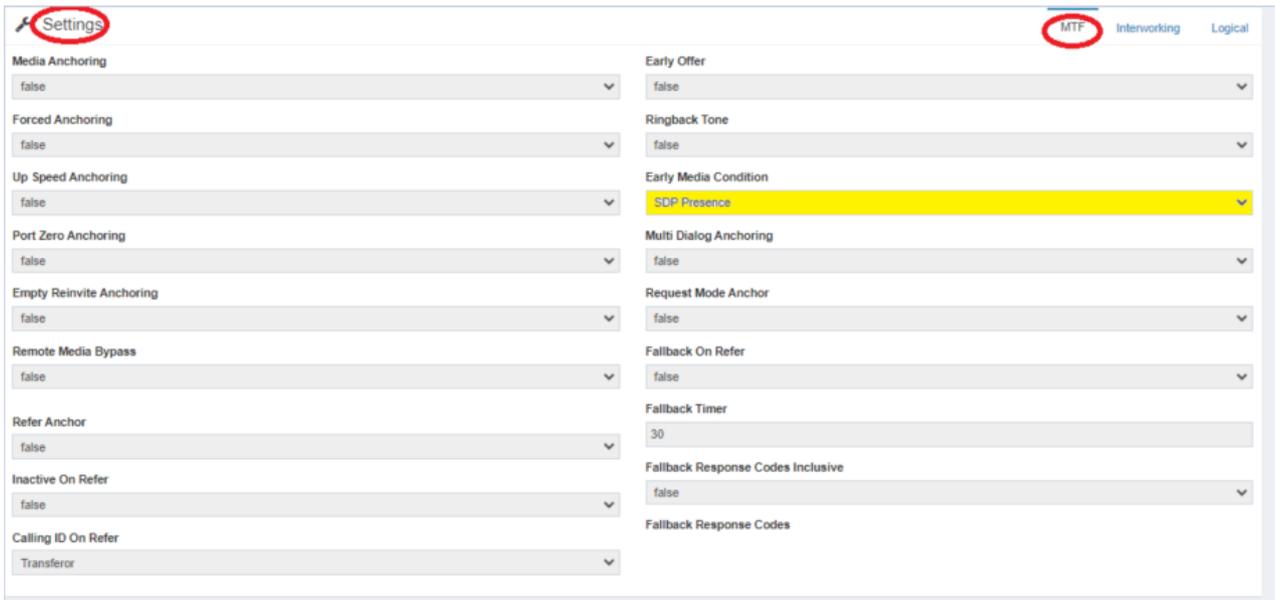
In the **Settings Interworking** section, you can provide the following information:



The **Forced Request URI** field sets the Forced replacement of the Request-URI with the IP address and destination port.

Parameter	Value
Forced Request URI	True

In the **Settings MTF** section, you can provide the following information:



The **Modality** and **Early Media Condition** fields are used to define the tone emission mode

Parameter	Value
Early Media Condition	SDP Presence

5.13.2.1 Adding SIP, Digit and SDP Manipulations

In addition, it is possible to associate to the SIP Domain one or more manipulation rules. You can associate SIP and Digit Manipulations previously configured and one or more SDP Manipulations. All these settings can be defined for Inbound and Outbound directions.

In the following example, for all call inbound side to the domain the previously imported SIP Manipulation “**userPhoneWebex**” is applied in Inbound Side:



At the end, you have (for example):

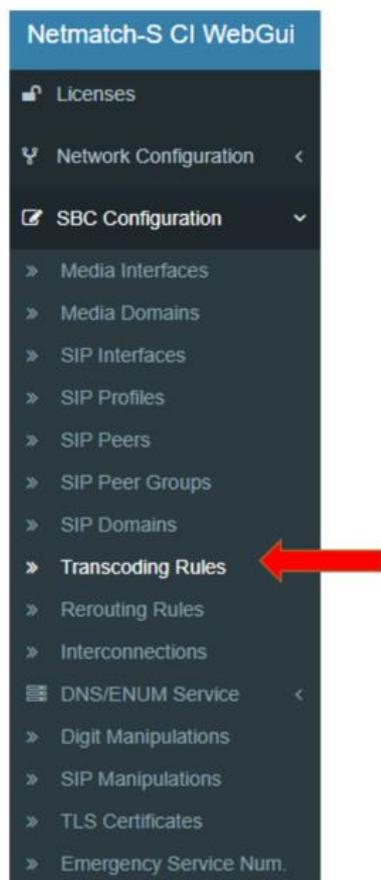
SIP Domains										+ New
Name	SIP Interface	Policy	SIP Profile	Media Interface	Media Domain	Def	TLS	Actions		
path.it	sip_sud_psth - [138.132.66.68 : 5060]	INGRESS_ADDRESS	SYSTEM_DEFAULTS	media_sud_psth - 138.132.66.68 [10000-12001]	dom_sud_psth	<input checked="" type="checkbox"/>	false	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
webex	sip_webex - [138.132.65.47 : 5060]	INGRESS_ADDRESS	norefer	media_north_webex - 138.132.65.47 [10000-12001]	dom_webex	<input type="checkbox"/>	true	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Showing 1 to 5 of 5 entries

5.14 How to create Transcoding Rules

In order to create a Transcoding Rule, select **SBC Configuration >> Transcoding Rules** in the main menu:

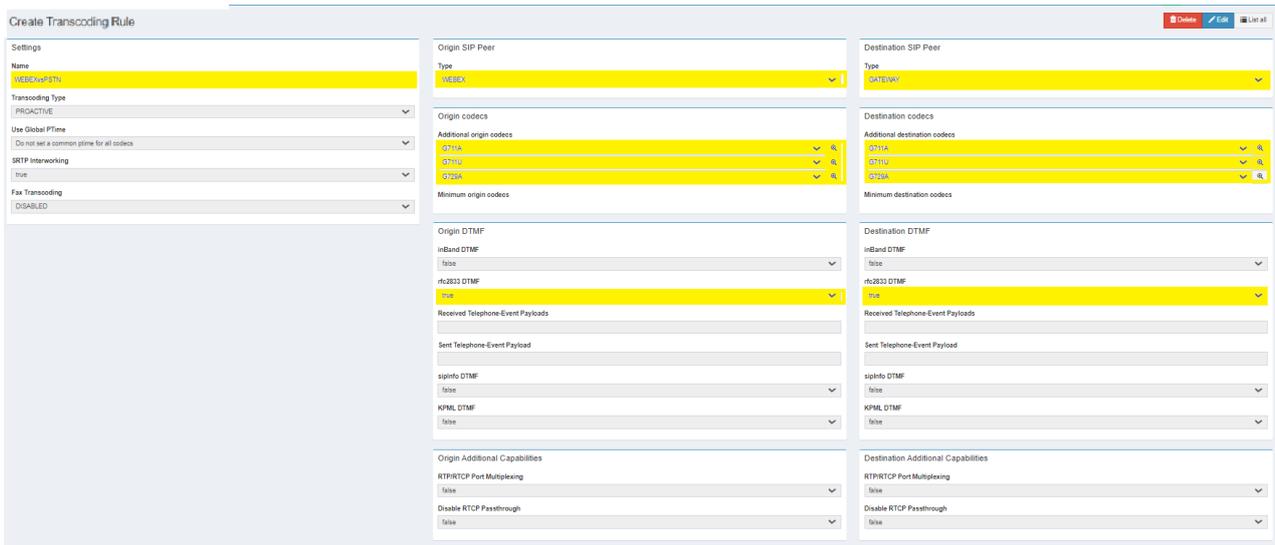
Then the **Transcoding Rules List** is displayed.



5.14.1 Create Transcoding Rules from Webex Calling to PSTN

Click  to create a new Transcoding Rule, the following view is displayed:

The following describes the information that you have to change or insert to create rule Webex Calling-PSTN:



In the **Settings** section, you can provide the following information:

Name, a string identifier.

In tab **Origin SIP Peer**, field **Type**, specify the type of terminals present on the call side in order to compose the SDP with attributes congruent to these terminals.

In tab **Destination SIP Peer**, field **Type**, specify the type of terminals present on the call side in order to compose the SDP with attributes congruent to these terminals.

In tab **Origin Codecs and Destination Codecs** add codecs G711A, G711U and G729A.

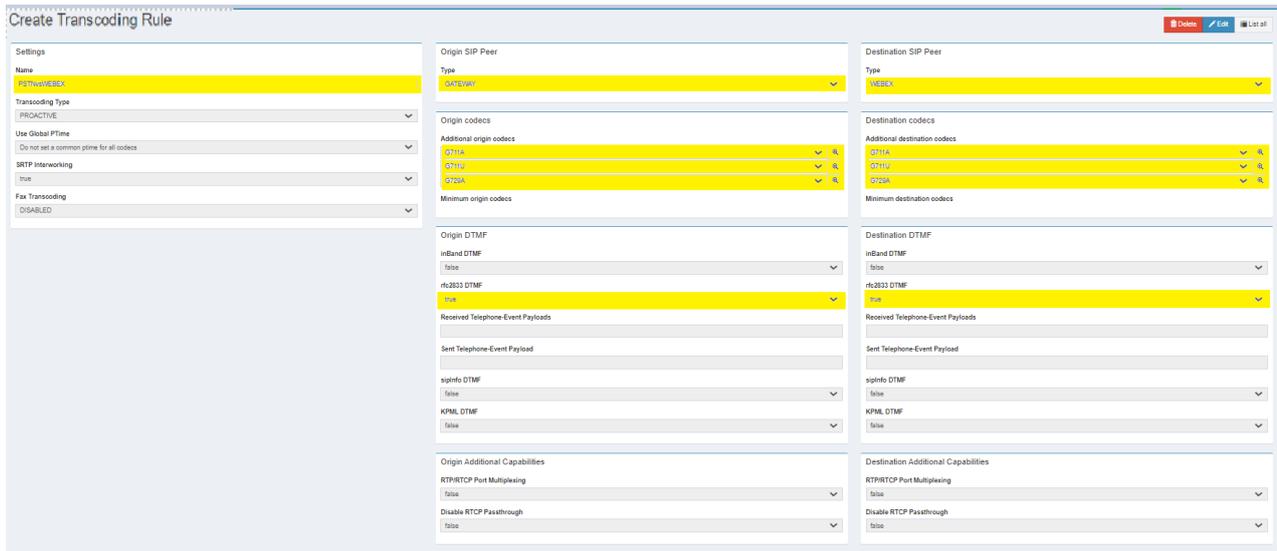
In tab **Origin DTMF and Destination DTMF**, in **rfc2833 DTMF** field set dual-tone multi-frequency (DTMF) signalling as specially marked RTP packets according to **RFC2833**.

Parameter	Value
Name	WEBEXvsPSTN
Origin SIP Peer Type	WEBEX
Destination SIP Peer Type	GATEWAY
Origin Codecs	G711A, G711U, G729A
Origin DTMF – rfc2833 DTMF	True
Destination Codecs	G711A, G711U, G729A
Destination DTMF – rfc2833 DTMF	True

5.14.2 Create Transcoding Rules from PSTN to Webex Calling

Click  to create a new Transcoding Rule, the following view is displayed.

The following describes the information that you have to change or insert to create rule PSTN-Webex Calling:



In the **Settings** section, you can provide the following information:

Name, a string identifier.

In tab **Origin SIP Peer** field **Type** specify the type of terminals present on the call side in order to compose the SDP with attributes congruent to these terminals.

In tab **Destination SIP Peer** field **Type** specify the type of terminals present on the call side in order to compose the SDP with attributes congruent to these terminals.

In tab **Origin Codecs** and **Destination Codecs** add codecs G711A, G711U and G729A.

In tab **Origin DTMF** and **Destination DTMF**, in field rfc2833 send dual-tone multi-frequency (DTMF) signalling as specially marked RTP packets according [to RFC2833](#).

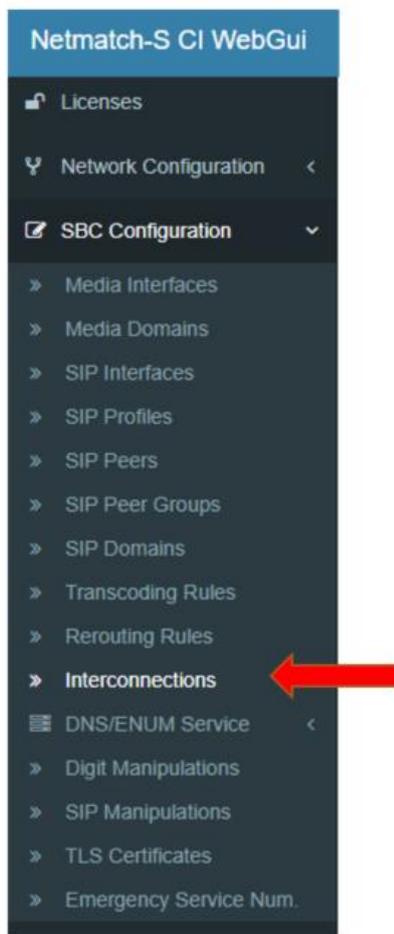
Parameter	Value
Name	PSTNvsWEBEX
Origin SIP Peer Type	GATEWAY
Destination SIP Peer Type	WEBEX
Origin Codecs	G711A, G711U, G729A
Origin DTMF – rfc2833 DTMF	True
Destination Codecs	G711A, G711U, G729A
Destination DTMF – rfc2833 DTMF	True

At the end, you will have (for example):

Transcoding Rules				+ New
Transcoding Rules				Search: <input type="text"/>
Show: 10 entries				
Name	Transcoding Type	Fax Transcoding Type	Actions	
PSTNtoWEBEX	PROACTIVE	DISABLED		
WEBEXtoPSTN	PROACTIVE	DISABLED		
Showing 1 to 4 of 4 entries				
				Previous 1 Next

5.15 How to create Interconnection

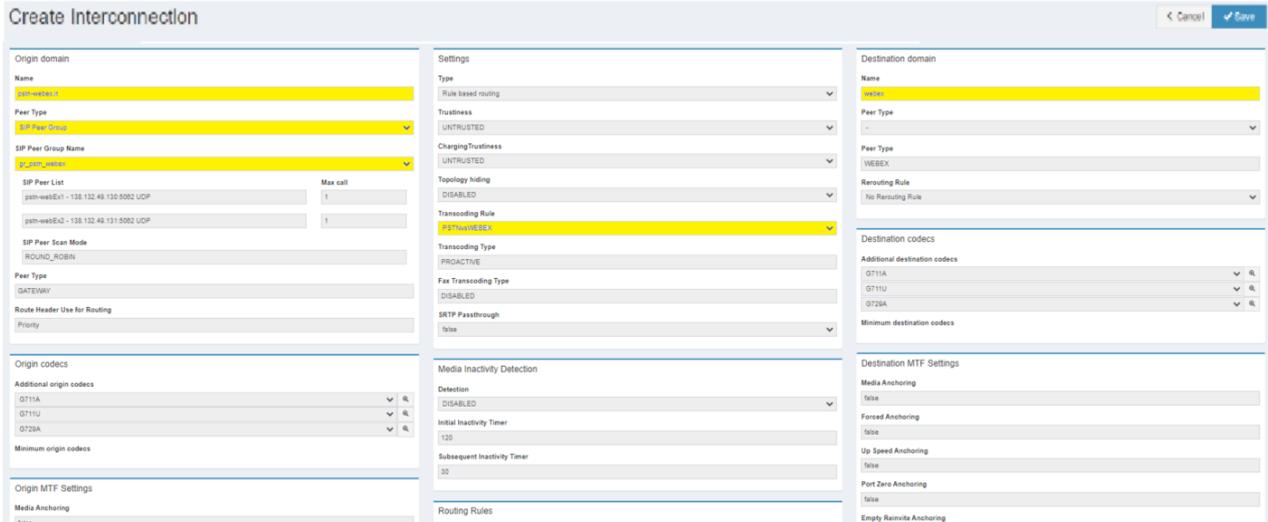
To create, customize and view the interconnections with additional (optional) features, select **SBC Configuration >> Interconnections** in the main menu.



Click  to create a new interconnection, the following view is displayed (with Transcoding Rule associated)

5.15.1 Create Interconnection from PSTN to Webex Calling for No Media Bypass option

The following describes the information that you have to change or insert to create an interconnection PSTN-Webex Calling:



In **Orig Domain** tab for **Name** field (select **pstn.com**).

In **Orig Domain** tab for **Peer Type** field (select **Sip Peer Group**).

In **Orig Domain** tab for **Sip Peer Group Name** field (select **gr_pstn**).

In **Settings** tab for **Transcoding Rule** field (select **PSTNvsWEBEX**).

In **Destination Domain** tab for **Name** field (select **sip_webex.com**).

In **Routing Rules** tab push on Add Number routing policy and insert REMOTE route number that identify destination .

Routing Rules

Route Number

Number

+39024388497

Route Number

Number

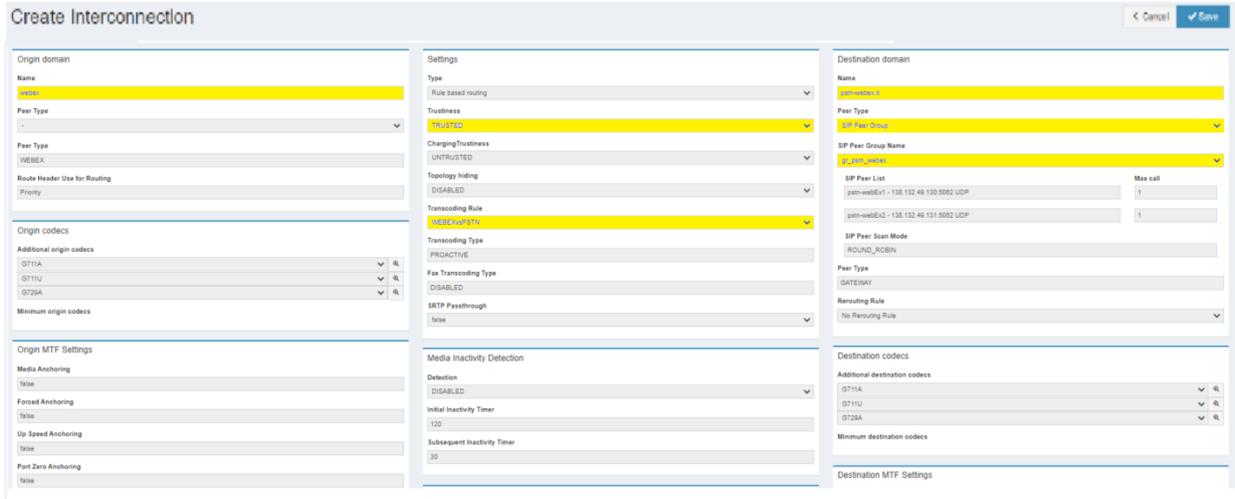
024388497

Parameter	Value
Origin Domain	

Name	pstn.com
Peer Type	Sip Peer Group
SIP Peer Group Name	gr_pstn
Settings	
Trascoding Rule	PSTNvsWEBEX
Type	Rule Based routing
Destination Domain	
Name	sip_webex.com

5.15.2 Create Interconnection from Webex Calling to PSTN for No Media Bypass option

The following describes the information that you have to change or insert to create an interconnection Webex Calling-PSTN:



In **Orig Domain** tab for **Name** field.

In **Settings** tab for **Trustiness** field.

In **Settings** tab for **Transcoding Rule** field.

In **Destination Domain** tab for **Name** field.

In **Destination Domain** tab for **Peer Type** field.

In **Destination Domain** tab for **Sip Peer Group** Name field.

In **Routing Rules** tab push on Add Number routing policy and insert REMOTE route number that identify destination .

Routing Rules

Route Number
Number
+39

Route Number
Number
0039

Route Number
Number
342

Parameter	Value
Origin Domain	
Name	Sip_webex.com
Settings	
Trustiness	TRUSTED
Transcoding Rule	WEBEXvsPSTN
Type	Rule Based routing
Destination Domain	
Name	pstn.com
Peer Type	Sip Peer Group
SIP Peer Group Name	gr_pstn

At the end, you will have (for example):

Interconnections + New

Interconnections Search:

Show 10 entries

Id	Origin	Origin Peer Group	Destination	Destination Peer Group	Type	Route Use	Routing Policy	Transcoding Rule	Trustiness	Charging/Trustiness	Topology Hiding	Media Inactivity Detection	Actions
0	pstn.com	gr_pstn	pstnhub-ppe.skype.net	-	Static routing	Priority	-	PSTNsTEAMS	TRUSTED	UNTRUSTED	DISABLED	DISABLED	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>
1	pstnhub- ppe.skype.net	-	pstn.com	gr_pstn	Static routing	Priority	-	TEAMSvsPSTN	TRUSTED	UNTRUSTED	HIDE_HEADER	DISABLED	<input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>

Showing 1 to 2 of 2 entries Previous 1 Next