Oracle SBC integration with Cisco CUCM and Twilio Elastic Sip Trunking

Technical Application Note

Disclaimer

The following is intended to outline our general product direction. It is intended for information purposes only, and may not be incorporated into any contract. It is not a commitment to deliver any material, code, or functionality, and should not be relied upon in making purchasing decisions. The development, release, and timing of any features or functionality described for Oracle’s products remains at the sole discretion of Oracle.

Revision History

Version	Description of Changes	Date Revision Completed
1.0	Oracle SBC integration with Cisco CUCM and Twilio Elastic SIP Trunking	21st May 2021

Table of Contents

1. Intended Audience

This document is intended for use by Oracle Systems Engineers, third party Systems Integrators, Oracle Enterprise customers and partners and end users of the Oracle Enterprise Session Border Controller (SBC). It is assumed that the reader is familiar with basic operations of the Oracle Enterprise Session Border Controller platform along with Cisco Call Manager (Cisco CUCM).

2. Document Overview

This Oracle technical application note outlines how to configure the Oracle SBC to interwork between Twilio Elastic Sip Trunk with on premises Cisco CUCM. The solution contained within this document has been tested using Oracle Communication SBC with **OS840p4A**.

Please find the related documentation links below:

2.1**.** Twilio Elastic SIP Trunking

Twilio Elastic SIP Trunking is a cloud-based solution that provides connectivity for IP-based communications infrastructure to connect to the PSTN for making and receiving telephone calls to the rest of the world via any broadband internet connection. Twilio's Elastic SIP Trunking service automatically scales, up or down, to meet your traffic needs with unlimited capacity. In just minutes you can deploy globally with Twilio's easy-to-use self-service tools without having to rely on slow providers.

Sign up for a free Twilio trial and learn more about configuring your Twilio Elastic SIP Trunk.

2.2**.** Cisco Call Manager (Cisco CUCM)

Cisco Unified Call Manager provides industry-leading reliability, security, scalability, efficiency, and enterprise call and session management and is the core call control application of the collaboration portfolio.

It should be noted that while this application note focuses on the optimal configurations for the Oracle SBC in an enterprise Cisco CUCM 11.5 environment, the same SBC configuration model can also be used for other enterprise applications with a few tweaks to the configuration for required features.

In addition, it should be noted that the SBC configuration provided in this guide focuses strictly on the Cisco CUCM Server associated parameters. Many SBC applications may have additional configuration requirements that are specific to individual customer requirements. These configuration items are not covered in this guide. Please contact your Oracle representative with any questions pertaining to this topic.

For additional information on CUCM 11.5, please visit

https://www.cisco.com/c/en/us/products/unified-communications/unified-communications-manager-version-11-5/index.html

Please note that the IP Addresses, FQDN and configuration names and details given in this document are used for reference purposes only. These same details cannot be used in customer configurations. End users of this document can use the configuration details according to their network requirements.

3. Introduction

3.1. Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring Cisco CUCM 11.5 version using Oracle Enterprise SBC. There will be steps that require navigating the CUCM 11.5 server configuration, Oracle SBC GUI interface, understanding the basic concepts of TCP/UDP, IP/Routing, DNS server and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

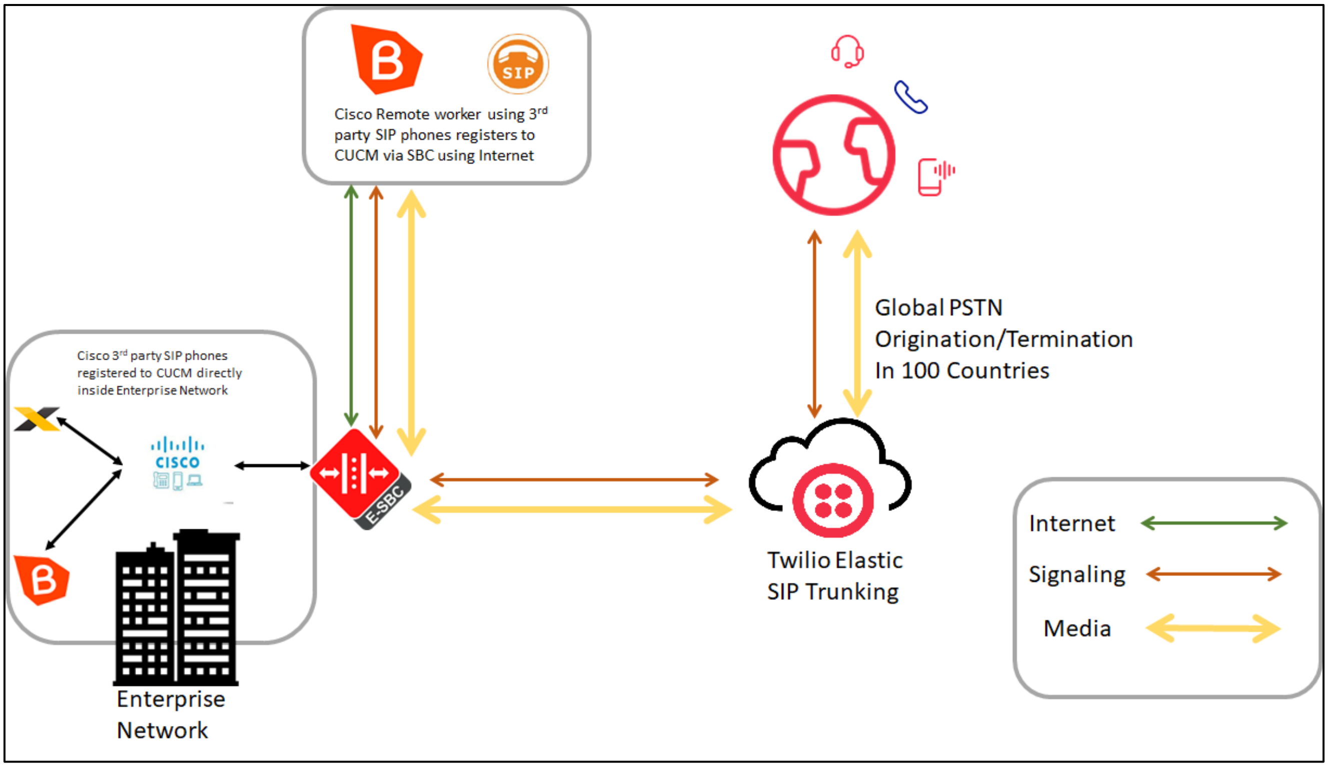
3.2. Requirements

- Fully functioning Cisco CUCM 11.5
- Oracle Enterprise Session Border Controller (hereafter Oracle SBC) running 8.4.0 version

The below revision table explains the versions of the software used for each component:
This table is Revision 1 as of now:

Software Used	SBC Version	Cisco CUCM Version
Revision 1	8.4.0	11.5

3.3. Architecture

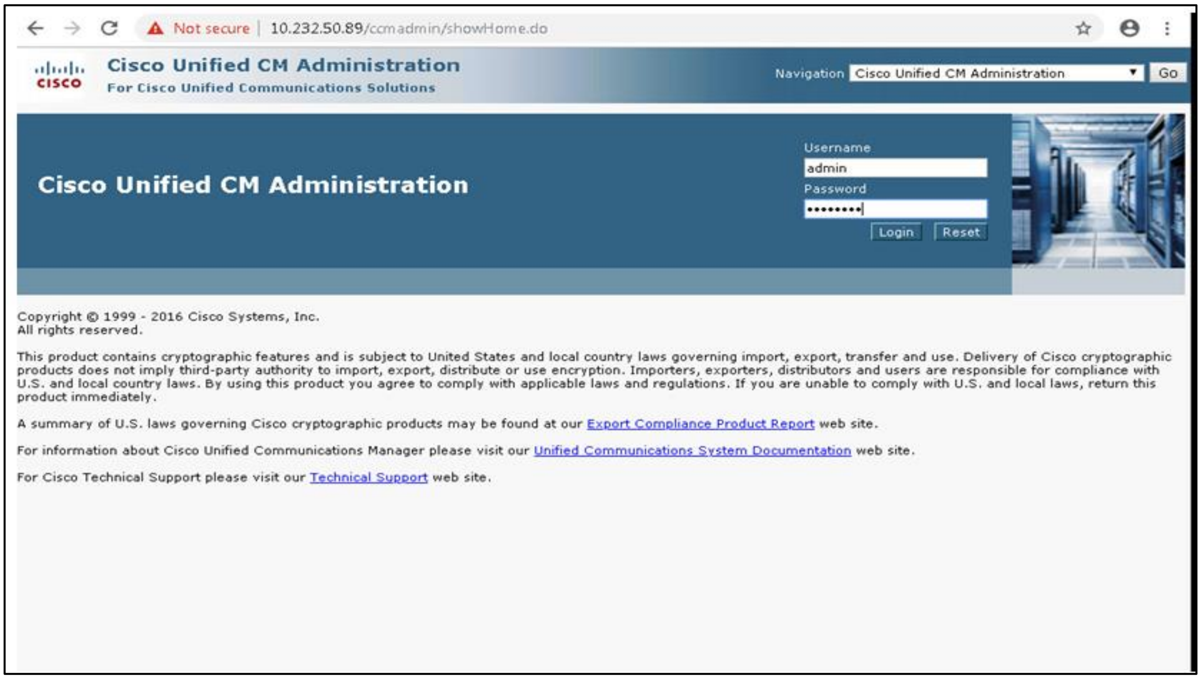

Cisco Remote worker using 3rd party SIP phones registers to CUCM via SBC using Internet
SIP
Cisco 3rd party SIP phones registered to CUCM directly inside Enterprise Network
cisco
E-SBC
Enterprise Network
Global PSTN Origination/Termination In 100 Countries
Twilio Elastic SIP Trunking
Internet
Signaling
Media

The configuration, validation and troubleshooting are the focuses of this document and will be described in three phases:

- Phase 1 – Configuring the Cisco Unified Call Manager v11.5 for Oracle SBC.
- Phase 2 – Configuring the Oracle SBC.
- Phase 3 – Configuring the Twilio Elastic SIP Trunk

4. Configuring the Cisco Call Manager (Cisco CUCM)

Please login to Cisco CUCM admin web GUI with proper login credentials (Username and password). After that, perform the steps below in the given order.



10.232.50.89/ccmadmin/showHome.do

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Cisco Unified CM Administration

Username: admin

Password: ••••••••

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

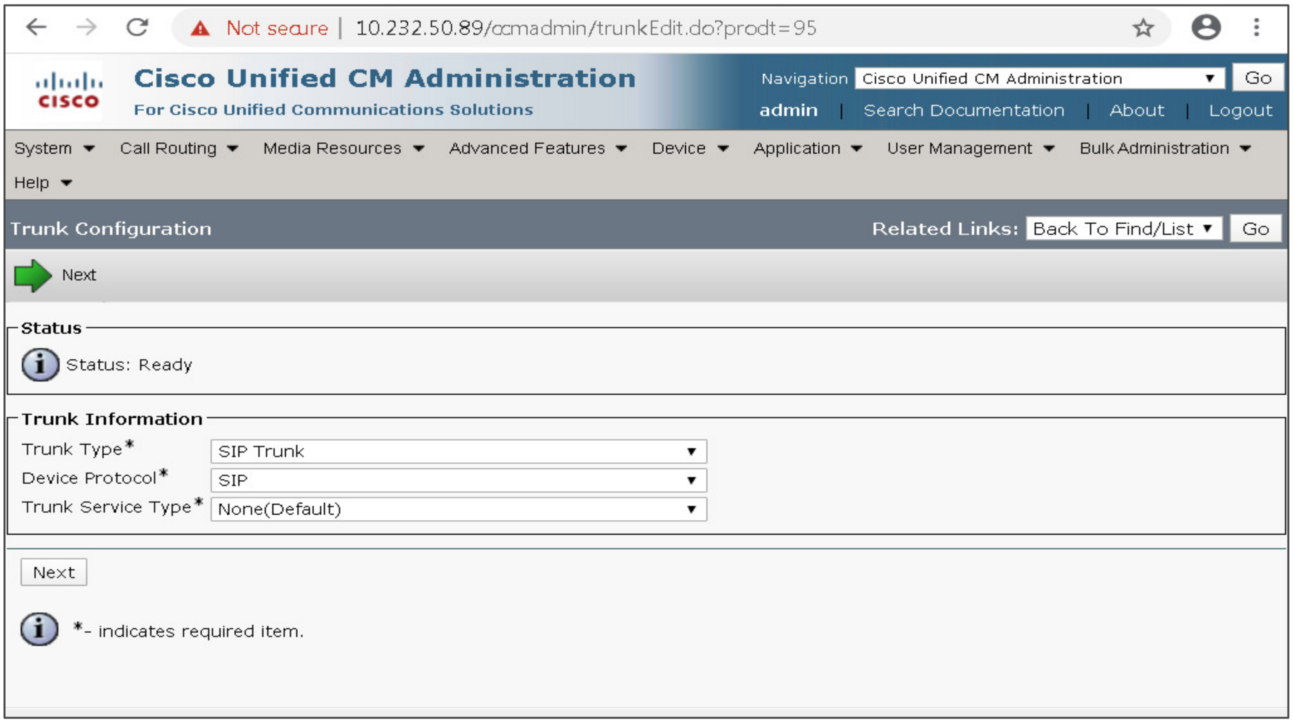
A summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.

For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.

4.1. Configuring a new SIP Trunk

01) Go to Device ----- Trunk ----- Add New
02) Select Trunk Type – SIP Trunk and then Click Next
03) In the Device Name field, enter the SIP Trunk name and optionally provide a description.
04) In the Device Pool drop-down list, select a device pool id created already else select Default
05) Enter the Destination Address and Destination Port of the SBC under SIP Information.
06) Select appropriate SIP profile and SIP trunk security profile from the dropdown menu.
07) Click Save


Not secure | 10.232.50.89/ccmadmin/trunkEdit.do?prodt=95
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Trunk Configuration

Status

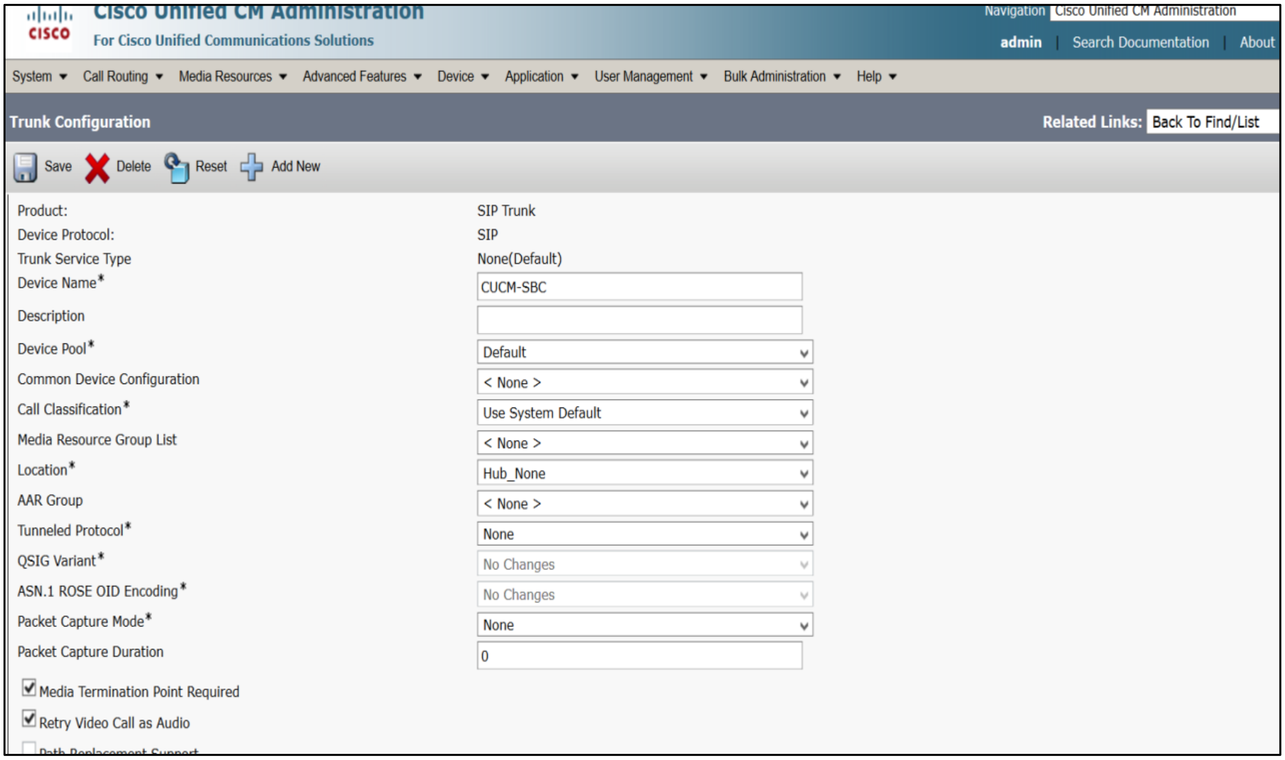
Status: Ready

Trunk Information

Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

Next

*- indicates required item.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Save Delete Reset Add New

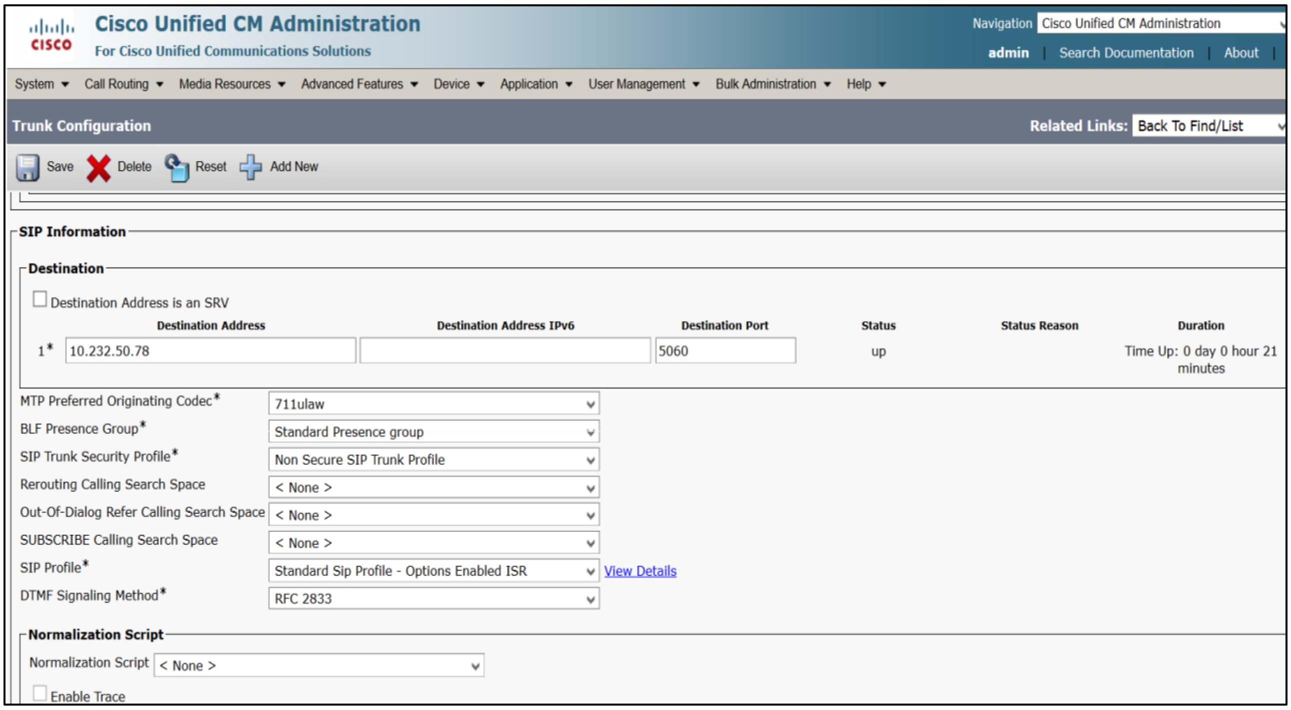
Trunk Configuration

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

☑ Media Termination Point Required

☑ Retry Video Call as Audio

□ Path Replacement Support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Related Links: Back To Find/List
Save Delete Reset Add New

Trunk Configuration

SIP Information

Destination

□ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	10.232.50.78		5060	up		Time Up: 0 day 0 hour 21 minutes

MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard Sip Profile - Options Enabled ISR View Details
DTMF Signaling Method*	RFC 2833

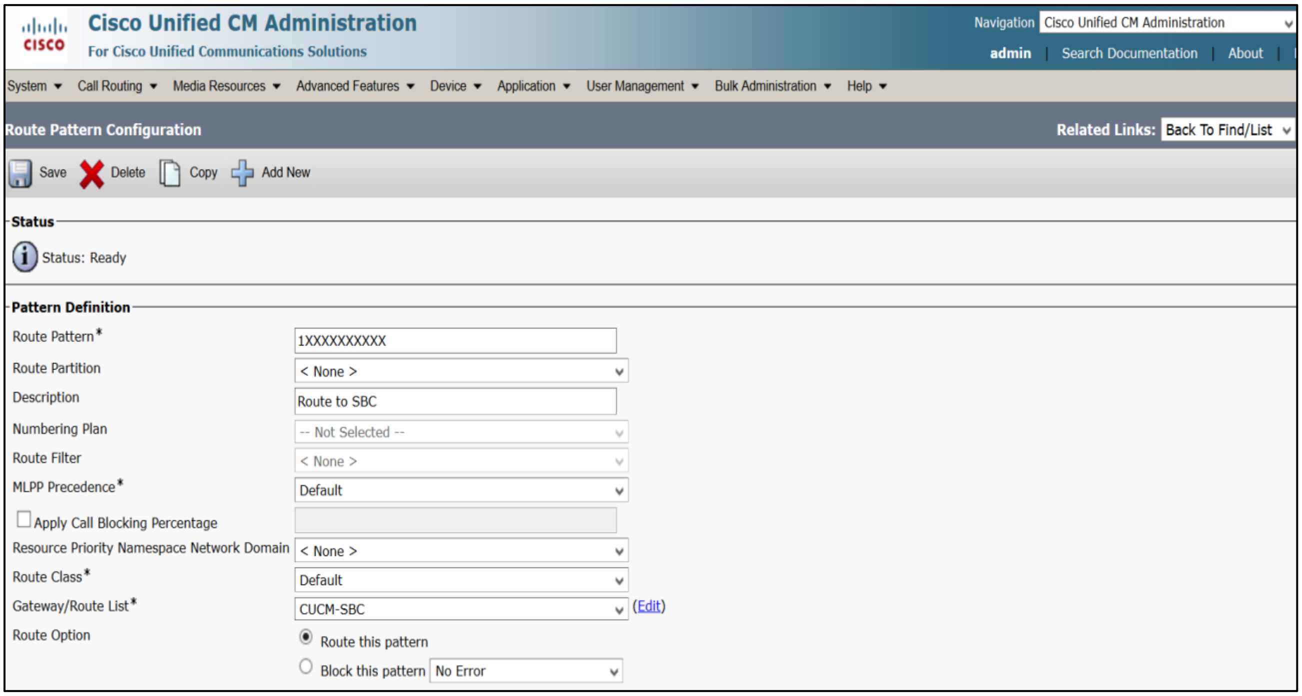
Normalization Script

Normalization Script < None >

□ Enable Trace

4.2. Configure a new Route Pattern

01) Go to Call Routing ------ Route/Hunt ------ Route Pattern and click Add New
02) Enter a Route Pattern according to the network requirements and calling plan.
03) From the Gateway/Route List drop-down list, select the created SIP Trunk device name.
04) Click Save. We can create other route patterns in the same way as shown below.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Related Links: Back To Find/List
Save Delete Copy Add New

Route Pattern Configuration

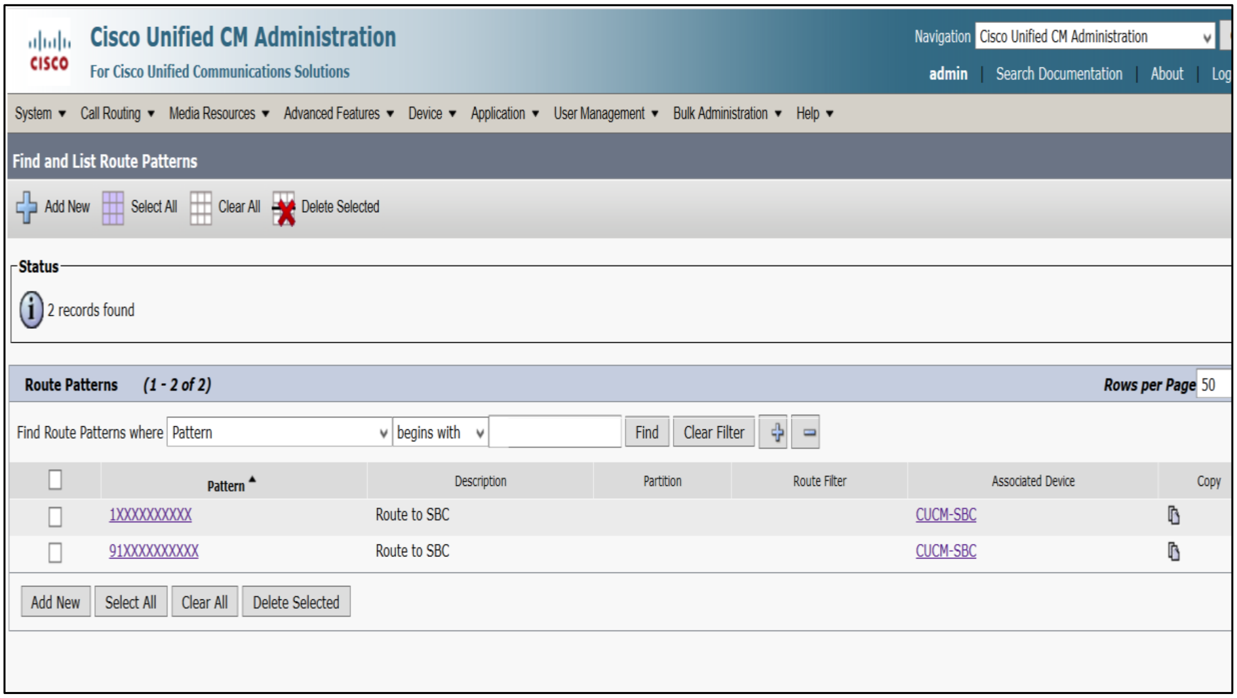
Status

Status: Ready

Pattern Definition

Route Pattern*	1XXXXXXXXXX
Route Partition	< None >
Description	Route to SBC
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
□ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CUCM-SBC (Edit)
Route Option	● Route this pattern
	○ Block this pattern No Error

The route patterns that has been created is shown below: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Add New
Select All
Clear All
Delete Selected
Find Route Patterns where Pattern begins with
Find
Clear Filter
Rows per Page 50

Find and List Route Patterns

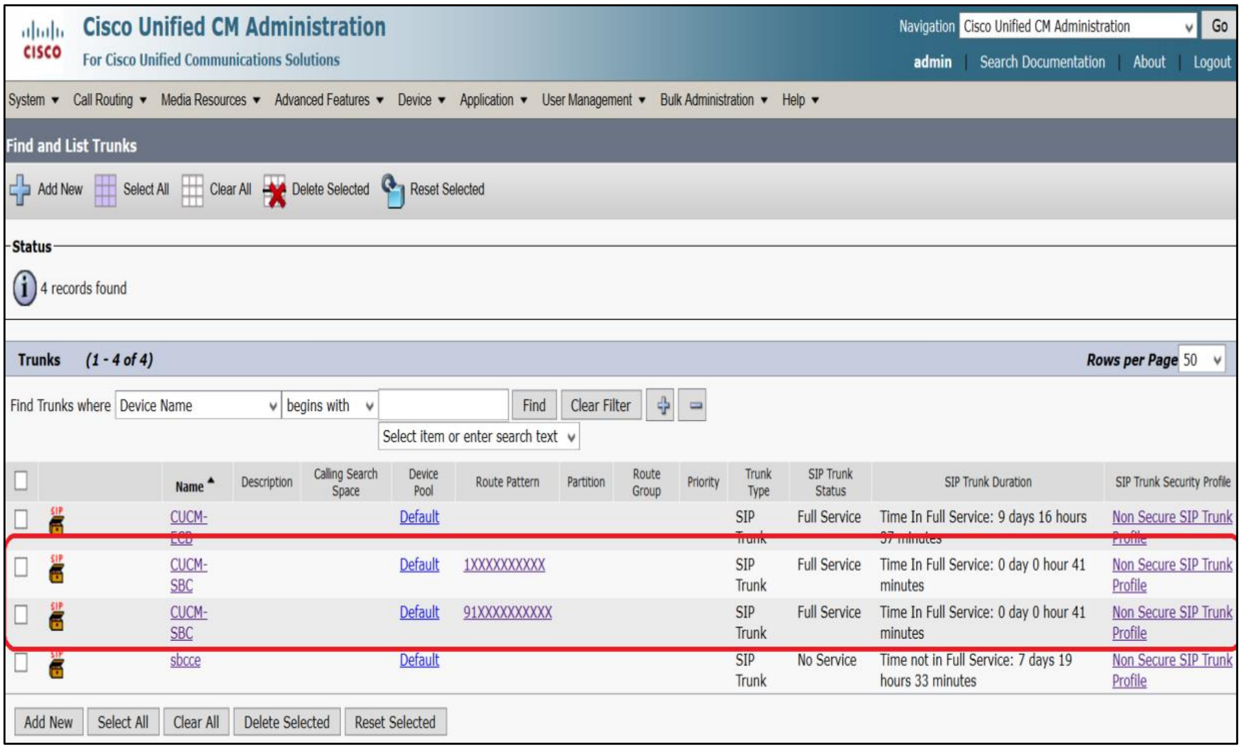
Status

2 records found

Route Patterns (1 - 2 of 2)

□	Pattern ▲	Description	Partition	Route Filter	Associated Device	Copy
□	1XXXXXXXXXX	Route to SBC			CUCM-SBC	
□	91XXXXXXXXXX	Route to SBC			CUCM-SBC	

The created SIP trunk associated with the route pattern is shown below: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Add New
Select All
Clear All
Delete Selected
Reset Selected
Find Trunks where Device Name begins with
Find
Clear Filter
Select item or enter search text
Rows per Page 50

Find and List Trunks

Status

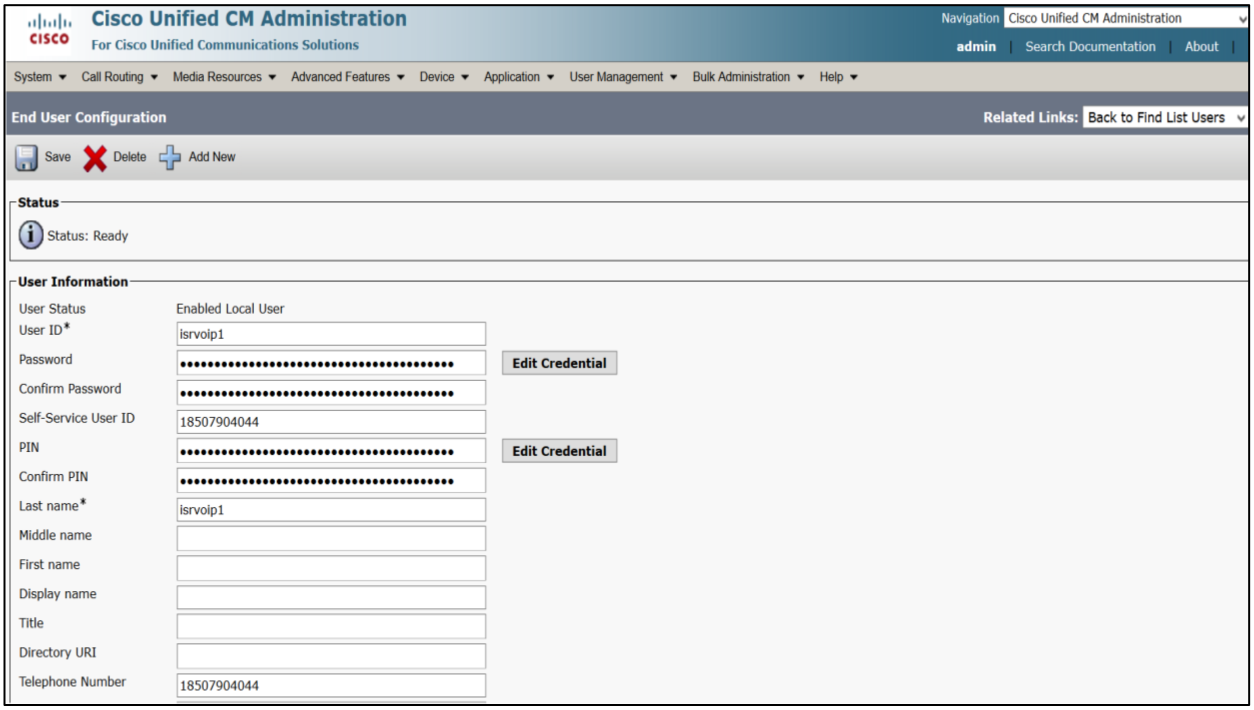
4 records found

Trunks (1 - 4 of 4)

□		Name ▲	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
□	SIP	CUCM-ECB			Default					SIP Trunk	Full Service	Time In Full Service: 9 days 16 hours 37 minutes	Non Secure SIP Trunk Profile
□	SIP	CUCM-SBC			Default	1XXXXXXXXXX				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 41 minutes	Non Secure SIP Trunk Profile
□	SIP	CUCM-SBC			Default	91XXXXXXXXXX				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 41 minutes	Non Secure SIP Trunk Profile
□	SIP	sbcce			Default					SIP Trunk	No Service	Time not in Full Service: 7 days 19 hours 33 minutes	Non Secure SIP Trunk Profile

4.3. End User Configuration

01) Go to User Management ---- End User and click Add New
02) Enter in your User ID, password, pin, and Last Name
03) You must also enter in a password in the Digest Credentials and Confirm.
04) Click Save (remember the User ID and Password and DN of the device)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Related Links: Back to Find List Users
Save
Delete
Add New
Edit Credential
Edit Credential

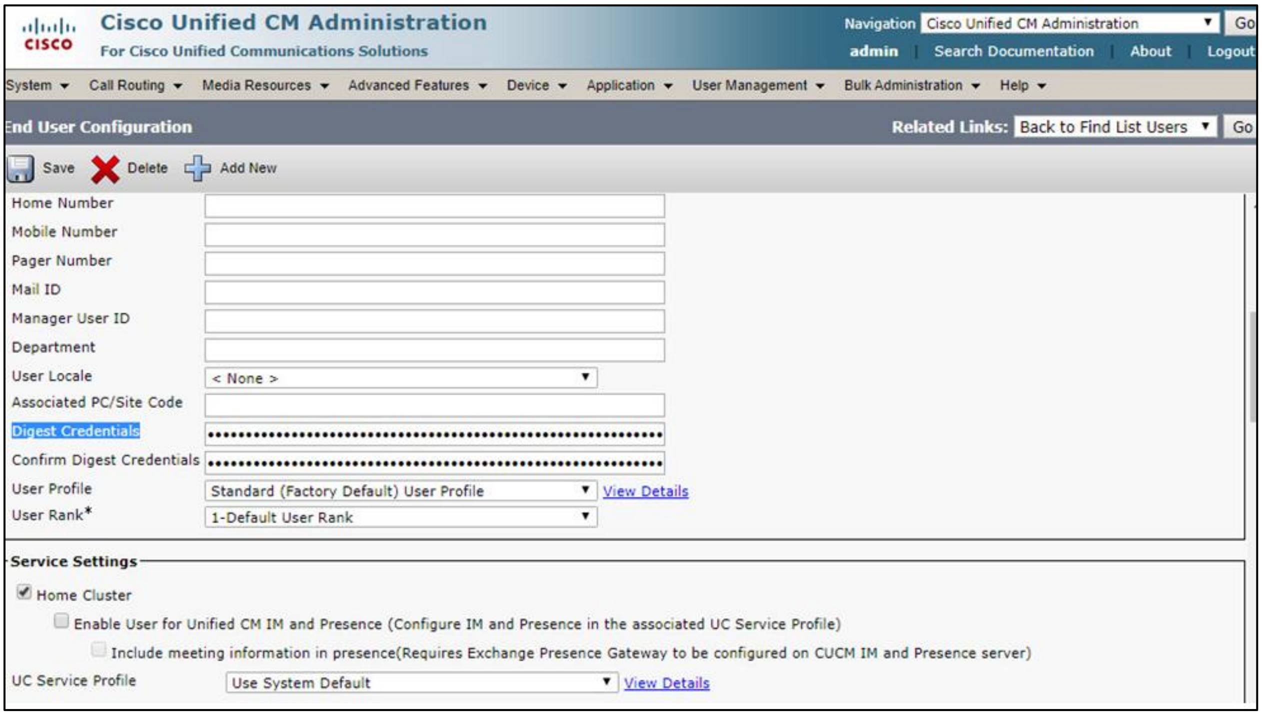
End User Configuration

Status

Status: Ready

User Information

User Status	Enabled Local User
User ID*	isrvoip1
Password	•••••••••••••••••••••••••••••••••••••••
Confirm Password	•••••••••••••••••••••••••••••••••••••••
Self-Service User ID	18507904044
PIN	•••••••••••••••••••••••••••••••••••••••
Confirm PIN	•••••••••••••••••••••••••••••••••••••••
Last name*	isrvoip1
Middle name	
First name	
Display name	
Title	
Directory URI	
Telephone Number	18507904044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Related Links: Back to Find List Users
Go
Save
Delete
Add New
View Details
View Details

End User Configuration

Home Number	
Mobile Number	
Pager Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None >
Associated PC/Site Code	
Digest Credentials	••
Confirm Digest Credentials	••
User Profile	Standard (Factory Default) User Profile
User Rank*	1-Default User Rank

Service Settings

☑ Home Cluster

☐ Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

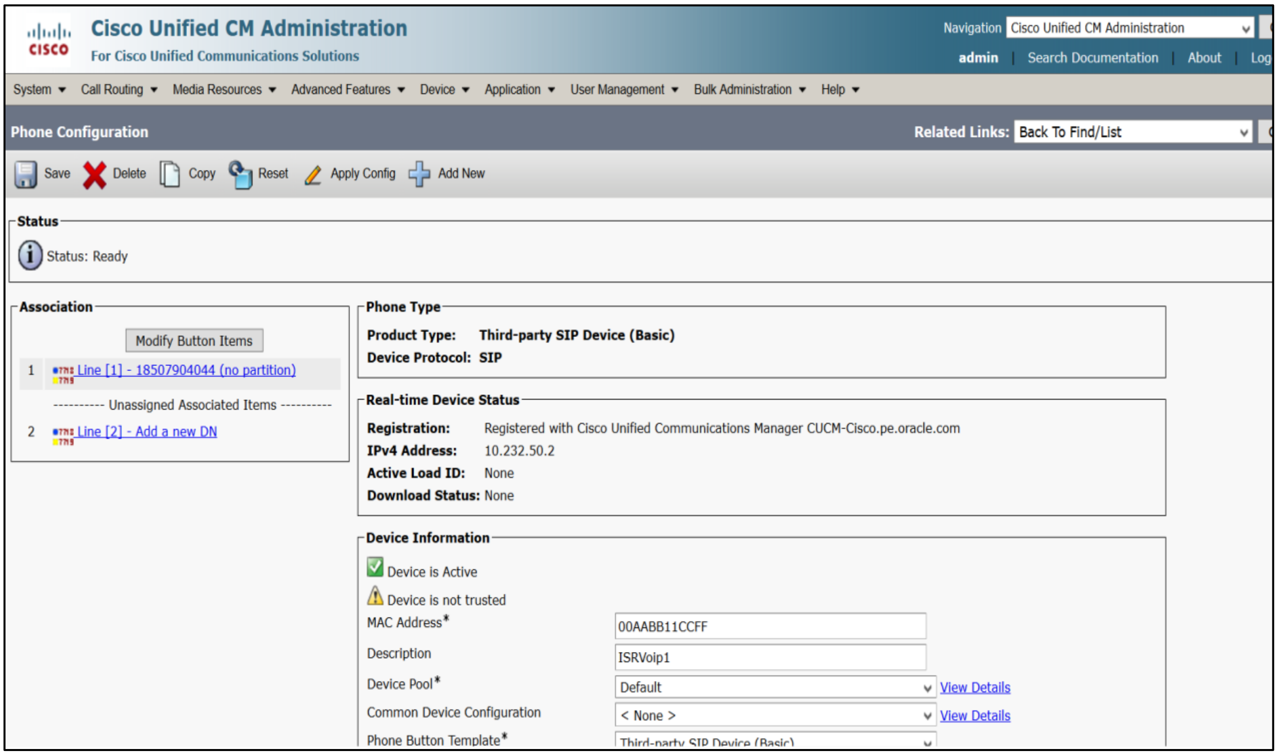
☐ Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

UC Service Profile: Use System Default

4.4. Adding SIP Phone in CUCM

01) Go to Device ---- Phone and click Add New
02) Select Third Party Sip Device (Basic) and click Next
03) Enter in a 12 digit MAC address (any dummy MAC address)
04) Enter the pertinent information for the SIP DEVICE settings – it should mostly be configured the same as
a standard phone on your system except for the following settings
a) in the owner user ID field select the user you created above
b) in the Device Security Profile field select the security profile you created above
c) in the Digest User field select the user you created above

05) Click Save.
06) Configure the line settings for the SIP device – the line settings should match the line settings of your standard user's Cisco IP phones
There are no special attributes that we need to worry about on the line configuration.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Related Links: Back To Find/List
Save Delete Copy Reset Apply Config Add New
Modify Button Items

Phone Configuration

Status
Status: Ready

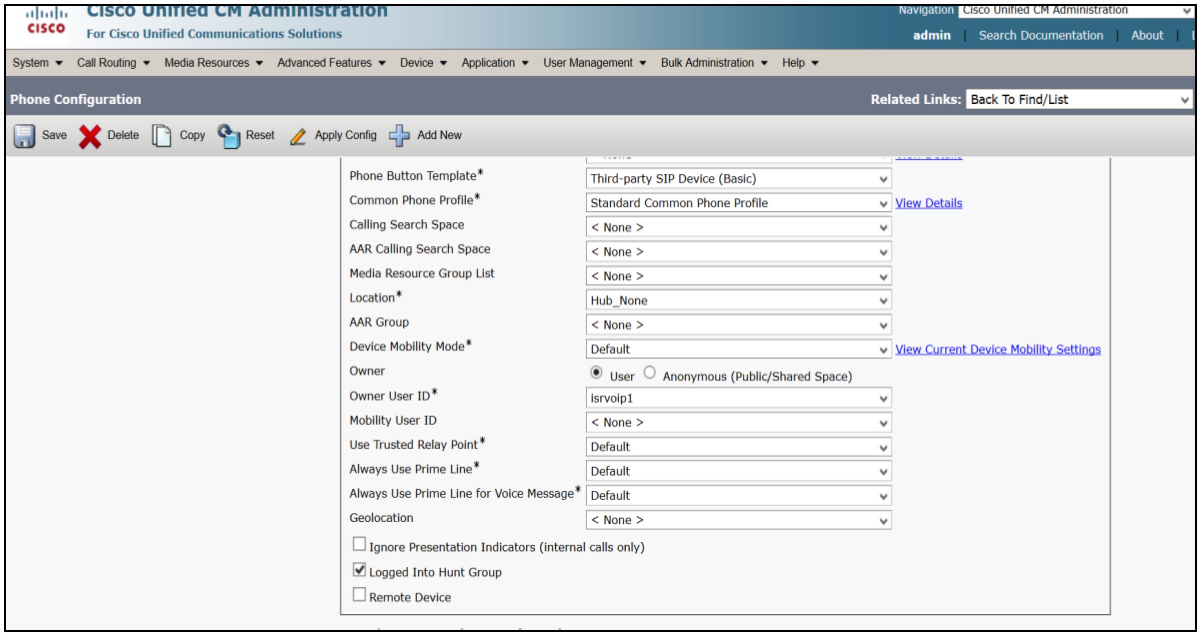
Association
1 Line [1] - 18507904044 (no partition)
---------- Unassigned Associated Items ----------
2 Line [2] - Add a new DN

Phone Type
Product Type: **Third-party SIP Device (Basic)**
Device Protocol: **SIP**

Real-time Device Status
Registration: Registered with Cisco Unified Communications Manager CUCM-Cisco.pe.oracle.com
IPv4 Address: 10.232.50.2
Active Load ID: None
Download Status: None

Device Information
☑ Device is Active
⚠ Device is not trusted

MAC Address*	00AABB11CCFF
Description	ISRVoip1
Device Pool*	Default View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Third-party SIP Device (Basic)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Phone Configuration
Related Links: Back To Find/List
Save Delete Copy Reset Apply Config Add New

Phone Button Template* Third-party SIP Device (Basic)

Common Phone Profile* Standard Common Phone Profile View Details

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List < None >

Location* Hub_None

AAR Group < None >

Device Mobility Mode* Default View Current Device Mobility Settings

Owner ● User ○ Anonymous (Public/Shared Space)

Owner User ID* isrvoip1

Mobility User ID < None >

Use Trusted Relay Point* Default

Always Use Prime Line* Default

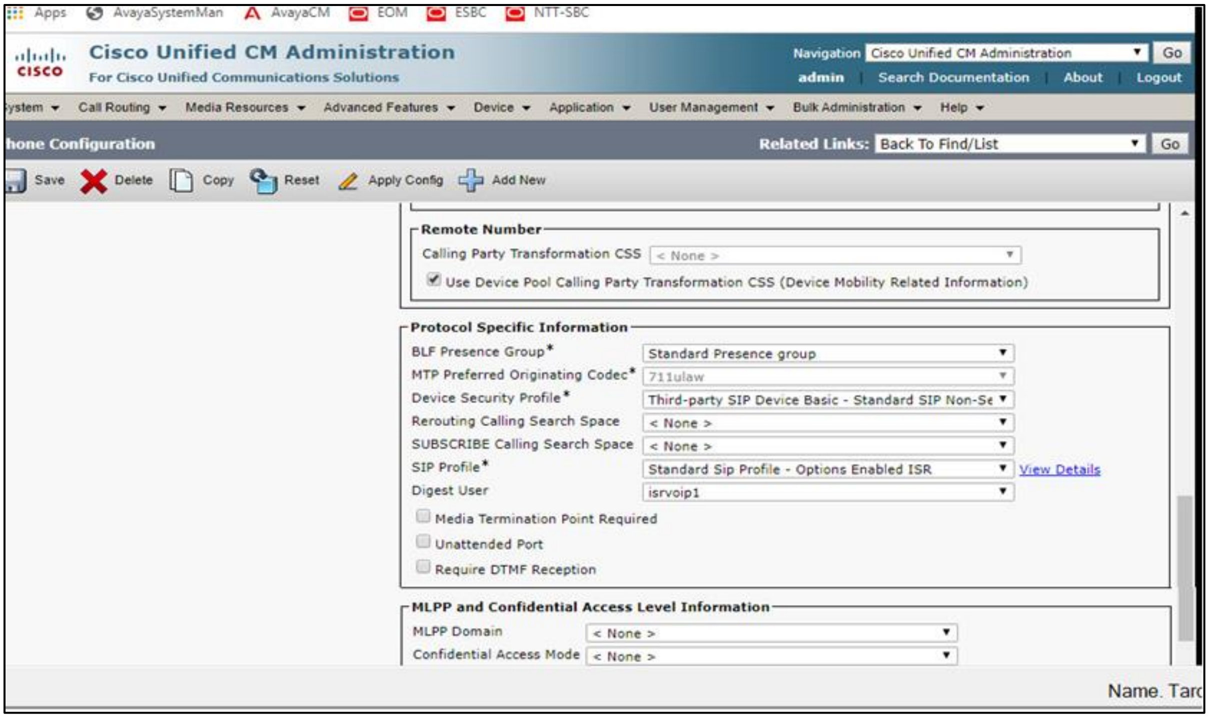
Always Use Prime Line for Voice Message* Default

Geolocation < None >

□ Ignore Presentation Indicators (internal calls only)

☑ Logged Into Hunt Group

□ Remote Device


Apps AvayaSystemMan AvayaCM EOM ESBC NTT-SBC
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hone Configuration
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Remote Number

Calling Party Transformation CSS < None >

☑ Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Protocol Specific Information

BLF Presence Group* Standard Presence group

MTP Preferred Originating Codec* 711ulaw

Device Security Profile* Third-party SIP Device Basic - Standard SIP Non-Se

Rerouting Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard Sip Profile - Options Enabled ISR View Details

Digest User isrvoip1

□ Media Termination Point Required

□ Unattended Port

□ Require DTMF Reception

MLPP and Confidential Access Level Information

MLPP Domain < None >

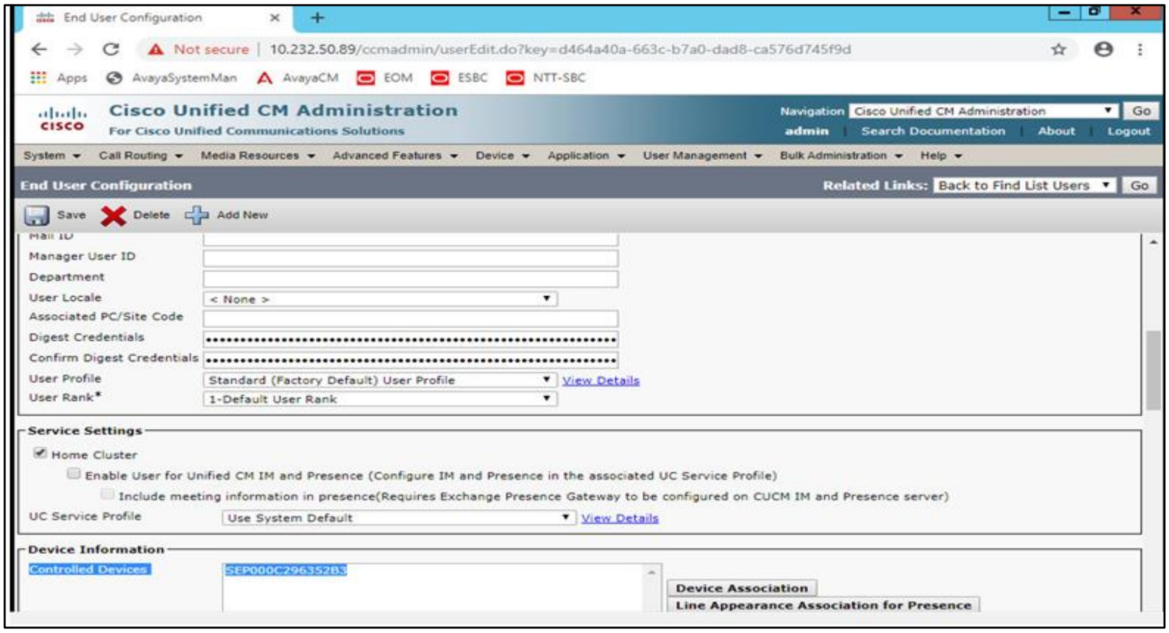
Confidential Access Mode < None >

Name. Tarc

4.5. Associating End User to Phone

01) Go to User Management ----- End Users and search for the sip user you created above, once you find it, click on it

02) Scroll down to Device Association and click on the Device Association button
03) Locate and select the sip device you created above
04) Check the checkbox next to this device and click Save Selected/Changes
05) Click Go next to the Back to User related link near the upper right-hand corner
06) Click Save one more time on the End User Configuration screen.



Mail ID
Manager User ID
Department
User Locale < None >
Associated PC/Site Code
Digest Credentials ••
Confirm Digest Credentials ••
User Profile Standard (Factory Default) User Profile View Details
User Rank* 1-Default User Rank

Service Settings

☑ Home Cluster
☐ Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)
☐ Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)
UC Service Profile Use System Default View Details

Device Information

Controlled Devices SEP000C296352B3

With these steps, the CUCM configuration is complete.

5. Configuring the SBC

This chapter provides step-by-step guidance on how to configure Oracle SBC for Cisco Call Manager (Cisco CUCM) and Twilio Elastic SIP Trunking. **In this SBC config, Twilio Elastic SIP trunk side is secure (TLS/SRTP) and Cisco Side is unsecure (UDP or TCP/RTP).**

5.1. Validated Oracle SBC version

Oracle conducted tests with Oracle SBC 8.4 software – this software with the configuration listed below can run on any of the following products:

- AP 1100
- AP 3900
- AP 4600
- AP 6300
- AP 6350
- VME

6. New SBC configuration

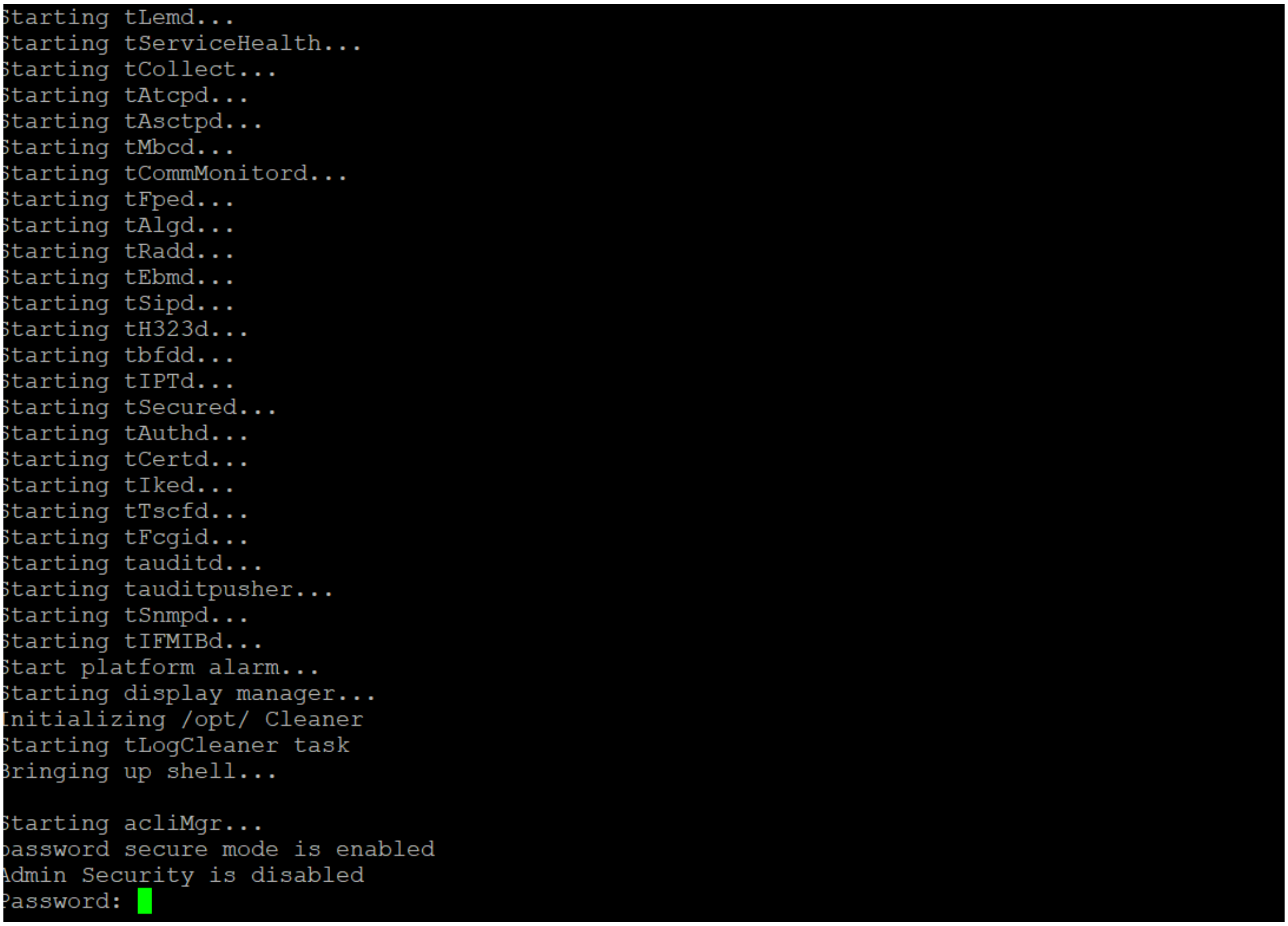
If the customer is looking to setup a new SBC from scratch, please follow the section below.

6.1. Establishing a serial connection to the SBC

Connect one end of a straight-through Ethernet cable to the front console port (which is active by default) on the SBC and the other end to console adapter that ships with the SBC, connect the console adapter (a DB-9 adapter) to the DB-9 port on a workstation, running a terminal emulator application such as Putty. Start the terminal emulation application using the following settings:

- Baud Rate=115200
- Data Bits=8
- Parity=None
- Stop Bits=1
- Flow Control=None

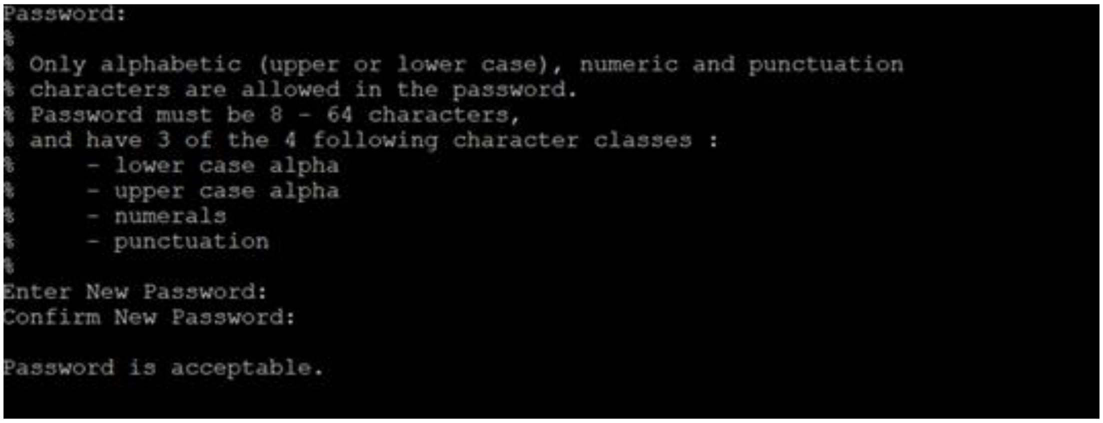
Power on the SBC and confirm that you see the following output from the boot-up sequence



```
Starting tLemd...
Starting tServiceHealth...
Starting tCollect...
Starting tAtcpd...
Starting tAsctpd...
Starting tMbcd...
Starting tCommMonitord...
Starting tFped...
Starting tAlgd...
Starting tRadd...
Starting tEbmd...
Starting tSipd...
Starting tH323d...
Starting tbfdd...
Starting tIPTd...
Starting tSecured...
Starting tAuthd...
Starting tCertd...
Starting tIked...
Starting tTscfd...
Starting tFcgid...
Starting tauditd...
Starting tauditpusher...
Starting tSnmpd...
Starting tIFMIBd...
Start platform alarm...
Starting display manager...
Initializing /opt/ Cleaner
Starting tLogCleaner task
Bringing up shell...

Starting acliMgr...
password secure mode is enabled
Admin Security is disabled
Password: 
```

Enter the default password to log in to the SBC. Note that the default SBC password is “acme” and the default super user password is “packet”.

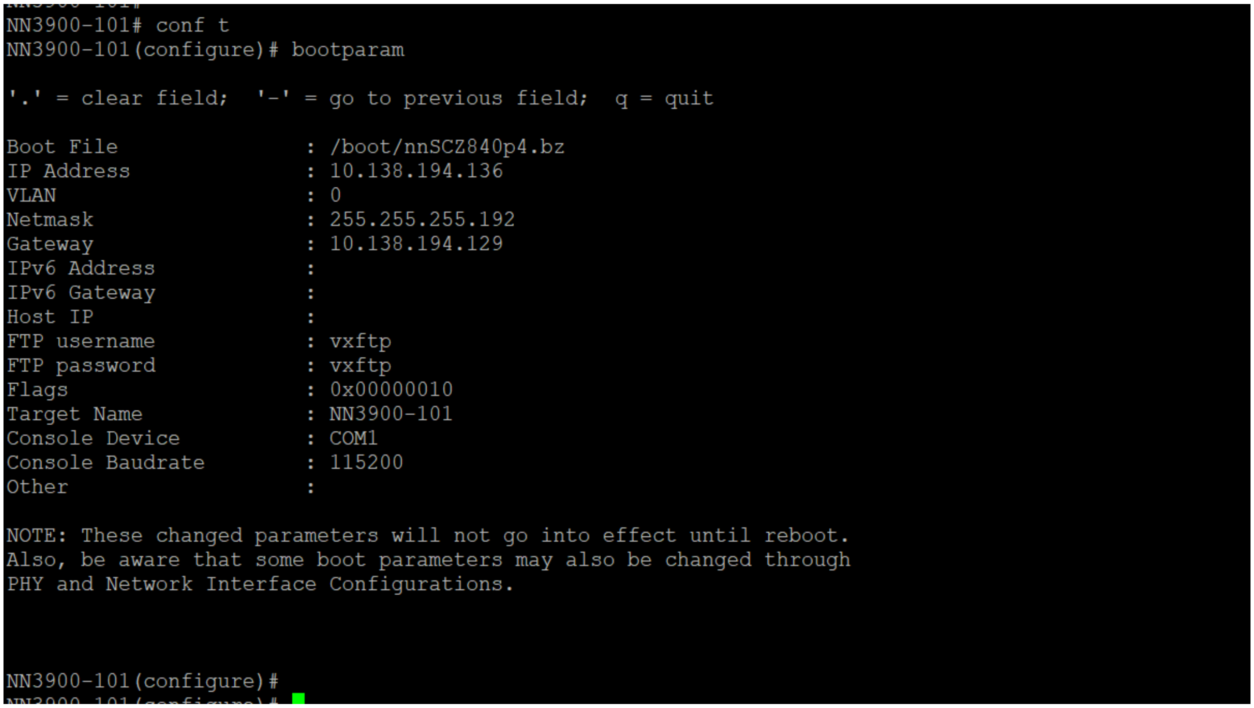
Both passwords have to be changed according to the rules shown below.



```
Password:
%
% Only alphabetic (upper or lower case), numeric and punctuation
% characters are allowed in the password.
% Password must be 8 - 64 characters,
% and have 3 of the 4 following character classes :
%      - lower case alpha
%      - upper case alpha
%      - numerals
%      - punctuation
%
Enter New Password:
Confirm New Password:

Password is acceptable.
```

Now set the management IP of the SBC by setting the IP address in bootparam.

To access bootparam. Go to Configure terminal->bootparam.

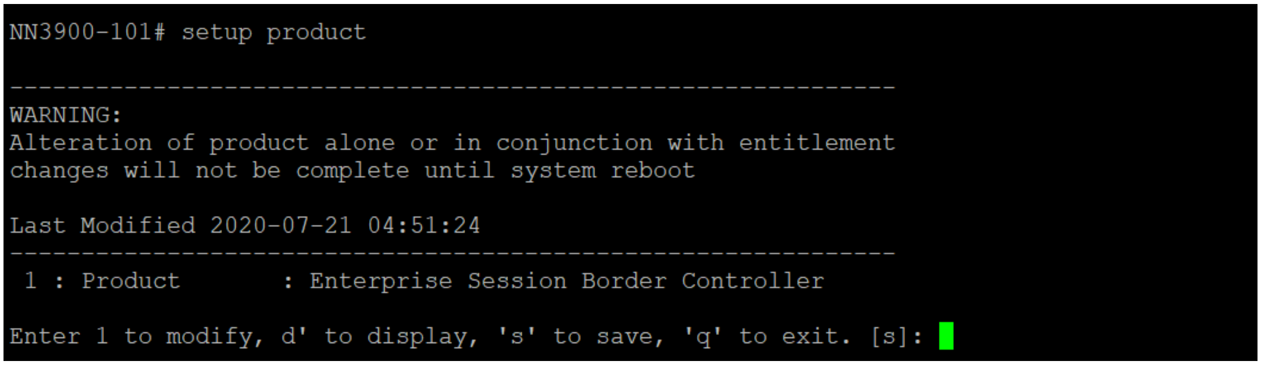


```
NN3900-101#
NN3900-101# conf t
NN3900-101(configure)# bootparam

'.' = clear field;  '-' = go to previous field;  q = quit

Boot File                : /boot/nnSCZ840p4.bz
IP Address               : 10.138.194.136
VLAN                     : 0
Netmask                  : 255.255.255.192
Gateway                  : 10.138.194.129
IPv6 Address             :
IPv6 Gateway             :
Host IP                  :
FTP username             : vxftp
FTP password             : vxftp
Flags                    : 0x00000010
Target Name              : NN3900-101
Console Device           : COM1
Console Baudrate         : 115200
Other                    :

NOTE: These changed parameters will not go into effect until reboot.
Also, be aware that some boot parameters may also be changed through
PHY and Network Interface Configurations.



NN3900-101(configure)#
NN3900-101(configure)#
```

Note: There is no management IP configured by default.

Setup product type to Enterprise Session Border Controller as shown below.

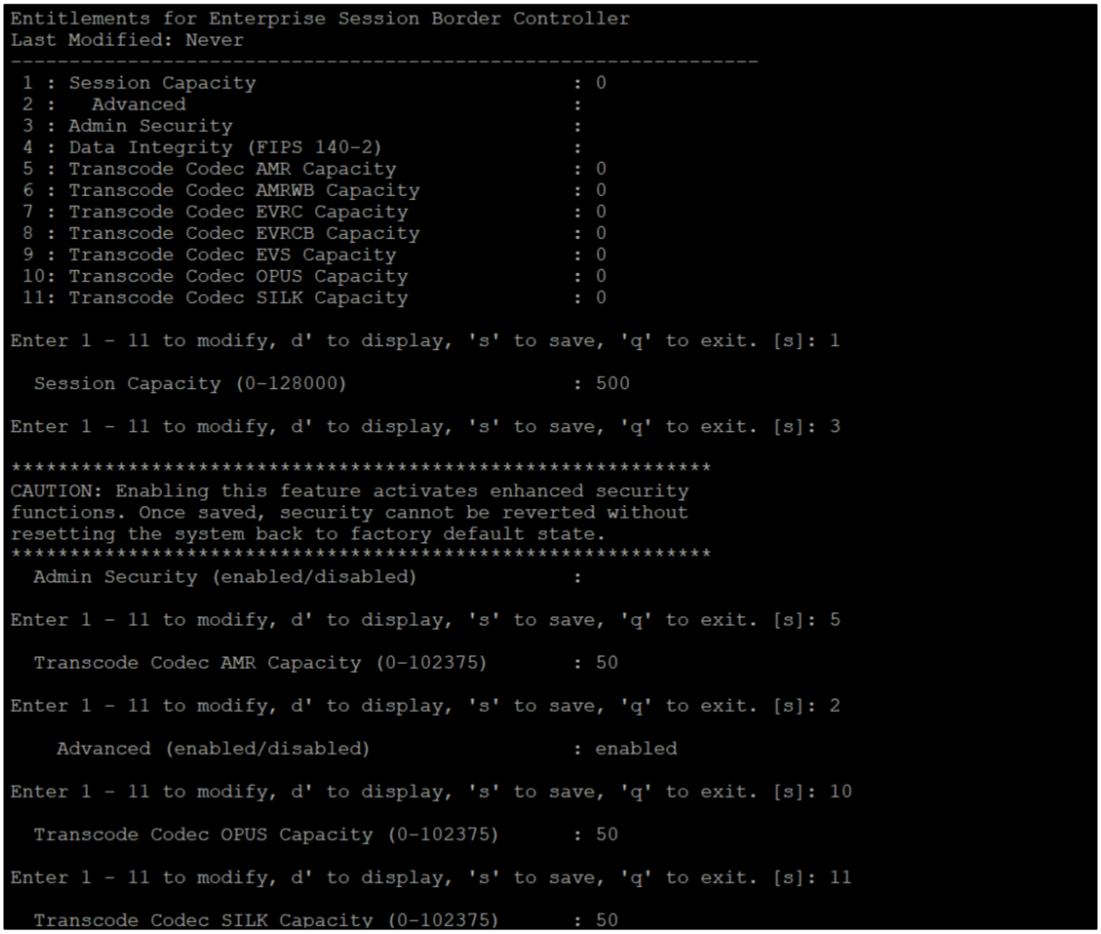
To configure product type, type in setup product in the terminal



```
NN3900-101# setup product

-------------------------------------------------------------
WARNING:
Alteration of product alone or in conjunction with entitlement
changes will not be complete until system reboot

Last Modified 2020-07-21 04:51:24
-------------------------------------------------------------
 1 : Product       : Enterprise Session Border Controller

Enter 1 to modify, d' to display, 's' to save, 'q' to exit. [s]:
```

Enable the features for the ESBC using the setup entitlements command as shown

Save the changes and reboot the SBC.

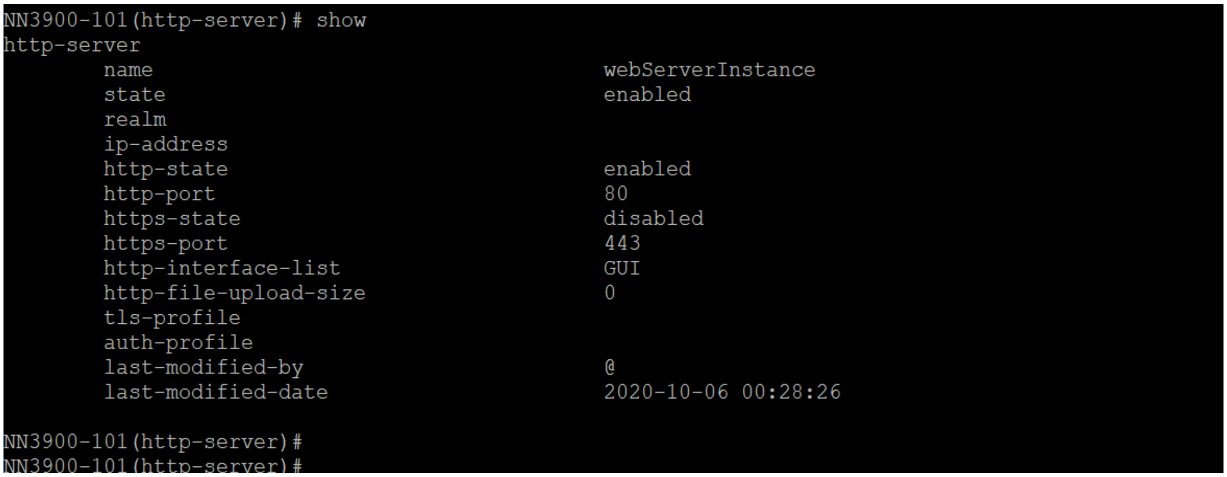


```
Entitlements for Enterprise Session Border Controller
Last Modified: Never
----------------------------------------------------------------
 1 : Session Capacity                           : 0
 2 :   Advanced                                 :
 3 : Admin Security                             :
 4 : Data Integrity (FIPS 140-2)                :
 5 : Transcode Codec AMR Capacity               : 0
 6 : Transcode Codec AMRWB Capacity             : 0
 7 : Transcode Codec EVRC Capacity              : 0
 8 : Transcode Codec EVRCB Capacity             : 0
 9 : Transcode Codec EVS Capacity               : 0
 10: Transcode Codec OPUS Capacity              : 0
 11: Transcode Codec SILK Capacity              : 0

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 1

  Session Capacity (0-128000)                   : 500

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 3

************************************************************
CAUTION: Enabling this feature activates enhanced security
functions. Once saved, security cannot be reverted without
resetting the system back to factory default state.
************************************************************
  Admin Security (enabled/disabled)             :

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 5

  Transcode Codec AMR Capacity (0-102375)       : 50

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 2

    Advanced (enabled/disabled)                 : enabled

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 10

  Transcode Codec OPUS Capacity (0-102375)      : 50

Enter 1 - 11 to modify, d' to display, 's' to save, 'q' to exit. [s]: 11

  Transcode Codec SILK Capacity (0-102375)      : 50
```

The SBC comes up after reboot and is now ready for configuration.

Go to configure terminal->system->http-server-config.

Enable the http-server-config to access the SBC using Web GUI. Save and activate the config.

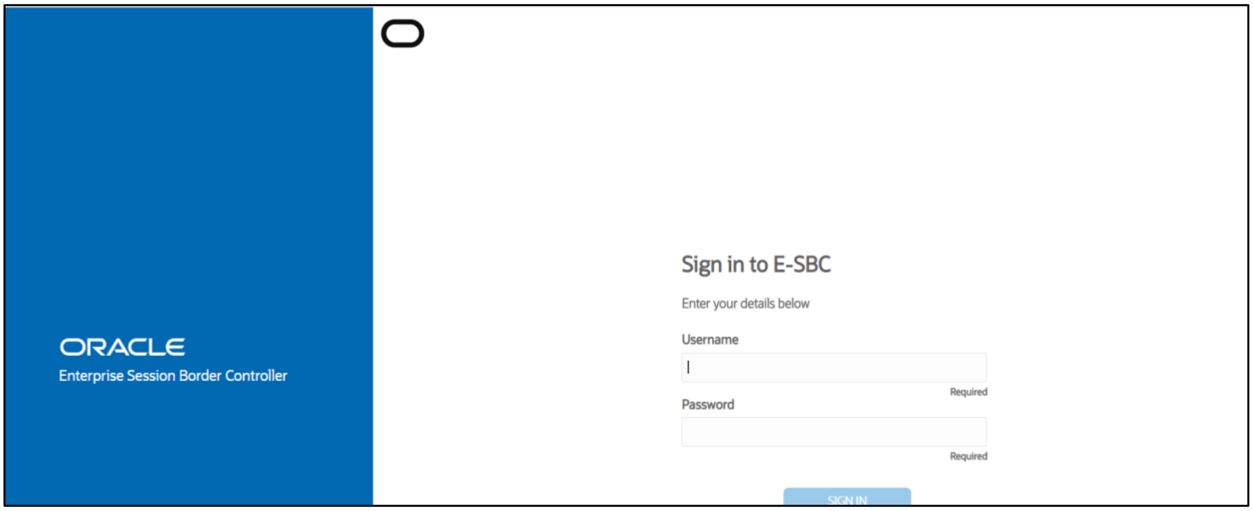


```
NN3900-101(http-server)# show
http-server
        name                                    webServerInstance
        state                                   enabled
        realm
        ip-address
        http-state                              enabled
        http-port                               80
        https-state                             disabled
        https-port                              443
        http-interface-list                     GUI
        http-file-upload-size                   0
        tls-profile
        auth-profile
        last-modified-by                        @
        last-modified-date                      2020-10-06 00:28:26

NN3900-101(http-server)#
NN3900-101(http-server)#
```

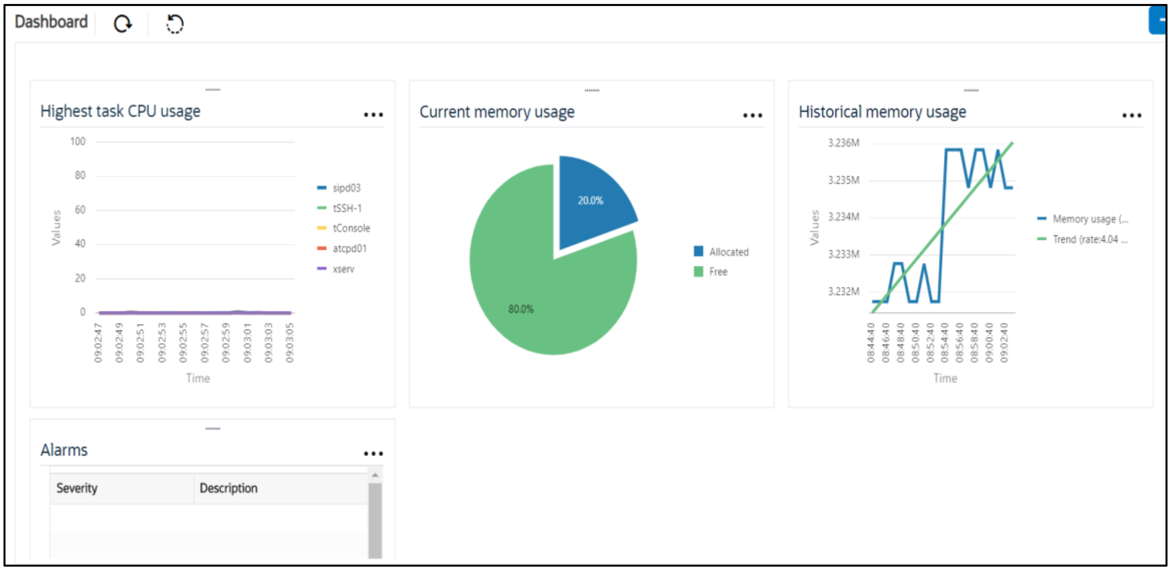
6.2. Configure SBC using Web GUI

In this app note, we configure SBC using the WebGUI.

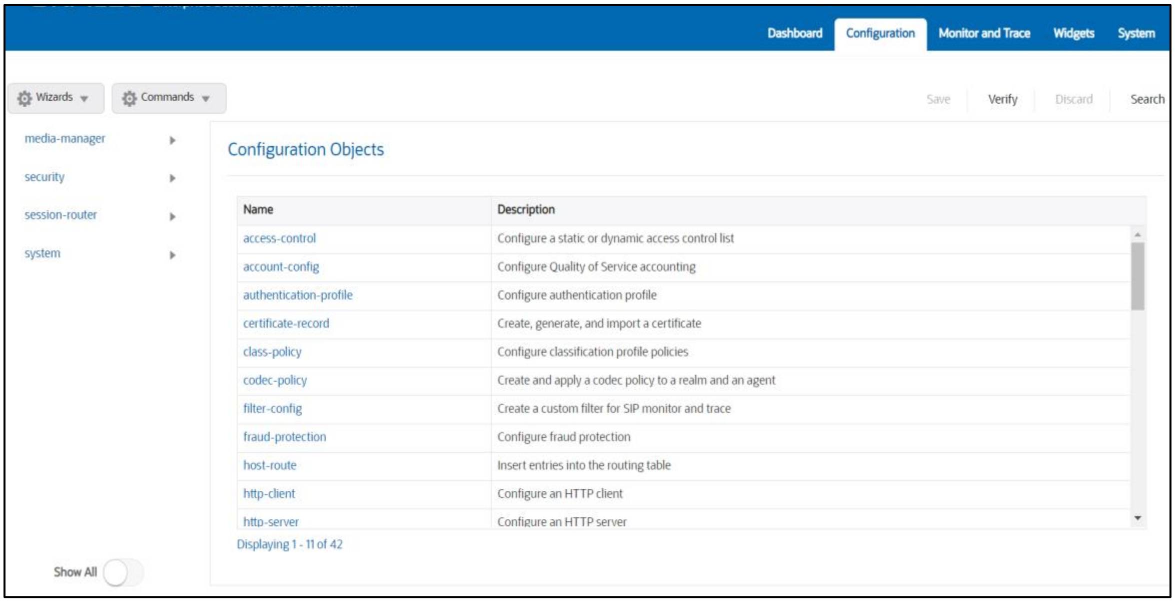
The Web GUI can be accessed through the url http://<SBC_MGMT_IP>.



The username and password is the same as that of CLI.



Go to Configuration as shown below, to configure the SBC



Configuration Objects

Name	Description
access-control	Configure a static or dynamic access control list
account-config	Configure Quality of Service accounting
authentication-profile	Configure authentication profile
certificate-record	Create, generate, and import a certificate
class-policy	Configure classification profile policies
codec-policy	Create and apply a codec policy to a realm and an agent
filter-config	Create a custom filter for SIP monitor and trace
fraud-protection	Configure fraud protection
host-route	Insert entries into the routing table
http-client	Configure an HTTP client
http-server	Configure an HTTP server

Displaying 1 - 11 of 42

Kindly refer to the GUI User Guide given below for more information.

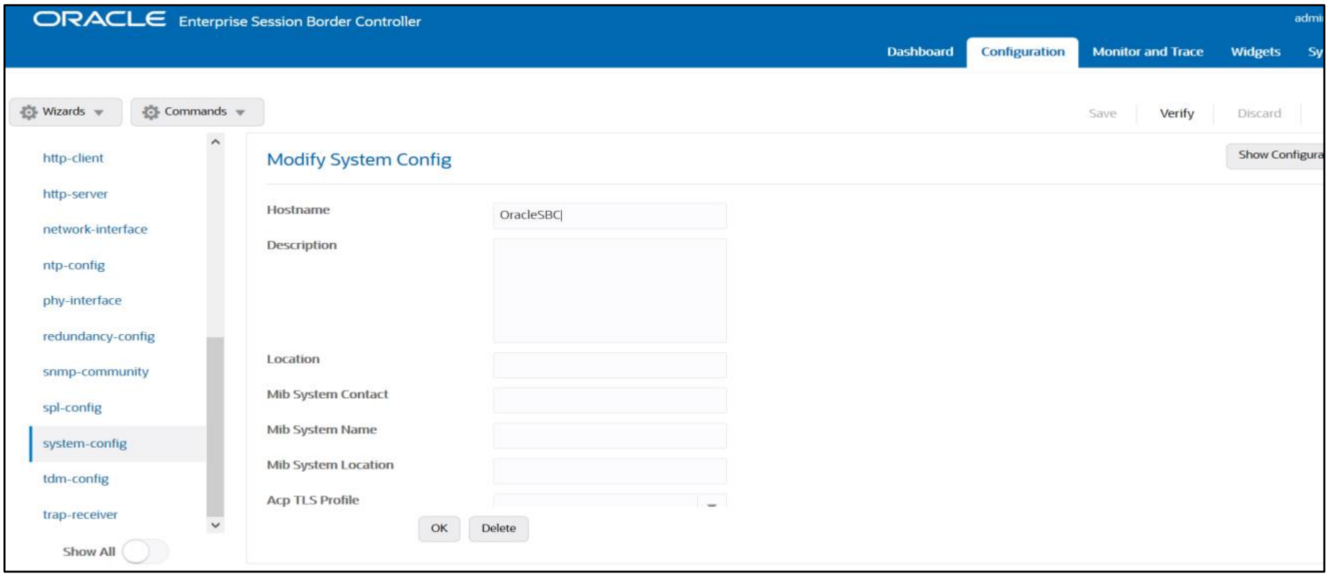
https://docs.oracle.com/en/industries/communications/enterprise-session-border-controller/8.4.0/webgui/esbc_scz840_webgui.pdf

The expert mode is used for configuration.

Tip: To make this configuration simpler, one can directly search the element to be configured, from the Objects tab available.

6.3. Configure system-config

Go to system->system-config



Modify System Config

Hostname: OracleSBC|

Description

Location

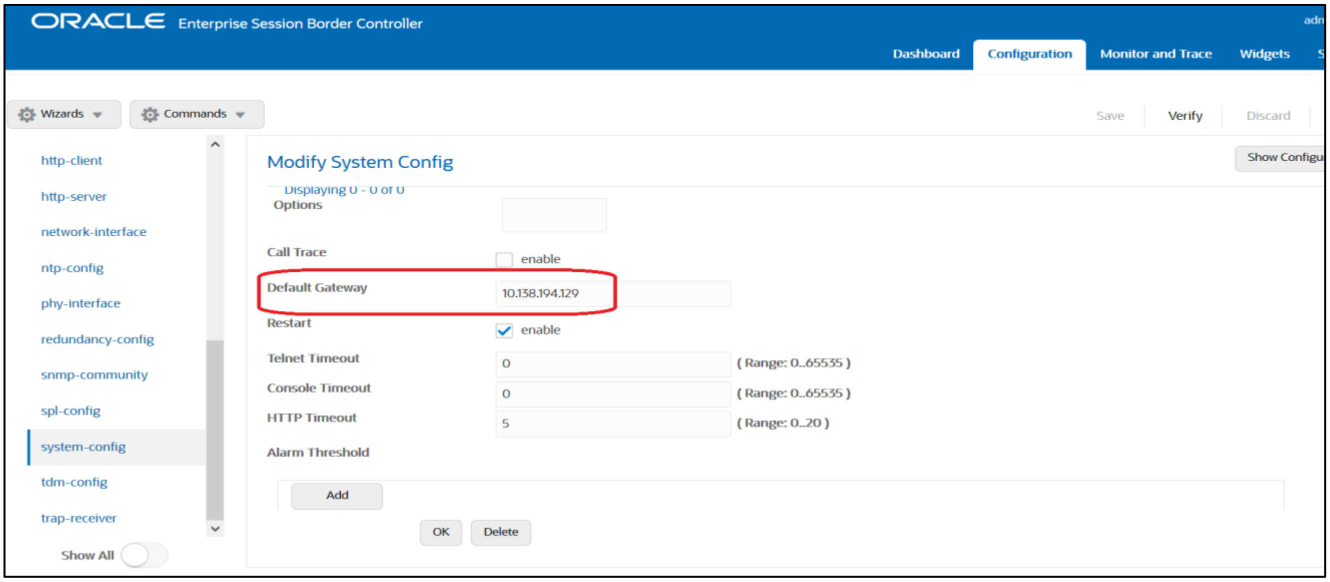
Mib System Contact

Mib System Name

Mib System Location

Acp TLS Profile

Please enter the default gateway value in the system config page.



Modify System Config

Options

Call Trace: □ enable

Default Gateway: 10.138.194.129

Restart: ☑ enable

Telnet Timeout: 0 (Range: 0..65535)

Console Timeout: 0 (Range: 0..65535)

HTTP Timeout: 5 (Range: 0..20)

Alarm Threshold

For VME, transcoding cores are required. Please refer the documentation here for more information

https://docs.oracle.com/en/industries/communications/enterprise-session-border-controller/8.4.0/releasenotes/esbc_scz840_releasenotes.pdf

The above step is needed only if any transcoding is used in the configuration.
If there is no transcoding involved, then the above step is not needed.

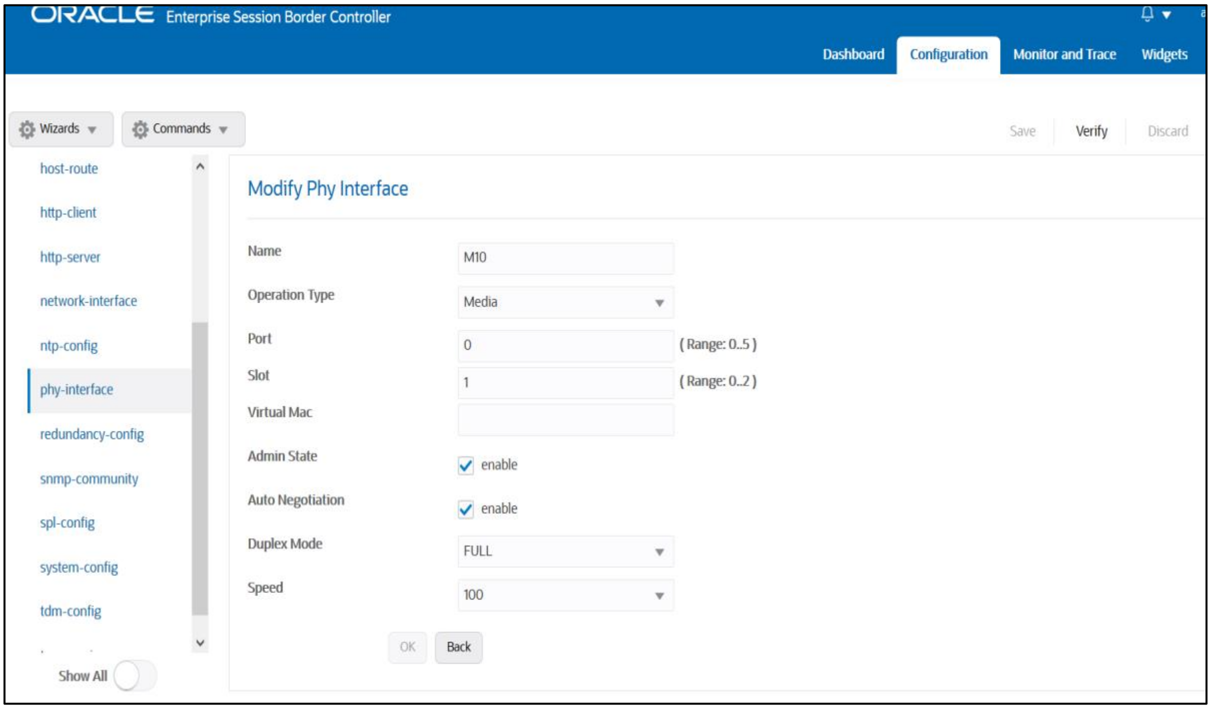
6.4. Configure Physical Interface values

To configure physical Interface values, go to System->phy-interface.

Please configure M10 for Twilio side and M11 for Cisco side.

Parameter Name	Twilio Elastic Sip Trunk side (M10)	Cisco side (M11)
Slot	1	1
Port	0	1
Operation Mode	Media	Media

Please configure M10 interface as below.



Modify Phy Interface

Name: M10

Operation Type: Media

Port: 0 (Range: 0..5)

Slot: 1 (Range: 0..2)

Virtual Mac:

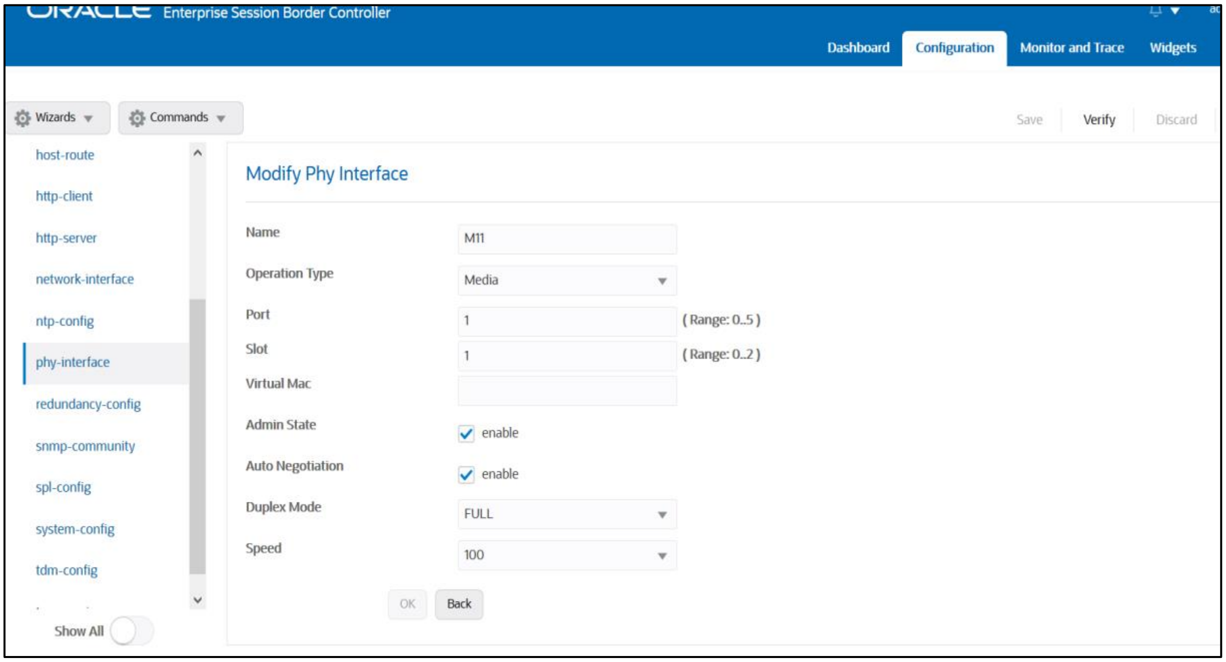
Admin State: ☑ enable

Auto Negotiation: ☑ enable

Duplex Mode: FULL

Speed: 100

Please configure M11 interface as below



Modify Phy Interface

Name: M11
Operation Type: Media
Port: 1 (Range: 0..5)
Slot: 1 (Range: 0..2)
Virtual Mac:
Admin State: ☑ enable
Auto Negotiation: ☑ enable
Duplex Mode: FULL
Speed: 100

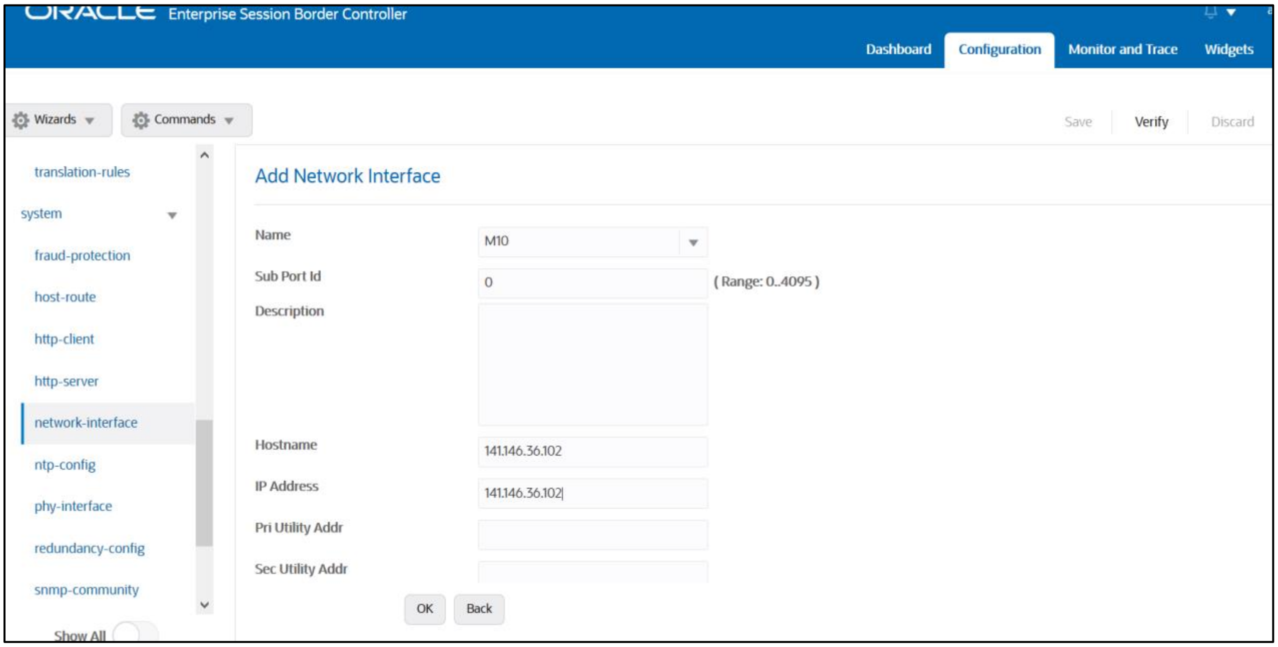
6.5. Configure Network Interface values

To configure network-interface, go to system->Network-Interface. Configure interface

The table below lists the parameters, to be configured for both the interfaces.

Parameter Name	Twilio side Network interface	Cisco side Network interface
Name	M10	M11
Host Name		
IP address	141.146.36.102	10.232.50.78
Netmask	255.255.255.192	255.255.255.0
Gateway	141.146.36.65	10.232.50.1

Please configure network interface M10 as below



Add Network Interface

Name: M10

Sub Port Id: 0 (Range: 0..4095)

Description:

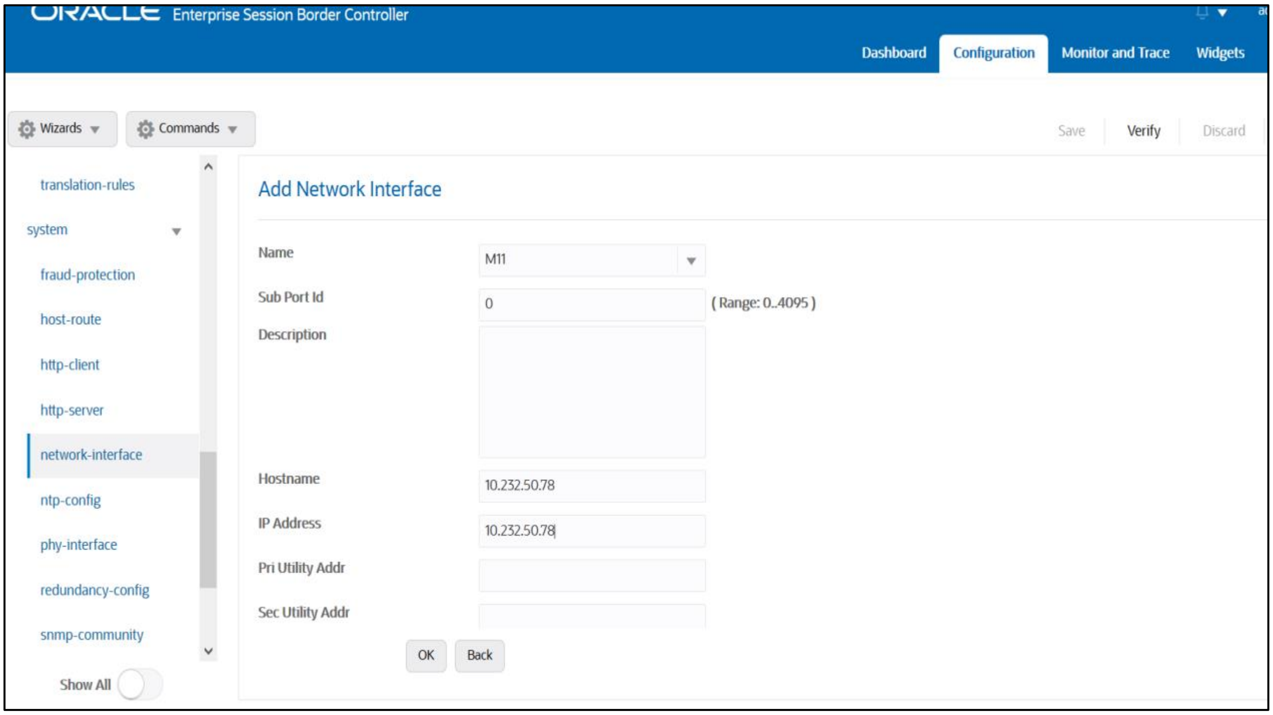
Hostname: 141.146.36.102

IP Address: 141.146.36.102

Pri Utility Addr:

Sec Utility Addr:

Similarly, configure network interface M11 as below



Add Network Interface

Name: M11

Sub Port Id: 0 (Range: 0..4095)

Description:

Hostname: 10.232.50.78

IP Address: 10.232.50.78

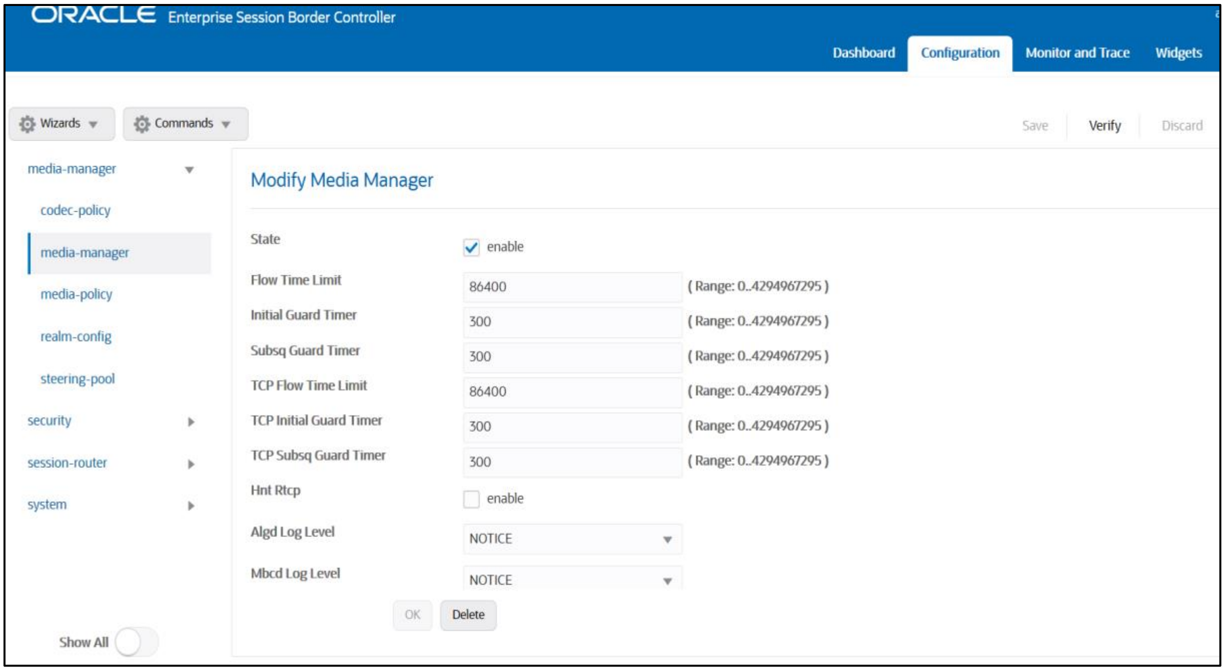
Pri Utility Addr:

Sec Utility Addr:

6.6. Enable media manager

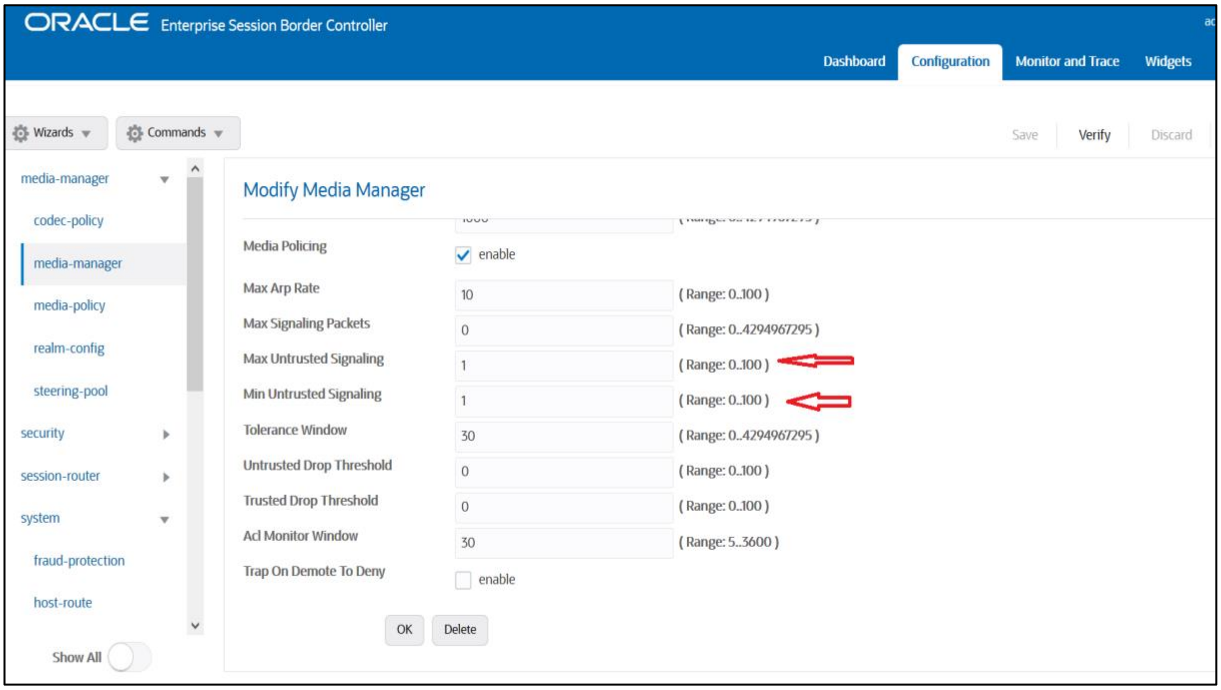
Media-manager handles the media stack required for SIP sessions on the SBC. Enable the media manager option as below.

In addition to the above config, please set the max and min untrusted signaling values to 1.
Go to Media-Manager->Media-Manager



Modify Media Manager

State	☑ enable	
Flow Time Limit	86400	(Range: 0..4294967295)
Initial Guard Timer	300	(Range: 0..4294967295)
Subsq Guard Timer	300	(Range: 0..4294967295)
TCP Flow Time Limit	86400	(Range: 0..4294967295)
TCP Initial Guard Timer	300	(Range: 0..4294967295)
TCP Subsq Guard Timer	300	(Range: 0..4294967295)
Hnt Rtcp	□ enable	
Algd Log Level	NOTICE	
Mbcd Log Level	NOTICE	



Modify Media Manager

Media Policing	☑ enable	
Max Arp Rate	10	(Range: 0..100)
Max Signaling Packets	0	(Range: 0..4294967295)
Max Untrusted Signaling	1	(Range: 0..100)
Min Untrusted Signaling	1	(Range: 0..100)
Tolerance Window	30	(Range: 0..4294967295)
Untrusted Drop Threshold	0	(Range: 0..100)
Trusted Drop Threshold	0	(Range: 0..100)
Acl Monitor Window	30	(Range: 5..3600)
Trap On Demote To Deny	□ enable	

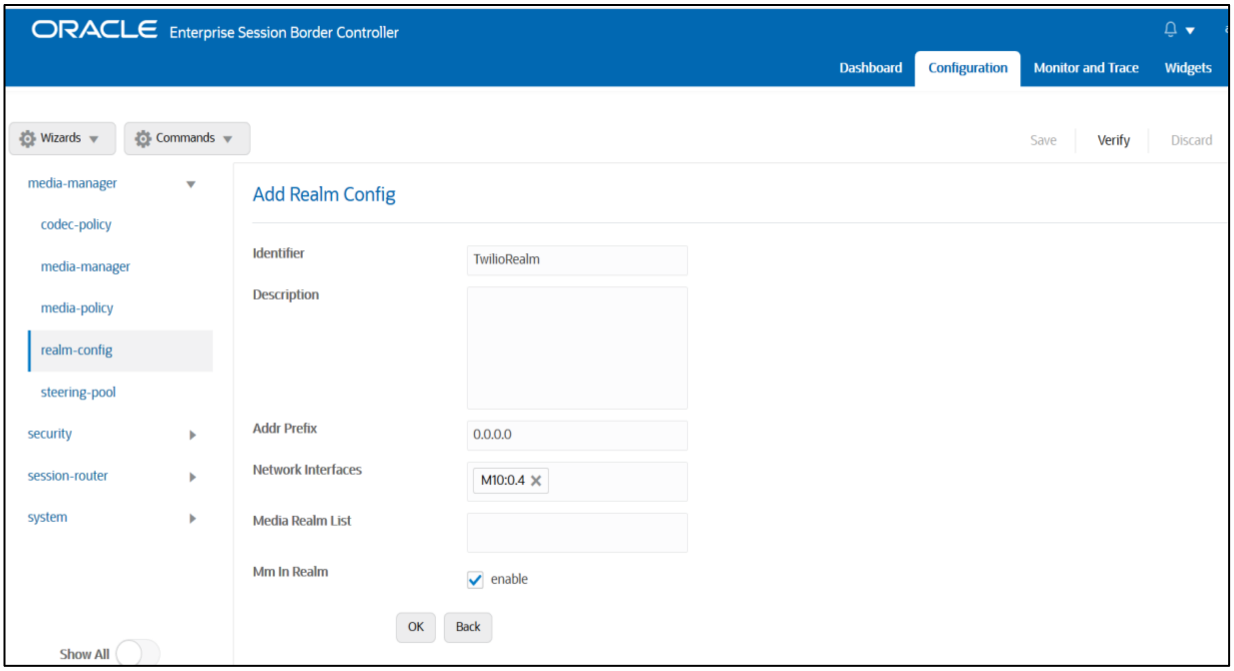
6.7. Configure Realms

Navigate to realm-config under media-manager and configure a realm as shown below
The name of the Realm can be any relevant name according to the user convenience.

Use the following table as a configuration example for the two realms used in this configuration:

Config Parameter	Twilio Side	Cisco Side
Identifier	TwilioRealm	CUCMRealm
Network Interface	M10	M11
Mm in realm	☑	☑
FQDN		
Media Sec policy	sdespolicy	RTP
Access Control Trust Level	High	High

In the below case, Realm name is given as TwilioRealm for Twilio Elastic SIP Trunking Side
Please set the Access Control Trust Level as high for this realm



Add Realm Config

Identifier: TwilioRealm

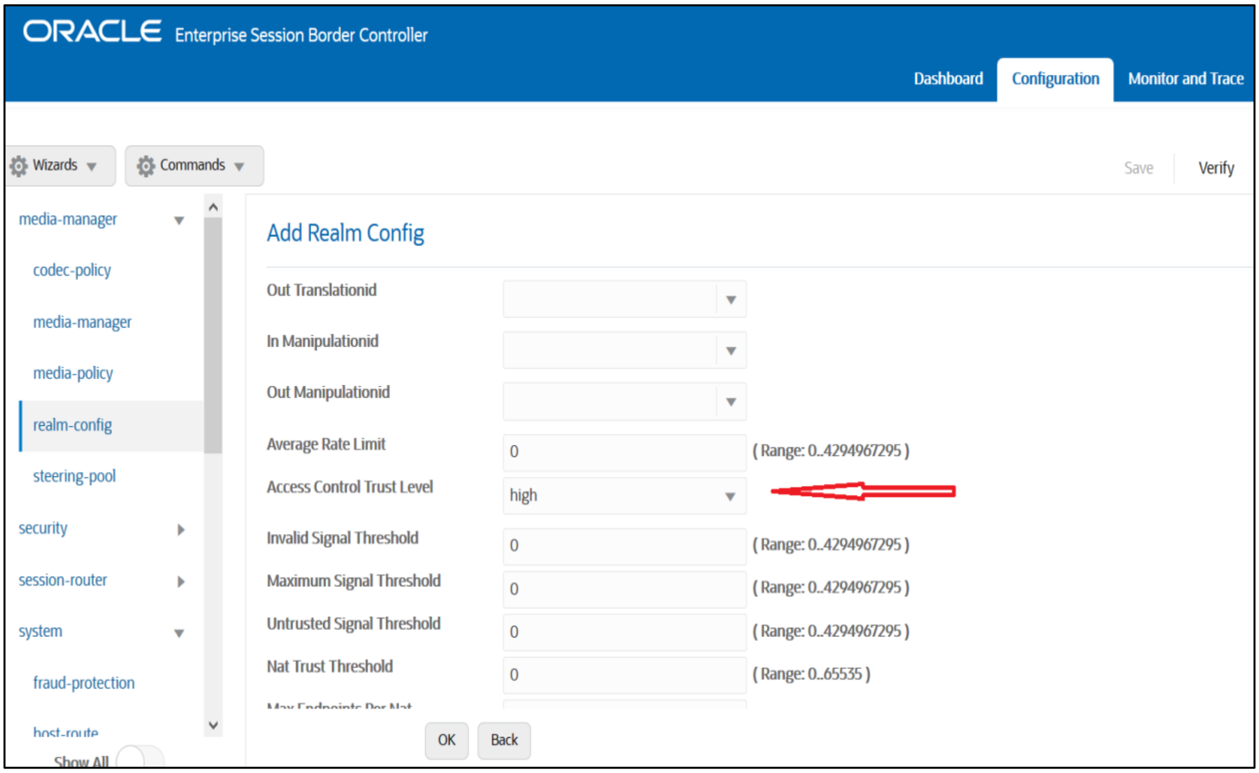
Description:

Addr Prefix: 0.0.0.0

Network Interfaces: M10:0.4 ×

Media Realm List:

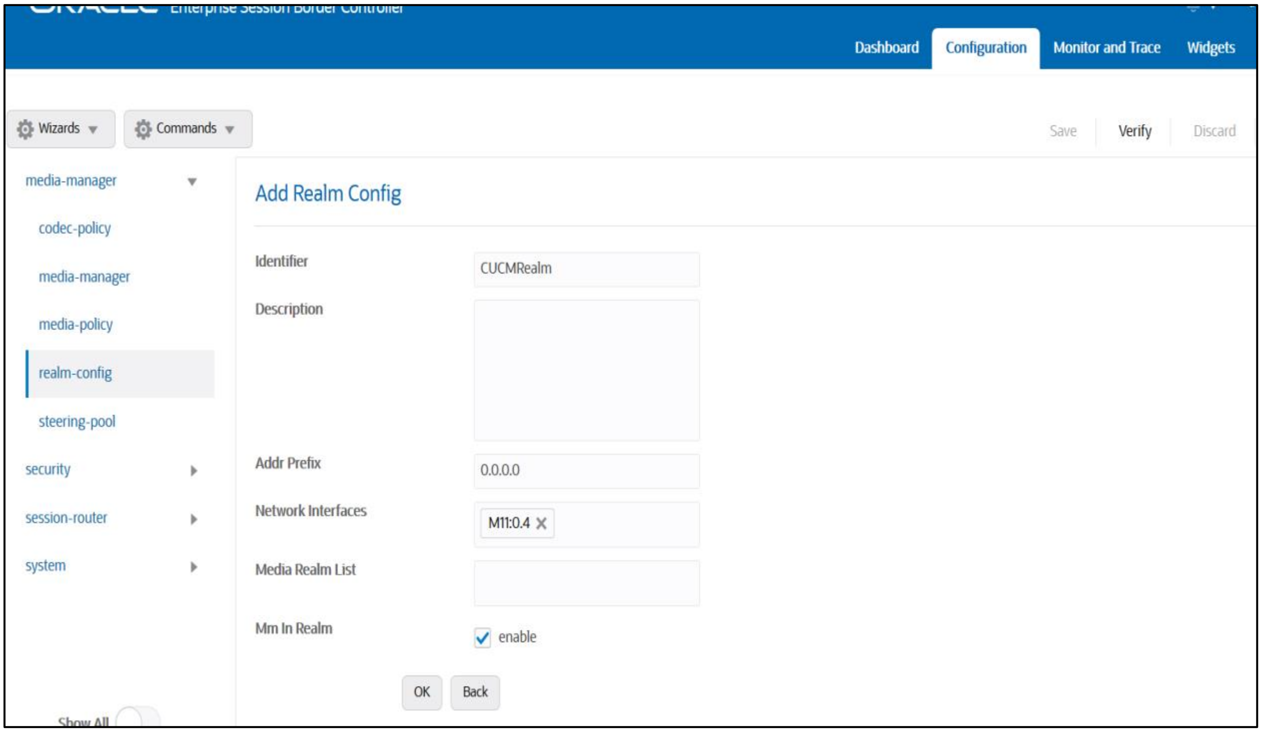
Mm In Realm: ☑ enable



Add Realm Config

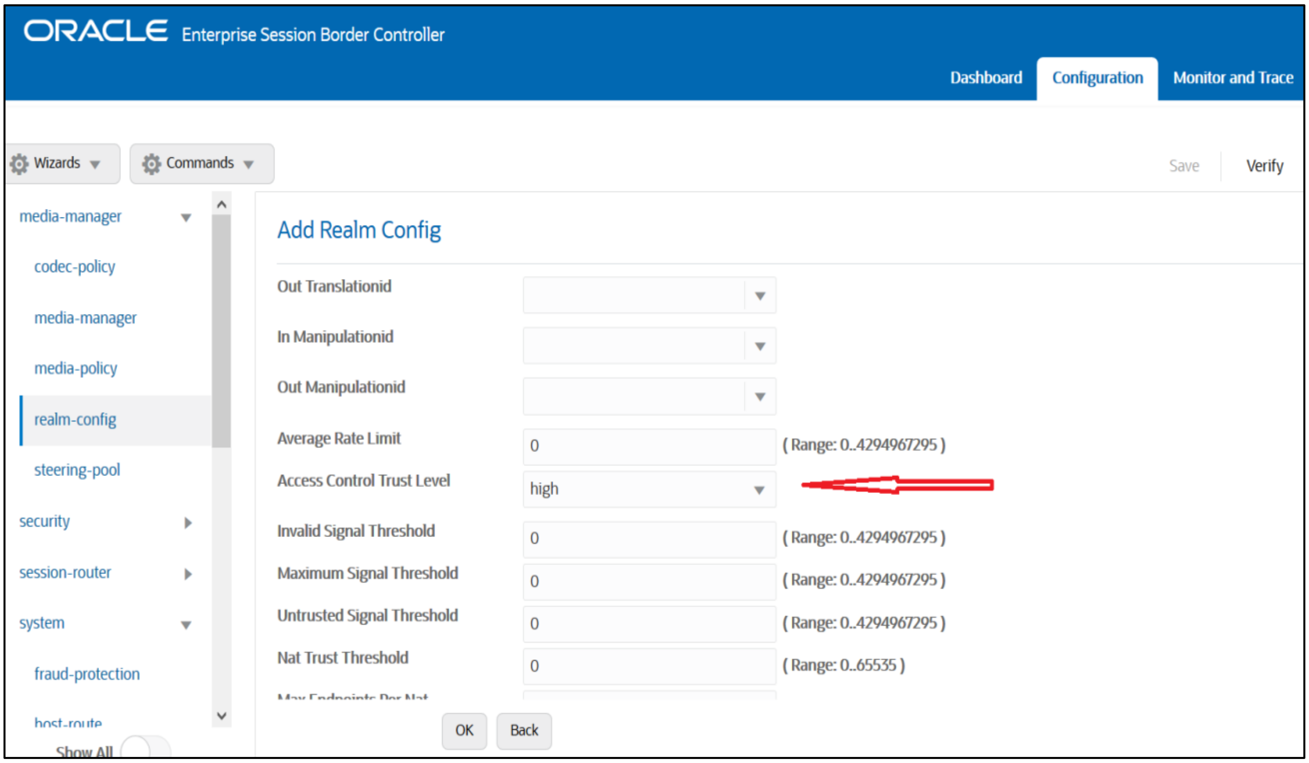
Out Translationid		
In Manipulationid		
Out Manipulationid		
Average Rate Limit	0	(Range: 0..4294967295)
Access Control Trust Level	high	
Invalid Signal Threshold	0	(Range: 0..4294967295)
Maximum Signal Threshold	0	(Range: 0..4294967295)
Untrusted Signal Threshold	0	(Range: 0..4294967295)
Nat Trust Threshold	0	(Range: 0..65535)
Max Endpoints Per Nat		

Similarly, Realm name is given as CUCMRealm for Cisco side.
Please set the Access Control Trust Level as high for this realm too.



Add Realm Config

Identifier	CUCMRealm
Description	
Addr Prefix	0.0.0.0
Network Interfaces	M11:0.4 ✕
Media Realm List	
Mm In Realm	☑ enable



Add Realm Config

Out Translationid		
In Manipulationid		
Out Manipulationid		
Average Rate Limit	0	(Range: 0..4294967295)
Access Control Trust Level	high	
Invalid Signal Threshold	0	(Range: 0..4294967295)
Maximum Signal Threshold	0	(Range: 0..4294967295)
Untrusted Signal Threshold	0	(Range: 0..4294967295)
Nat Trust Threshold	0	(Range: 0..65535)
Max Endpoints Per Nat		

For more information on Access Control Trust Level, please refer to SBC Security guide link given below:

https://docs.oracle.com/en/industries/communications/session-border-controller/8.4.0/security/sbc_scz840_security.pdf

6.8. Configuring a certificate for SBC

This section describes how to configure the SBC for TLS and SRTP communication for Twilio Elastic SIP Trunking.

Twilio Elastic SIP Trunking allows TLS connections from SBC’s for SIP traffic, and SRTP for media traffic. It requires a certificate signed by one of the trusted Certificate Authorities.
The process includes the following steps:

1) Create a certificate-record – “Certificate-record” are configuration elements on Oracle SBC which captures information for a TLS certificate – such as common-name, key-size, key-usage etc.

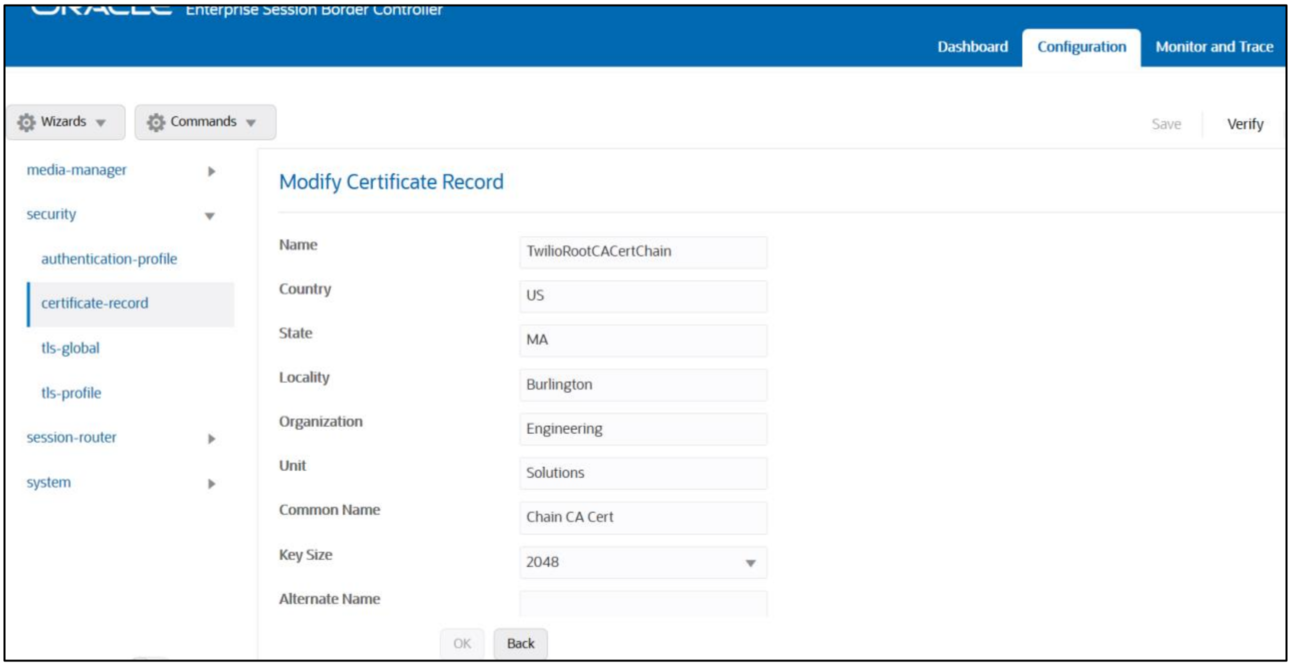
- SBC – 1 certificate-record assigned to SBC
- Root – 1 certificate-record for root cert

2) Deploy the SBC and Root certificates on the SBC

Step 1 – Creating the certificate record

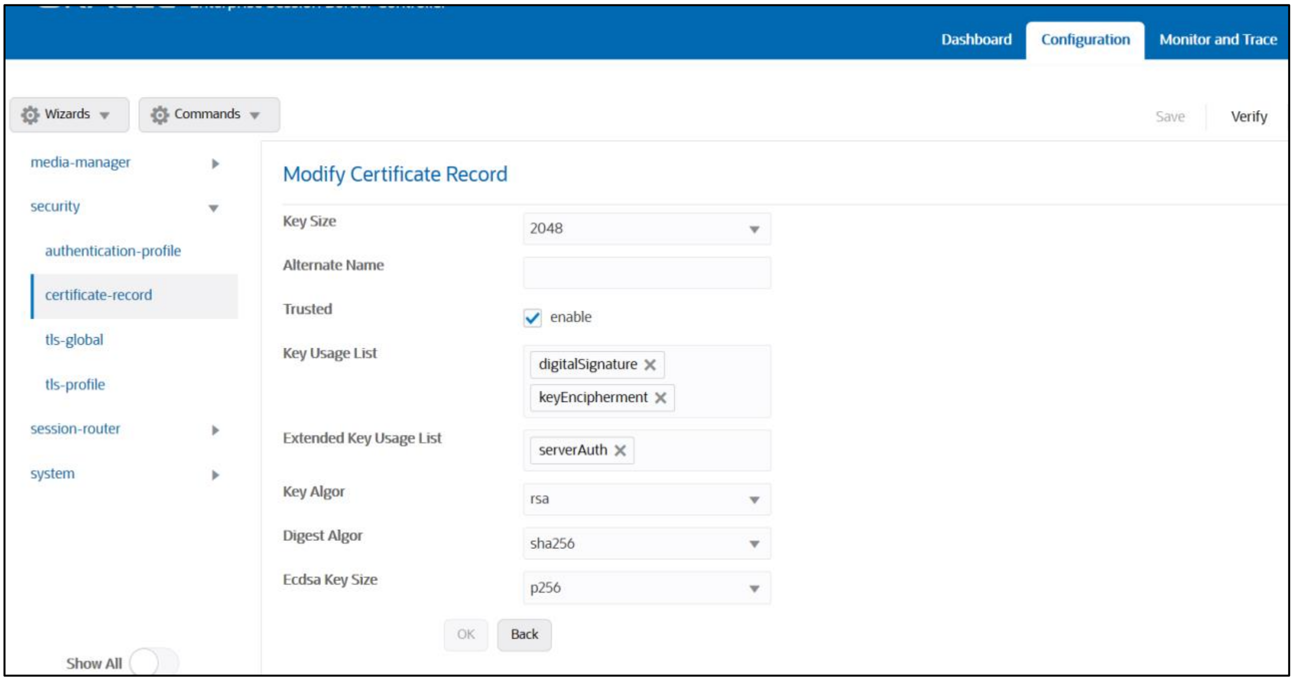
Twilio Elastic SIP Trunking uses certificates from a CA (Certificate Authority) for establishing the TLS connections from SBC's for SIP traffic, and SRTP for media traffic. It is important that you add the following root certificate to establish TLS connection from the link given below:

https://www.twilio.com/docs/sip-trunking#rootCA



Modify Certificate Record

Name	TwilioRootCACertChain
Country	US
State	MA
Locality	Burlington
Organization	Engineering
Unit	Solutions
Common Name	Chain CA Cert
Key Size	2048
Alternate Name	



Modify Certificate Record

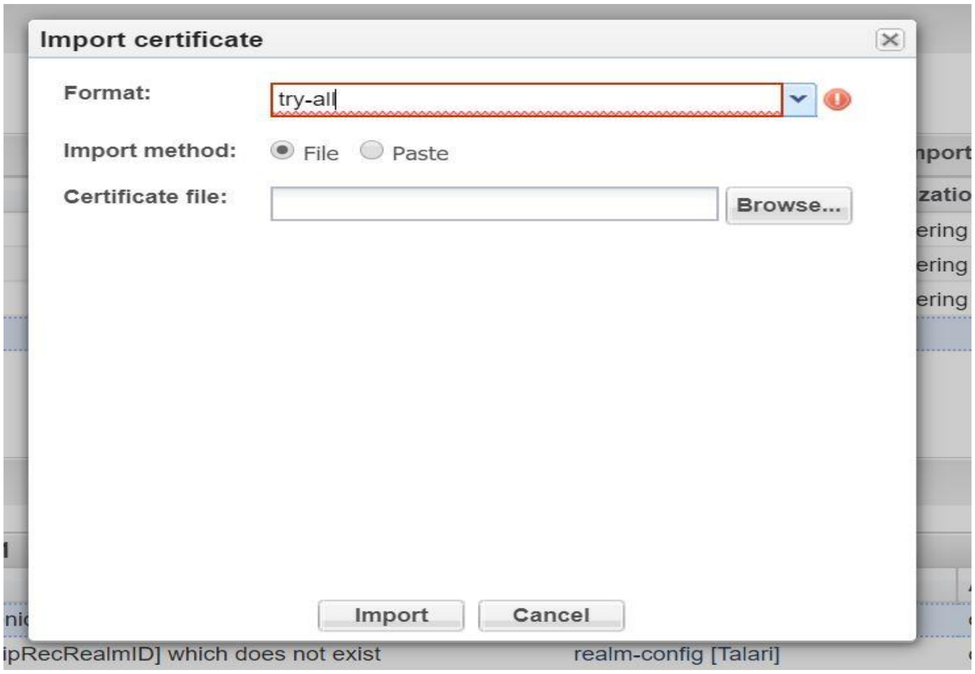
Key Size	2048
Alternate Name	
Trusted	☑ enable
Key Usage List	digitalSignature keyEncipherment
Extended Key Usage List	serverAuth
Key Algor	rsa
Digest Algor	sha256
Ecdsa Key Size	p256

The table below specifies the parameters required for certificate configuration.
Modify the configuration according to the certificates in your environment.

Config Parameter	DigiCert Root CA
Common Name	DigiCert Global Root CA
Key Size	2048
Key-Usage-List	digitalSignature keyEncipherment
Extended Key Usage List	serverAuth
Key algor	rsa
Digest-algor	Sha256

Step 2 – Deploy SBC & root certificates

Once certificate record has been created – import the signed certificate to the SBC.
Please note – all certificates including root certificates are required to be imported to the SBC.
Once done, issue save/activate from the WebGUI



Import certificate

Format: try-all

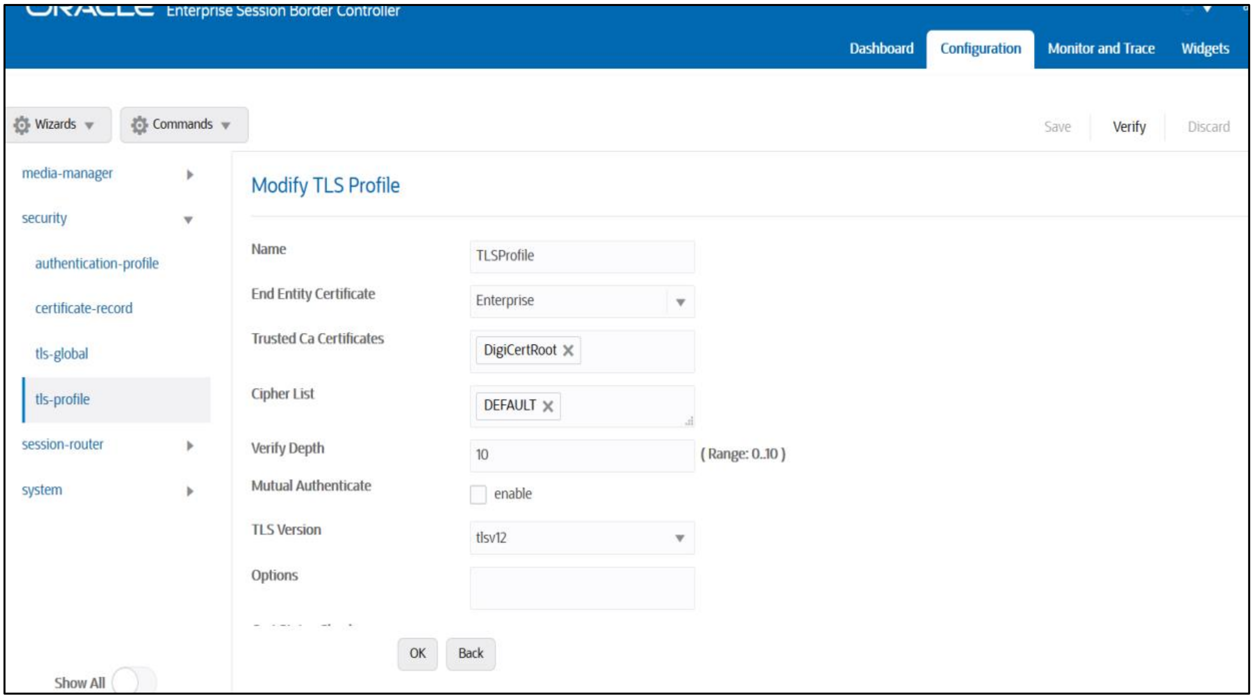
Import method: ● File ○ Paste

Certificate file:

Repeat these steps to import all the root certificates into the SBC:
At this stage all the required certificates have been imported to the SBC for Twilio Elastic SIP Trunk.

6.9. TLS-Profile

A TLS profile configuration on the SBC allows for specific certificates to be assigned.
Go to security-> TLS-profile config element and configure the tls-profile as shown below
The below is the TLS profile configured for the Twilio Elastic SIP Trunk side:



Modify TLS Profile

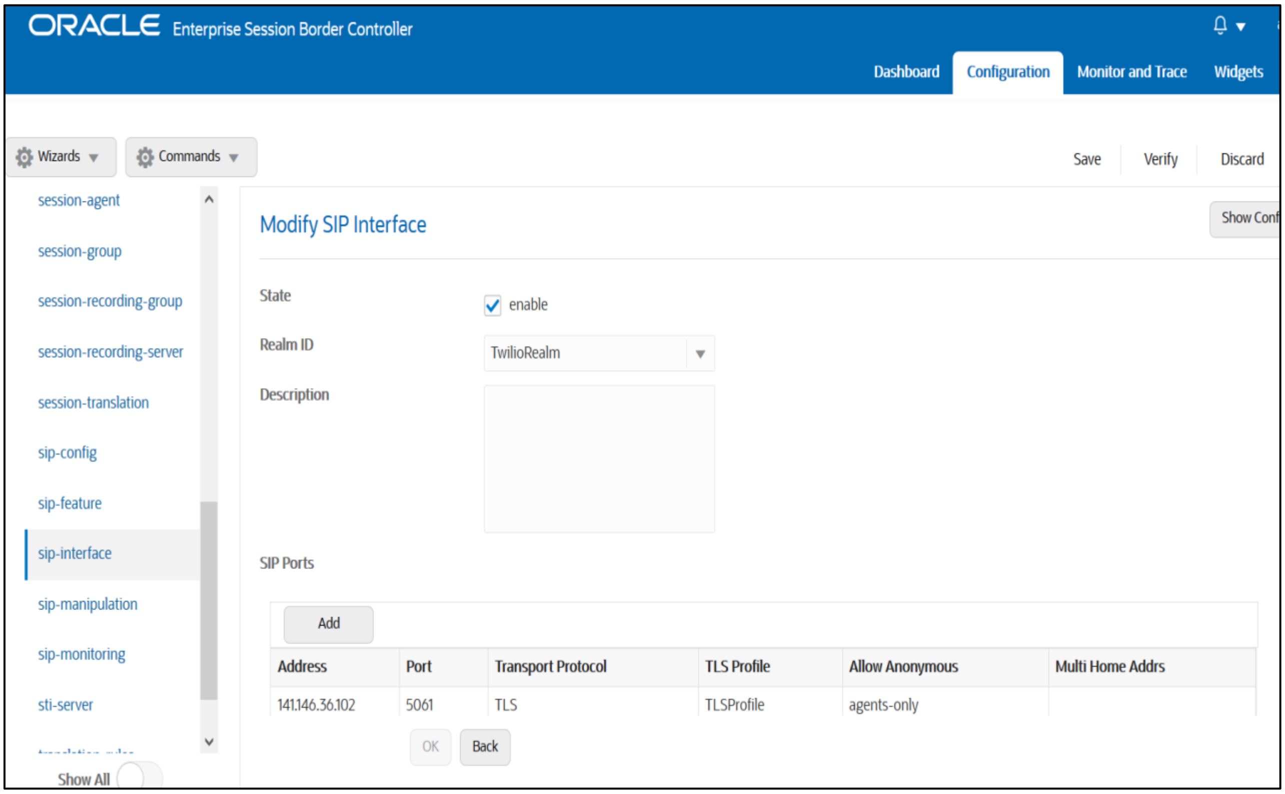
Name	TLSProfile
End Entity Certificate	Enterprise
Trusted Ca Certificates	DigiCertRoot
Cipher List	DEFAULT
Verify Depth	10 (Range: 0..10)
Mutual Authenticate	□ enable
TLS Version	tlsv12
Options	

6.10. Configure SIP Interfaces

Navigate to sip-interface under session-router and configure the sip-interface as shown below.
Please configure the below settings under the sip-interface.

Please Configure sip-interface for the Twilio Elastic SIP Trunk side as below:

- Tls-profile needs to match the name of the tls-profile previously created
- Set allow-anonymous to agents-only to ensure traffic to this sip-interface only comes from the particular Session agents added to the SBC.



Modify SIP Interface

State ☑ enable

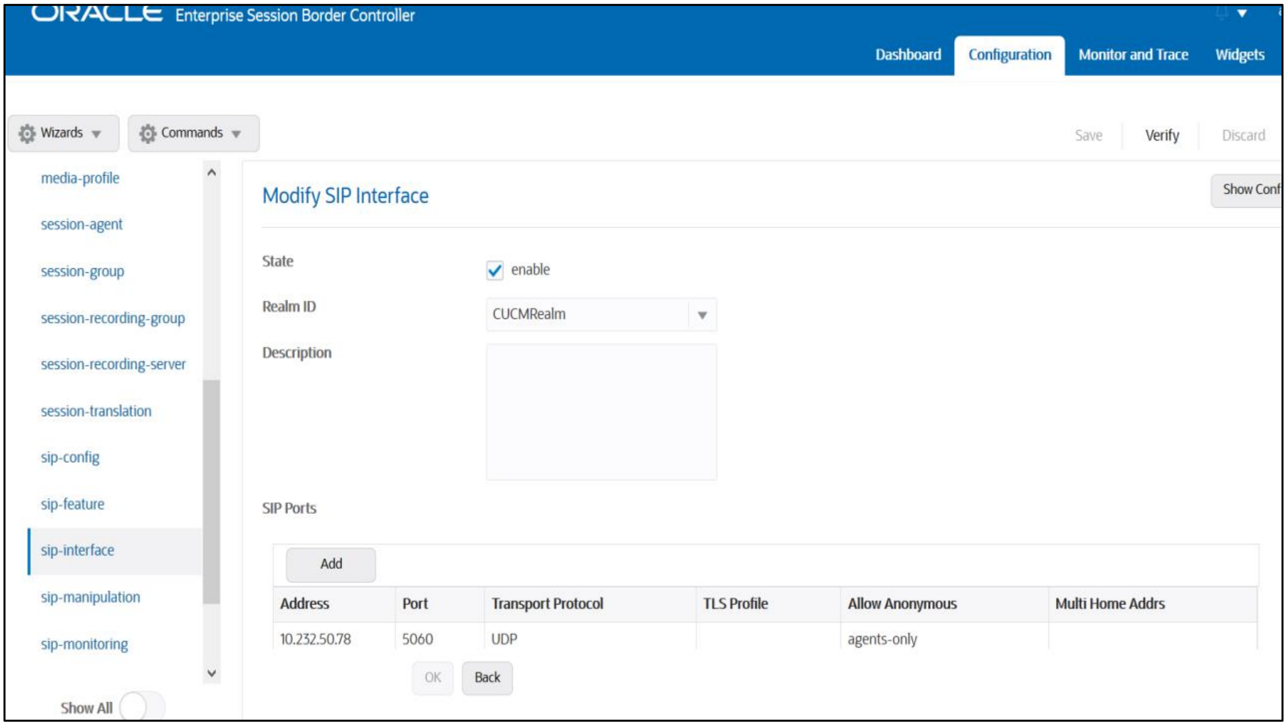
Realm ID TwilioRealm

Description

SIP Ports

Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addrs
141.146.36.102	5061	TLS	TLSProfile	agents-only	

Similarly, Please Configure sip-interface for the Cisco side as below:



Modify SIP Interface

State ☑ enable

Realm ID CUCMRealm

Description

SIP Ports

Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addrs
10.232.50.78	5060	UDP		agents-only	

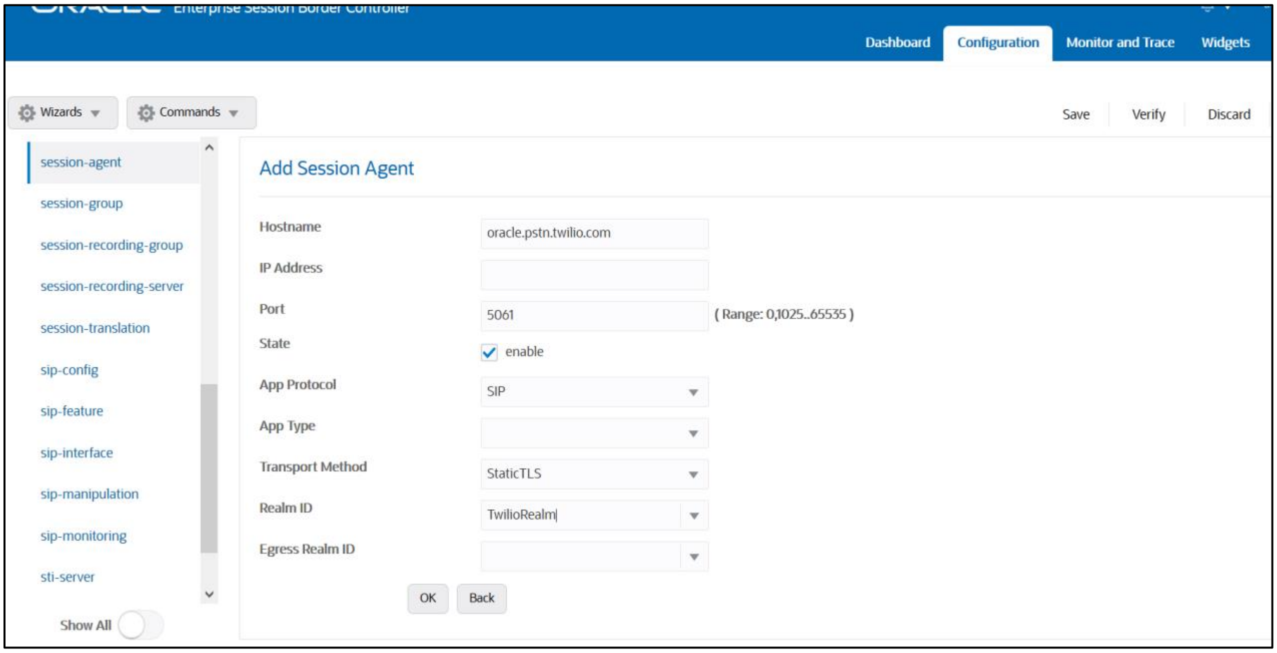
Once sip-interface is configured – the SBC is ready to accept traffic on the allocated IP address.

6.11. Configure session-agent

Session-agents are config elements which are trusted agents who can send/receive traffic from the SBC with direct access to trusted data path. Session-agents are config elements which are trusted agents who can send/receive traffic from the SBC with direct access to trusted data path.

Go to session-router->Session-Agent and Configure the session-agents for the Twilio Elastic SIP Trunk

- Host name to “oracle.pstn.twilio.com”, port to 5061
- realm-id – needs to match the realm created for the Twilio Elastic SIP Trunk
- transport set to “staticTLS”



Add Session Agent

Hostname: oracle.pstn.twilio.com
IP Address:
Port: 5061 (Range: 0,1025..65535)
State: ☑ enable
App Protocol: SIP
App Type:
Transport Method: StaticTLS
Realm ID: TwilioRealm
Egress Realm ID:

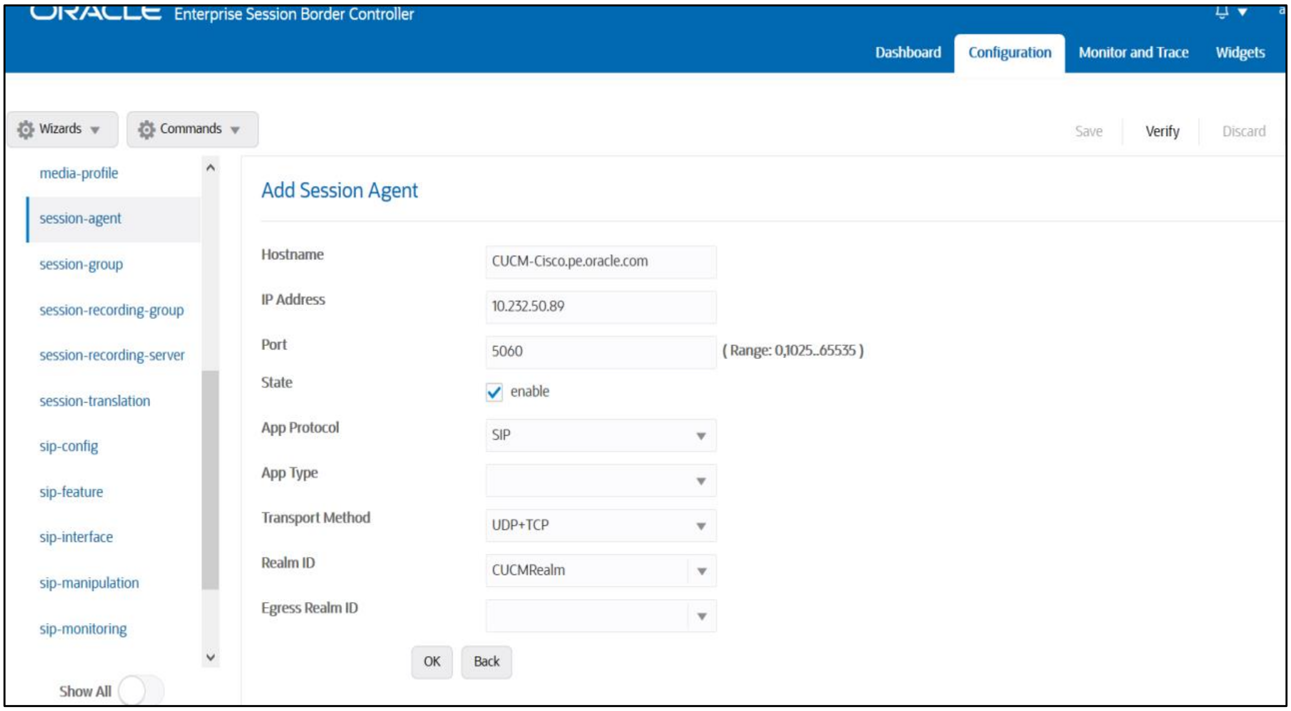
****NOTE: Connection to Twilio Elastic SIP Trunking is available in multiple geographic edge locations. If you wish to manually connect to a specific geographic edge location that is closest to the location of your communications infrastructure, you may do so by pointing your communications infrastructure to any of the following localized Termination SIP URIs:**

- {example}.pstn.ashburn.twilio.com (North America Virginia)
- {example}.pstn.umatilla.twilio.com (North America Oregon)
- {example}.pstn.dublin.twilio.com (Europe Ireland)
- {example}.pstn.frankfurt.twilio.com (Europe Frankfurt)
- {example}.pstn.singapore.twilio.com (Asia Pacific Singapore)
- {example}.pstn.tokyo.twilio.com (Asia Pacific Tokyo)
- {example}.pstn.sao-paulo.twilio.com (South America São Paulo)
- {example}.pstn.sydney.twilio.com (Asia Pacific Sydney)

Click here for more information on Twilio Elastic SIP Trunking IP Address

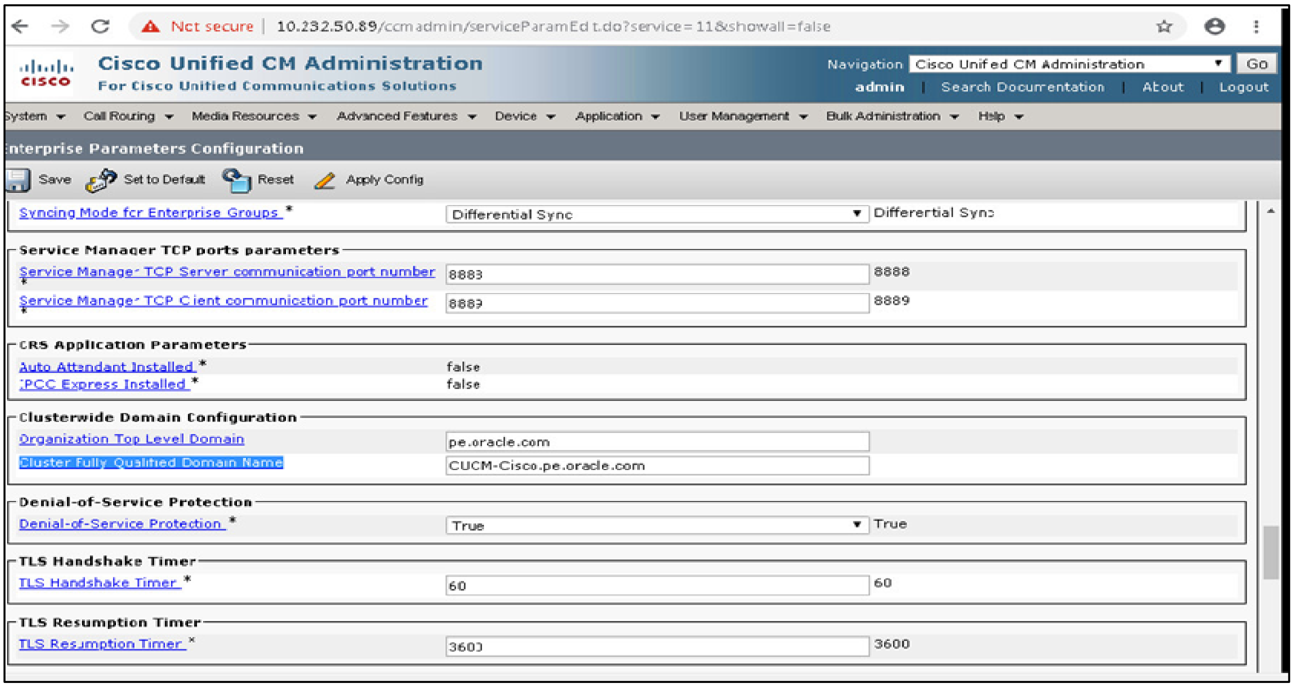
Similarly, configure the session-agents for the Cisco Side as below:

- Host name to FQDN of CUCM which is “CUCM-Cisco.pe.oracle.com” in our example. **We can also give Cisco CUCM IP address if there is no host name configured.**
- The same FQDN value should be configured in Cisco CUCM under System ---Enterprise Parameter ----Cluster FQDN.



Add Session Agent

Hostname	CUCM-Cisco.pe.oracle.com
IP Address	10.232.50.89
Port	5060 (Range: 0,1025..65535)
State	☑ enable
App Protocol	SIP
App Type	
Transport Method	UDP+TCP
Realm ID	CUCMRealm
Egress Realm ID	


Nct secure | 10.232.50.89/ccmadmin/serviceParamEdt.do?service=11&showall=false
Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unifed CM Administration Go
admin | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Save Set to Default Reset Apply Config

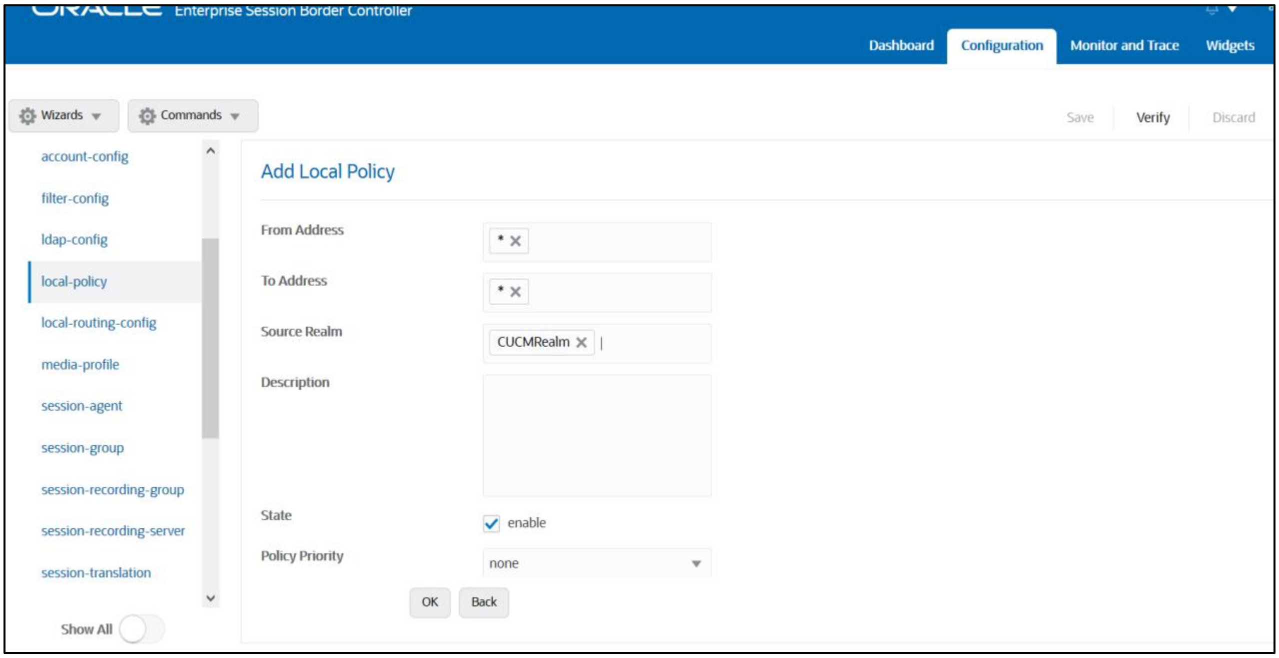
nterprise Parameters Configuration

Syncing Mode fcr Enterprise Groups *	Differential Sync	Differertial Sync
Service Manager TCP ports parameters		
Service Manage· TCP Server communication port number *	8883	8888
Service Manage· TCP Cient communication port number *	8889	8889
CRS Application Parameters		
Auto Attendant Installed *	false	
IPCC Express Installed *	false	
Clusterwide Domain Configuration		
Organization Top Level Domain	pe.oracle.com	
Cluster Fully Qualified Domain Name	CUCM-Cisco.pe.oracle.com	
Denial-of-Service Protection		
Denial-of-Service Protection *	True	True
TLS Handshake Timer		
TLS Handshake Timer *	60	60
TLS Resumption Timer		
TLS Resumption Timer *	3600	3600

6.12. Configure local-policy

Local policy config allows for the SBC to route calls from one end of the network to the other based on routing criteria. To configure local-policy, go to Session-Router->local-policy.

To route the calls from Cisco side to Twilio side, Use the below local –policy



Add Local Policy

From Address: *

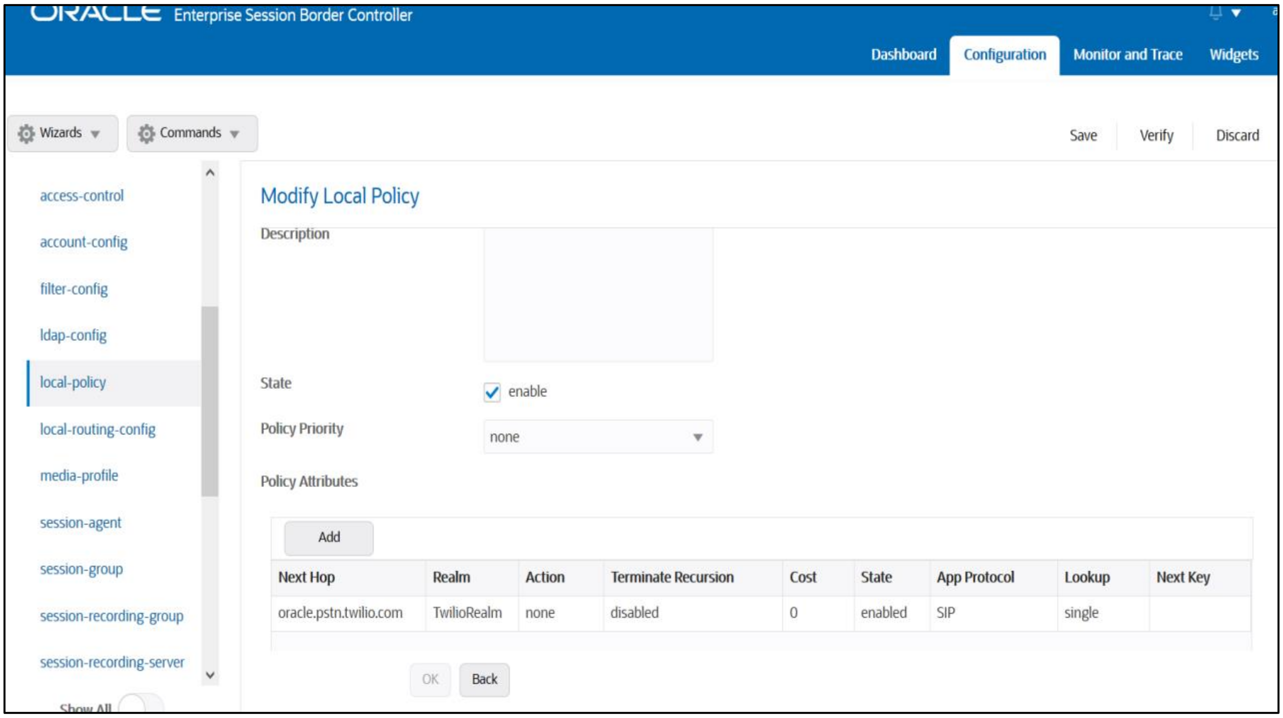
To Address: *

Source Realm: CUCMRealm

Description

State: ☑ enable

Policy Priority: none



Modify Local Policy

Description

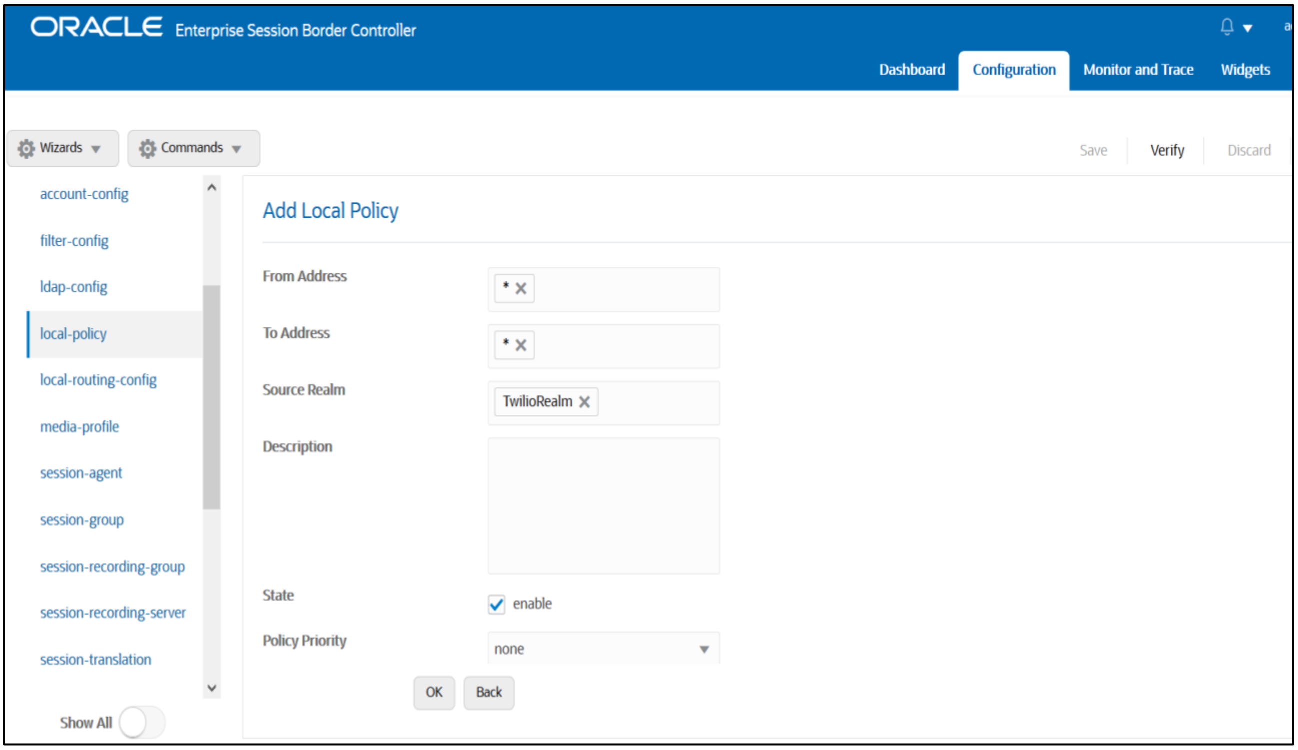
State: ☑ enable

Policy Priority: none

Policy Attributes

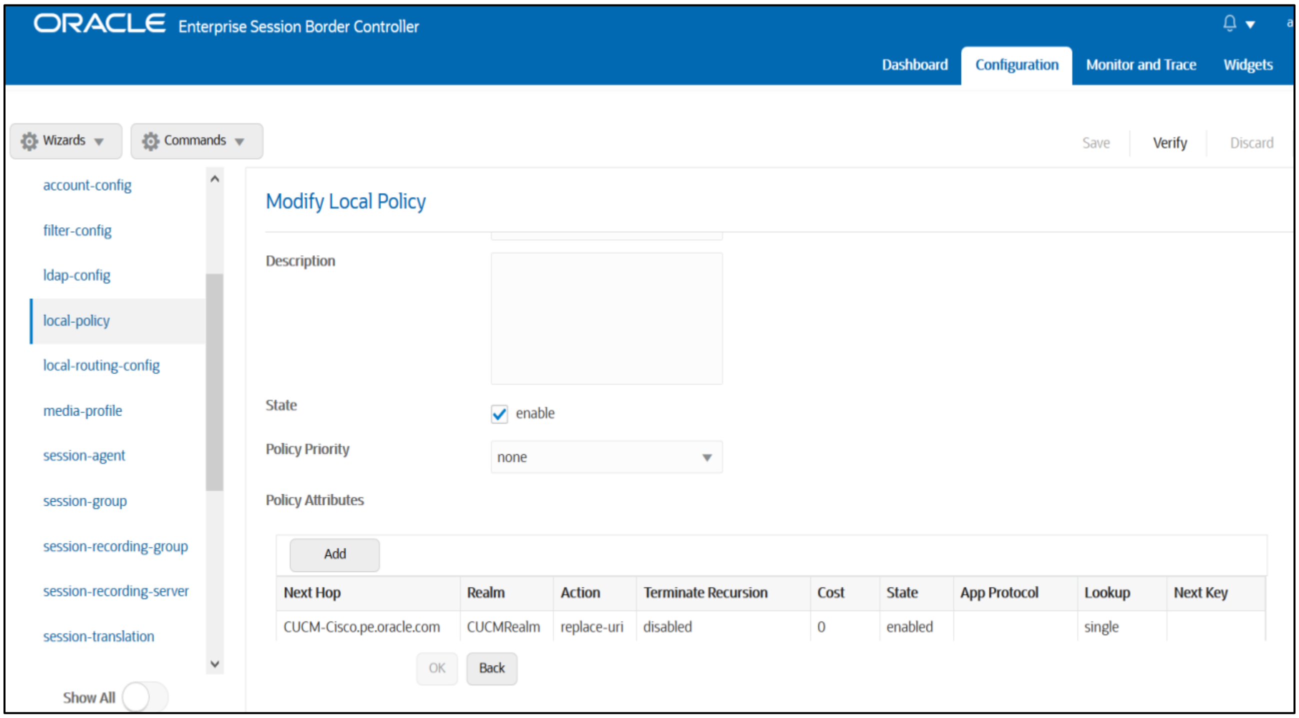
Next Hop	Realm	Action	Terminate Recursion	Cost	State	App Protocol	Lookup	Next Key
oracle.pstn.twilio.com	TwilioRealm	none	disabled	0	enabled	SIP	single	

To route the calls from the Twilio Elastic SIP Trunk side to Cisco side, Use the below local –policy



Add Local Policy

From Address	*
To Address	*
Source Realm	TwilioRealm
Description	
State	☑ enable
Policy Priority	none



Modify Local Policy

Description	
State	☑ enable
Policy Priority	none

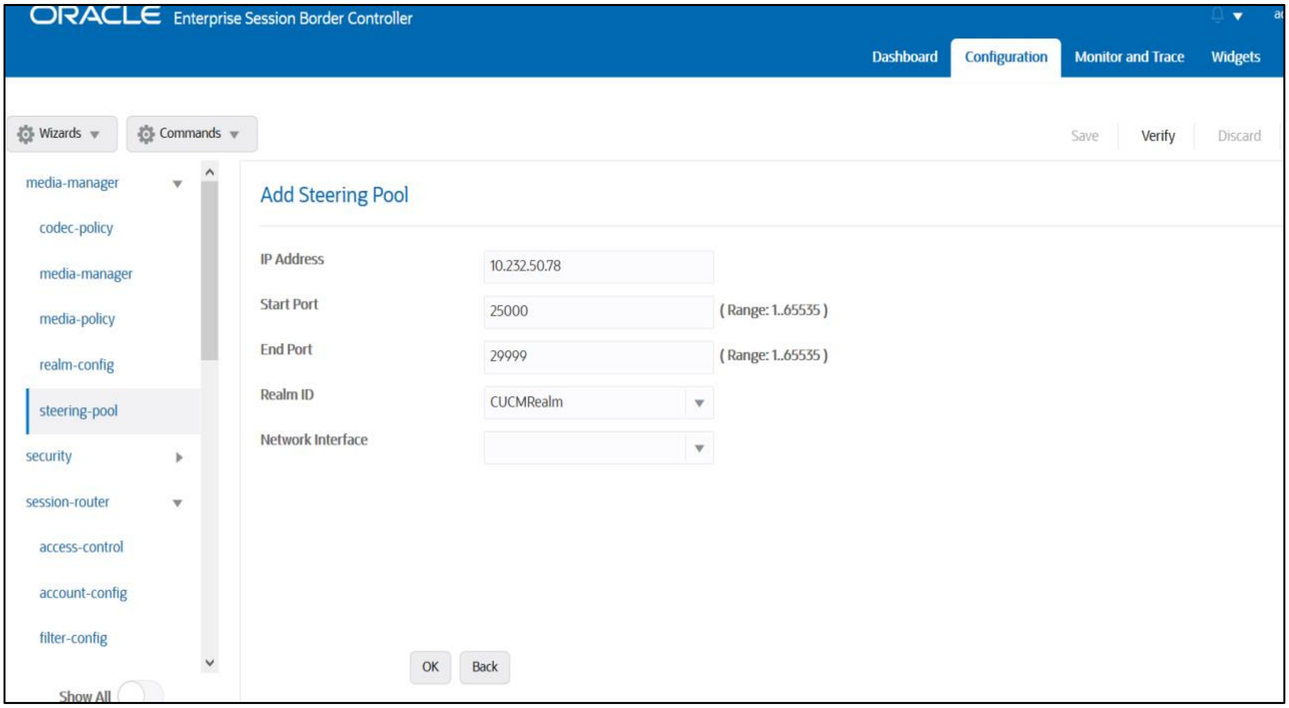
Policy Attributes

Next Hop	Realm	Action	Terminate Recursion	Cost	State	App Protocol	Lookup	Next Key
CUCM-Cisco.pe.oracle.com	CUCMRealm	replace-uri	disabled	0	enabled		single	

6.13. Configure steering-pool

Steering-pool config allows configuration to assign IP address(es), ports & a realm.

Cisco side steering pool.



Add Steering Pool

IP Address: 10.232.50.78

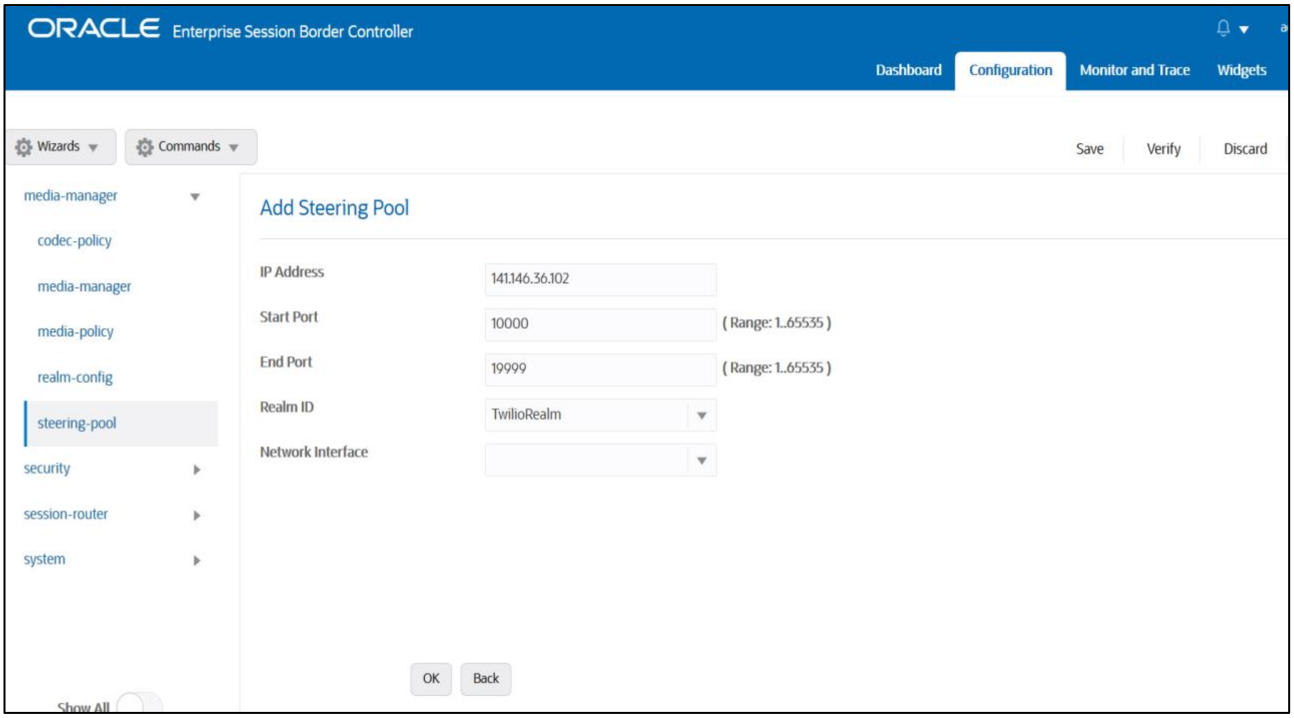
Start Port: 25000 (Range: 1..65535)

End Port: 29999 (Range: 1..65535)

Realm ID: CUCMRealm

Network Interface:

Twilio side steering pool.



Add Steering Pool

IP Address: 141.146.36.102

Start Port: 10000 (Range: 1..65535)

End Port: 19999 (Range: 1..65535)

Realm ID: TwilioRealm

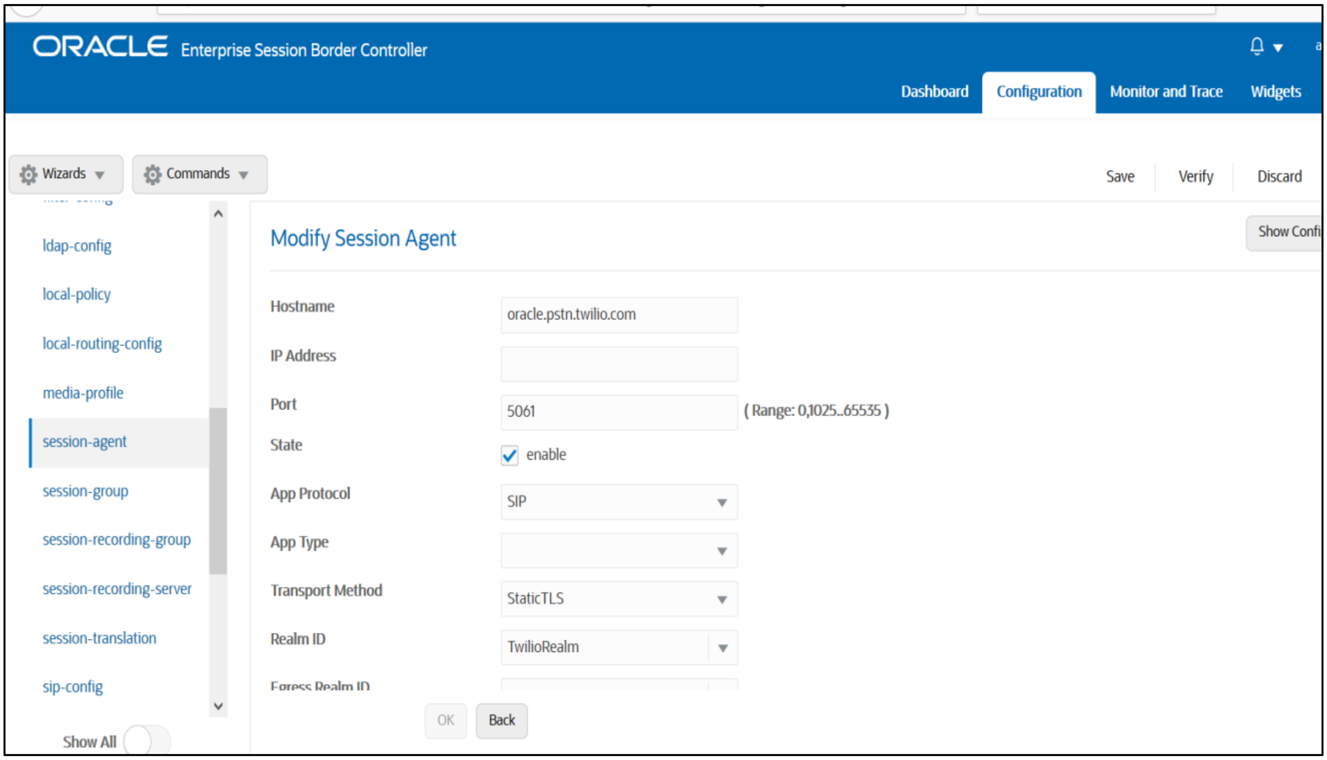
Network Interface:

6.14. Configure Ping Response

To simplify the ORACLE SBC configuration, from GA Release SCZ830m1p7, there is a new parameter introduced under the **Session agent** configuration element. The parameter name is **Ping response**.

Ping Response:

When this parameter is enabled, the SBC responds with a 200 OK to all Sip Options Pings it receives from trusted agents. This takes the place of the current Sip Manipulation, RepondOptions.



Modify Session Agent

Hostname: oracle.pstn.twilio.com

IP Address:

Port: 5061 (Range: 0,1025..65535)

State: ☑ enable

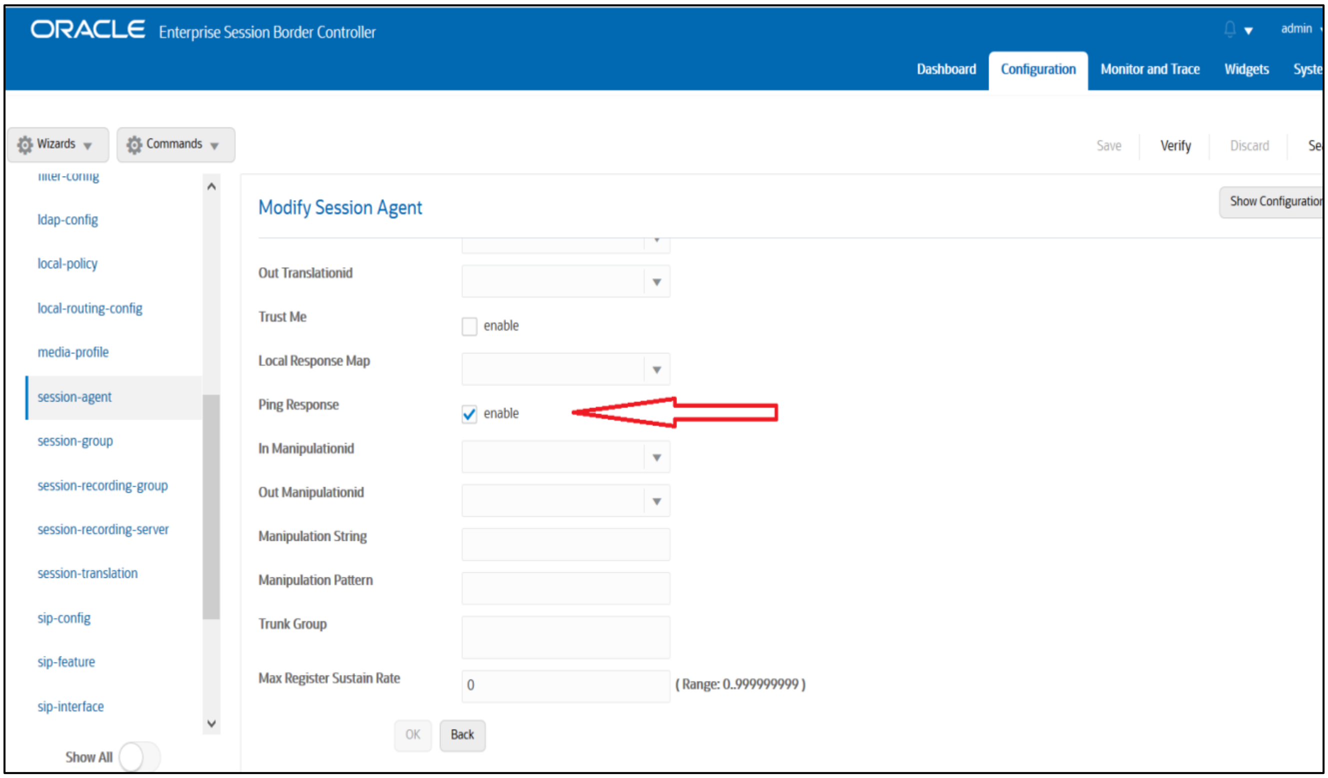
App Protocol: SIP

App Type:

Transport Method: StaticTLS

Realm ID: TwilioRealm

Egress Realm ID



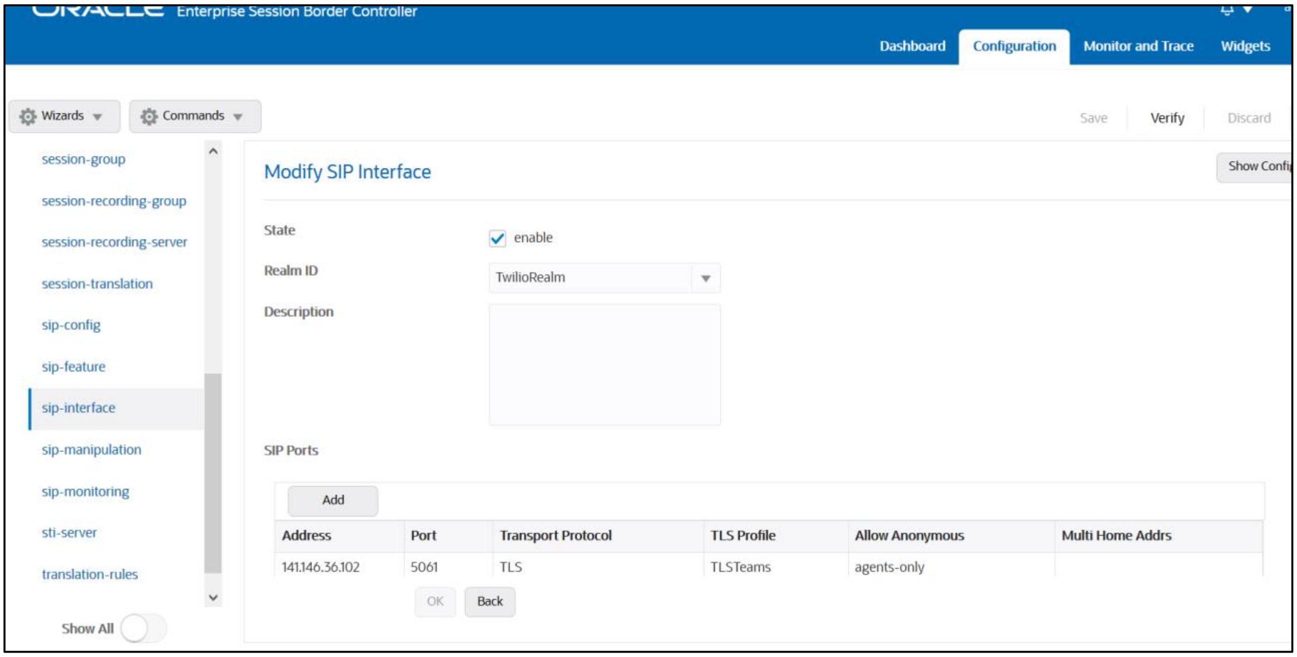
Modify Session Agent

Out Translationid	
Trust Me	☐ enable
Local Response Map	
Ping Response	☑ enable
In Manipulationid	
Out Manipulationid	
Manipulation String	
Manipulation Pattern	
Trunk Group	
Max Register Sustain Rate	0 (Range: 0..999999999)

6.15. SBC config for Cisco Offer less INVITE

When CUCM sends INVITE without SDP towards SBC and in that case, SBC needs to send out INVITE with SDP towards Twilio Elastic SIP trunk and vice versa. To do that, please set the parameter "**Add SDP Invite**" as both under Twilio sip interface as highlighted below. When this option is enabled, codecs have to be configured under the parameter "**Add SDP profiles**”. The configured codecs is also shown below.

Note: this is an optional config – configure this only if CUCM sends offer less INVITE towards SBC.



Modify SIP Interface

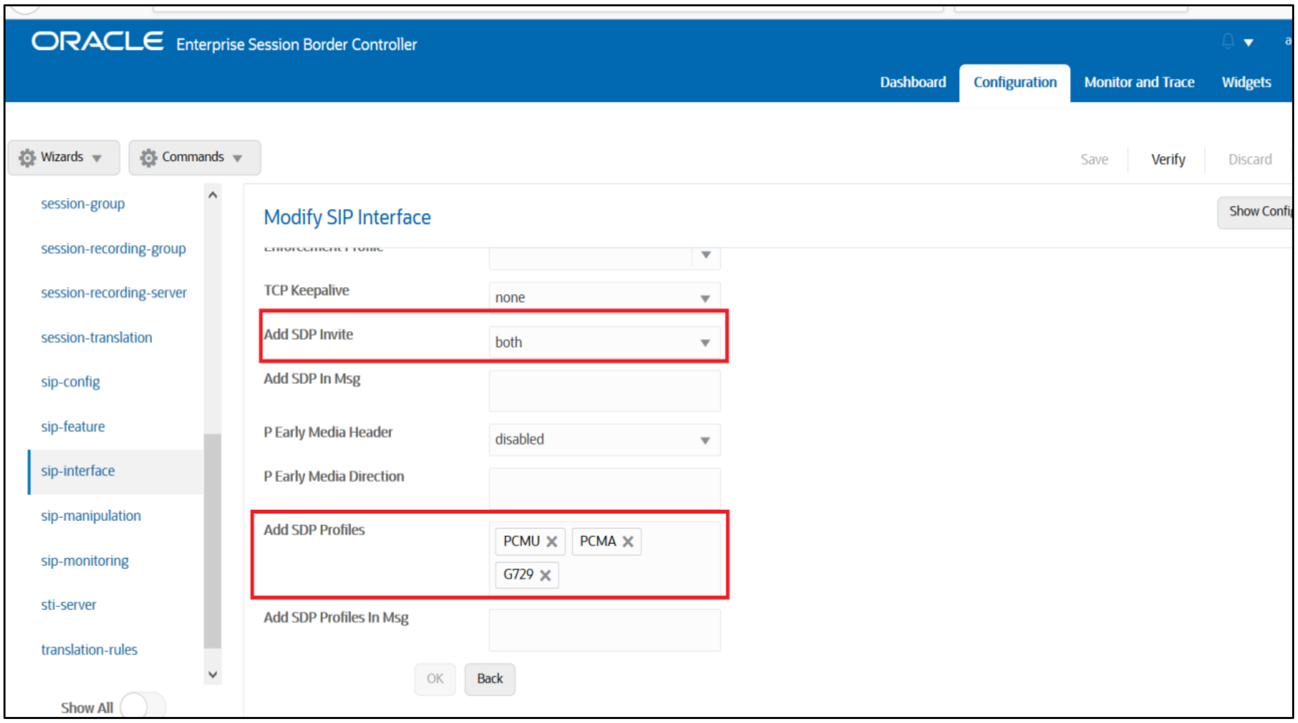
State ☑ enable

Realm ID: TwilioRealm

Description

SIP Ports

Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addrs
141.146.36.102	5061	TLS	TLSTeams	agents-only	



Modify SIP Interface

TCP Keepalive: none

Add SDP Invite: both

Add SDP In Msg

P Early Media Header: disabled

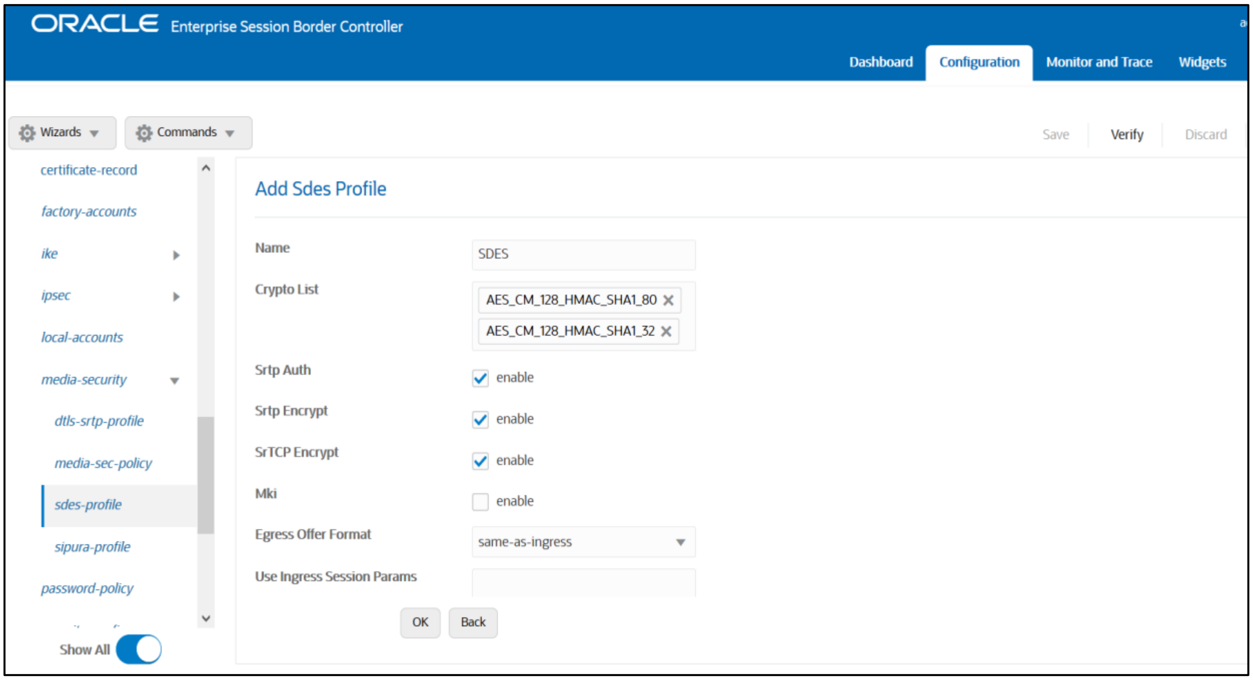
P Early Media Direction

Add SDP Profiles: PCMU ✕ PCMA ✕ G729 ✕

Add SDP Profiles In Msg

6.16. Configure sdes profile

Please go to →Security → Media Security →sdes profile and create the policy as below.

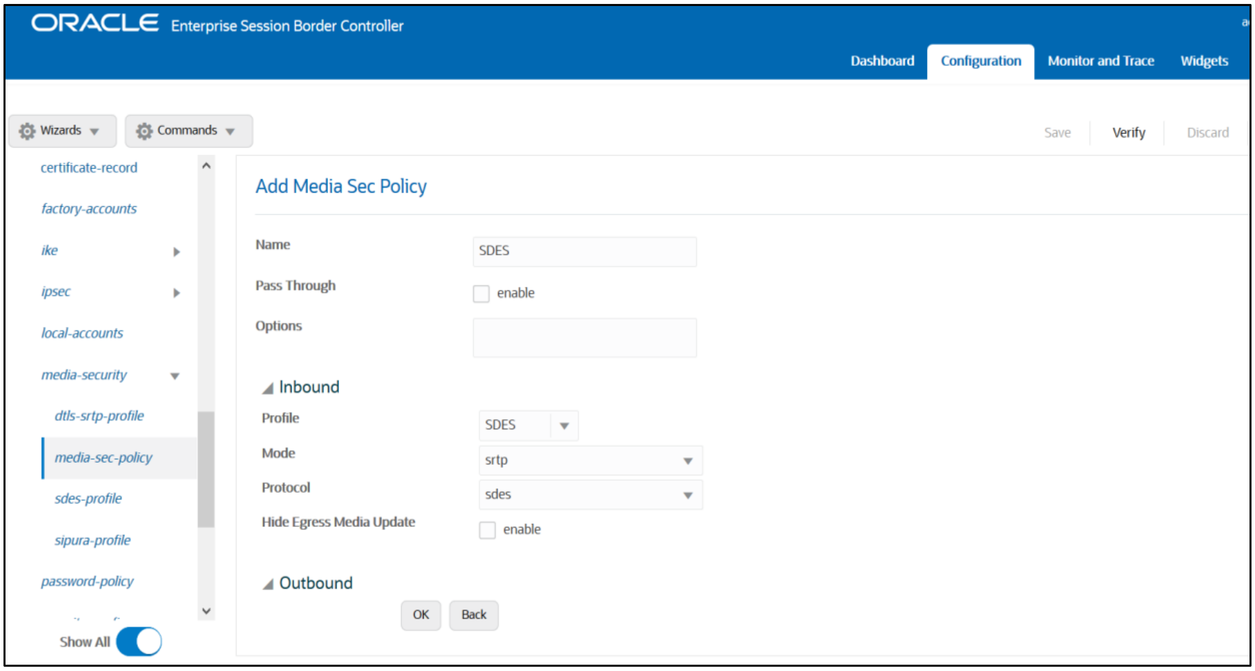


Add Sdes Profile

Name	SDES
Crypto List	AES_CM_128_HMAC_SHA1_80 AES_CM_128_HMAC_SHA1_32
Srtp Auth	☑ enable
Srtp Encrypt	☑ enable
SrTCP Encrypt	☑ enable
Mki	□ enable
Egress Offer Format	same-as-ingress
Use Ingress Session Params	

6.17. Configure Media Security Profile

Please go to →Security → Media Security →media Sec policy and create the policy as below:
Create Media Sec policy with name SDES which will have the sdes profile created above.
Assign this media policy to Twilio Realm as it use TLS/SRTP.



Add Media Sec Policy

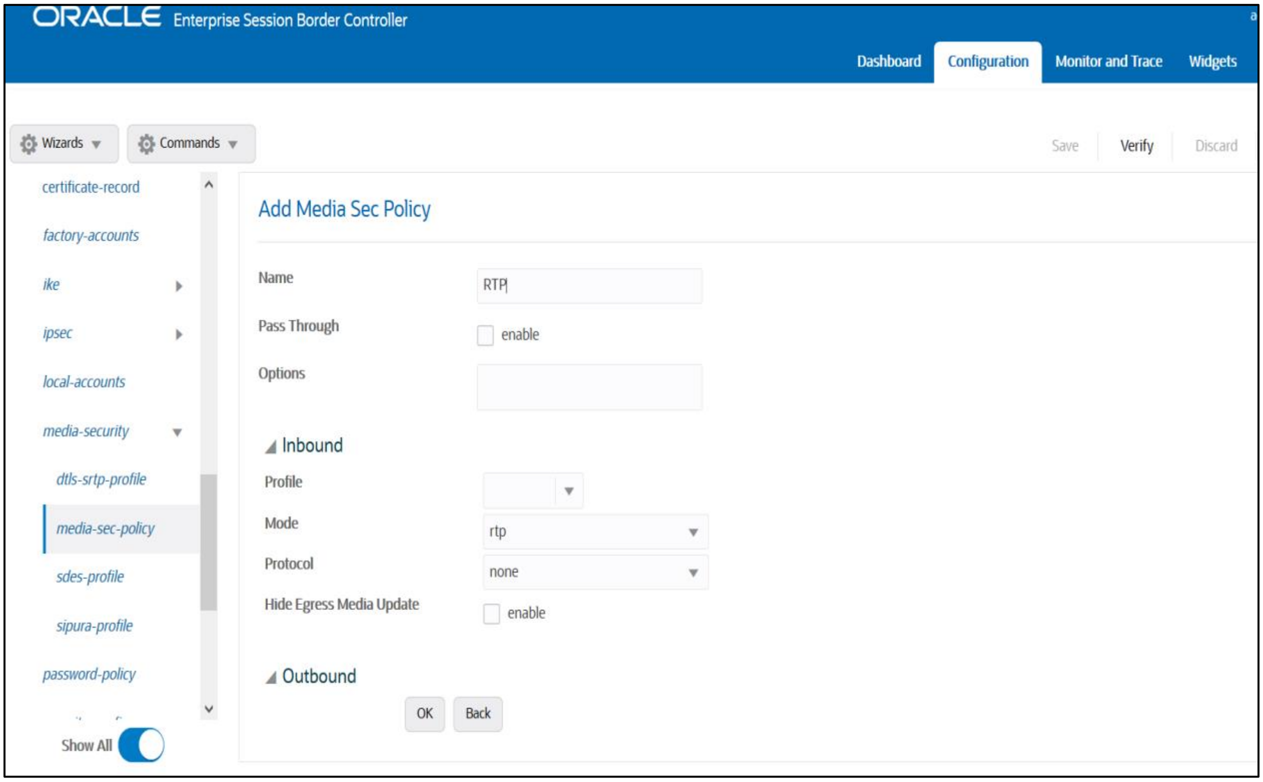
Name	SDES
Pass Through	□ enable
Options	

Inbound

Profile	SDES
Mode	srtp
Protocol	sdes
Hide Egress Media Update	□ enable

Outbound

Similarly, Create Media Sec policy with name RTP to convert srtp to rtp for the Cisco side which will use only TCP/UDP as transport protocol. **Assign this media policy to the Cisco Realm.**



Add Media Sec Policy

Name: RTP

Pass Through: □ enable

Options:

Inbound

Profile:

Mode: rtp

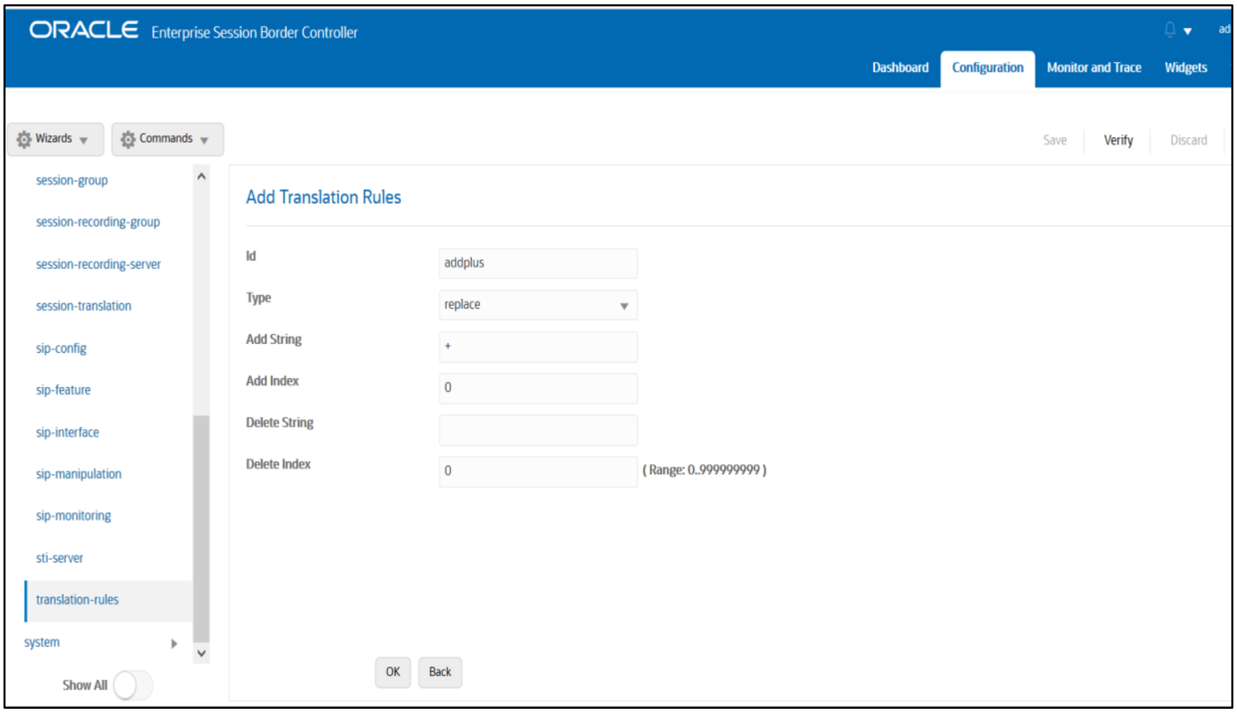
Protocol: none

Hide Egress Media Update: □ enable

Outbound

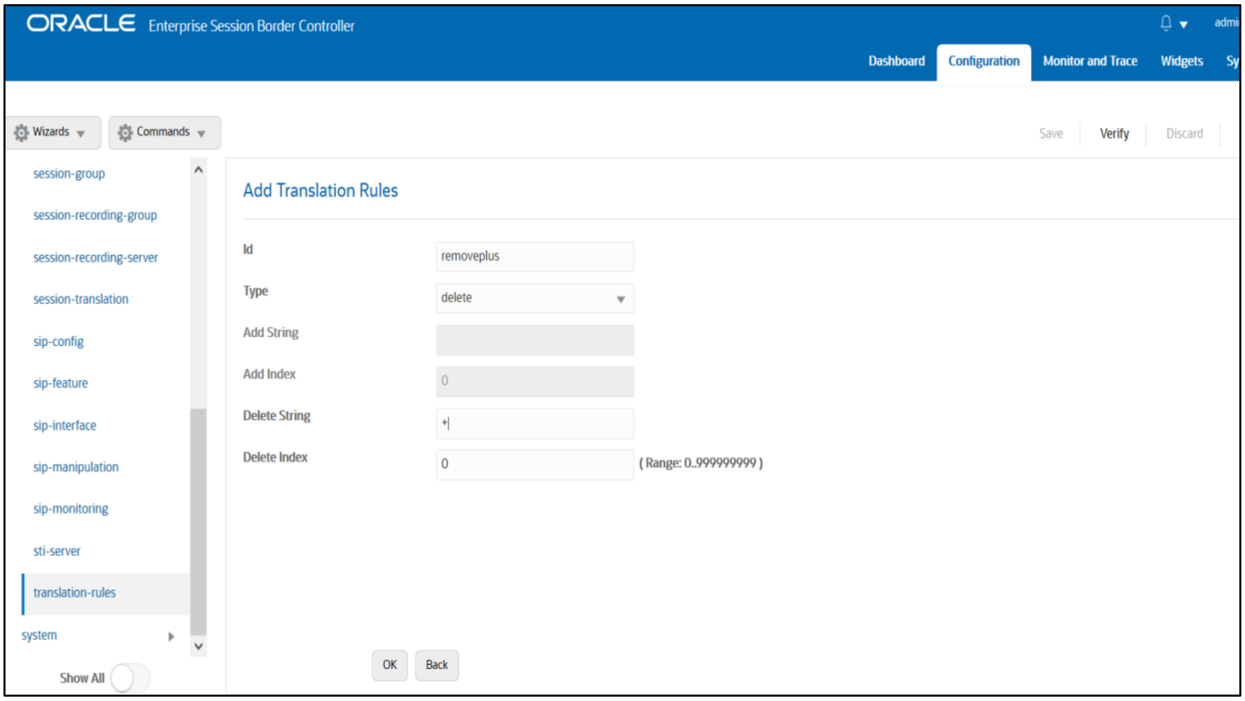
6.18. Configure Translation Rules

The translation rules sub-element is where the actual translation rules are created.
Go to Session router → translation-rules and create the below rule.



Add Translation Rules

Id	addplus
Type	replace
Add String	+
Add Index	0
Delete String	
Delete Index	0 (Range: 0..999999999)



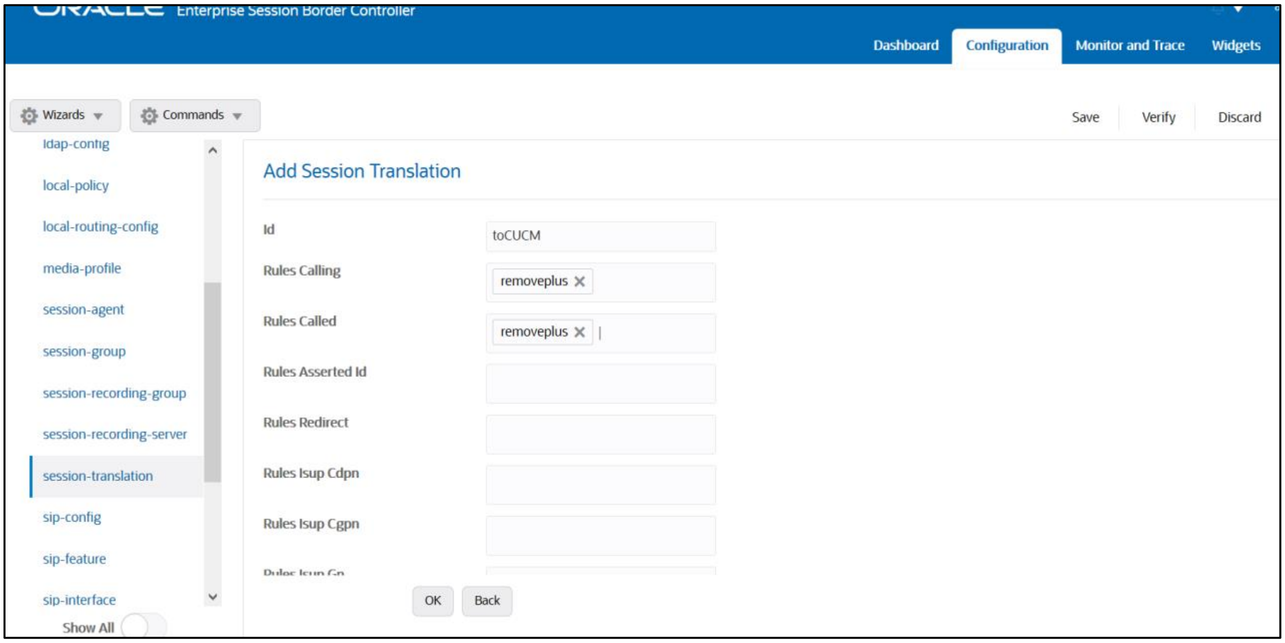
Add Translation Rules

Id	removeplus
Type	delete
Add String	
Add Index	0
Delete String	+
Delete Index	0 (Range: 0..999999999)

6.19. Configure Session Translation Rules

A session translation defines how translation rules are applied to calling and called numbers.
Go to Session Router → session-translation and configure the below translation rules.

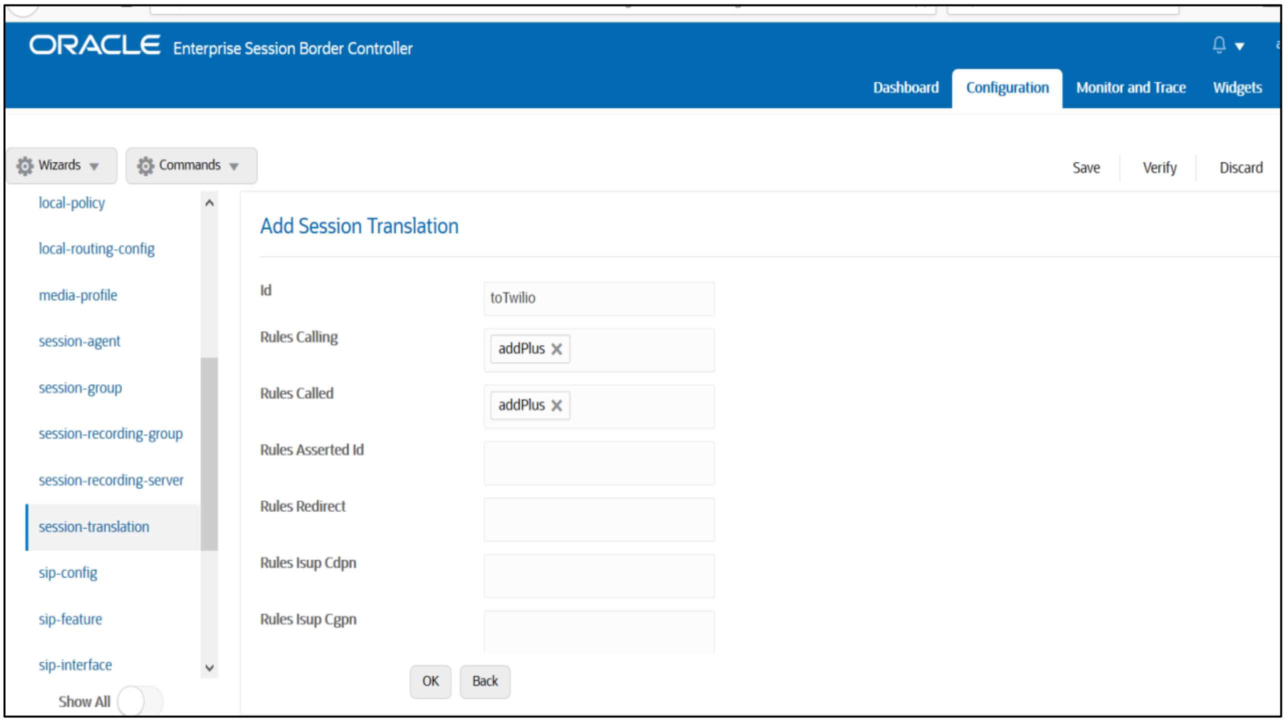
Add the below translation rule to Cisco side.



Add Session Translation

Id	toCUCM
Rules Calling	removeplus
Rules Called	removeplus
Rules Asserted Id	
Rules Redirect	
Rules Isup Cdpn	
Rules Isup Cgpn	

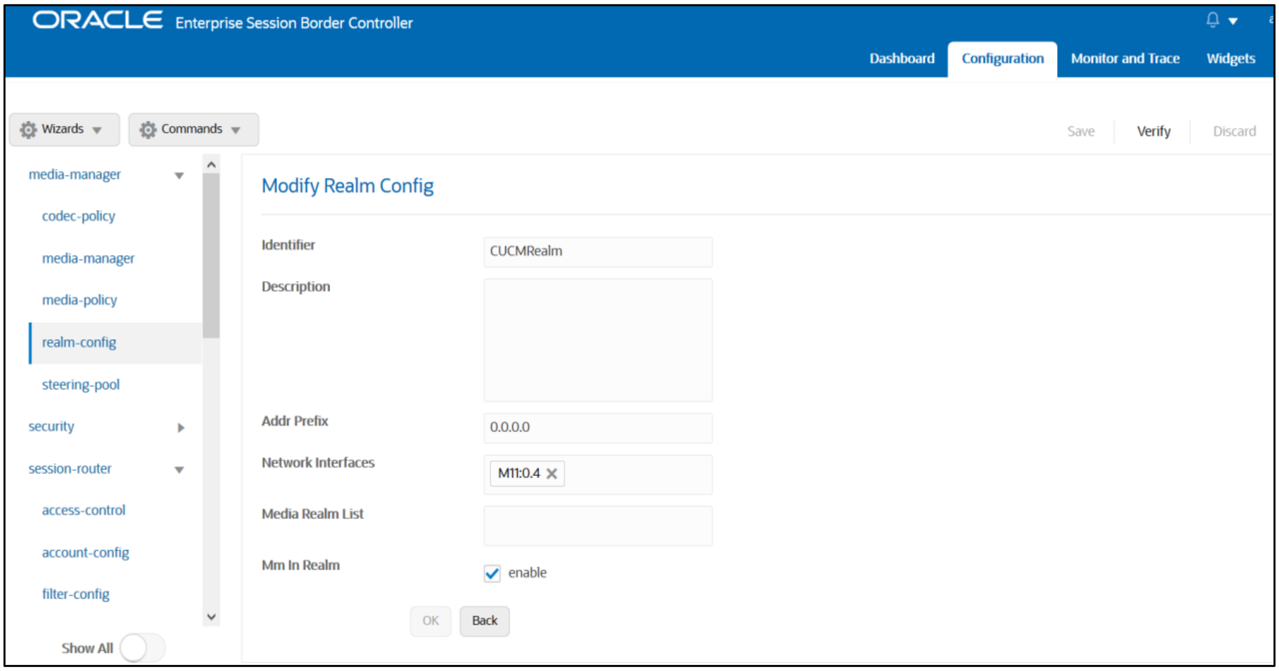
Add the below translation rule to Twilio side as PSTN expects call with + sign.



Add Session Translation

Id	toTwilio
Rules Calling	addPlus
Rules Called	addPlus
Rules Asserted Id	
Rules Redirect	
Rules Isup Cdpn	
Rules Isup Cgpn	

Please add the above session translation rules to Cisco realm as shown below



Modify Realm Config

Identifier: CUCMRealm

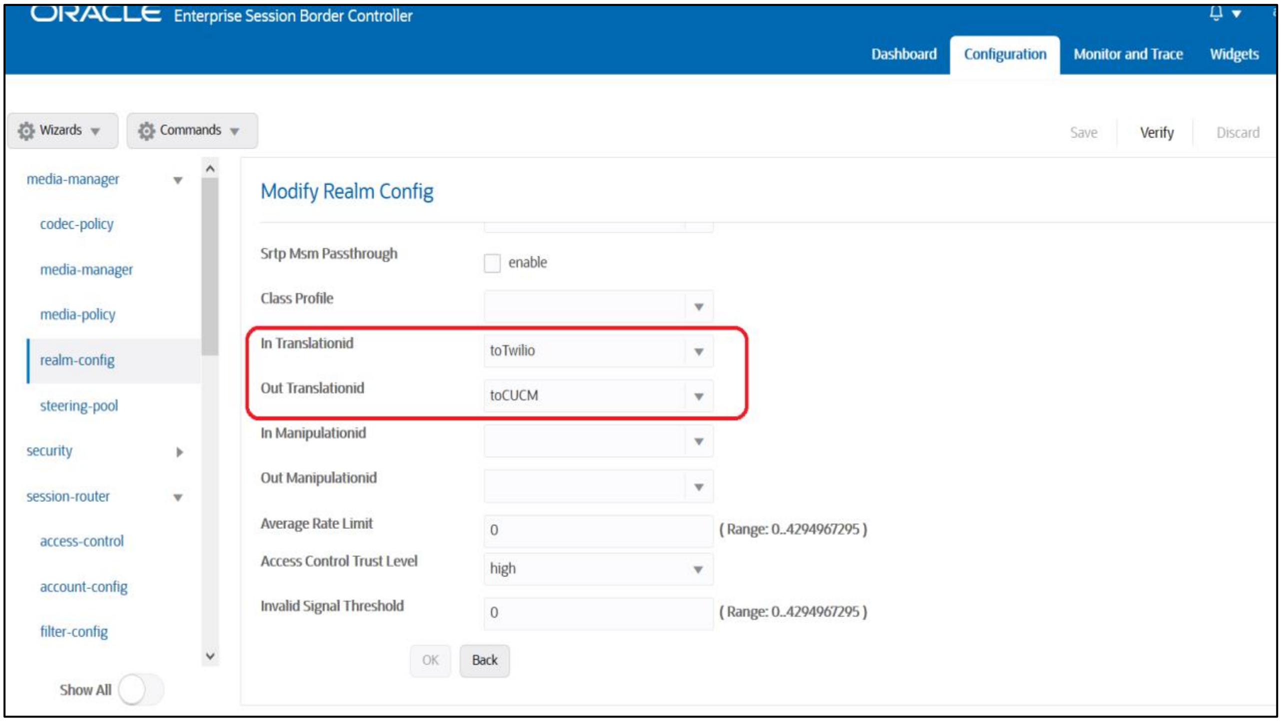
Description:

Addr Prefix: 0.0.0.0

Network Interfaces: M11:0.4

Media Realm List:

Mm In Realm: ☑ enable



Modify Realm Config

Srtp Msm Passthrough: □ enable

Class Profile:

In Translationid: toTwilio

Out Translationid: toCUCM

In Manipulationid:

Out Manipulationid:

Average Rate Limit: 0 (Range: 0..4294967295)

Access Control Trust Level: high

Invalid Signal Threshold: 0 (Range: 0..4294967295)

With this, SBC configuration is complete

7. SBC configuration for Cisco Remote Worker

This section of Cisco Remote Worker configuration is included for Cisco remote endpoints that register through the Oracle SBC to the Cisco Call Manager (Cisco CUCM). This would require additional configuration to be configured on the Oracle SBC along with the SIP trunking config as mentioned in the earlier description of the test bed. To complete the particular testing we have configured Cisco endpoints which will register to Cisco CUCM through the SBC. SBC will handle the calls based on the registration information present in the cache. **Please note that Cisco Remote worker Access side is secured (TLS/SRTP) and Cisco Core side is unsecured (UDP or TCP/RTP)**

In order to achieve the requirement we have made below configuration on the Oracle SBC

Access and Core Realm for Cisco Remote worker
Steering Pool associated with the Realm for Cisco Remote worker
Sip-interface associated with the Realm for Cisco Remote worker
(Optional) A local-policy to route the registration requests from this Realm to the SIP Server.

Note -The local-policy element is optional as we can enable the Route to registrar parameter on the sip-interface config to route the requests to the Registrar.
The registrar host and port is configured in the sip-config element on the SBC. The remote endpoint sends register requests from Cisco Access Realm onto the SBC and then SBC registers these endpoints onto the Cisco Core Realm maintaining the registration cache in its database to route inbound calls to these endpoint.

Below are the snippets from the Oracle SBC Web GUI for the Remote worker configuration.

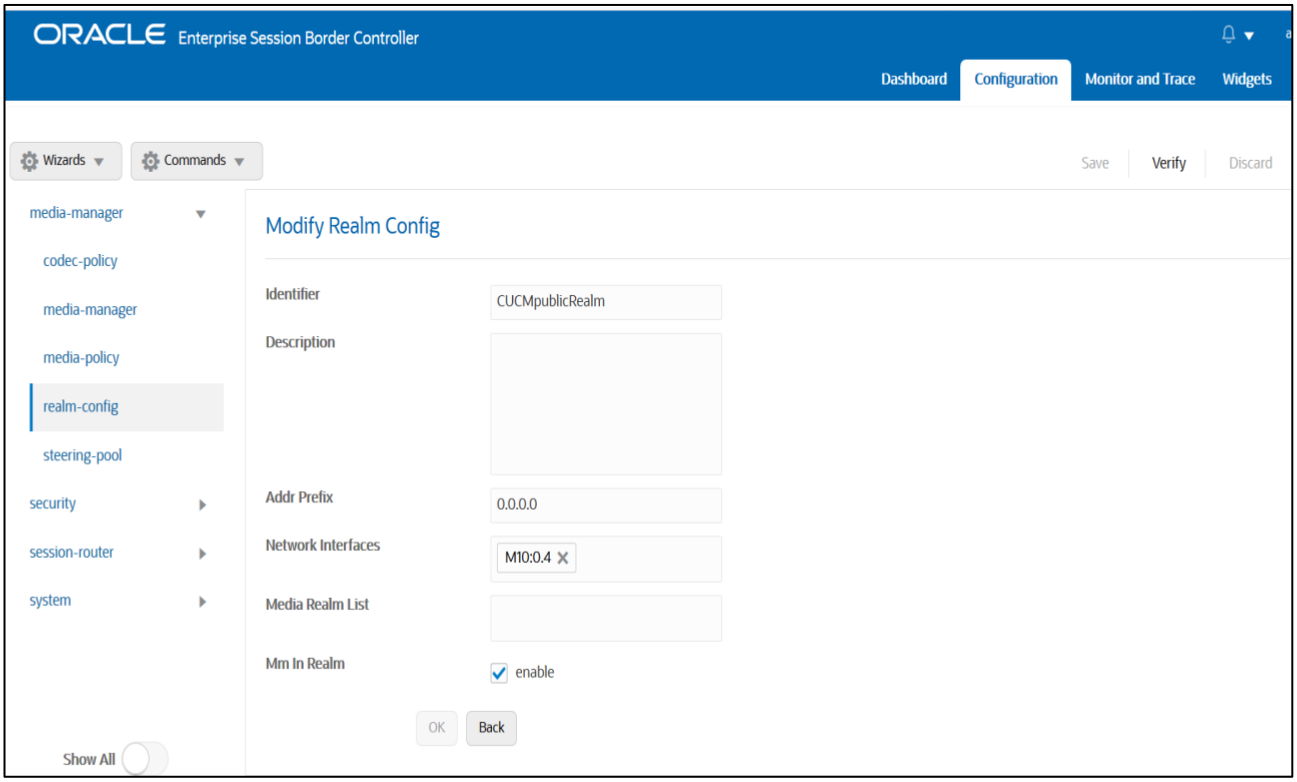
7.1. Configure Realms

Navigate to realm-config under media-manager and configure a realm as shown below
The name of the Realm can be any relevant name according to the user convenience.

Use the following table as a configuration example for the two realms used in this configuration:

Config Parameter	Cisco Access Side	Cisco Core Side
Identifier	CUCMpublicRealm	CUCMCoreRealm
Network Interface	M10	M11
Mm in realm	☑	☑
FQDN		
Media Sec policy	sdespolicy	RTP
Access Control Trust Level	High	High

In the below example, Realm name is given as CUCMpublicRealm for Cisco Access Side.
Please set the Access Control Trust Level as medium for this realm



Modify Realm Config

Identifier: CUCMpublicRealm

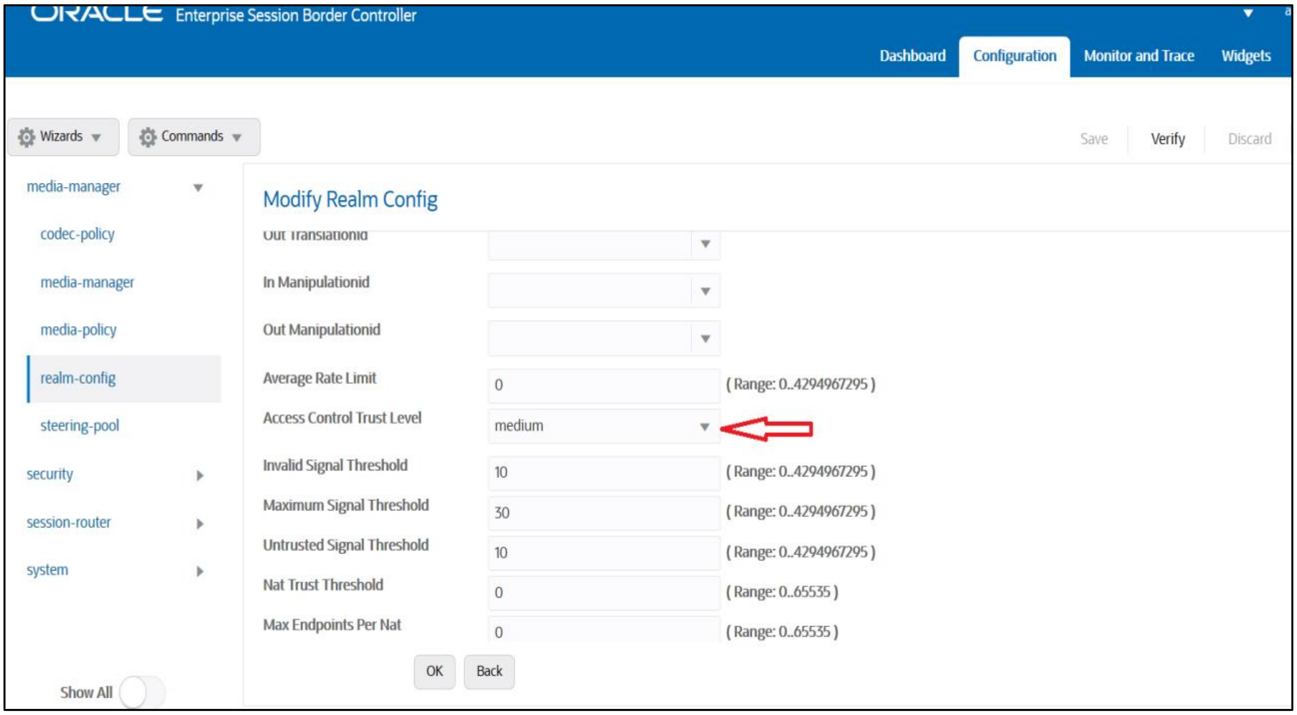
Description:

Addr Prefix: 0.0.0.0

Network Interfaces: M10:0.4 ✕

Media Realm List:

Mm In Realm: ☑ enable



Modify Realm Config

Out Translationid:

In Manipulationid:

Out Manipulationid:

Average Rate Limit: 0 (Range: 0..4294967295)

Access Control Trust Level: medium

Invalid Signal Threshold: 10 (Range: 0..4294967295)

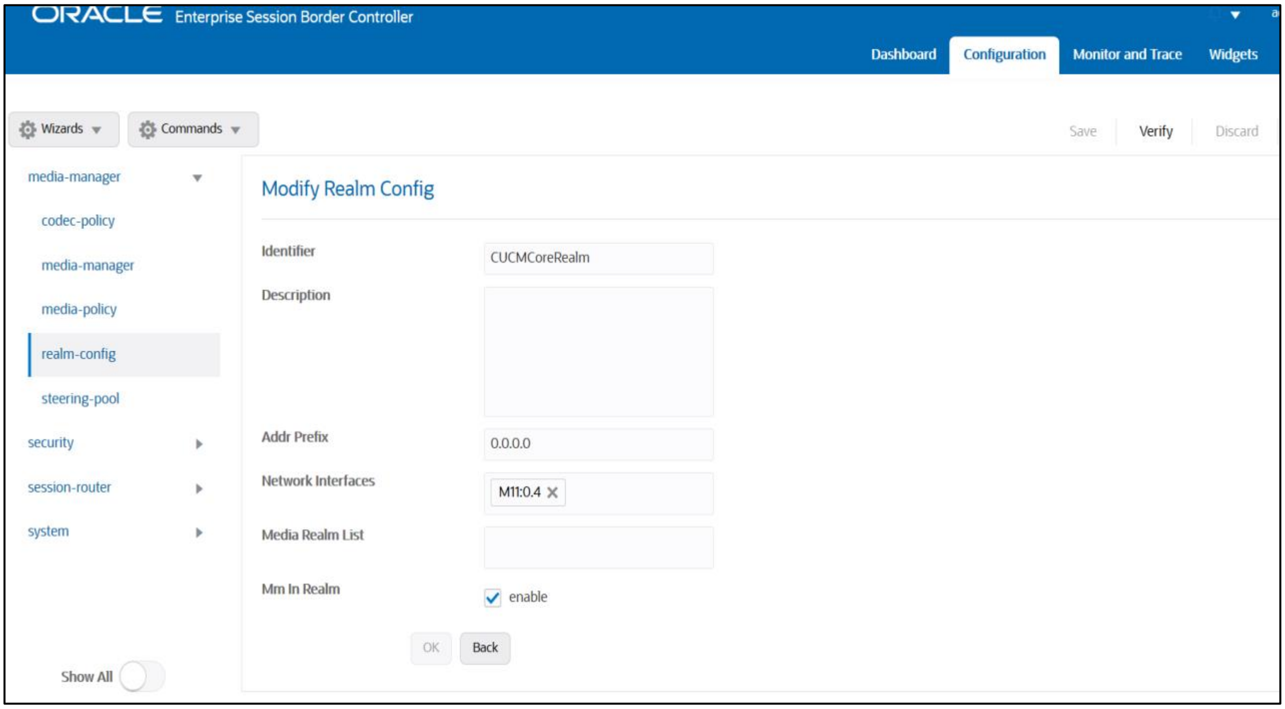
Maximum Signal Threshold: 30 (Range: 0..4294967295)

Untrusted Signal Threshold: 10 (Range: 0..4294967295)

Nat Trust Threshold: 0 (Range: 0..65535)

Max Endpoints Per Nat: 0 (Range: 0..65535)

Similarly, Realm name is given as CUCMCoreRealm for Cisco Core side



Modify Realm Config

Identifier: CUCMCoreRealm

Description:

Addr Prefix: 0.0.0.0

Network Interfaces: M11:0.4 ✕

Media Realm List:

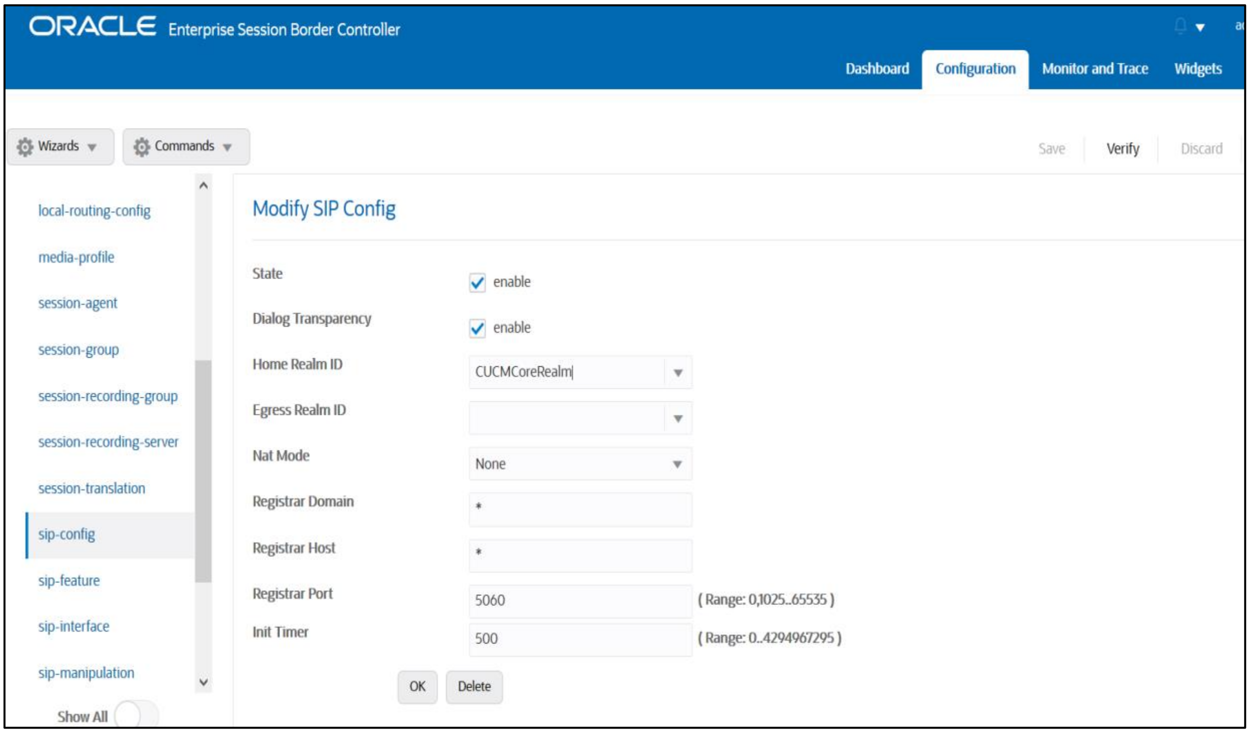
Mm In Realm: ☑ enable

7.2. Enable sip-config

SIP config enables SIP handling in the SBC.
Make sure the home realm-id, registrar-domain and registrar-host are configured.
Also add the options to the sip-config as shown below.

To configure sip-config, Go to Session-Router->sip-config and in options, add the below

- add max-udp-length =0
- reg-cach-mode=from



Modify SIP Config

State ☑ enable

Dialog Transparency ☑ enable

Home Realm ID CUCMCoreRealm

Egress Realm ID

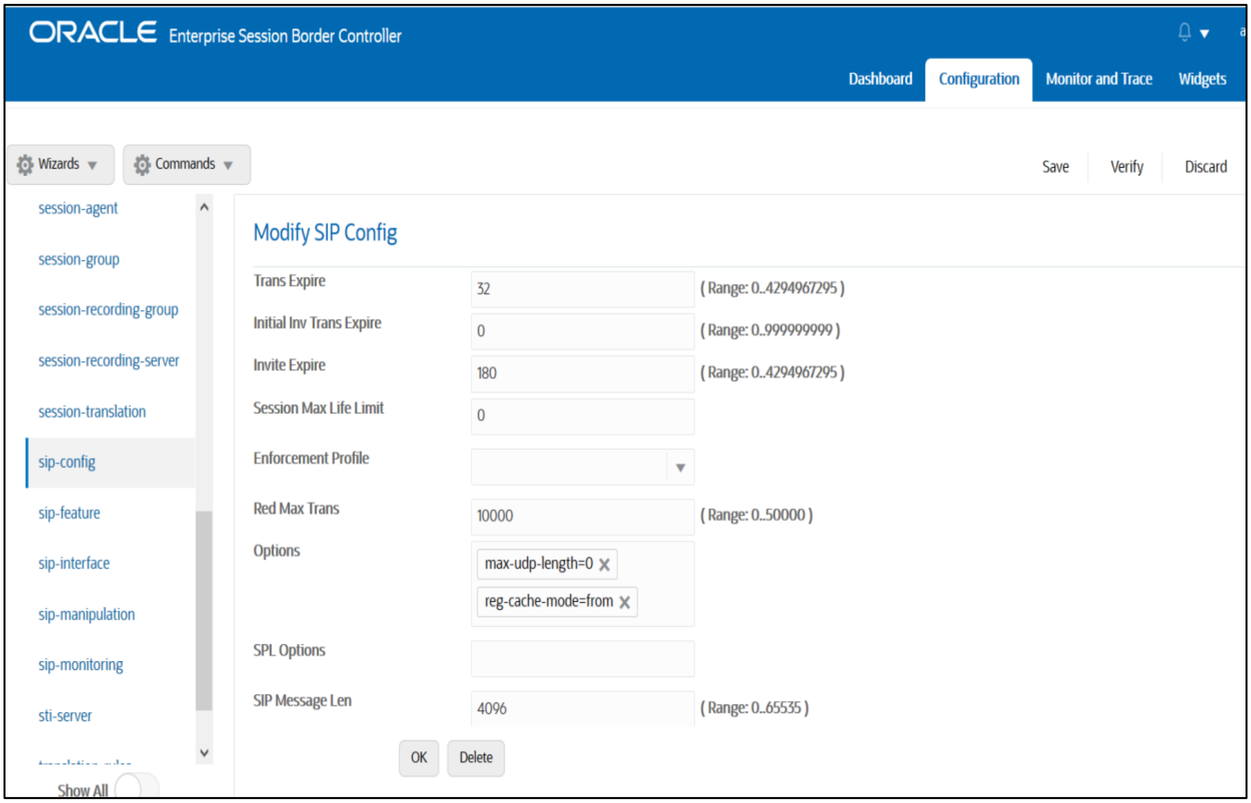
Nat Mode None

Registrar Domain *

Registrar Host *

Registrar Port 5060 (Range: 0,1025..65535)

Init Timer 500 (Range: 0..4294967295)



Modify SIP Config

Trans Expire 32 (Range: 0..4294967295)

Initial Inv Trans Expire 0 (Range: 0..999999999)

Invite Expire 180 (Range: 0..4294967295)

Session Max Life Limit 0

Enforcement Profile

Red Max Trans 10000 (Range: 0..50000)

Options max-udp-length=0 ✕ reg-cache-mode=from ✕

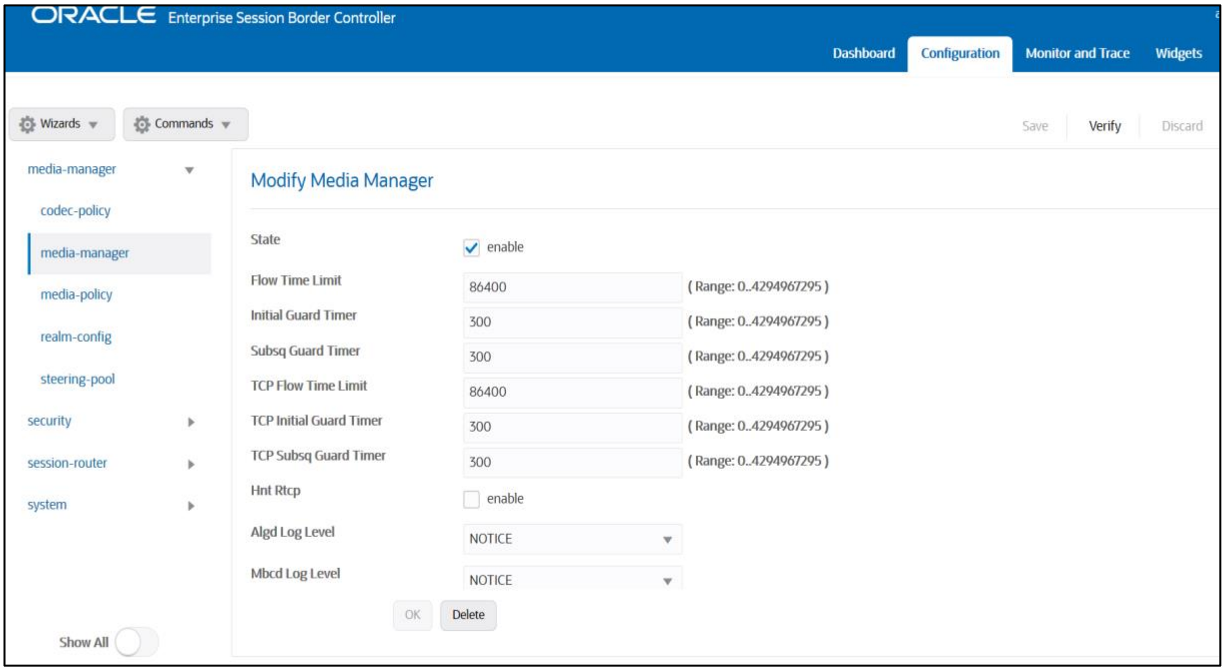
SPL Options

SIP Message Len 4096 (Range: 0..65535)

7.3. Enable media manager

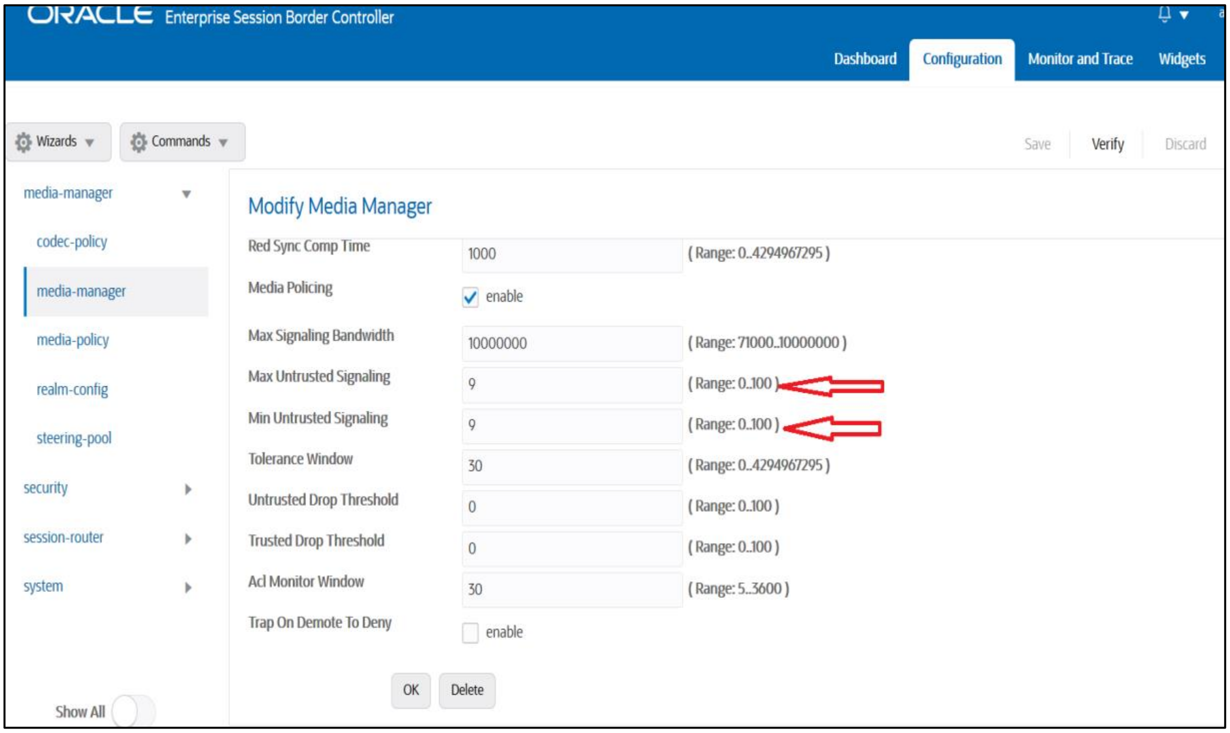
Media-manager handles the media stack required for SIP sessions on the SBC. Enable the media manager option as below.

In addition to the above config, please set the max and min untrusted signaling values to 9 which takes care of Access Realm. Go to Media-Manager->Media-Manager



Modify Media Manager

State	☑ enable	
Flow Time Limit	86400	(Range: 0..4294967295)
Initial Guard Timer	300	(Range: 0..4294967295)
Subsq Guard Timer	300	(Range: 0..4294967295)
TCP Flow Time Limit	86400	(Range: 0..4294967295)
TCP Initial Guard Timer	300	(Range: 0..4294967295)
TCP Subsq Guard Timer	300	(Range: 0..4294967295)
Hnt Rtcp	☐ enable	
Algd Log Level	NOTICE	
Mbcd Log Level	NOTICE	



Modify Media Manager

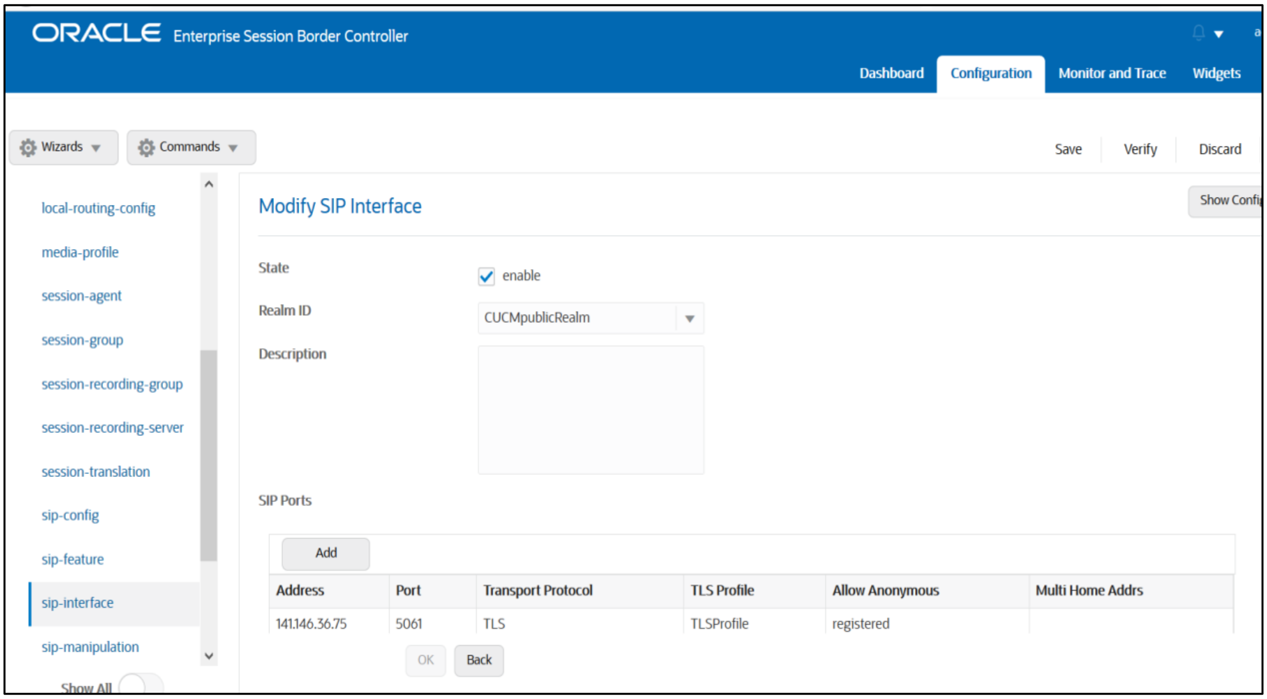
Red Sync Comp Time	1000	(Range: 0..4294967295)
Media Policing	☑ enable	
Max Signaling Bandwidth	10000000	(Range: 71000..10000000)
Max Untrusted Signaling	9	(Range: 0..100)
Min Untrusted Signaling	9	(Range: 0..100)
Tolerance Window	30	(Range: 0..4294967295)
Untrusted Drop Threshold	0	(Range: 0..100)
Trusted Drop Threshold	0	(Range: 0..100)
Acl Monitor Window	30	(Range: 5..3600)
Trap On Demote To Deny	☐ enable	

7.4. Configure SIP Interfaces

Navigate to sip-interface under session-router and configure the sip-interface as shown below. Please configure the below settings under the sip-interface.

Please Configure sip-interface for the for Cisco Access side as below:

- Tls-profile needs to match the name of the tls-profile created earlier.
- Set allow-anonymous to Registered to ensure traffic to this sip-interface only comes from the registered user.
- Set NAT traversal to always for the remote workers to register.
- Enable Registration Caching and Route to Register



Modify SIP Interface

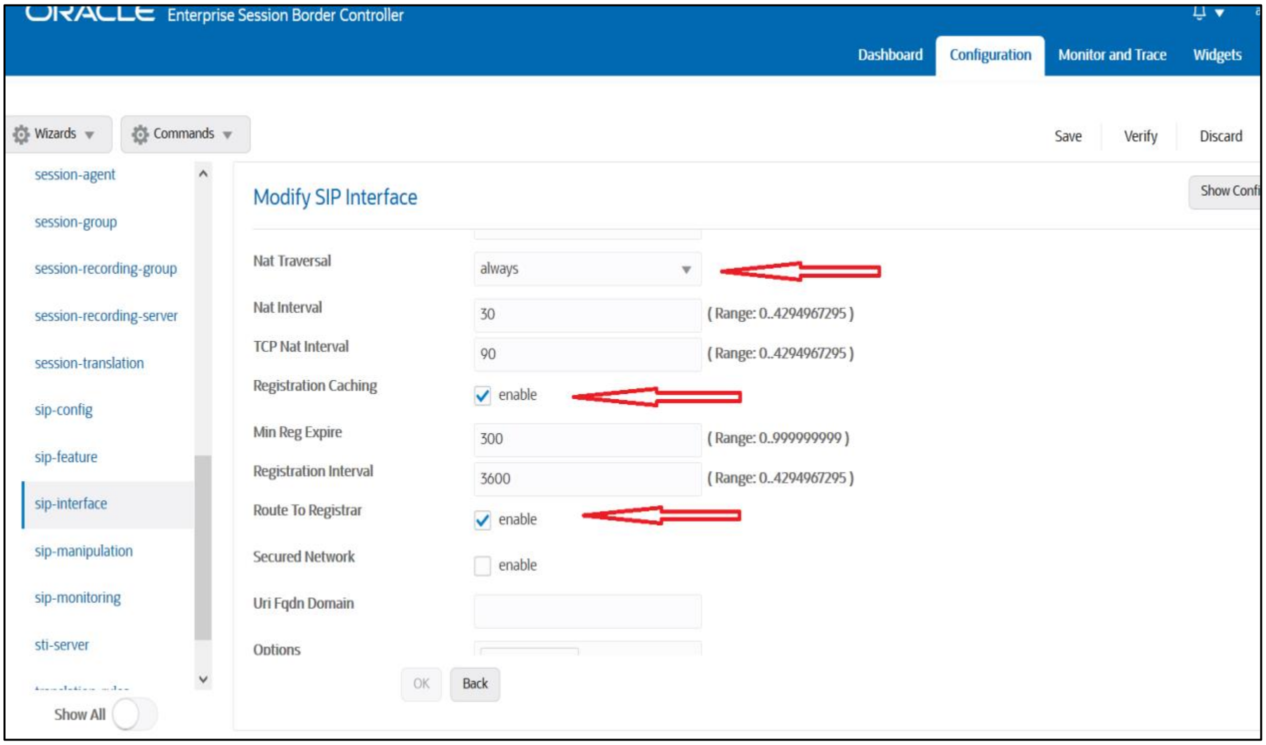
State ☑ enable

Realm ID CUCMpublicRealm

Description

SIP Ports

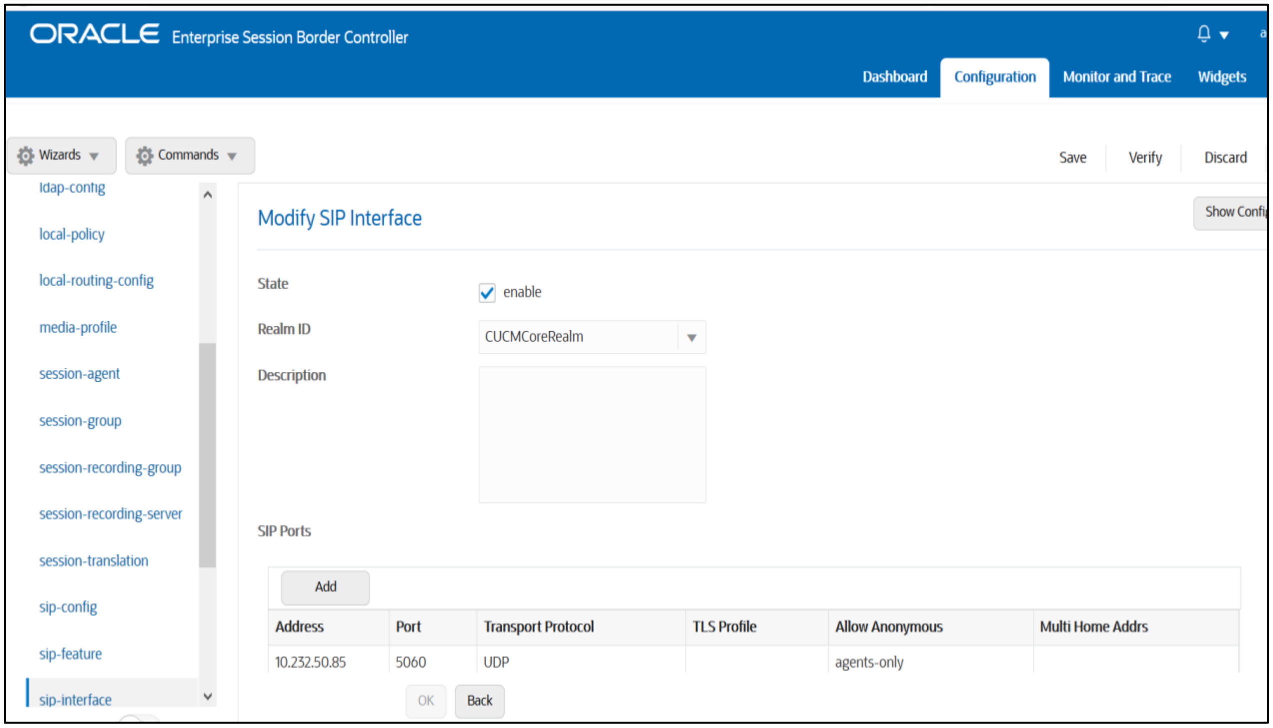
Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addrs
141.146.36.75	5061	TLS	TLSProfile	registered	



Modify SIP Interface

Nat Traversal	always	
Nat Interval	30	(Range: 0..4294967295)
TCP Nat Interval	90	(Range: 0..4294967295)
Registration Caching	☑ enable	
Min Reg Expire	300	(Range: 0..999999999)
Registration Interval	3600	(Range: 0..4294967295)
Route To Registrar	☑ enable	
Secured Network	☐ enable	
Uri Fqdn Domain		
Options		

Similarly, Please Configure sip-interface for the Cisco Core side as below:



Modify SIP Interface

State	☑ enable
Realm ID	CUCMCoreRealm
Description	

SIP Ports

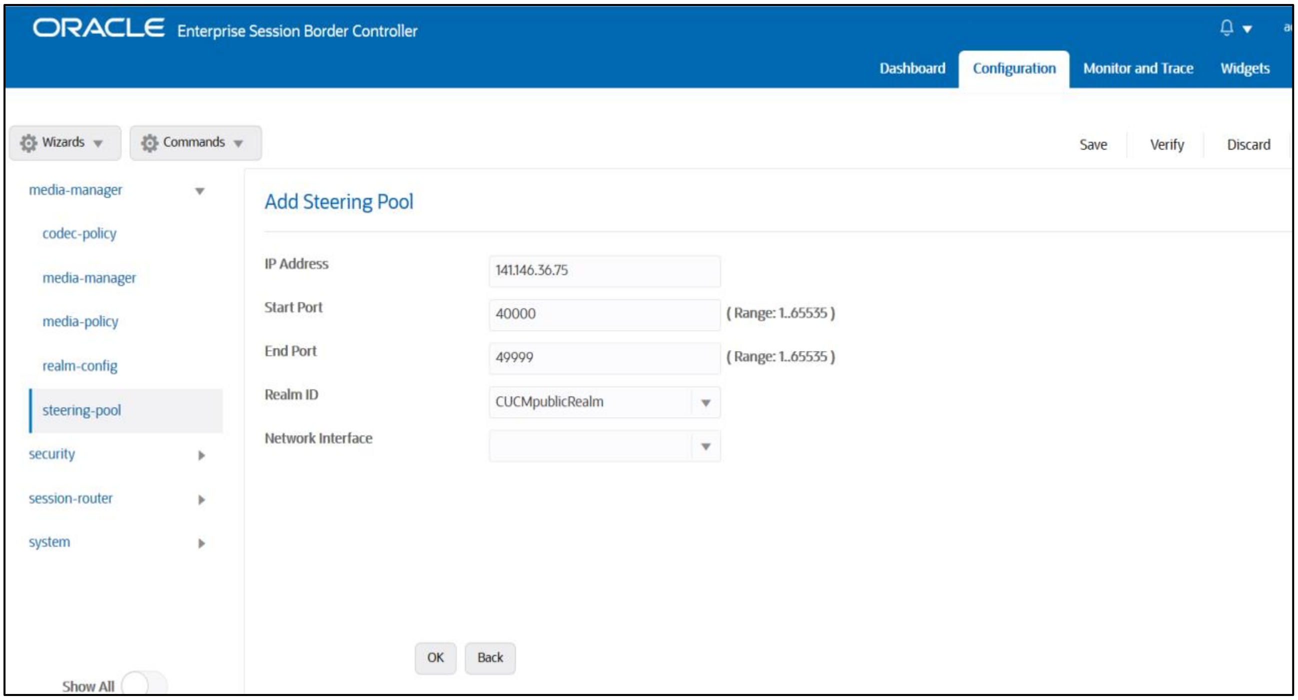
Address	Port	Transport Protocol	TLS Profile	Allow Anonymous	Multi Home Addrs
10.232.50.85	5060	UDP		agents-only	

Once sip-interface is configured – the SBC is ready to accept traffic on the allocated IP address.

7.5. Configure steering-pool

Steering-pool config allows configuration to assign IP address(es), ports & a realm.

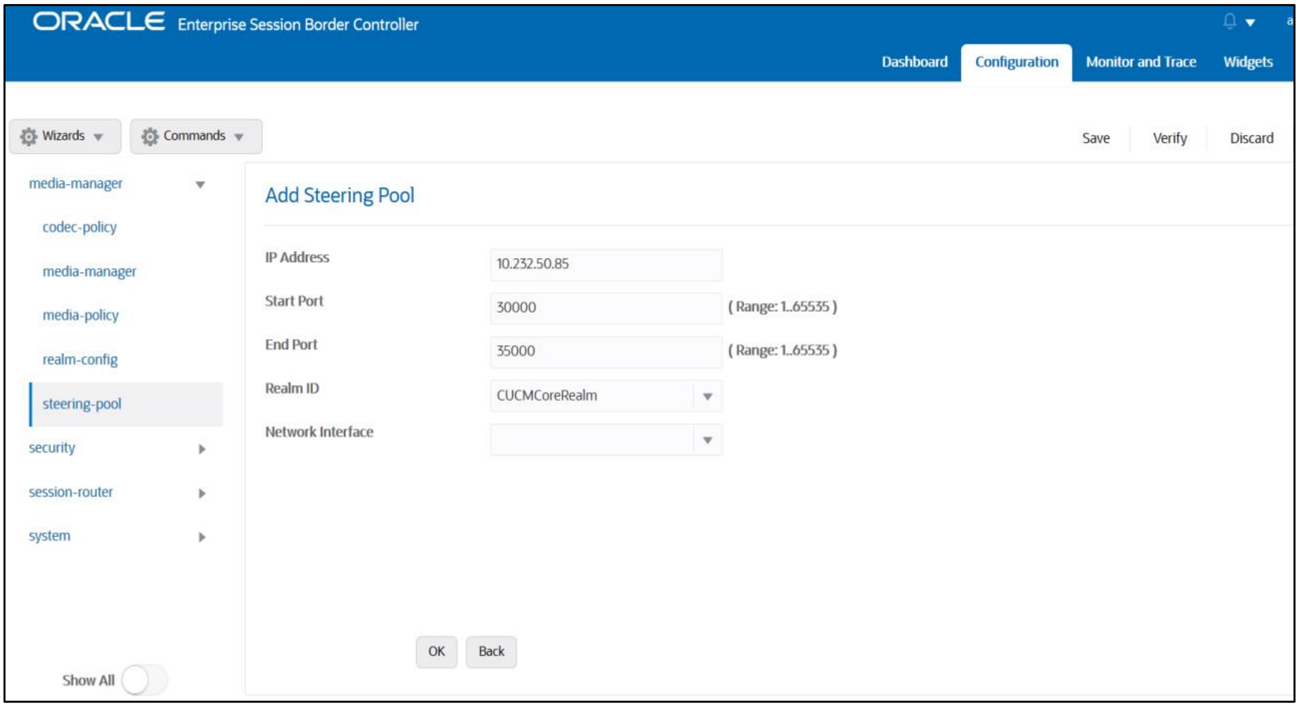
Cisco Access side steering pool.



Add Steering Pool

IP Address: 141.146.36.75
Start Port: 40000 (Range: 1..65535)
End Port: 49999 (Range: 1..65535)
Realm ID: CUCMpublicRealm
Network Interface:

Cisco Core side steering pool.



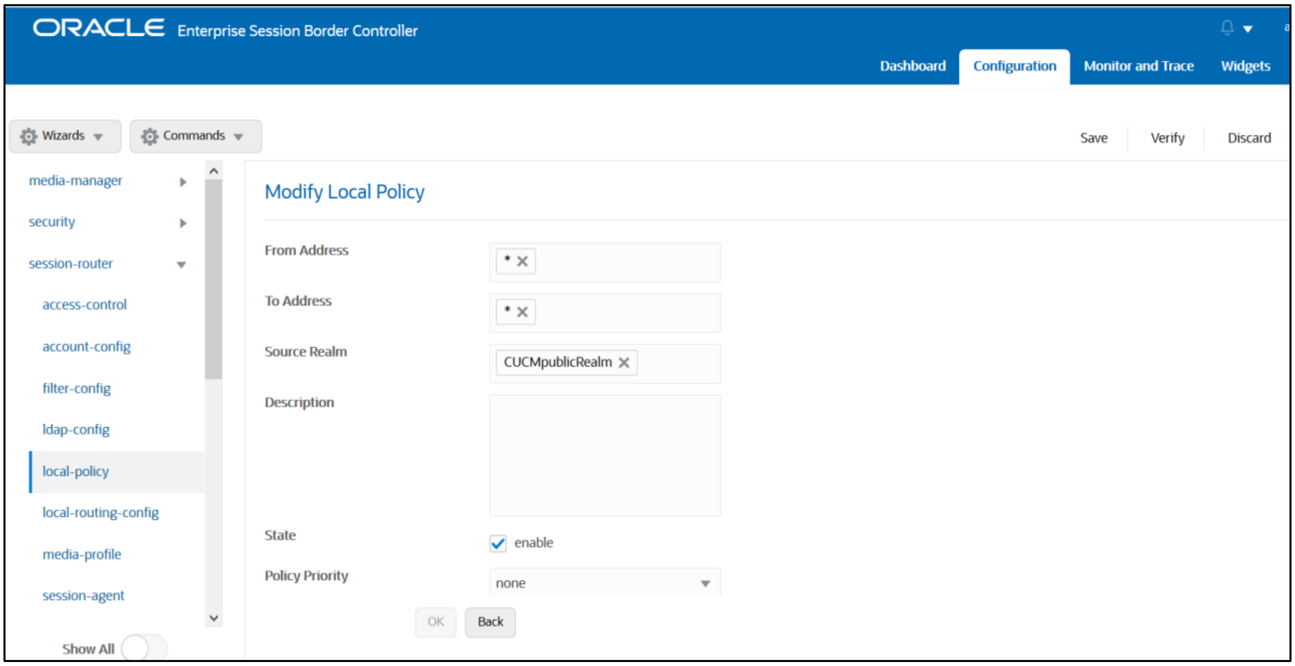
Add Steering Pool

IP Address: 10.232.50.85
Start Port: 30000 (Range: 1..65535)
End Port: 35000 (Range: 1..65535)
Realm ID: CUCMCoreRealm
Network Interface:

7.6. Configure local-policy (Optional)

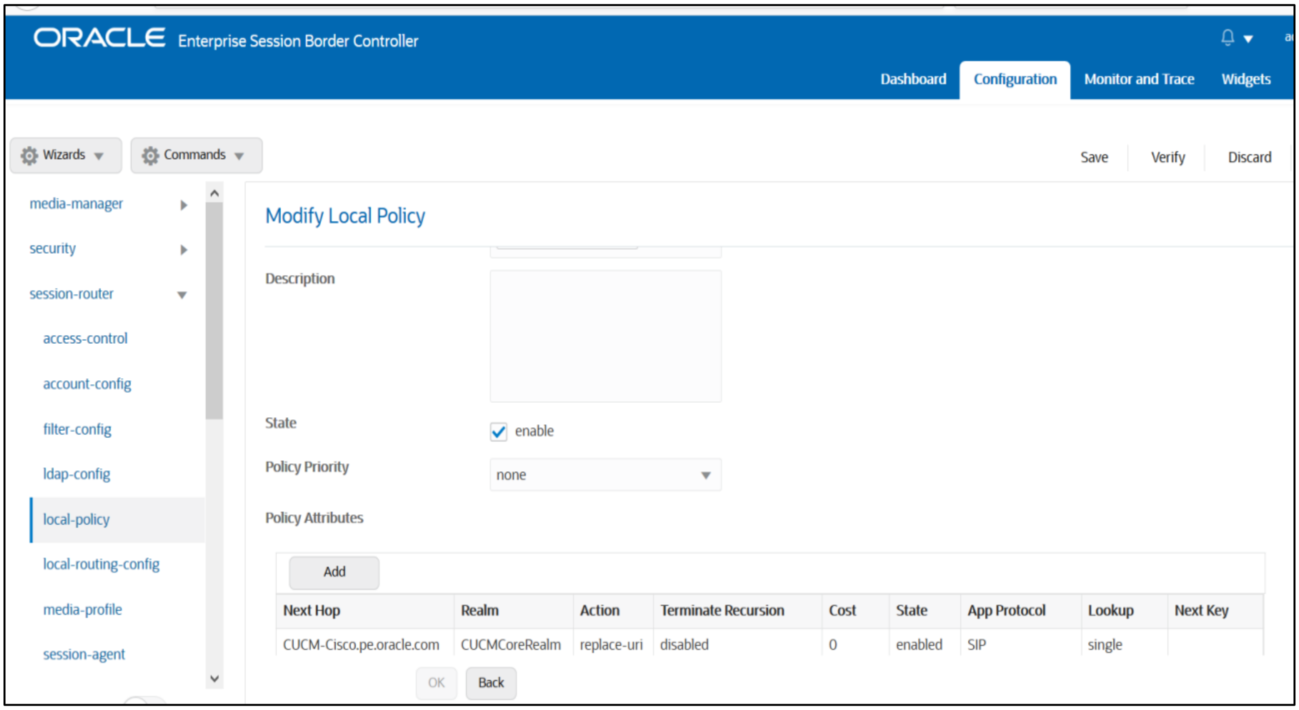
Local policy config allows for the SBC to route calls from one end of the network to the other based on routing criteria. To configure local-policy, go to Session-Router->local-policy.

To route the calls from Cisco Access side to Cisco Core side and vice versa, Use the below local –policy



Modify Local Policy

From Address	*
To Address	*
Source Realm	CUCMpublicRealm
Description	
State	☑ enable
Policy Priority	none



Modify Local Policy

Description	
State	☑ enable
Policy Priority	none

Policy Attributes

Next Hop	Realm	Action	Terminate Recursion	Cost	State	App Protocol	Lookup	Next Key
CUCM-Cisco.pe.oracle.com	CUCMCoreRealm	replace-uri	disabled	0	enabled	SIP	single	

Cisco Offer less INVITE can happen in the Remote worker scenarios too.
In that case, please set the parameter "**Add SDP Invite**" as both and "**Add SDP profiles**" under Cisco Access side sip-interface. The configuration is similar to what we have done in Sec 6.15

8. Existing SBC configuration

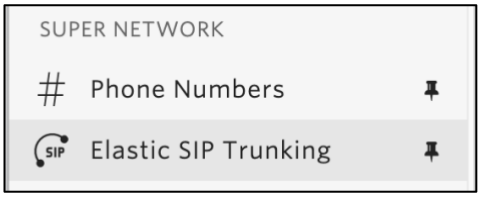
If the SBC being used is an existing SBC with functional configuration, following configuration elements are required:

- New realm-config
- Configuring a certificate for SBC Interface
- TLS-Profile
- New sip-interface
- New session-agent
- New steering-pools
- New local-policy
- SDES Profile
- Media-sec-Policy
- New Translation Rules
- Session Translation Rules

Please follow the steps mentioned in the above chapters to configure these elements.

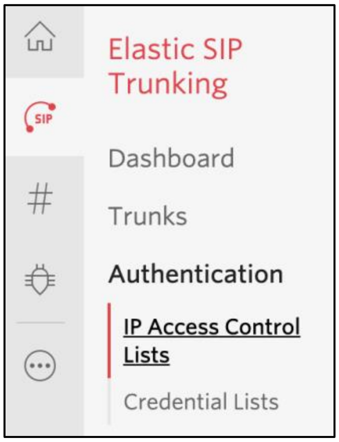
9. Twilio Elastic SIP Trunking Configuration

From your Twilio Console, navigate to the Elastic SIP Trunking area (or click on the SIP icon on the left vertical navigation bar).

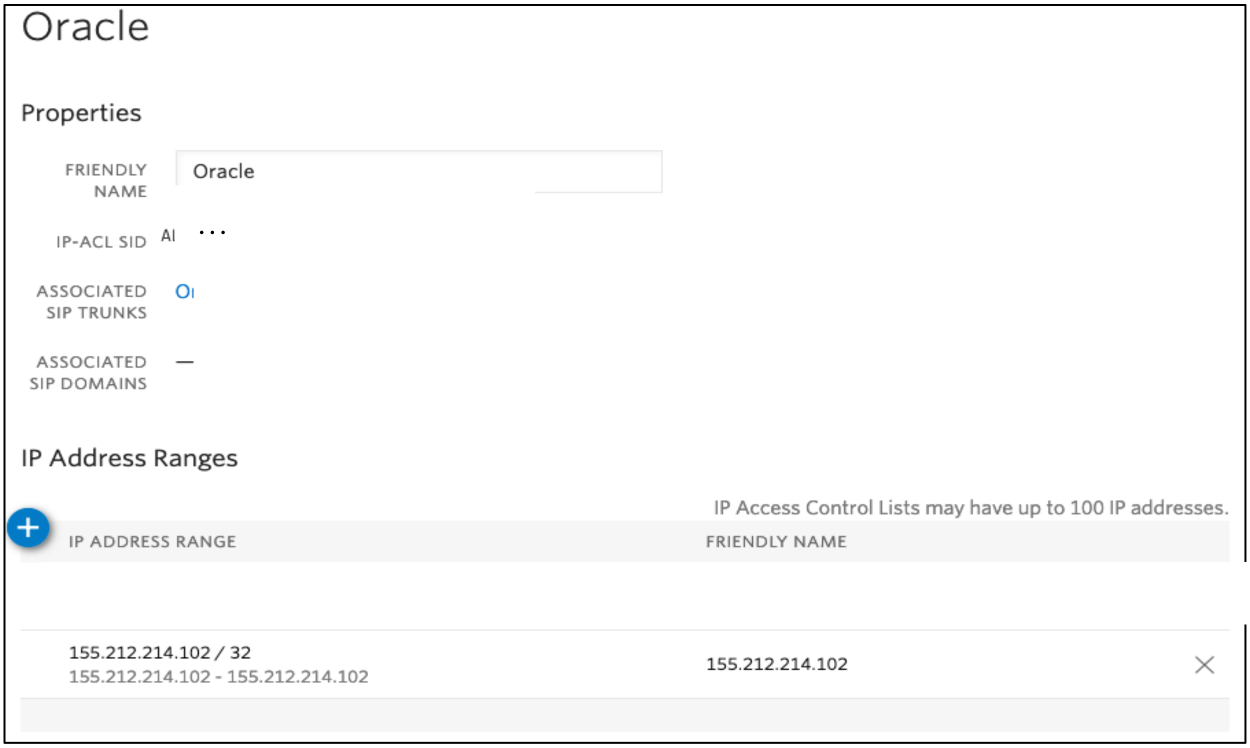


9.1. Create an IP-ACL rule

Click on Authentication in the left navigation, and then click on IP Access Control Lists.



Create a new IP-ACL, for example call it "Oracle" and add your SBCs IP addresses.



Oracle

Properties

FRIENDLY NAME	Oracle
IP-ACL SID	AI ...
ASSOCIATED SIP TRUNKS	Oı
ASSOCIATED SIP DOMAINS	—

IP Address Ranges

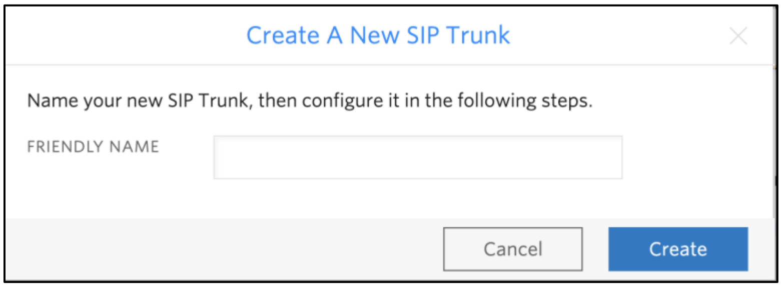
IP Access Control Lists may have up to 100 IP addresses.

IP ADDRESS RANGE	FRIENDLY NAME
155.212.214.102 / 32 155.212.214.102 - 155.212.214.102	155.212.214.102

9.2. Create a new Trunk

For each geographical region desired (e.g., North America, Europe), create a new Elastic SIP Trunk.

Now click on Trunks again on the left vertical navigation bar, and create a new Trunk.

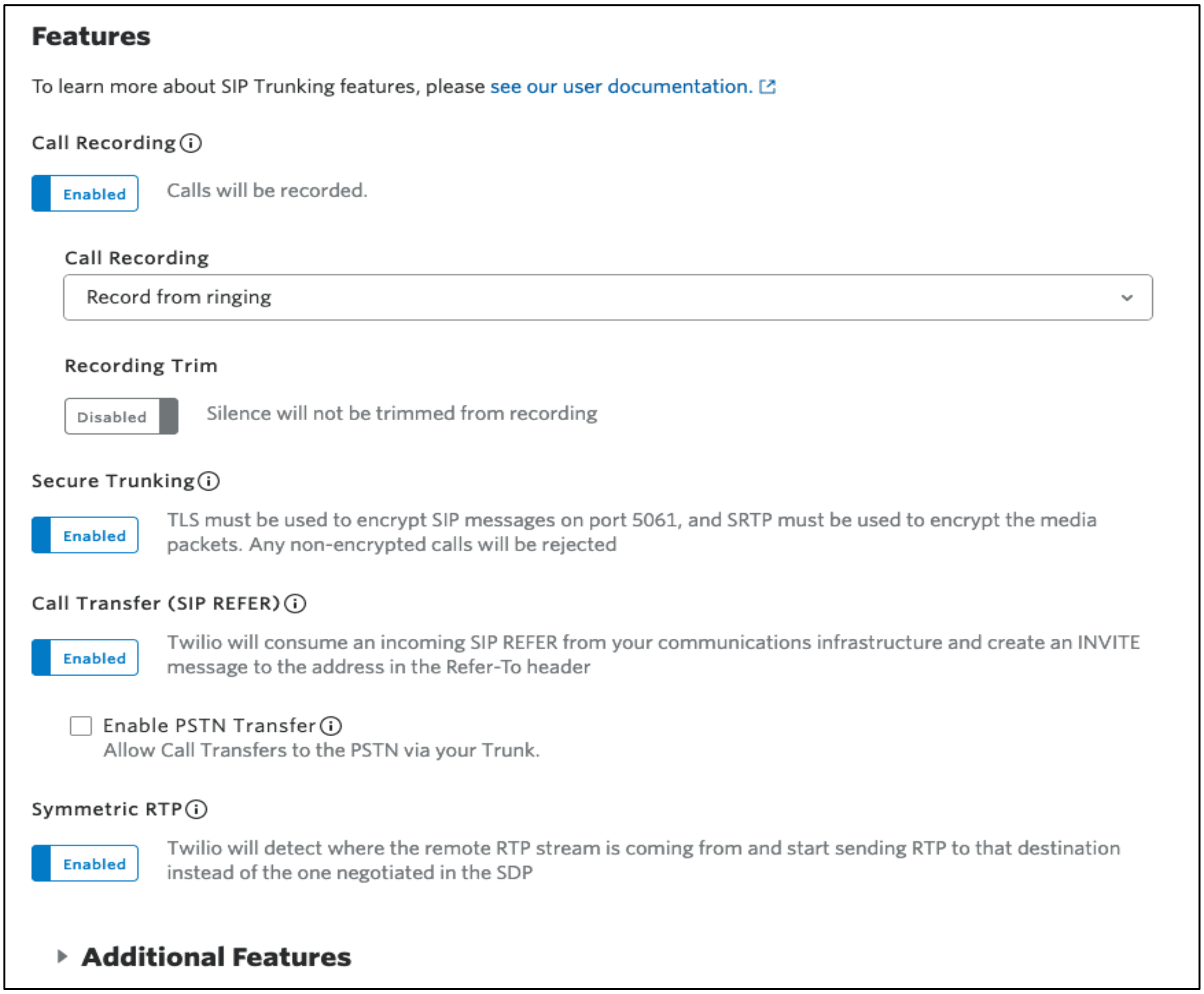


Create A New SIP Trunk

Name your new SIP Trunk, then configure it in the following steps.

FRIENDLY NAME

Under the **General Settings** you can enable different features as desired.



Features

To learn more about SIP Trunking features, please see our user documentation.

Call Recording ⓘ

Enabled — Calls will be recorded.

Call Recording: Record from ringing

Recording Trim

Disabled — Silence will not be trimmed from recording

Secure Trunking ⓘ

Enabled — TLS must be used to encrypt SIP messages on port 5061, and SRTP must be used to encrypt the media packets. Any non-encrypted calls will be rejected

Call Transfer (SIP REFER) ⓘ

Enabled — Twilio will consume an incoming SIP REFER from your communications infrastructure and create an INVITE message to the address in the Refer-To header

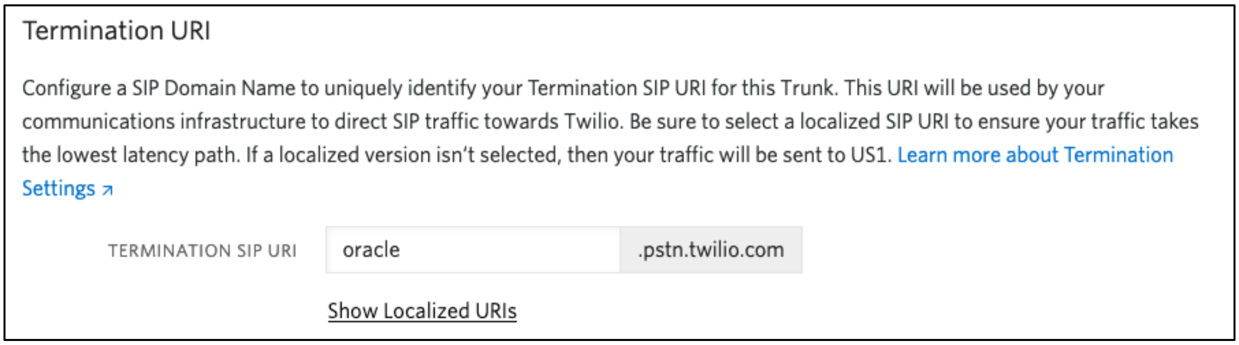
□ Enable PSTN Transfer ⓘ
Allow Call Transfers to the PSTN via your Trunk.

Symmetric RTP ⓘ

Enabled — Twilio will detect where the remote RTP stream is coming from and start sending RTP to that destination instead of the one negotiated in the SDP

▸ **Additional Features**

In the **Termination** section, select a Termination SIP URI.

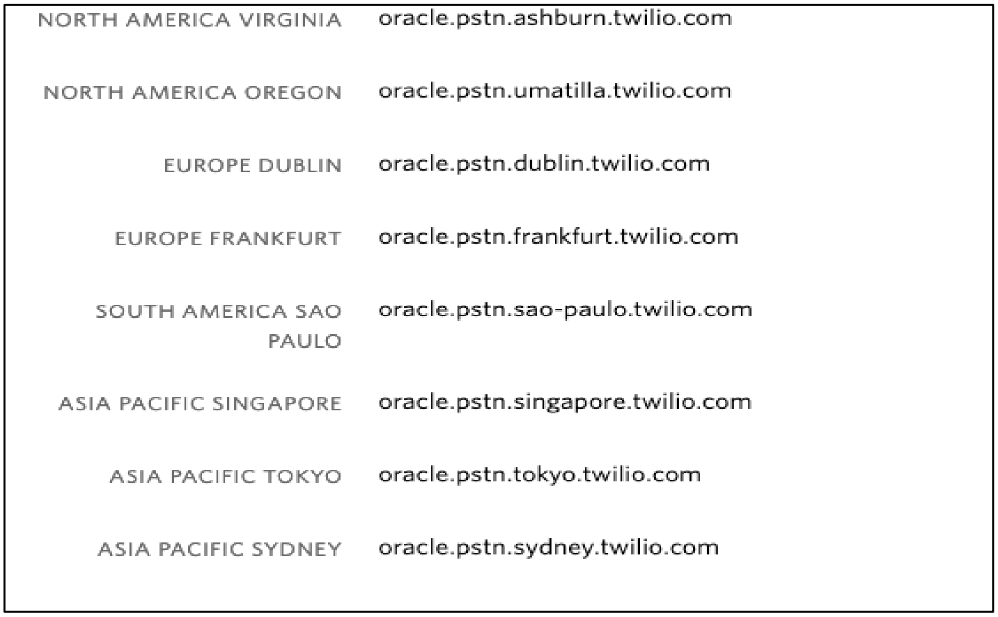


Termination URI

Configure a SIP Domain Name to uniquely identify your Termination SIP URI for this Trunk. This URI will be used by your communications infrastructure to direct SIP traffic towards Twilio. Be sure to select a localized SIP URI to ensure your traffic takes the lowest latency path. If a localized version isn't selected, then your traffic will be sent to US1. Learn more about Termination Settings ↗

TERMINATION SIP URI: oracle .pstn.twilio.com

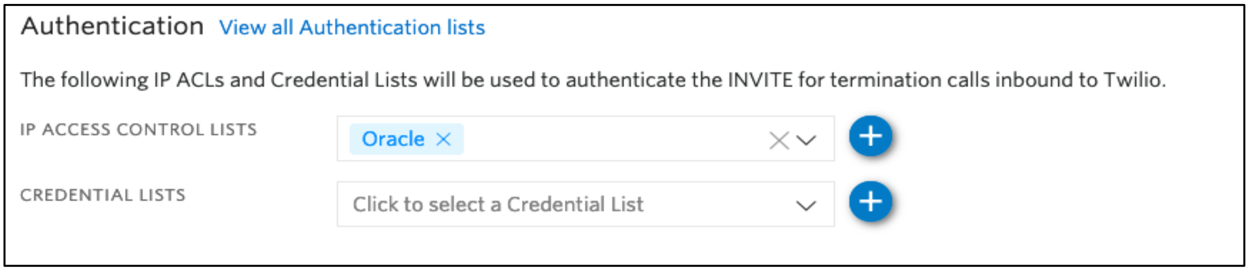
Click on "Show localized URI's" and copy and paste this information as you will use this on your SBC to configure your Trunk.



NORTH AMERICA VIRGINIA	oracle.pstn.ashburn.twilio.com
NORTH AMERICA OREGON	oracle.pstn.umatilla.twilio.com
EUROPE DUBLIN	oracle.pstn.dublin.twilio.com
EUROPE FRANKFURT	oracle.pstn.frankfurt.twilio.com
SOUTH AMERICA SAO PAULO	oracle.pstn.sao-paulo.twilio.com
ASIA PACIFIC SINGAPORE	oracle.pstn.singapore.twilio.com
ASIA PACIFIC TOKYO	oracle.pstn.tokyo.twilio.com
ASIA PACIFIC SYDNEY	oracle.pstn.sydney.twilio.com

or

Assign the IP ACL ("Oracle") that you created in the previous step.



Authentication View all Authentication lists

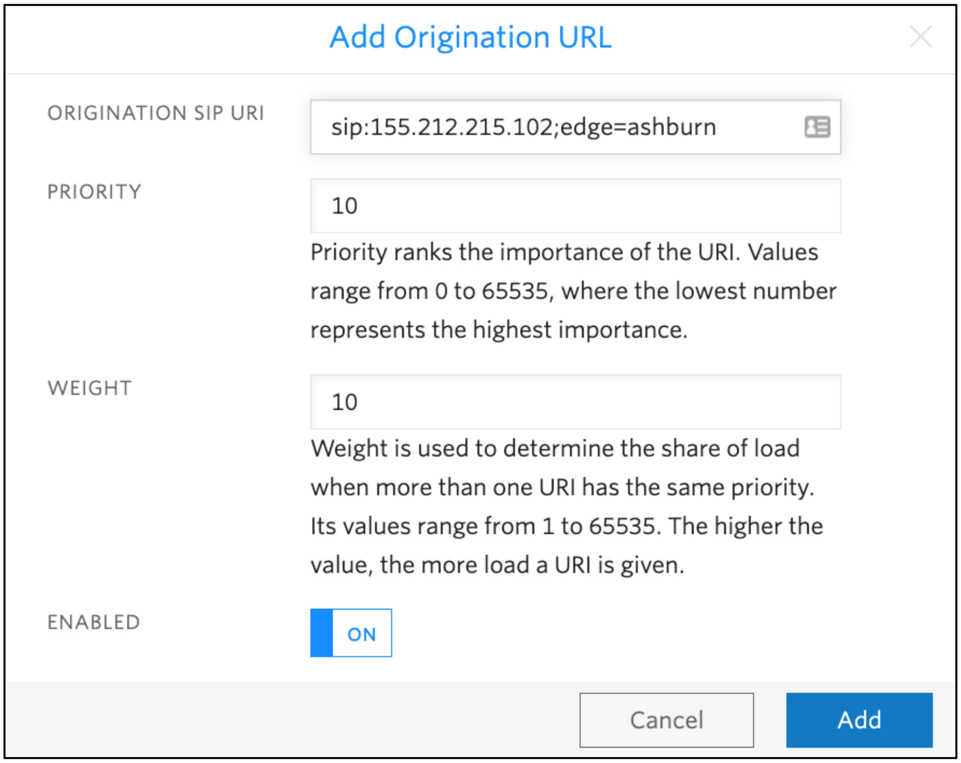
The following IP ACLs and Credential Lists will be used to authenticate the INVITE for termination calls inbound to Twilio.

IP ACCESS CONTROL LISTS: Oracle ×

CREDENTIAL LISTS: Click to select a Credential List

In the **Origination** section, we'll need to add Origination URI's to route traffic towards your Oracle SBC. The recommended practice is to configure a redundant mesh per geographic region (in this context a region is one of North America, Europe, etc.). In this case, we configure two Origination URIs, each egressing from a different Twilio Edge.

Click on ‘Add New Origination URI’, we'll depict the configuration for North America:



Add Origination URL

ORIGINATION SIP URI: sip:155.212.215.102;edge=ashburn

PRIORITY: 10

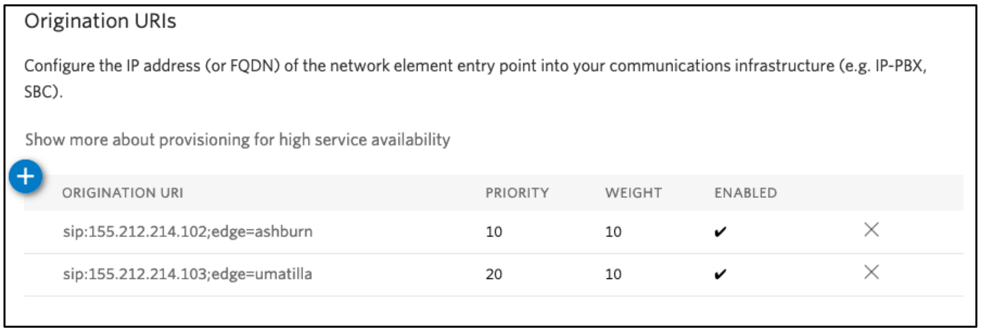
Priority ranks the importance of the URI. Values range from 0 to 65535, where the lowest number represents the highest importance.

WEIGHT: 10

Weight is used to determine the share of load when more than one URI has the same priority. Its values range from 1 to 65535. The higher the value, the more load a URI is given.

ENABLED: ON

Continue to add the other Origination URIs, so you have the following configuration:



Origination URIs

Configure the IP address (or FQDN) of the network element entry point into your communications infrastructure (e.g. IP-PBX, SBC).

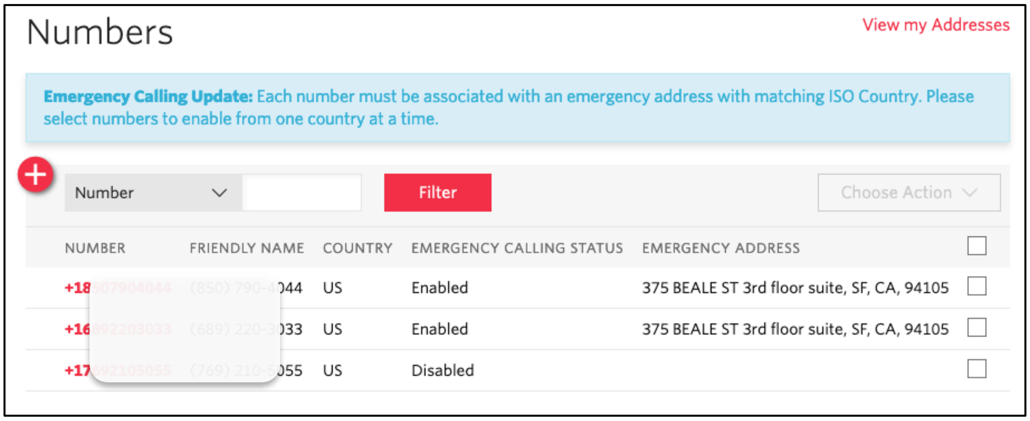
Show more about provisioning for high service availability

ORIGINATION URI	PRIORITY	WEIGHT	ENABLED	
sip:155.212.214.102;edge=ashburn	10	10	✔	×
sip:155.212.214.103;edge=umatilla	20	10	✔	×

In this example, Origination traffic is first routed via Twilio’s Ashburn edge, if that fails then we'll route from Twilio’s Umatilla edge.

9.3. Associate Phone Numbers on your Trunk

In the **Numbers** section of your Trunk, add the Phone Numbers that you want to associate with each Trunk. Remember to associate the Numbers from a given country in the right Trunk. For example, associate US & Canada Numbers with the North American Trunk and European Numbers with the European Trunk etc.



Numbers

View my Addresses

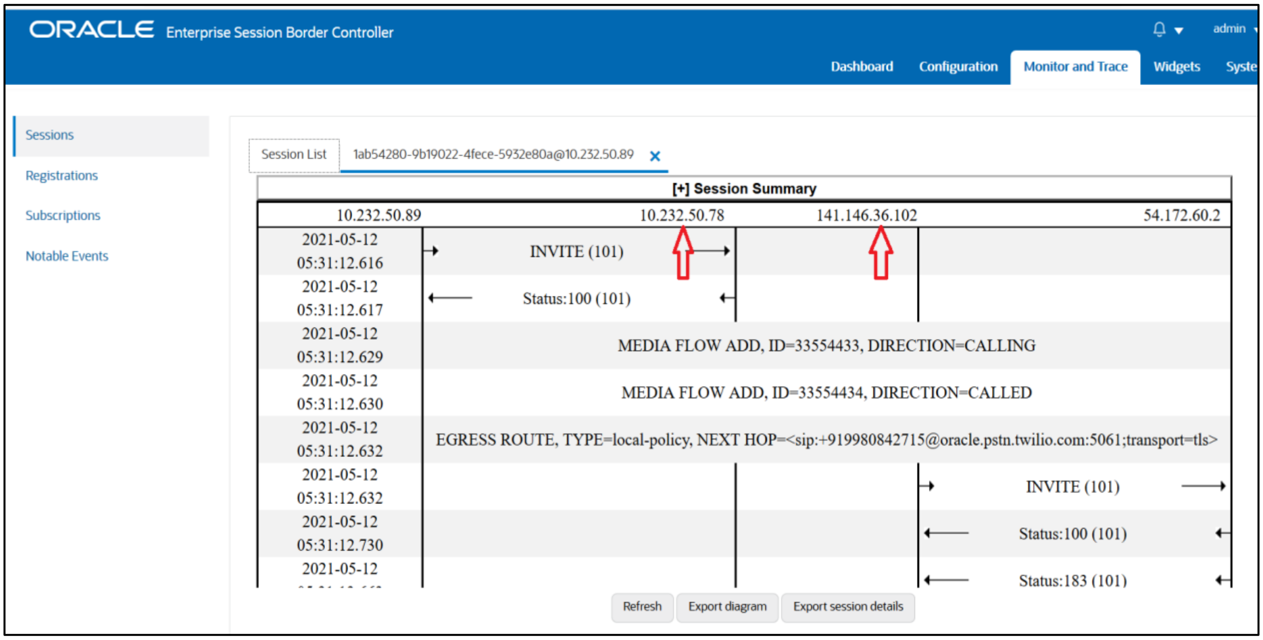
Emergency Calling Update: Each number must be associated with an emergency address with matching ISO Country. Please select numbers to enable from one country at a time.

NUMBER	FRIENDLY NAME	COUNTRY	EMERGENCY CALLING STATUS	EMERGENCY ADDRESS	□
+18[redacted]	[redacted]44	US	Enabled	375 BEALE ST 3rd floor suite, SF, CA, 94105	□
+16[redacted]	[redacted]33	US	Enabled	375 BEALE ST 3rd floor suite, SF, CA, 94105	□
+17[redacted]	[redacted]55	US	Disabled		□

10. Verification of Sample Call flows

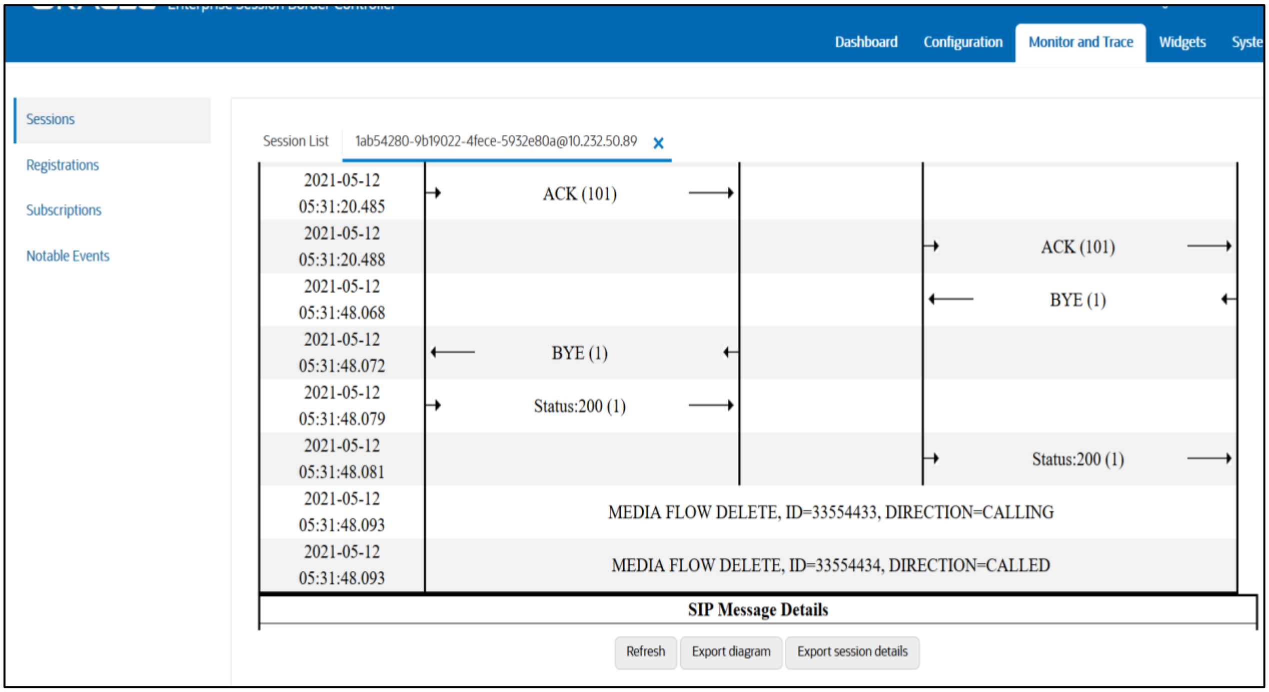
Once the configuration is complete, we can try making sample calls and can check the signaling path between Twilio Elastic Sip Trunk (PSTN Users) and Cisco Users

1. Make Call from Cisco user to the Twilio Elastic Sip Trunk and check the call flow.
   The calls flow from 10.232.50.78 (Cisco SIP Interface) to 141.146.36.102 (Twilio Elastic SIP Trunking Interface) and to Twilio Session Agent and the call reaches the PSTN user after that.



[+] Session Summary

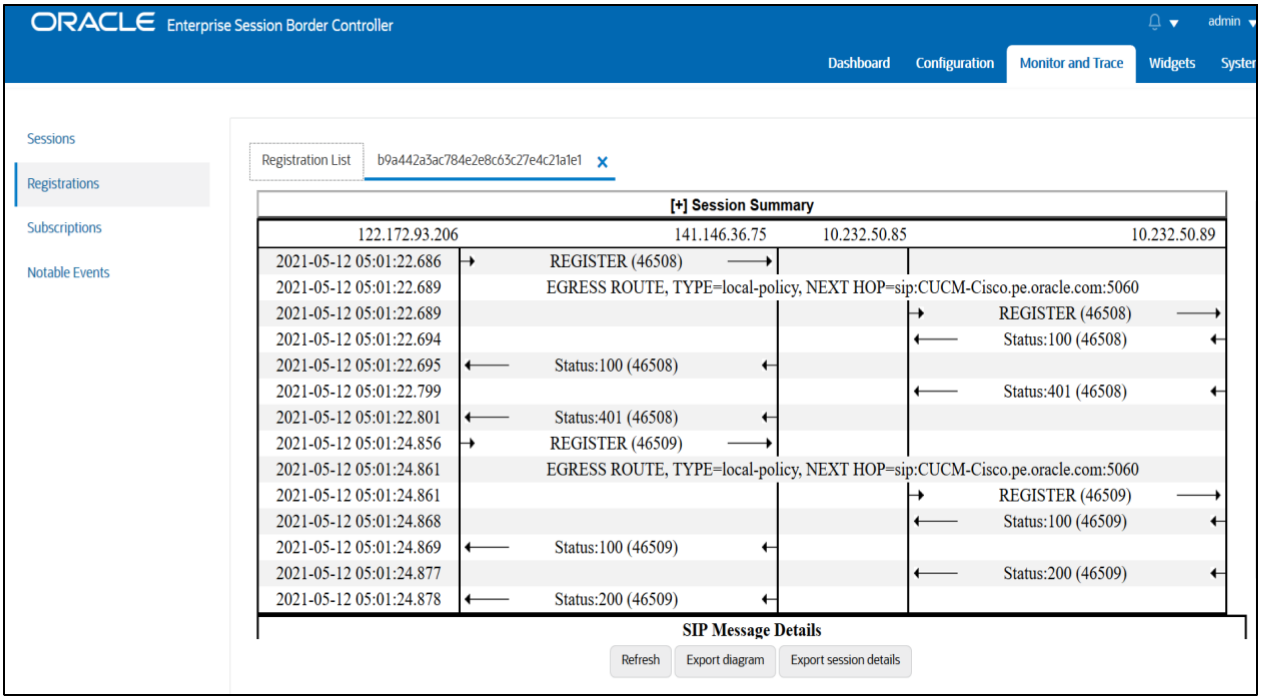
	10.232.50.89	10.232.50.78	141.146.36.102	54.172.60.2
2021-05-12 05:31:12.616	INVITE (101) →			
2021-05-12 05:31:12.617	← Status:100 (101)			
2021-05-12 05:31:12.629	MEDIA FLOW ADD, ID=33554433, DIRECTION=CALLING			
2021-05-12 05:31:12.630	MEDIA FLOW ADD, ID=33554434, DIRECTION=CALLED			
2021-05-12 05:31:12.632	EGRESS ROUTE, TYPE=local-policy, NEXT HOP=<sip:+919980842715@oracle.pstn.twilio.com:5061;transport=tls>			
2021-05-12 05:31:12.632			INVITE (101) →	
2021-05-12 05:31:12.730			← Status:100 (101)	
2021-05-12			← Status:183 (101)	



2021-05-12 05:31:20.485	ACK (101) →		
2021-05-12 05:31:20.488			ACK (101) →
2021-05-12 05:31:48.068			← BYE (1)
2021-05-12 05:31:48.072	← BYE (1)		
2021-05-12 05:31:48.079	Status:200 (1) →		
2021-05-12 05:31:48.081			Status:200 (1) →
2021-05-12 05:31:48.093	MEDIA FLOW DELETE, ID=33554433, DIRECTION=CALLING		
2021-05-12 05:31:48.093	MEDIA FLOW DELETE, ID=33554434, DIRECTION=CALLED		

SIP Message Details

2. When we register Cisco Remote Worker, we can see the registration happening through Oracle SBC to Cisco CUCM as given below.

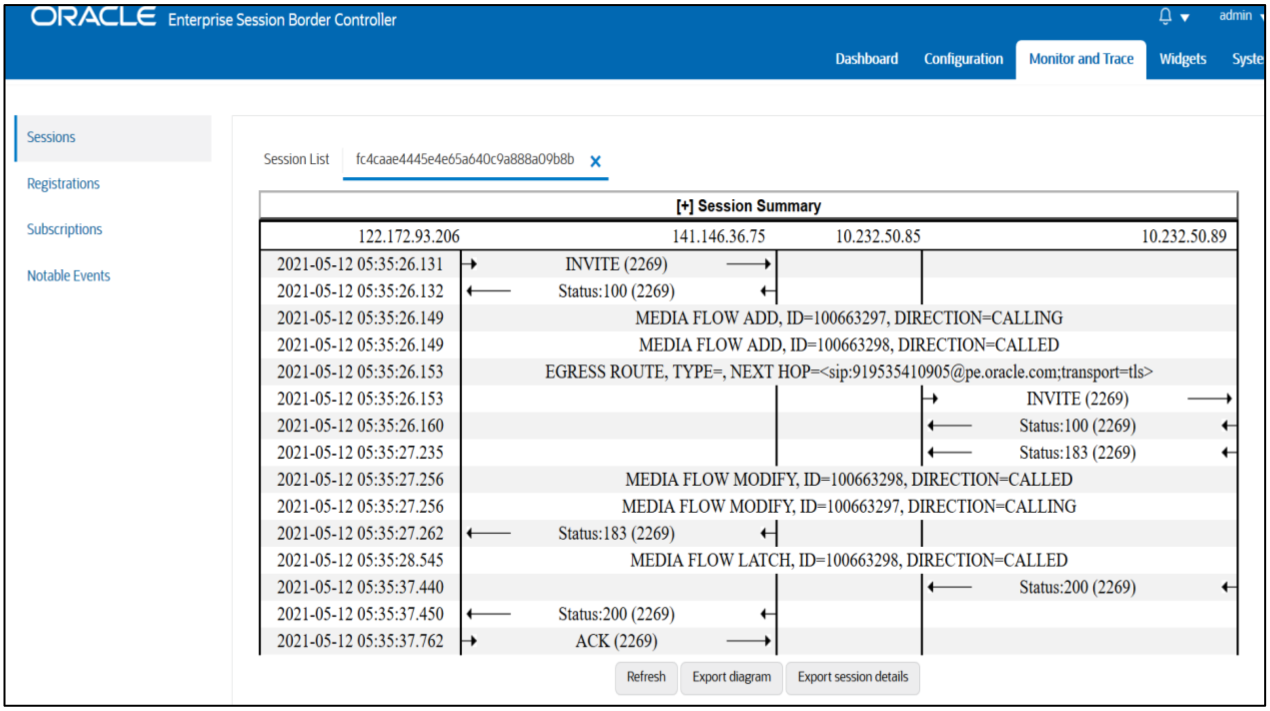


[+] Session Summary

	122.172.93.206	141.146.36.75	10.232.50.85	10.232.50.89
2021-05-12 05:01:22.686	→ REGISTER (46508) →			
2021-05-12 05:01:22.689	EGRESS ROUTE, TYPE=local-policy, NEXT HOP=sip:CUCM-Cisco.pe.oracle.com:5060			
2021-05-12 05:01:22.689			→ REGISTER (46508) →	
2021-05-12 05:01:22.694			← Status:100 (46508) ←	
2021-05-12 05:01:22.695	← Status:100 (46508) ←			
2021-05-12 05:01:22.799			← Status:401 (46508) ←	
2021-05-12 05:01:22.801	← Status:401 (46508) ←			
2021-05-12 05:01:24.856	→ REGISTER (46509) →			
2021-05-12 05:01:24.861	EGRESS ROUTE, TYPE=local-policy, NEXT HOP=sip:CUCM-Cisco.pe.oracle.com:5060			
2021-05-12 05:01:24.861			→ REGISTER (46509) →	
2021-05-12 05:01:24.868			← Status:100 (46509) ←	
2021-05-12 05:01:24.869	← Status:100 (46509) ←			
2021-05-12 05:01:24.877			← Status:200 (46509) ←	
2021-05-12 05:01:24.878	← Status:200 (46509) ←			

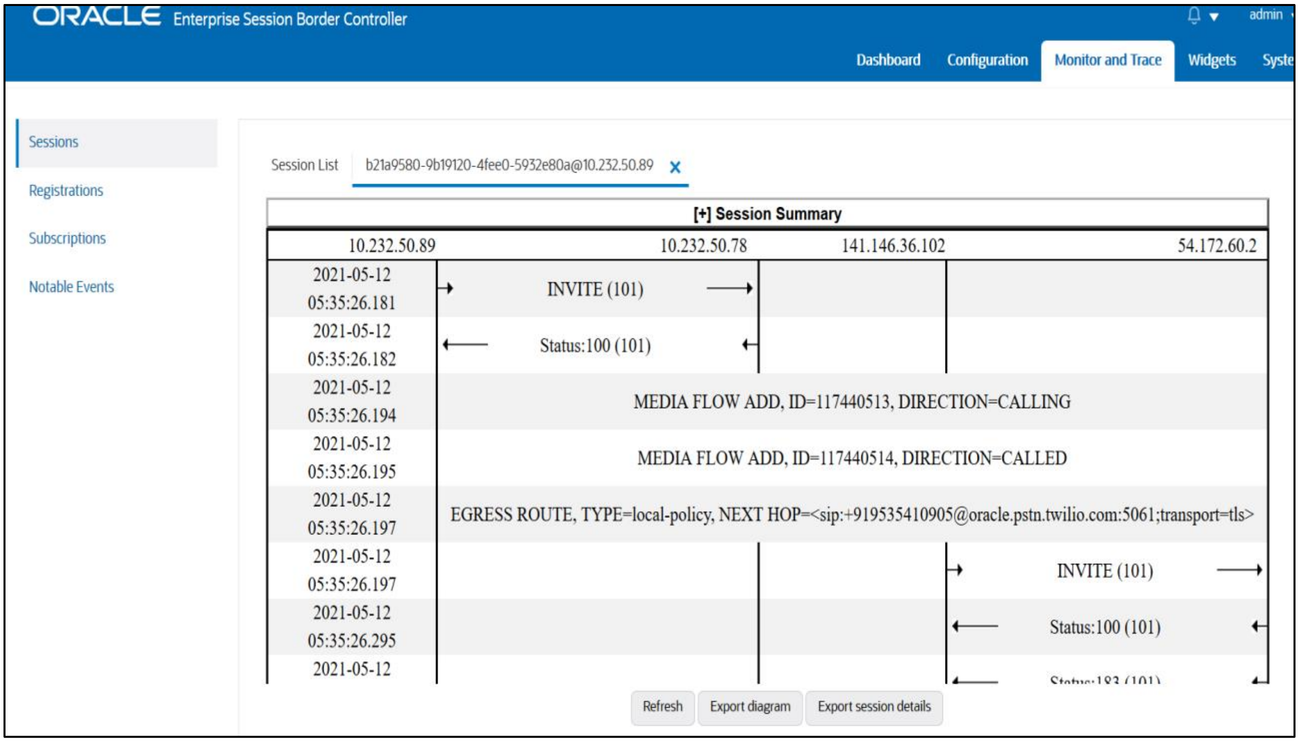
SIP Message Details

3. Make Call from Cisco Remote user to the Twilio Elastic Sip Trunk user and check the call flow. Now, there will be 2 call legs (hair pinned call) as the call reaches Cisco CUCM first and then reaches Twilio trunk user after that as given below.



[+] Session Summary

	122.172.93.206	141.146.36.75	10.232.50.85	10.232.50.89
2021-05-12 05:35:26.131	→ INVITE (2269) →			
2021-05-12 05:35:26.132	← Status:100 (2269) ←			
2021-05-12 05:35:26.149	MEDIA FLOW ADD, ID=100663297, DIRECTION=CALLING			
2021-05-12 05:35:26.149	MEDIA FLOW ADD, ID=100663298, DIRECTION=CALLED			
2021-05-12 05:35:26.153	EGRESS ROUTE, TYPE=, NEXT HOP=<sip:919535410905@pe.oracle.com;transport=tls>			
2021-05-12 05:35:26.153			→ INVITE (2269) →	
2021-05-12 05:35:26.160			← Status:100 (2269) ←	
2021-05-12 05:35:27.235			← Status:183 (2269) ←	
2021-05-12 05:35:27.256	MEDIA FLOW MODIFY, ID=100663298, DIRECTION=CALLED			
2021-05-12 05:35:27.256	MEDIA FLOW MODIFY, ID=100663297, DIRECTION=CALLING			
2021-05-12 05:35:27.262	← Status:183 (2269) ←			
2021-05-12 05:35:28.545	MEDIA FLOW LATCH, ID=100663298, DIRECTION=CALLED			
2021-05-12 05:35:37.440			← Status:200 (2269) ←	
2021-05-12 05:35:37.450	← Status:200 (2269) ←			
2021-05-12 05:35:37.762	→ ACK (2269) →			

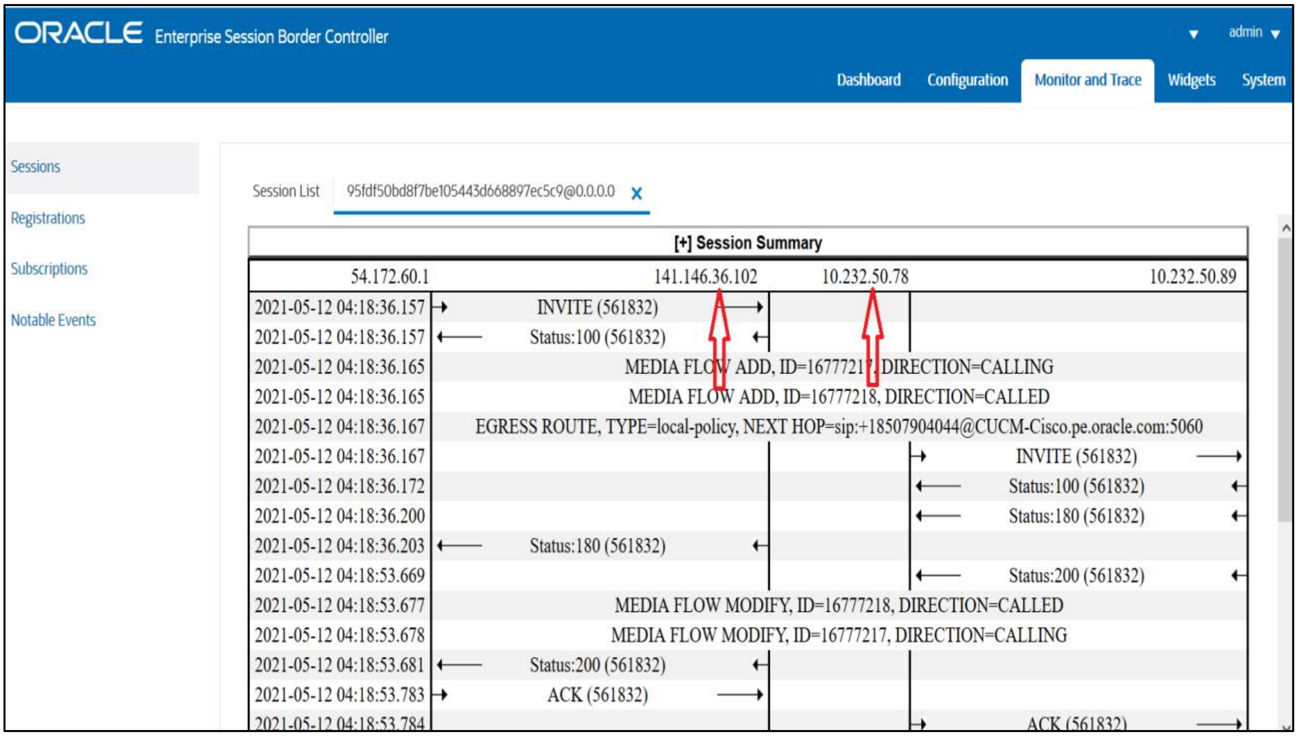


Session List b21a9580-9b19120-4fee0-5932e80a@10.232.50.89

[+] Session Summary

	10.232.50.89	10.232.50.78	141.146.36.102	54.172.60.2
2021-05-12 05:35:26.181	INVITE (101) →			
2021-05-12 05:35:26.182	← Status:100 (101)			
2021-05-12 05:35:26.194	MEDIA FLOW ADD, ID=117440513, DIRECTION=CALLING			
2021-05-12 05:35:26.195	MEDIA FLOW ADD, ID=117440514, DIRECTION=CALLED			
2021-05-12 05:35:26.197	EGRESS ROUTE, TYPE=local-policy, NEXT HOP=<sip:+919535410905@oracle.pstn.twilio.com:5061;transport=tls>			
2021-05-12 05:35:26.197			INVITE (101) →	
2021-05-12 05:35:26.295			← Status:100 (101)	
2021-05-12			← Status:183 (101)	

4. Make Call from the Twilio Elastic Sip Trunk to Cisco User and check the call flow.
   The calls flow from 141.146.36.102 (Twilio Elastic SIP Trunking Interface) to 10.232.50.78 (Cisco SIP Interface) and the call reaches the Cisco user after that.

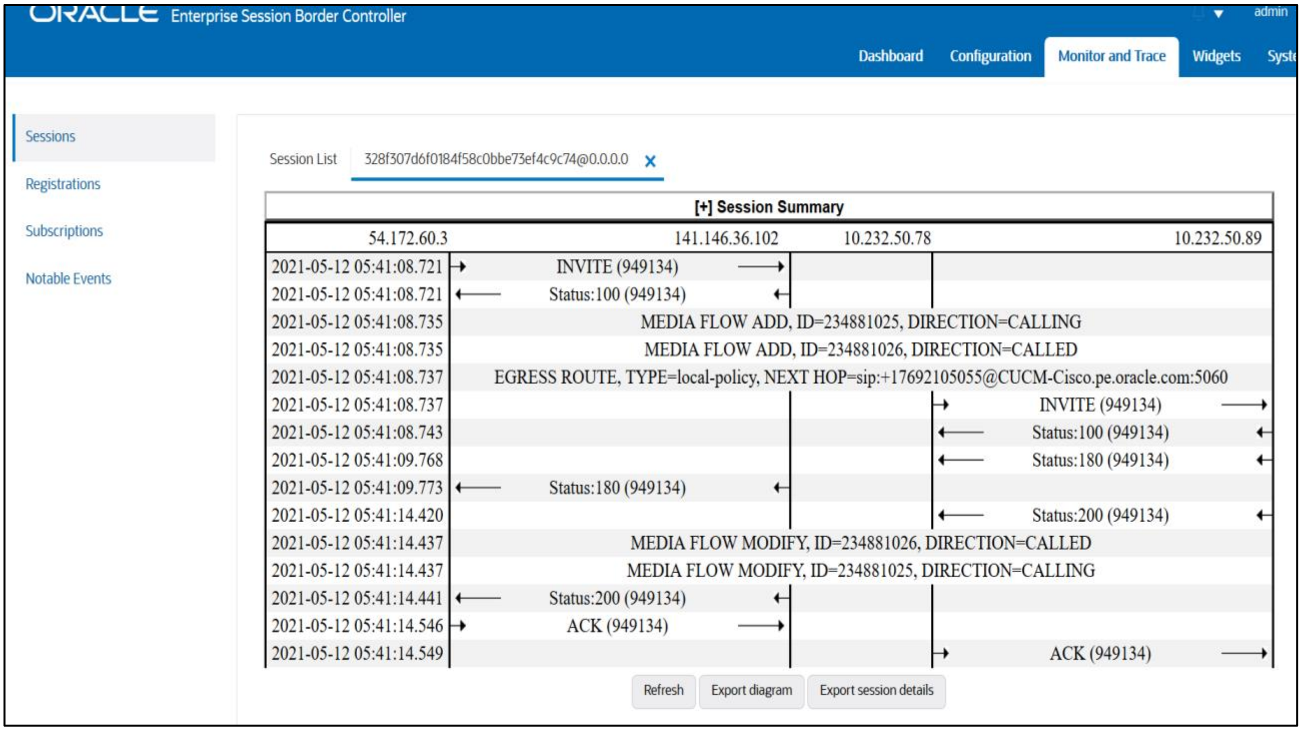


Session List 95fdf50bd8f7be105443d668897ec5c9@0.0.0.0

[+] Session Summary

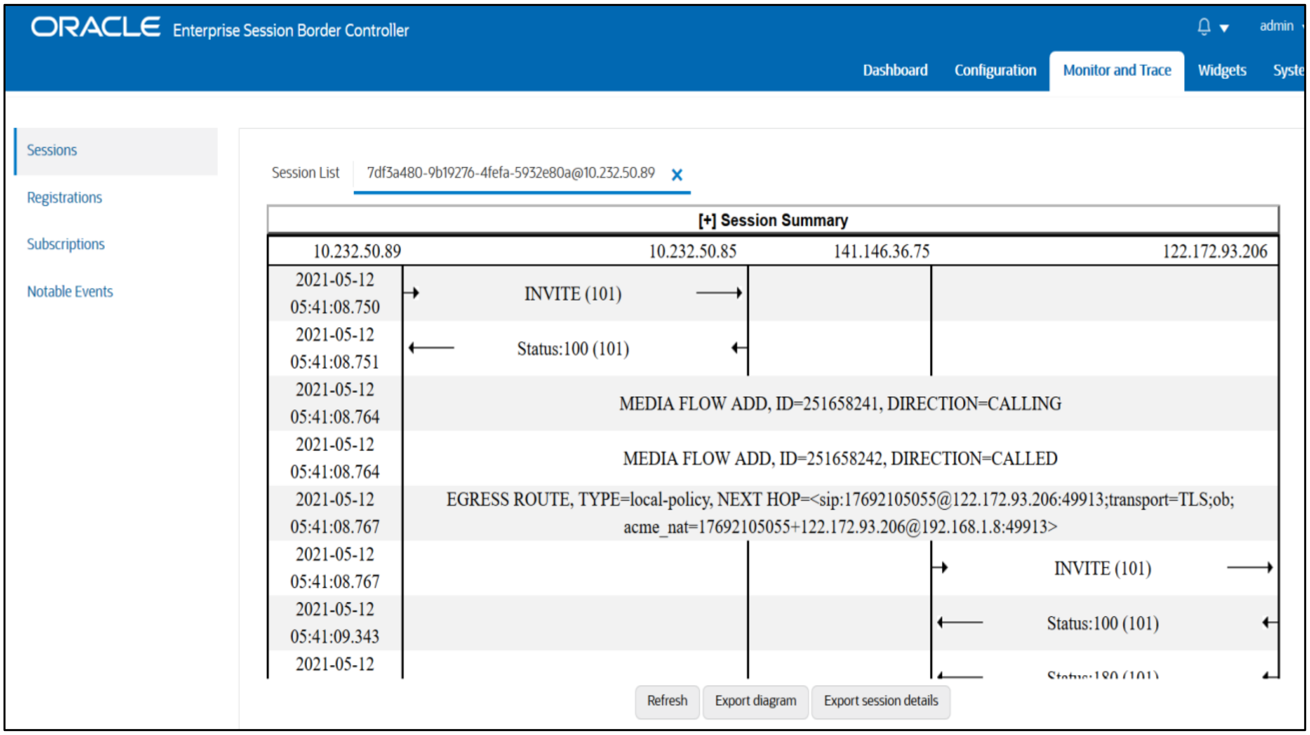
	54.172.60.1	141.146.36.102	10.232.50.78	10.232.50.89
2021-05-12 04:18:36.157	INVITE (561832) →			
2021-05-12 04:18:36.157	← Status:100 (561832)			
2021-05-12 04:18:36.165	MEDIA FLOW ADD, ID=16777217, DIRECTION=CALLING			
2021-05-12 04:18:36.165	MEDIA FLOW ADD, ID=16777218, DIRECTION=CALLED			
2021-05-12 04:18:36.167	EGRESS ROUTE, TYPE=local-policy, NEXT HOP=sip:+18507904044@CUCM-Cisco.pe.oracle.com:5060			
2021-05-12 04:18:36.167			INVITE (561832) →	
2021-05-12 04:18:36.172			← Status:100 (561832)	
2021-05-12 04:18:36.200			← Status:180 (561832)	
2021-05-12 04:18:36.203	← Status:180 (561832)			
2021-05-12 04:18:53.669			← Status:200 (561832)	
2021-05-12 04:18:53.677	MEDIA FLOW MODIFY, ID=16777218, DIRECTION=CALLED			
2021-05-12 04:18:53.678	MEDIA FLOW MODIFY, ID=16777217, DIRECTION=CALLING			
2021-05-12 04:18:53.681	← Status:200 (561832)			
2021-05-12 04:18:53.783	ACK (561832) →			
2021-05-12 04:18:53.784			ACK (561832) →	

5. Make Call from Twilio Elastic Sip Trunk user to Cisco Remote user and check the call flow. Now, there will be 2 call legs (hair pinned call) as the call reaches Cisco CUCM first and then reaches Cisco Remote user after that as given below.



[+] Session Summary

```
                        54.172.60.3          141.146.36.102   10.232.50.78                 10.232.50.89
2021-05-12 05:41:08.721 →      INVITE (949134)      →
2021-05-12 05:41:08.721 ←      Status:100 (949134)  ←
2021-05-12 05:41:08.735         MEDIA FLOW ADD, ID=234881025, DIRECTION=CALLING
2021-05-12 05:41:08.735         MEDIA FLOW ADD, ID=234881026, DIRECTION=CALLED
2021-05-12 05:41:08.737  EGRESS ROUTE, TYPE=local-policy, NEXT HOP=sip:+17692105055@CUCM-Cisco.pe.oracle.com:5060
2021-05-12 05:41:08.737                                        →    INVITE (949134)      →
2021-05-12 05:41:08.743                                        ←    Status:100 (949134)  ←
2021-05-12 05:41:09.768                                        ←    Status:180 (949134)  ←
2021-05-12 05:41:09.773 ←      Status:180 (949134)  ←
2021-05-12 05:41:14.420                                        ←    Status:200 (949134)  ←
2021-05-12 05:41:14.437         MEDIA FLOW MODIFY, ID=234881026, DIRECTION=CALLED
2021-05-12 05:41:14.437         MEDIA FLOW MODIFY, ID=234881025, DIRECTION=CALLING
2021-05-12 05:41:14.441 ←      Status:200 (949134)  ←
2021-05-12 05:41:14.546 →      ACK (949134)         →
2021-05-12 05:41:14.549                                        →    ACK (949134)         →
```



[+] Session Summary

```
                        10.232.50.89          10.232.50.85   141.146.36.75               122.172.93.206
2021-05-12 05:41:08.750 →      INVITE (101)        →
2021-05-12 05:41:08.751 ←      Status:100 (101)    ←
2021-05-12 05:41:08.764         MEDIA FLOW ADD, ID=251658241, DIRECTION=CALLING
2021-05-12 05:41:08.764         MEDIA FLOW ADD, ID=251658242, DIRECTION=CALLED
2021-05-12 05:41:08.767  EGRESS ROUTE, TYPE=local-policy, NEXT HOP=<sip:17692105055@122.172.93.206:49913;transport=TLS;ob;
                         acme_nat=17692105055+122.172.93.206@192.168.1.8:49913>
2021-05-12 05:41:08.767                                       →    INVITE (101)        →
2021-05-12 05:41:09.343                                       ←    Status:100 (101)    ←
2021-05-12                                                    ←    Status:180 (101)    ←
```

Appendix A

Following are the test cases that are executed between Cisco User with the Twilio Elastic SIP Trunk (PSTN user). **Please note that Cisco User here refers both Cisco User inside Enterprise network as well as Cisco Remote worker.**

Serial Number	Test Cases Executed	Result
1	Cisco user disconnects an inbound connected call	Pass
2	Cisco user disconnects an outbound connected call	Pass
3	Twilio Elastic SIP Trunk user disconnects an inbound connected call	Pass
4	Twilio Elastic SIP Trunk User disconnects an outbound connected call	Pass
5	Cisco user places inbound call from Twilio Elastic SIP Trunk user on hold and then resumes	Pass
6	Cisco user makes outbound call to Twilio Elastic SIP Trunk user and put that call on hold and then resumes	Pass
7	Twilio Elastic SIP Trunk user places inbound call from Cisco user on hold and then resumes	Pass
8	Twilio Elastic SIP Trunk user makes outbound call to Cisco user and put that call on hold and then resumes	Pass
9	Cisco user places inbound call from Twilio Elastic SIP Trunk user on hold for over 15/30 minutes and then resumes	Pass
10	Cisco user makes outbound call to Twilio Elastic SIP Trunk user and places the call on hold for over 15/30 minutes and then resumes	Pass
11	Inbound Twilio Elastic SIP Trunk call to Cisco blind transferred to second Cisco/ PSTN User	Pass
12	Outbound Twilio Elastic SIP Trunk call from Cisco user blind transferred to second Cisco/ PSTN User	Pass
13	Inbound Twilio Elastic SIP Trunk Call to Cisco consultatively transferred to Cisco/ PSTN User	Pass
14	Outbound Twilio Elastic SIP Trunk call from Cisco user consultatively transferred to Cisco/ PSTN User	Pass
15	Cisco user makes outbound call to Twilio Elastic SIP Trunk user and makes a conference call by adding another Cisco/ PSTN user.	Pass

16	Twilio Elastic SIP Trunk user makes outbound call to Cisco user and Cisco user makes a conference call by adding another Cisco/ PSTN user.	Pass
17	Cisco user mutes inbound call from Twilio Elastic SIP Trunk user and then unmutes	Pass
18	Cisco user mutes outbound call made to Twilio Elastic SIP Trunk user and then unmutes	Pass
19	Twilio Elastic SIP Trunk user mutes inbound call from Cisco user and then unmutes	Pass
20	Twilio Elastic SIP Trunk user mutes outbound call made to Cisco user and then unmutes	Pass
21	Twilio Elastic SIP Trunk User disconnects outbound call to Cisco user before it is answered	Pass
22	Cisco user disconnects outbound call to Twilio Elastic SIP Trunk user before it is answered	Pass

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