Configuring Oracle SBC with Genesys SIP Server

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		06/04/20

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 Session Border Controller (SBC). It is assumed that the reader is familiar with basic operations of the Oracle Session Border Controller.

2. Document Overview

In this document we will provide the steps to navigate the Oracle SBC configuration and to configure relevant sections through the webGUI interface.

Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP is necessary to be able to utilize the document in the intended manner.

SIP Server is the Genesys software component that provides an interface between your telephony hardware and the rest of the Genesys software components in your enterprise. It translates and keeps track of events and requests that come from and are sent to the telephony device.
SIP Server is a TCP/IP-based server that can also act as a messaging interface between SIP Server clients. It is the critical point in allowing your Genesys solution to facilitate and track the contacts that flow through your enterprise. This reduces the cost and complexity of extending an enterprise's telephony system outside its network borders.
Oracle Session Border Controllers (Oracle SBCs) play an important role in SIP trunking as they are used by many ITSPs and Enterprises as part of their SIP trunking infrastructure.

This application note has been prepared as a means of ensuring that SIP trunking between Genesys SIP Server, Oracle SBCs and IP Trunking services are configured in the optimal manner.

3. Introduction

3.1. Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Session Border Controller and the Genesys SIP Server. There will be steps that require navigating the Oracle SBC WebGUI.

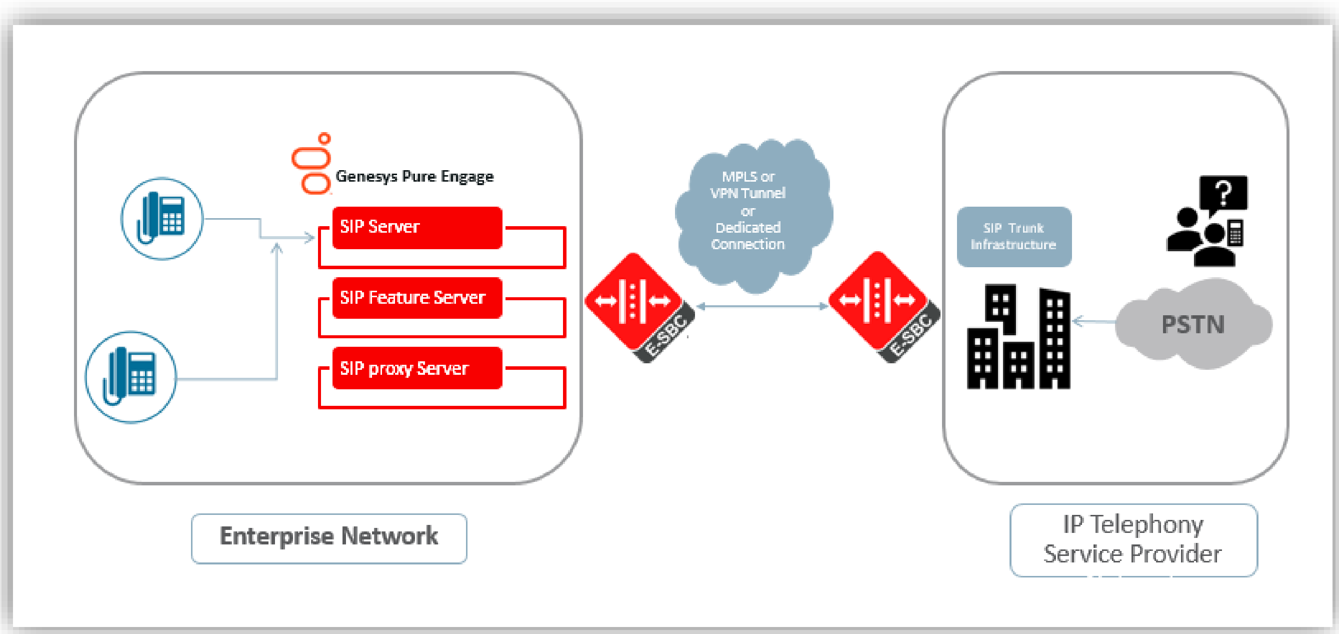
Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

3.2. Requirements

Fully functioning Genesys SIP Server deployment, including Media Server, SIP Proxy and SIP Feature Server. Testing is performed as per below product release version.

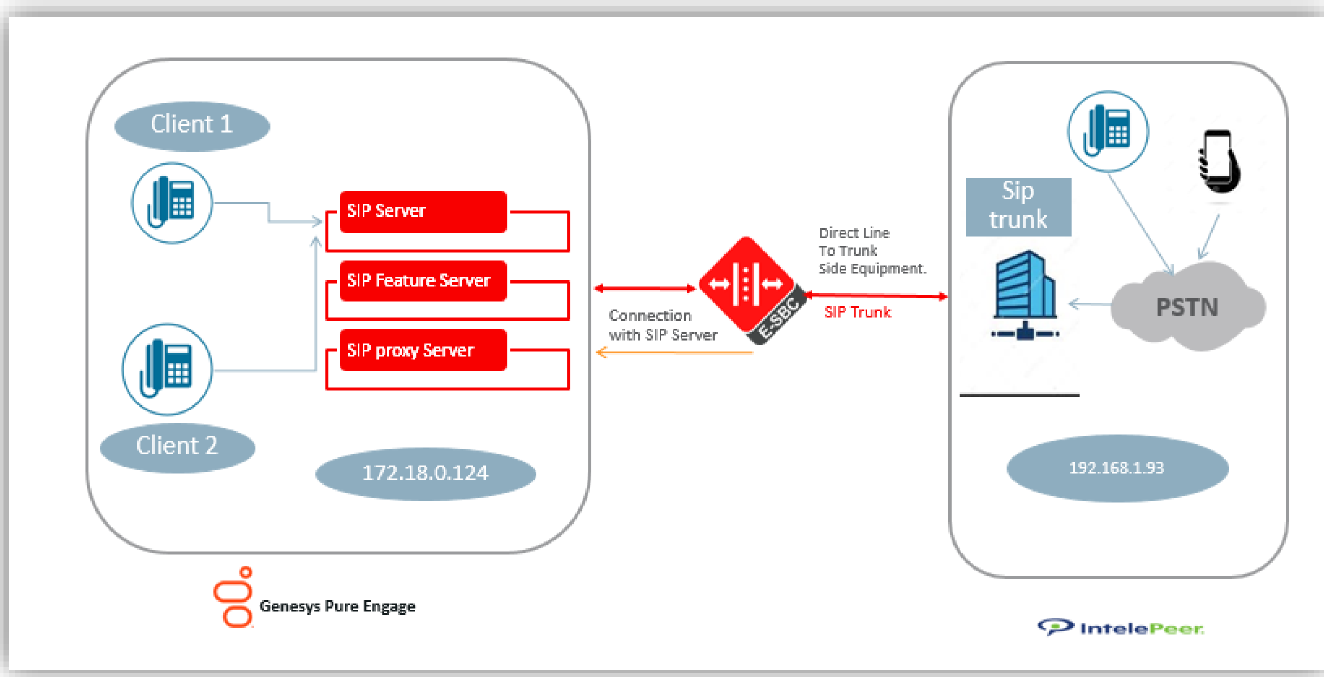
- Genesys SIP Server, Version 8.1.1
- Genesys Media Control Platform, Version 9.0.013.61
- Genesys SIP Proxy Server, Version 8.1.100.76
- Genesys SIP Feature Server, Version 8.1.202.11
- Oracle Enterprise Session Border Controller – All Oracle SBC models including Virtual Machine Edition,4600,1100,3900,6300,6350 platform running SCZ830m1p2 or above

3.3. Architecture



The Genesys SIP Server and the Oracle SBC are the edge components that form the boundary of the SIP trunk. The configuration, validation and troubleshooting of the Oracle SBC to work with the Genesys SIP Server will be described in this document.

3.4. Lab Configuration



The following diagram, similar to the Reference Architecture described earlier in this document, illustrates the lab environment created to facilitate certification testing.

All network parameters, ip addresses, hostnames etc. are specific to Oracle Labs, and cannot be used outside of the Oracle Lab environment. They are for example purposes only!!!

As per the Test Bed the connections made is as below -

- s0p3 – Connection to SIP Trunk
- s0p0 – Connection to Genesys SIP Server

In the setup the Oracle SBC sits in between the Genesys SIP Server and the SIP Trunk.

Client 1 and Client 2 are softphones registered on the SIP Server. The calls are made from PSTN Network which land onto the endpoints registered on Genesys SIP Server via the SBC.

We also have remote endpoints which register onto the SIP Server via the SBC which is not illustrated in the Diagram and is covered in another section of the documentation.

Calls made from Genesys Internal endpoints to external world are directed to SBC which then sends the call to the Trunk to terminate on PSTN Network.

4. Deploy the Oracle SBC

In this section we describe the steps for configuring an Oracle Session Border Controller, formally known as the Acme Packet Net-Net Session Director ("SBC"), for use with Genesys SIP Server in a SIP Trunking scenario.

4.1. In Scope

The following guide configuring the Oracle SBC assumes that this is a newly deployed device dedicated to a single customer. If a service provider currently has the Oracle SBC deployed and is adding SIP Server customers, then all the mentioned configuration may not be necessary and only the relevant sections must be configured.

Below are the Links to the Oracle Session Border Controller Configuration Guide which can be used as a reference point for configuring the Oracle SBC.

Web GUI User Guide

https://docs.oracle.com/en/industries/communications/enterprise-session-border-controller/8.3.0/webgui/Oracle SBC_scz830_webgui.pdf

ACLI Configuration Guide

https://docs.oracle.com/en/industries/communications/enterprise-session-border-controller/8.3.0/configuration/Oracle SBC_scz830_configuration.pdf

Note that Oracle offers several models of Oracle SBCs. This document covers the setup for Oracle SBC 4600 platform running SCZ830m1p2 or later. If instructions are needed for other Oracle SBC models, please contact your Oracle representative.

4.2 Out of Scope

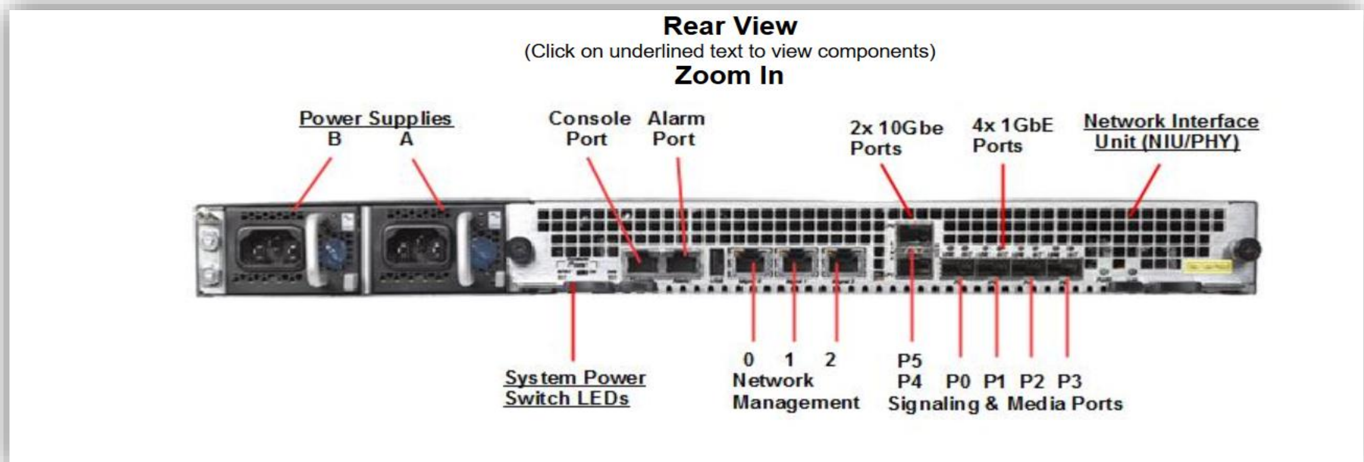
Configuration of Network management including SNMP and RADIUS

4.3 Booting the SBC

Once the Oracle SBC is racked and the power cable connected, you are ready to set up physical network connectivity.

In the Lab environment we have setup the 4600 SBC and the below Figure illustrates the Rear view of the SBC which is used to setup Physical Connectivity of management and media cables.

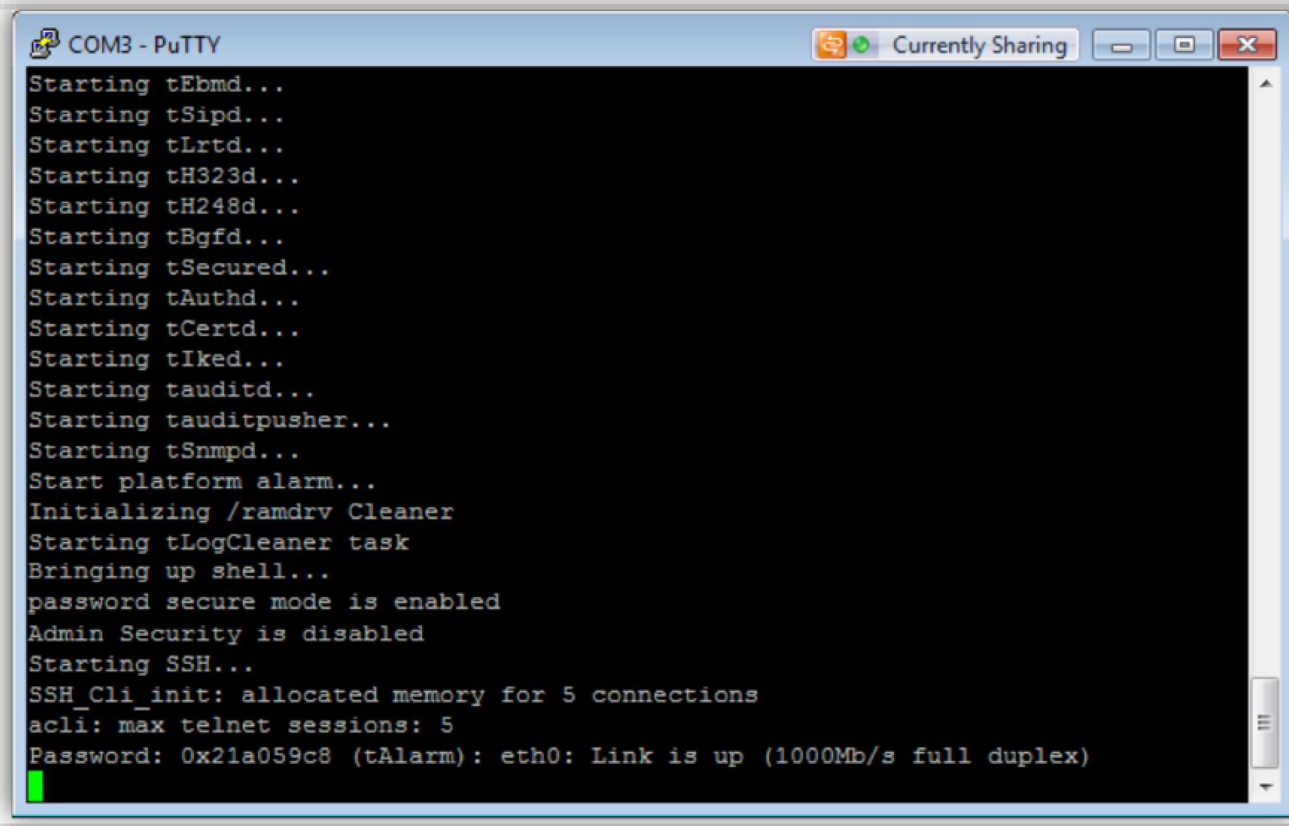
The port layout may differ depending upon the SBC model being used and must be configured accordingly.



To access the console port:
Connect the serial console cable to the Oracle SBC to 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 Oracle SBC and confirm that you see the following output from the bootup sequence.



Enter the following commands to login to the Oracle SBC and move to the configuration mode. Note that the default Oracle SBC password is “acme” and the default super user password is “packet”.

Password: acme

Oracle SBC-Genesys > enable

Password: packet

Oracle SBC-Genesys # configure terminal

Oracle SBC-Genesys (configure) #

You are now in the global configuration mode.

4.5. Initial Configuration

i) Assigning the management Interface an IP address

To assign an IP address, one has to configure the bootparams on the Oracle SBC by going to

Oracle SBC-Genesys #configure terminal --- >bootparams

```
NN4600-138# conf t
NN4600-138(configure)# bootparam

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

Boot File               : /boot/nnSCZ830m1p2.bz
IP Address              : 10.138.194.138
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                   :
Target Name             : NN4600-138
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.
```

A reboot is required if changes are made to the existing boot parameters.

Once you have gained access to the SBC you can further configure the system through the WEB-GUI Interface.

5. Configuring SBC using WEBGUI

There are two methods for configuring the Oracle SBC, ACLI, or GUI.
For the purposes of this note, we'll be using the Oracle SBC GUI for all configuration examples.

The WebGUI can be accessed through the url :-

http://<SBC_MGMT_IP>

web-server-config is enabled by default on the Oracle SBC. If not then one can make the web-server-config on the SBC by navigating to **system> web-server-config**

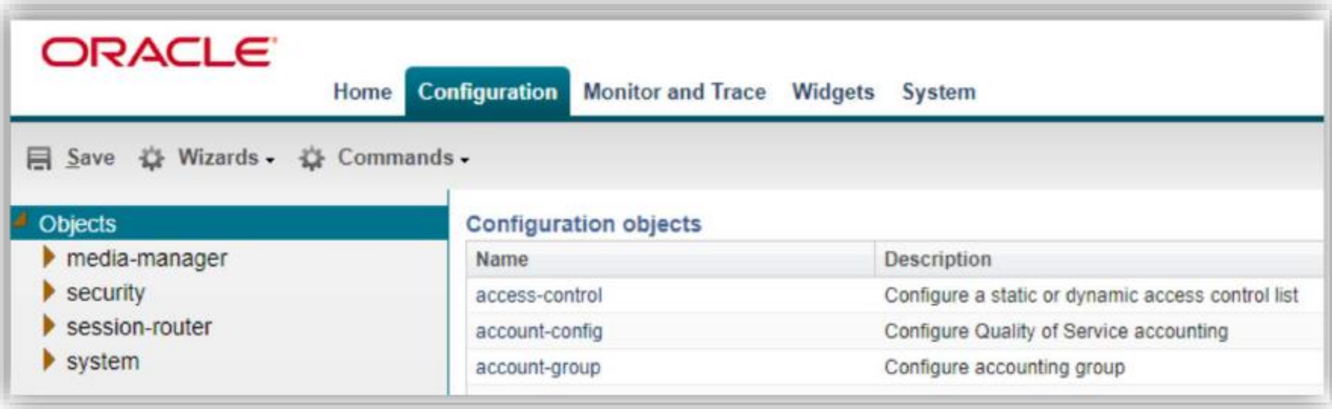
```
web-server-config
    state                   enabled
    inactivity-timeout          5
    http-state                enabled
    http-port                 80
    https-state               disabled
    https-port                443
    http-interface-list
    tls-profile
    last-modified-by              admin@console
```

Please refer to the Web GUI Guide for more information.

https://docs.oracle.com/en/industries/communications/enterprise-session-border-controller/8.3.0/webgui/Oracle SBC_scz830_webgui.pdf

The expert mode is used for configuration.

Once you have accessed the Oracle SBC, at the top, click the Configuration Tab. This will bring up the OCSBC Configuration Objects List on the left hand side of the screen.



You may now follow the further sections of the Document to configure the SBC as desired to Connect with Genesys Pure SIP Server.

5.1 SIP Trunking Configuration for the Oracle SBC

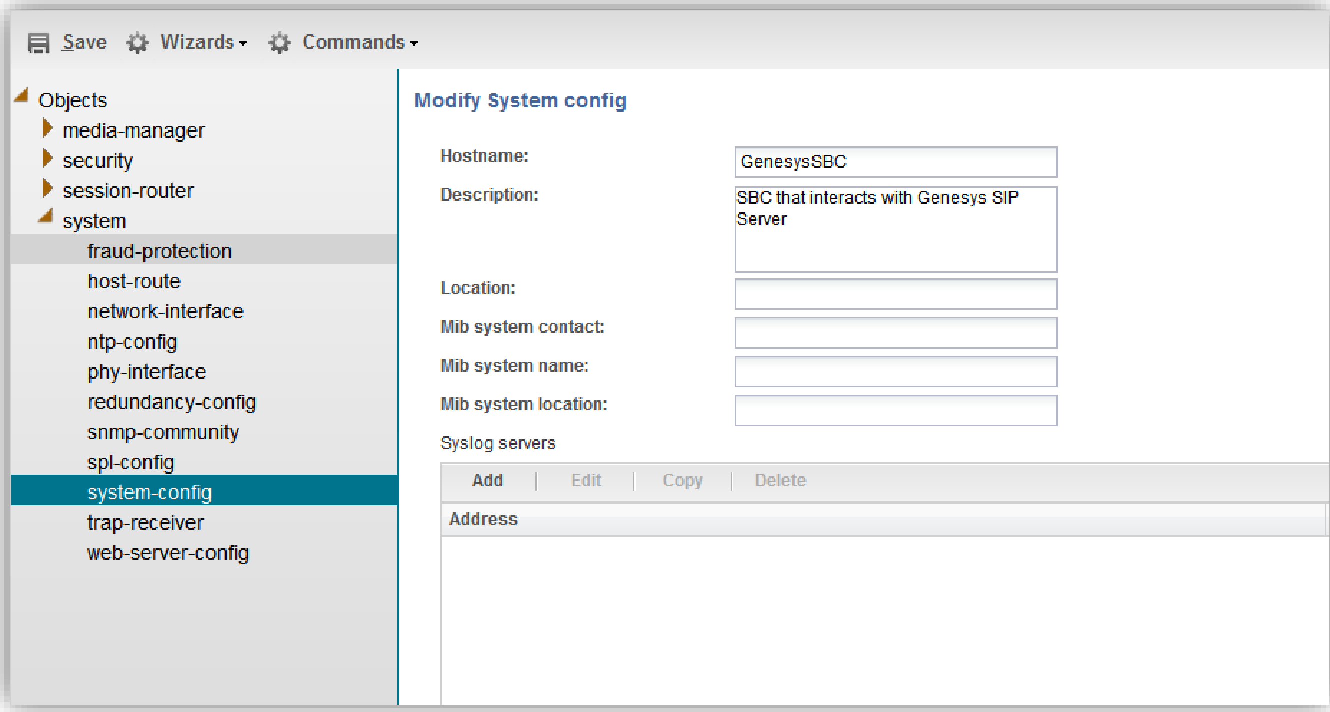
The following section shows the Oracle SBC configuration required to work with Genesys SIP Server and the SIP trunk. The protocol used between the Oracle SBC and SIP server is UDP for signaling and RTP for media; the SIP trunk is configured for UDP in this interop testing.

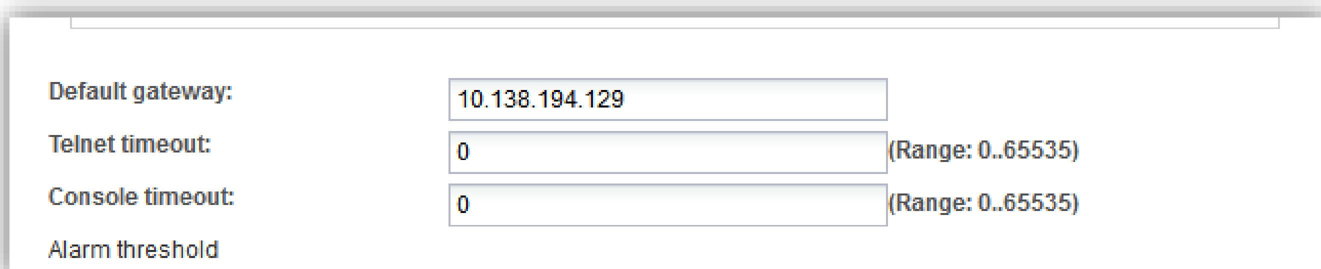
It is outside the scope of this document to include all the interoperability working information as it will differ in every deployment. The Document should be used as reference for the basic configuration objects required to interwork Oracle SBC with Genesys SIP Server.

5.2 Configure system element values

To configure system element values, use the system-config command under the system branch. Then enter values appropriate to your environment, including your default gateway IP address for your management Ethernet interface.

Here we have configured the SBC Hostname, Description and the Default Gateway. These can be used as minimal settings to configure the system-config element.





Click the **OK** at the bottom of the screen.

5.3 Configure Physical element values

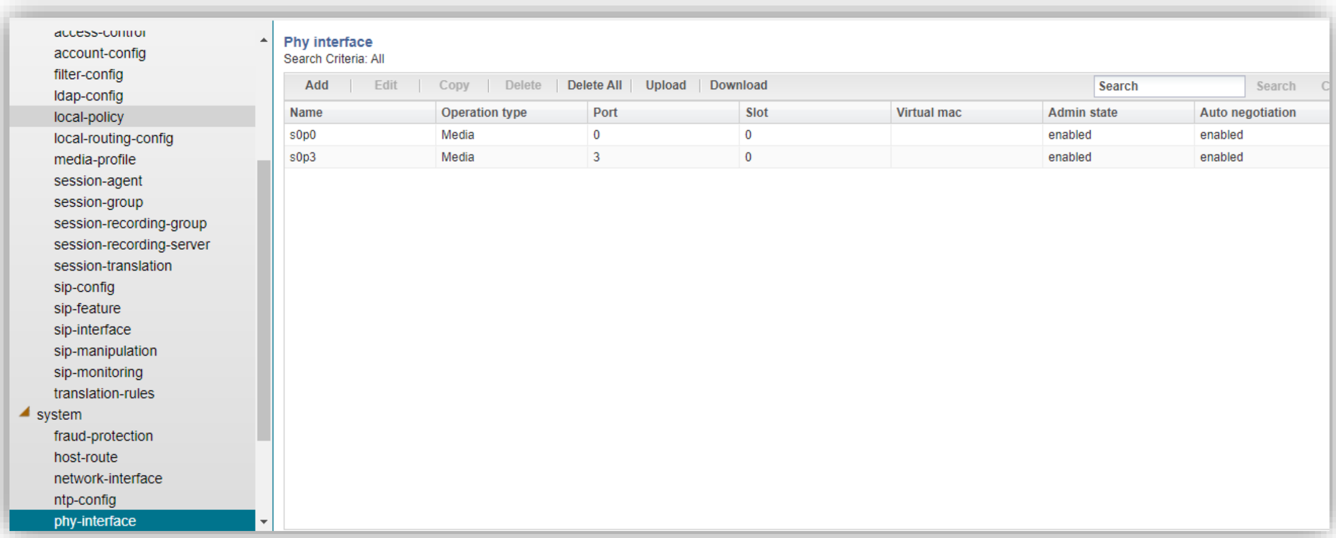
The phy-interface configuration element:

- Defines some Layers 1-2 properties (speed, half/full duplex, MAC address, and so on)
- Must be created for each physical connector that you plan to use.

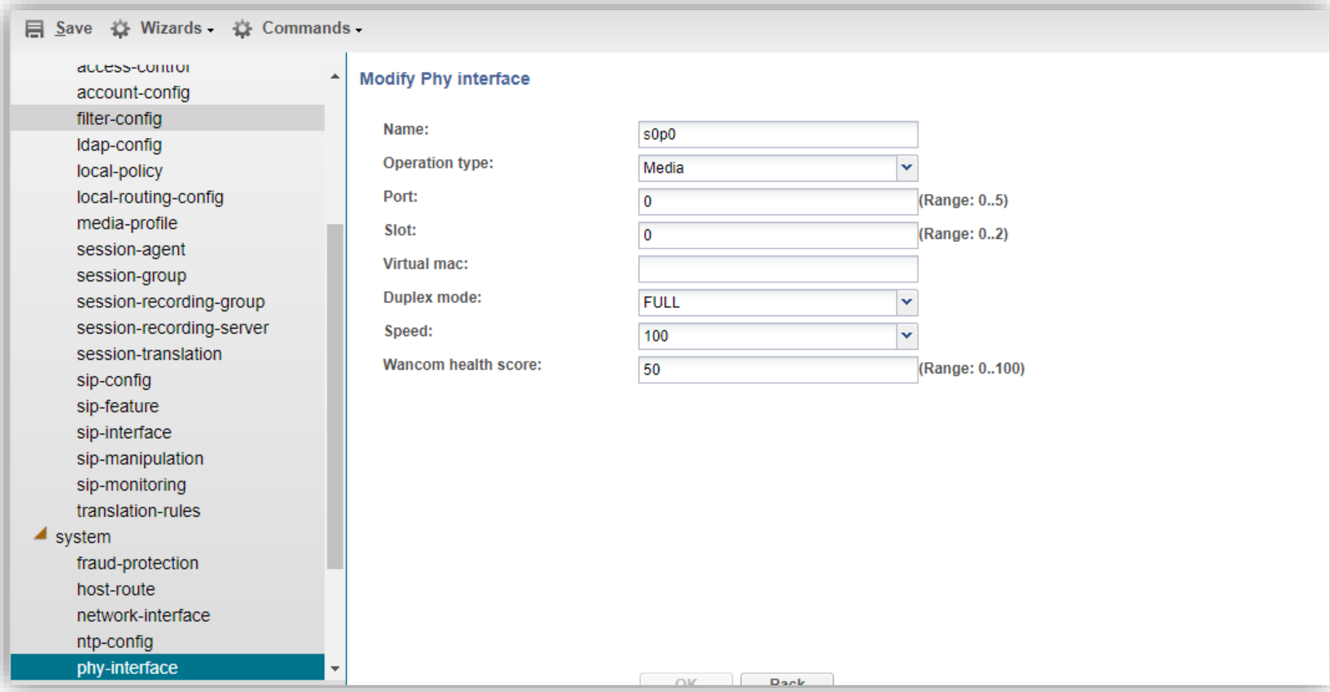
To configure physical Interface values, navigate to **system->phy-interface** on the Oracle SBC Web GUI Configure the physical interface for s0p0 and s1p0 for connectivity with the Trunk and the Genesys SIP Server Environment.

As per the Test Bed the connections made is as below -

- s0p3 – Connection to SIP Trunk
- s0p0 – Connection to Genesys SIP Server



Sample physical interface configuration.



5.4 Configure Network Interface

The network-interface configuration element:

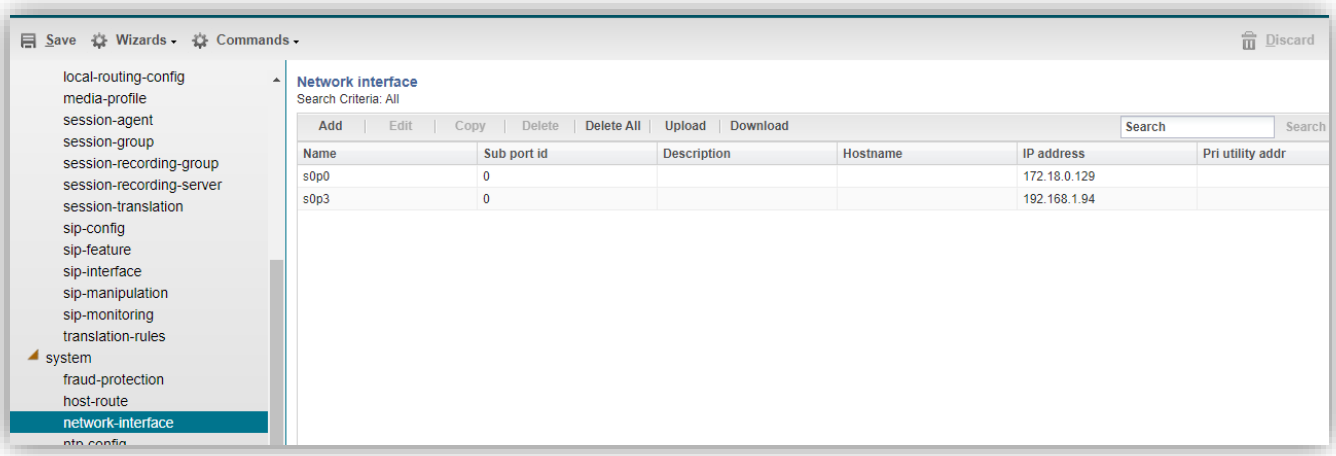
- Must be created and refers to a specific phy-interface
- Defines Layers 2-3 properties (VLAN, IP address, mask, default gateway, and so on)

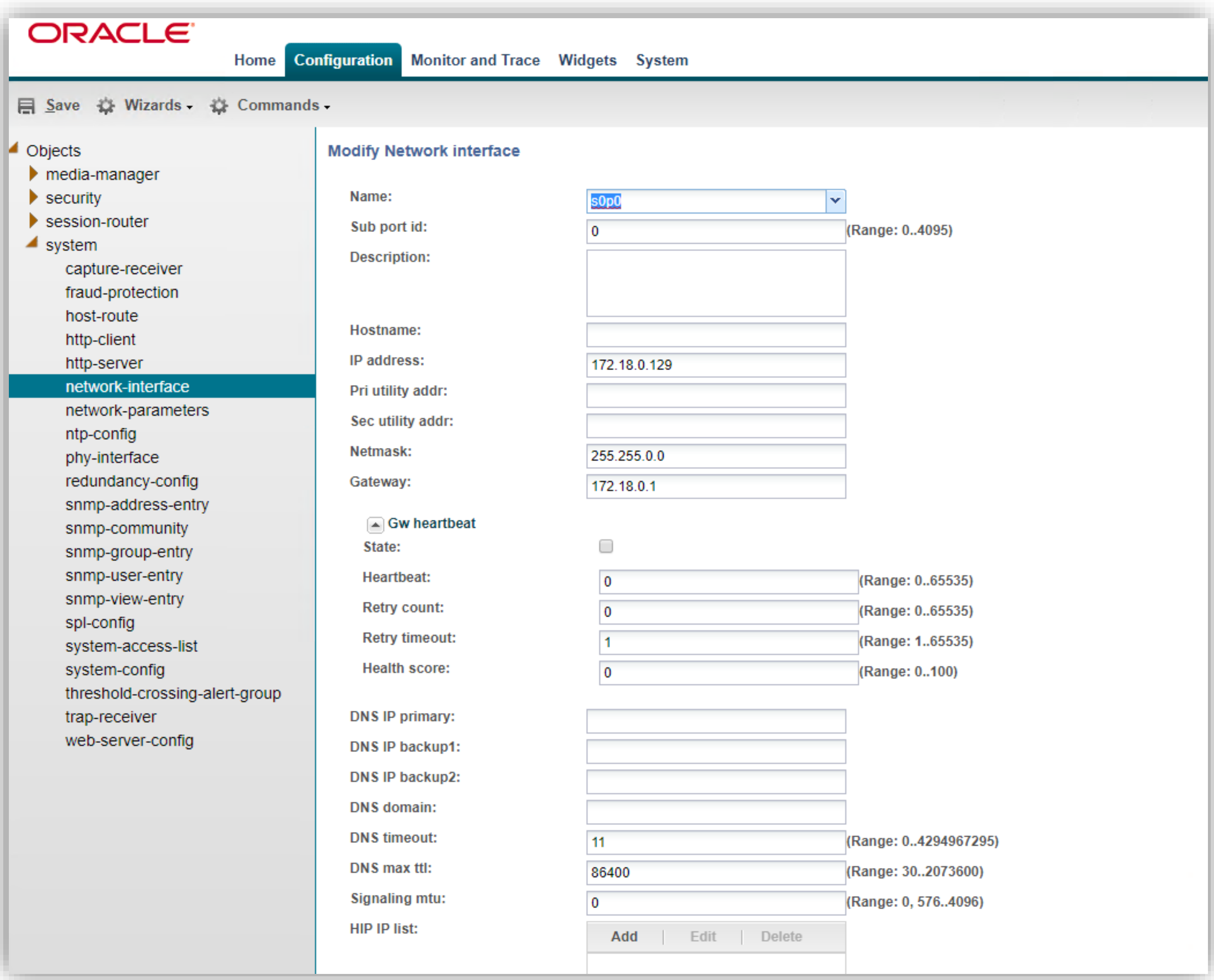
To configure network-interface, navigate to **system->Network-Interface**. Configure two interfaces, one for PSTN Trunk and one for Genesys SIP Server.

Below is the example from test bed for the network-interface configuration.

Here 2 Network interfaces are configured where-

- s0p3 – Connection to SIP Trunk
- s0p0 – Connection to Genesys SIP Server


Save
Wizards
Commands
Discard
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
sip-interface
sip-manipulation
sip-monitoring
translation-rules
system
fraud-protection
host-route
network-interface
Network interface
Search Criteria: All
Add
Edit
Copy
Delete
Delete All
Upload
Download
Search
Search
Name
Sub port id
Description
Hostname
IP address
Pri utility addr
s0p0
0
172.18.0.129
s0p3
0
192.168.1.94


ORACLE
Home
Configuration
Monitor and Trace
Widgets
System
Save
Wizards
Commands
Objects
media-manager
security
session-router
system
capture-receiver
fraud-protection
host-route
http-client
http-server
network-interface
network-parameters
ntp-config
phy-interface
redundancy-config
snmp-address-entry
snmp-community
snmp-group-entry
snmp-user-entry
snmp-view-entry
spl-config
system-access-list
system-config
threshold-crossing-alert-group
trap-receiver
web-server-config
Modify Network interface
Name: s0p0
Sub port id: 0 (Range: 0..4095)
Description:
Hostname:
IP address: 172.18.0.129
Pri utility addr:
Sec utility addr:
Netmask: 255.255.0.0
Gateway: 172.18.0.1
Gw heartbeat
State:
Heartbeat: 0 (Range: 0..65535)
Retry count: 0 (Range: 0..65535)
Retry timeout: 1 (Range: 1..65535)
Health score: 0 (Range: 0..100)
DNS IP primary:
DNS IP backup1:
DNS IP backup2:
DNS domain:
DNS timeout: 11 (Range: 0..4294967295)
DNS max ttl: 86400 (Range: 30..2073600)
Signaling mtu: 0 (Range: 0, 576..4096)
HIP IP list:
Add
Edit
Delete

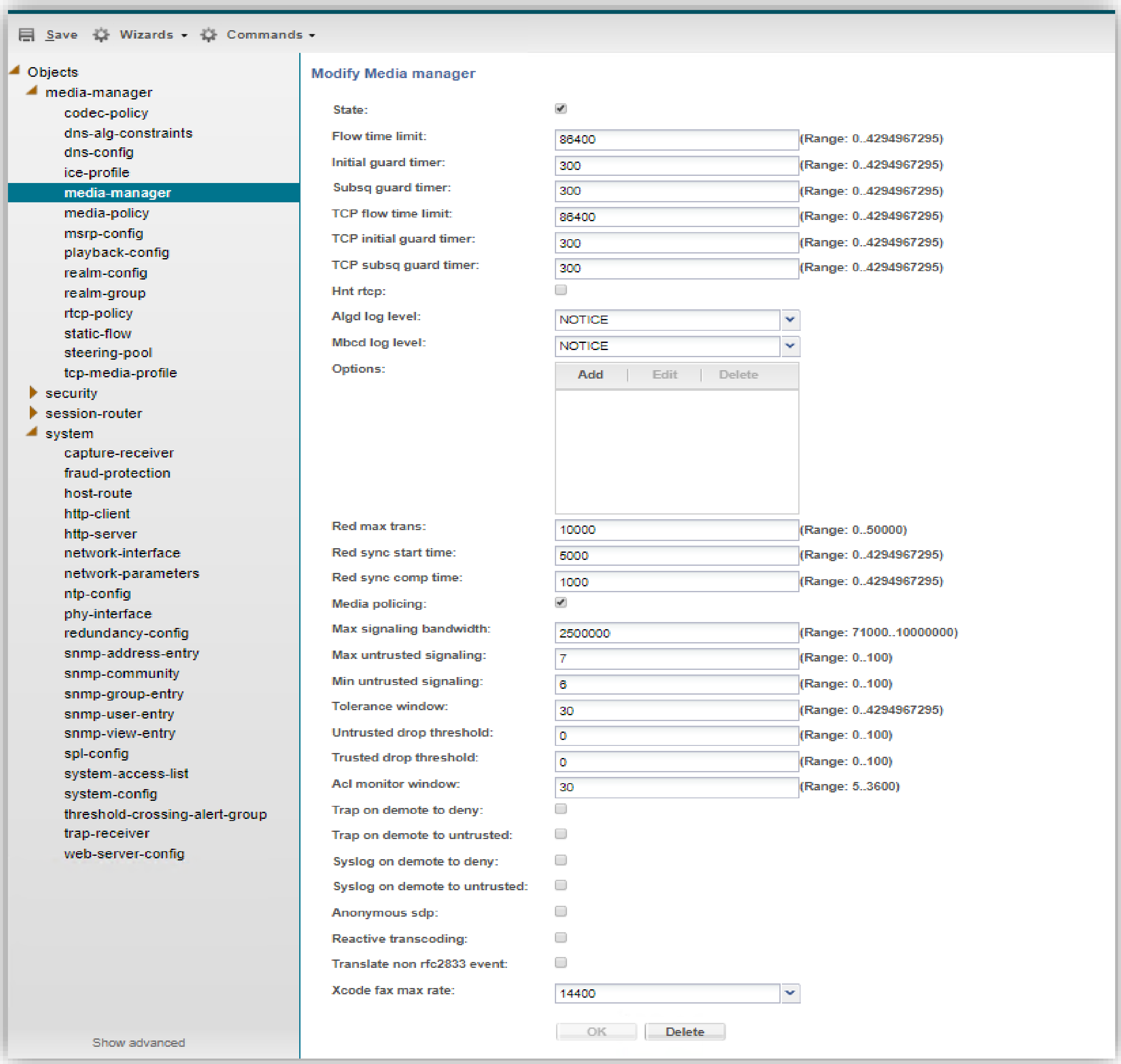
5.5 Enable media manager

Media-manager handles the media stack required for SIP sessions on the SBC. Enable the media manager and configure the below option for generating rtcp reports.

To configure navigate to **Media-Manager->Media-Manager** and enable the configuration.

Below is the example from test bed for the Media-Manager configuration. Just Checking on State as Yes Is sufficient for SBC to enable the Media Manager configuration and handle media traffic (RTP)

Other parameters are not required but are relevant for settings like Latching, DDOS Protection etc. to be enabled on the SBC. These parameters are outside the scope of the document and are left to their default values.


Save Wizards Commands
Objects
media-manager
codec-policy
dns-alg-constraints
dns-config
ice-profile
media-manager
media-policy
msrp-config
playback-config
realm-config
realm-group
rtcp-policy
static-flow
steering-pool
tcp-media-profile
security
session-router
system
capture-receiver
fraud-protection
host-route
http-client
http-server
network-interface
network-parameters
ntp-config
phy-interface
redundancy-config
snmp-address-entry
snmp-community
snmp-group-entry
snmp-user-entry
snmp-view-entry
spl-config
system-access-list
system-config
threshold-crossing-alert-group
trap-receiver
web-server-config
Show advanced
Modify Media manager
State:
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:
Algd log level: NOTICE
Mbcd log level: NOTICE
Options: Add Edit Delete
Red max trans: 10000 (Range: 0..50000)
Red sync start time: 5000 (Range: 0..4294967295)
Red sync comp time: 1000 (Range: 0..4294967295)
Media policing:
Max signaling bandwidth: 2500000 (Range: 71000..10000000)
Max untrusted signaling: 7 (Range: 0..100)
Min untrusted signaling: 6 (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:
Trap on demote to untrusted:
Syslog on demote to deny:
Syslog on demote to untrusted:
Anonymous sdp:
Reactive transcoding:
Translate non rfc2833 event:
Xcode fax max rate: 14400
OK Delete

5.6. Enable Sip Config

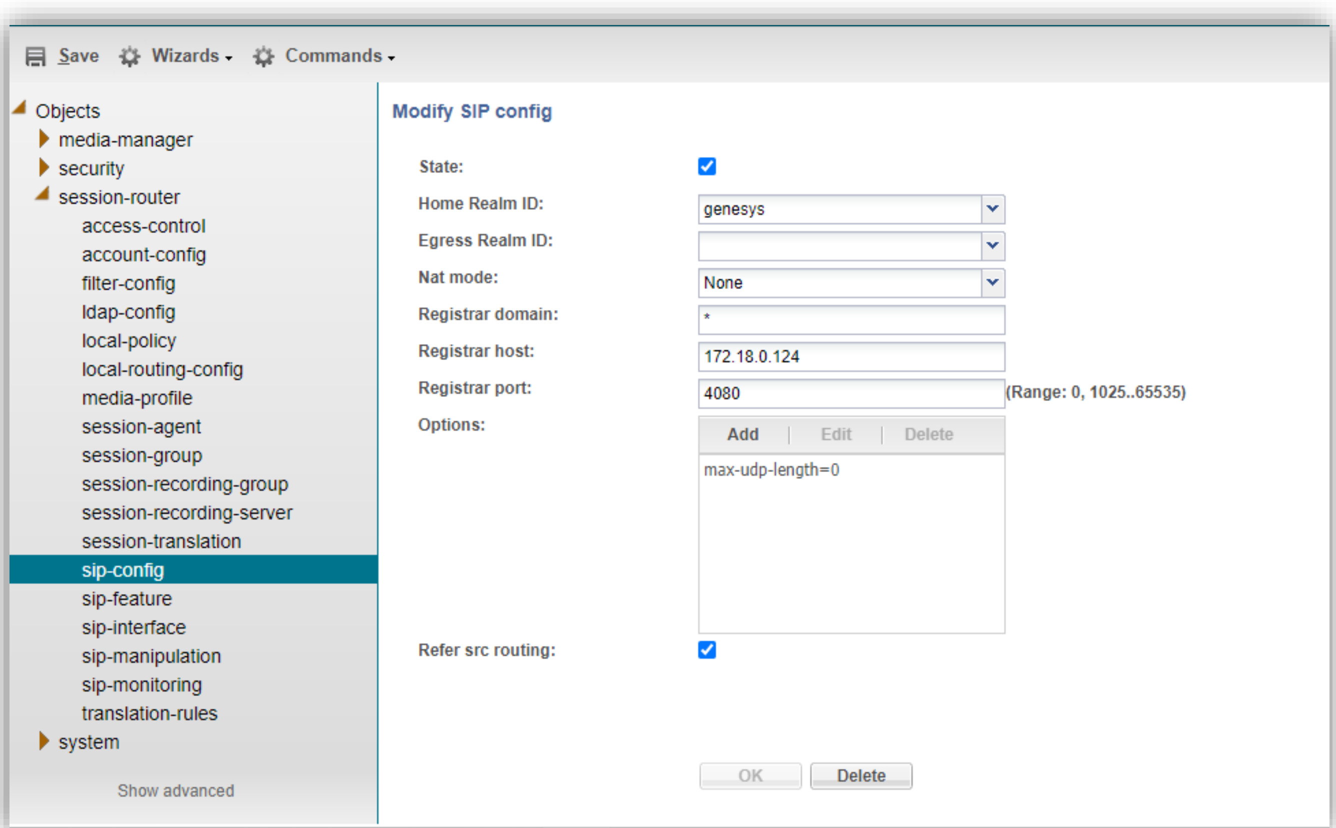
SIP config enables SIP handling in the SBC. Make sure the **home realm-id, registrar-domain and registrar-host are configured**. Also add the options to the sip-config as shown below.

To configure sip-config navigate to Session-Router->sip-config on the Oracle SBC Web GUI.

Below are the important parameters under sip-config that need to be configured.

- Registrar-host is the Genesys SIP Server IP.
- The domain is put as * as we have not specified any specific domain on the test bed.
- The Genesys SIP Server port is configured as the Registrar-port on the Oracle SBC.
- The options "max-udp-length=0" should be configured if the SIP messages are of larger size to avoid SBC failing the calls with "513 message too large"

Setting this option to zero (0) forces sipd to send fragmented UDP packets. Using this option, you override the default value of the maximum UDP datagram size (1500 bytes; sipd requires the use of SIP/TCP at 1300bytes).You can set the global SIP configuration's max-udp-length=x option for global use in your SIP configuration, or you can override it on a per-interface basis by configuring this option in a SIP interface configuration



5.7 Configure Realms

A Ream

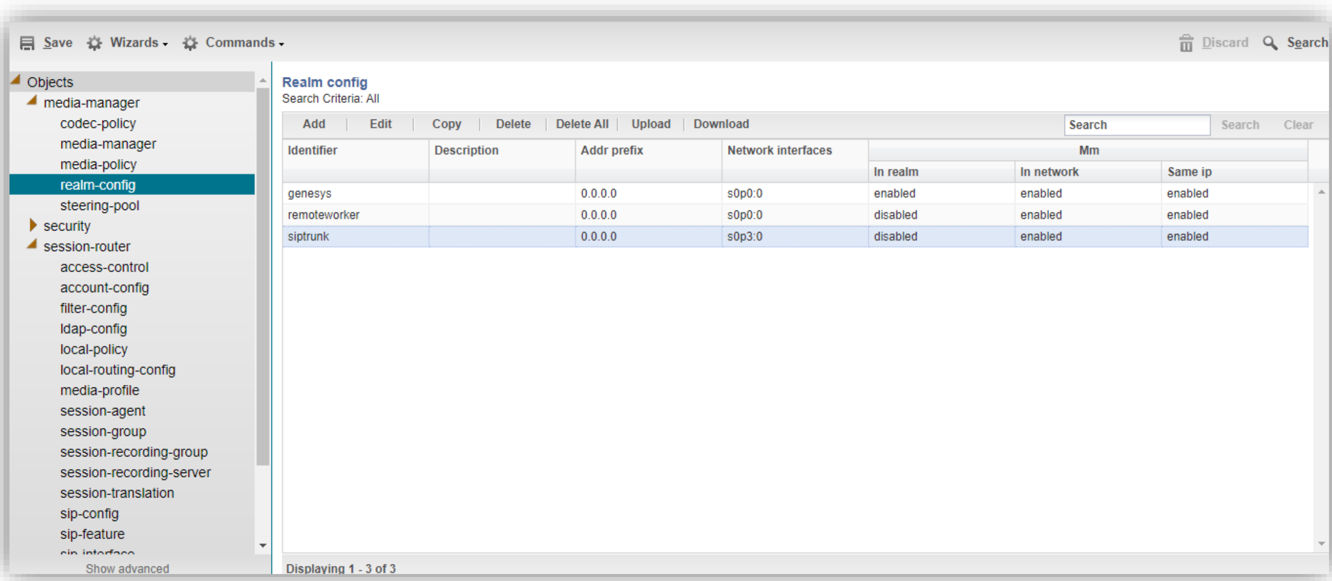
- Is a collection of VoIP entities residing in one or more networks.
- Typically maps to a service provider, enterprise, or end-user population environment.
- It is defined by a configuration element that contains many parameters that apply to the environment.
- Is considered as a “Layer 5” definition and a “container” of Resources.
- On the SBC, you configure realms (plus their associated configuration objects) to identify the interfaces, resources, and policies that apply to the signaling and media going through them.

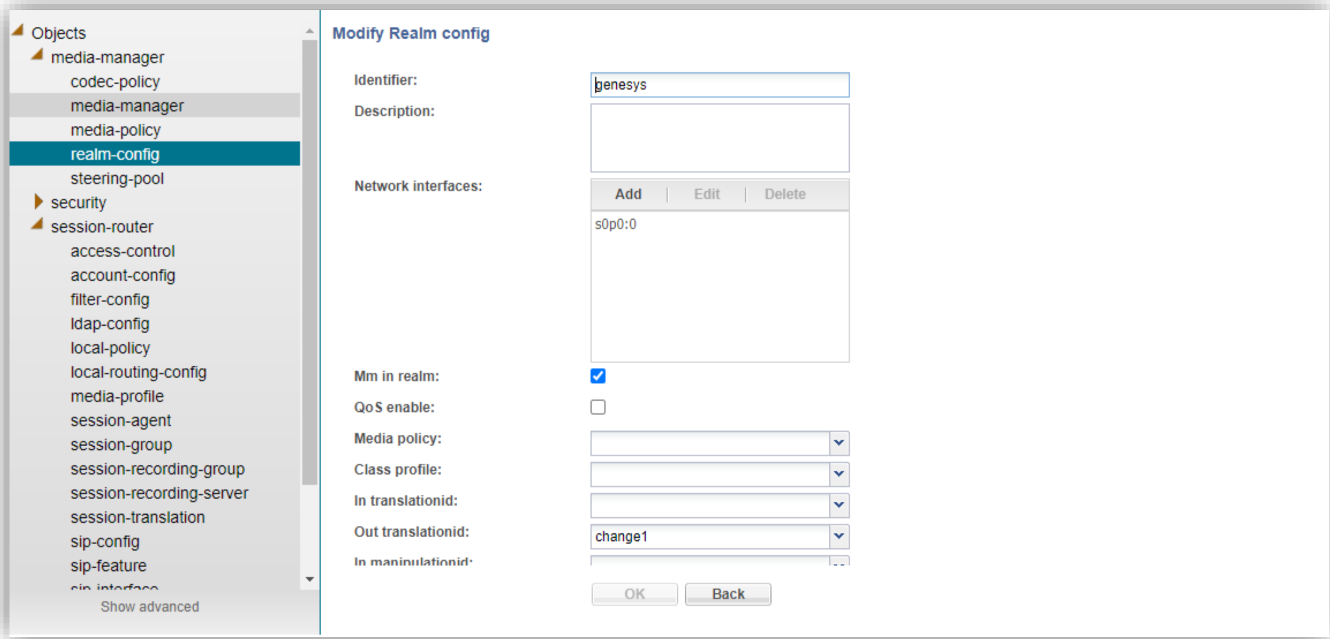
To configure Realm Navigate to **realm-config under media-manager** and configure a realm as shown in the picture.

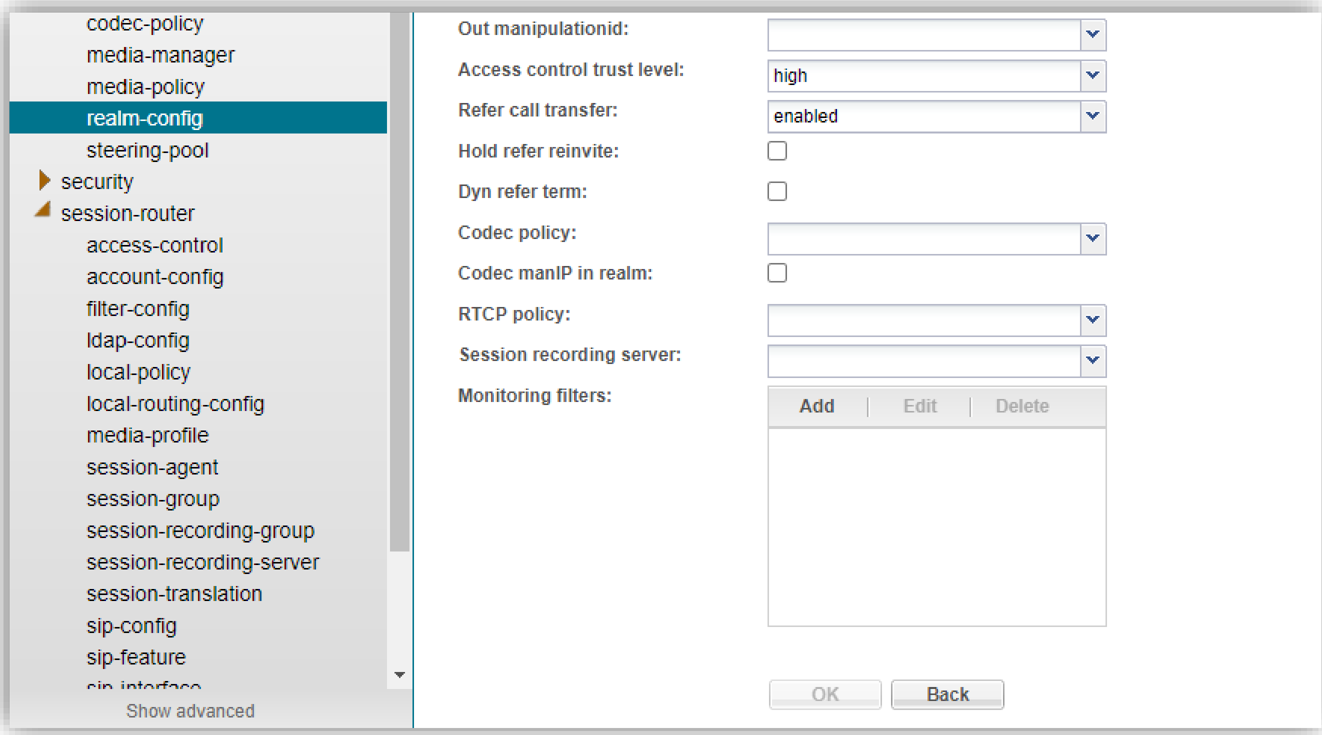
In this setup we have configured 3 Realms configured where –

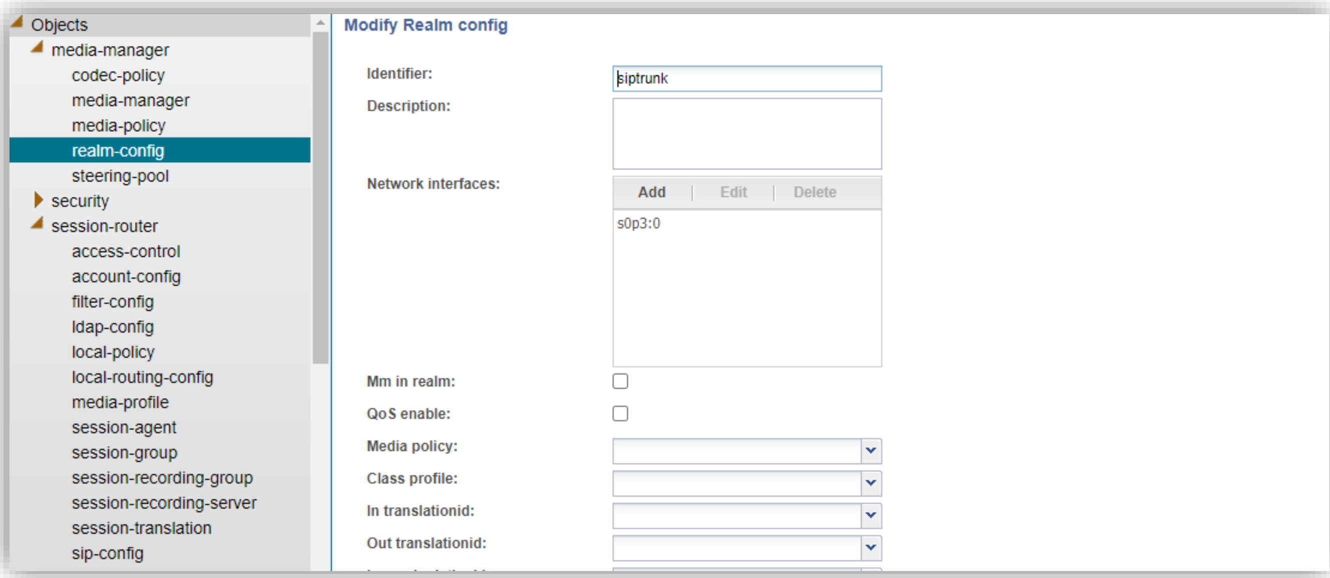
‘**siptrunk** is the realm for the connection to PSTN Trunk and is configured on s1p0 network interface.
‘**genesys’** is the Realm for connection to the to Genesys SIP Server and is configured on s0p0 network interface.

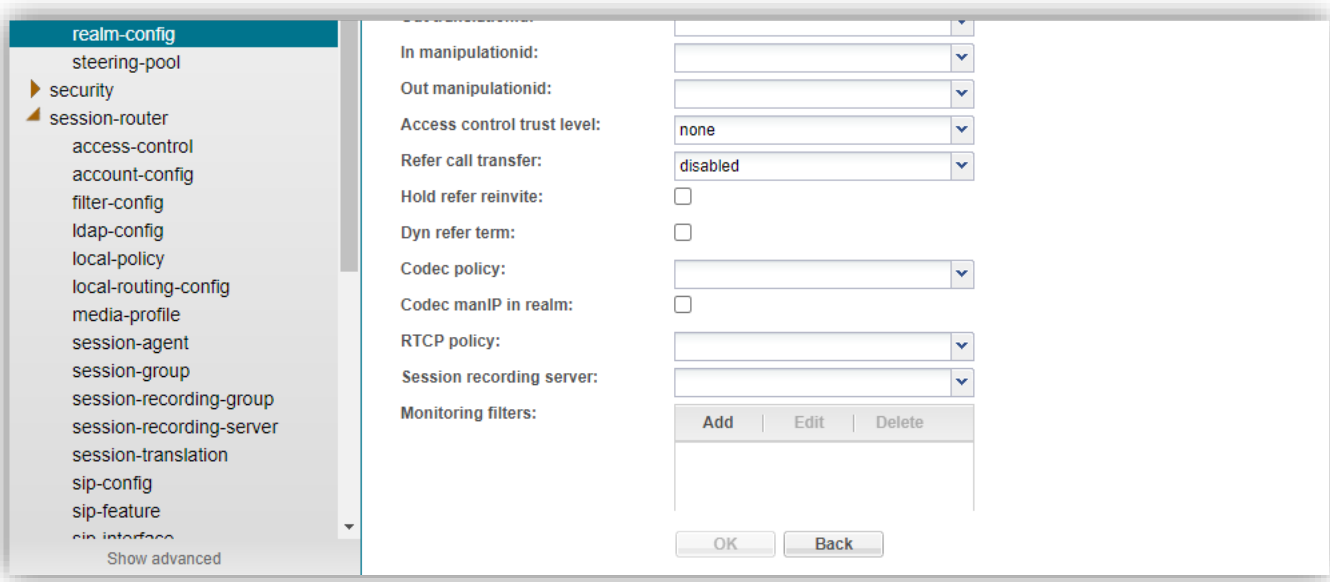
Another Realm ‘**remoteworker’** is configured to register remote endpoints and is described in a different section on the document.




Objects
media-manager
codec-policy
media-manager
media-policy
realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
Show advanced
Modify Realm config
Identifier:
genesys
Description:
Network interfaces:
Add
Edit
Delete
s0p0:0
Mm in realm:
QoS enable:
Media policy:
Class profile:
In translationid:
Out translationid:
change1
OK
Back


codec-policy
media-manager
media-policy
realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
Show advanced
Out manipulationid:
Access control trust level:
high
Refer call transfer:
enabled
Hold refer reinvite:
Dyn refer term:
Codec policy:
Codec manIP in realm:
RTCP policy:
Session recording server:
Monitoring filters:
Add
Edit
Delete
OK
Back


Objects
media-manager
codec-policy
media-manager
media-policy
realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
Modify Realm config
Identifier:
siptrunk
Description:
Network interfaces:
Add
Edit
Delete
s0p3:0
Mm in realm:
QoS enable:
Media policy:
Class profile:
In translationid:
Out translationid:


realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
Show advanced
In manipulationid:
Out manipulationid:
Access control trust level:
none
Refer call transfer:
disabled
Hold refer reinvite:
Dyn refer term:
Codec policy:
Codec manIP in realm:
RTCP policy:
Session recording server:
Monitoring filters:
Add
Edit
Delete
OK
Back

5.8 Configure Steering Pool

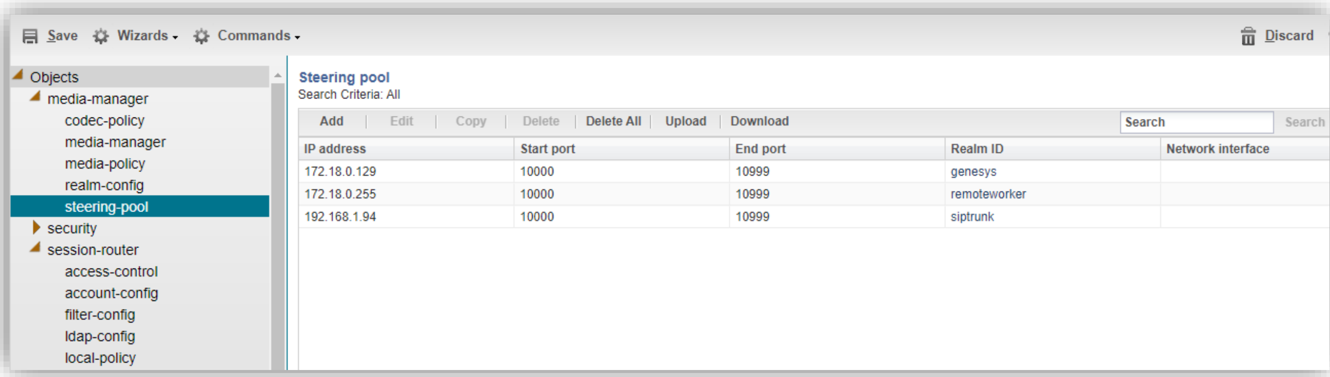
The steering-pool:

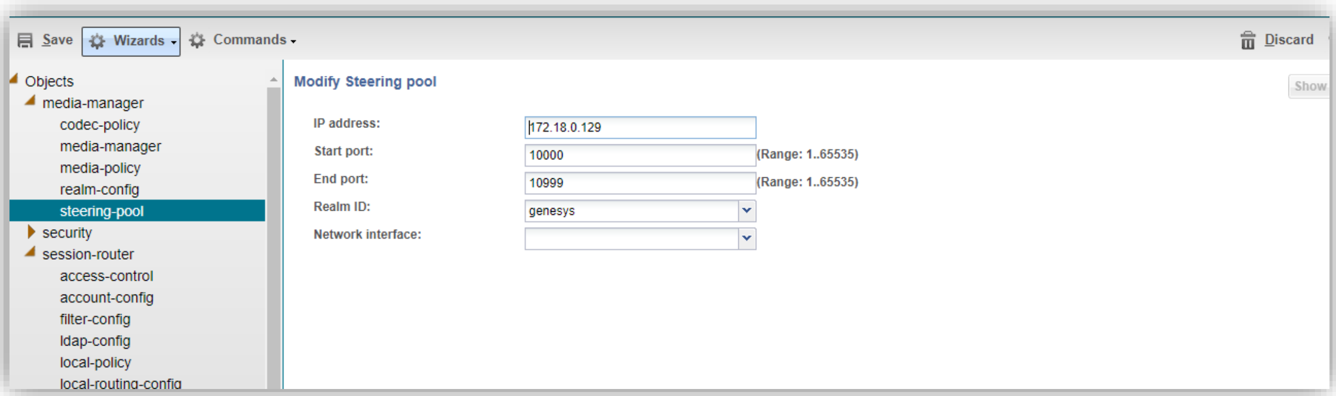
- Is the SBC's media interface (for a given realm)
- Receives and transmits RTP packets
- Defines a media IP address and a pool (range) of ports from which port(s) are dynamically allocated for every established session.
- Provides call admission control (CAC) by setting a limit of sessions going into and out of a realm
- A realm can have more than one steering-pool.

To configure steering pool navigate to **media-manager->steering pool**.

In this setup we have configured 3 steering pool against 3 Realms.

Below is the example from test bed for the steering-pool configuration.





5.9 Configure sip-interface

The sip-interface:

- Is the SBC's Edge Proxy Function
- Receives and transmits SIP signaling messages
- Provides a service pipe to the SIP daemon (sipd)
- Defines SIP signaling IP addresses, ports, transport protocols, and various SIP processing policies

To configure sip-interface, navigate to **session-router->sip-Interface**. Configure the interfaces for the PSTN and Genesys SIP Server.

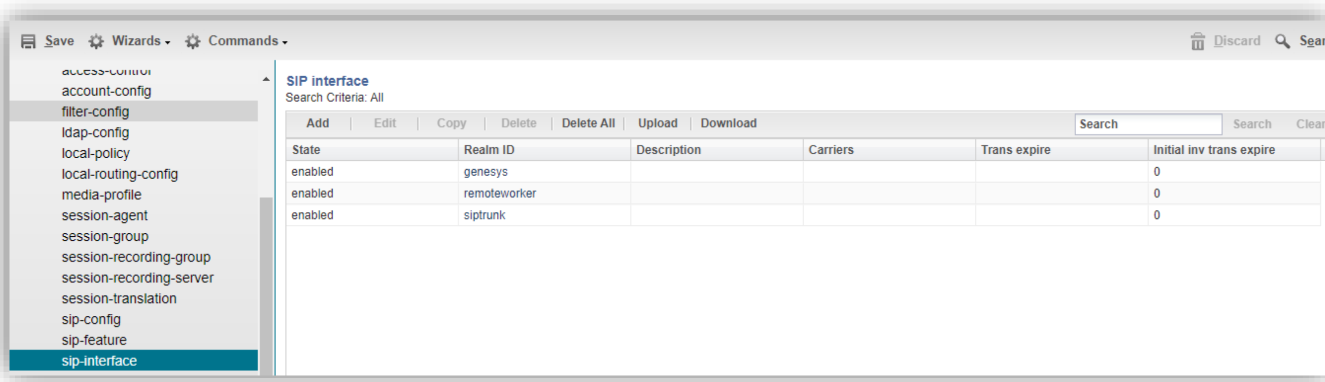
Below is the example from test bed on the sip-interface configuration.

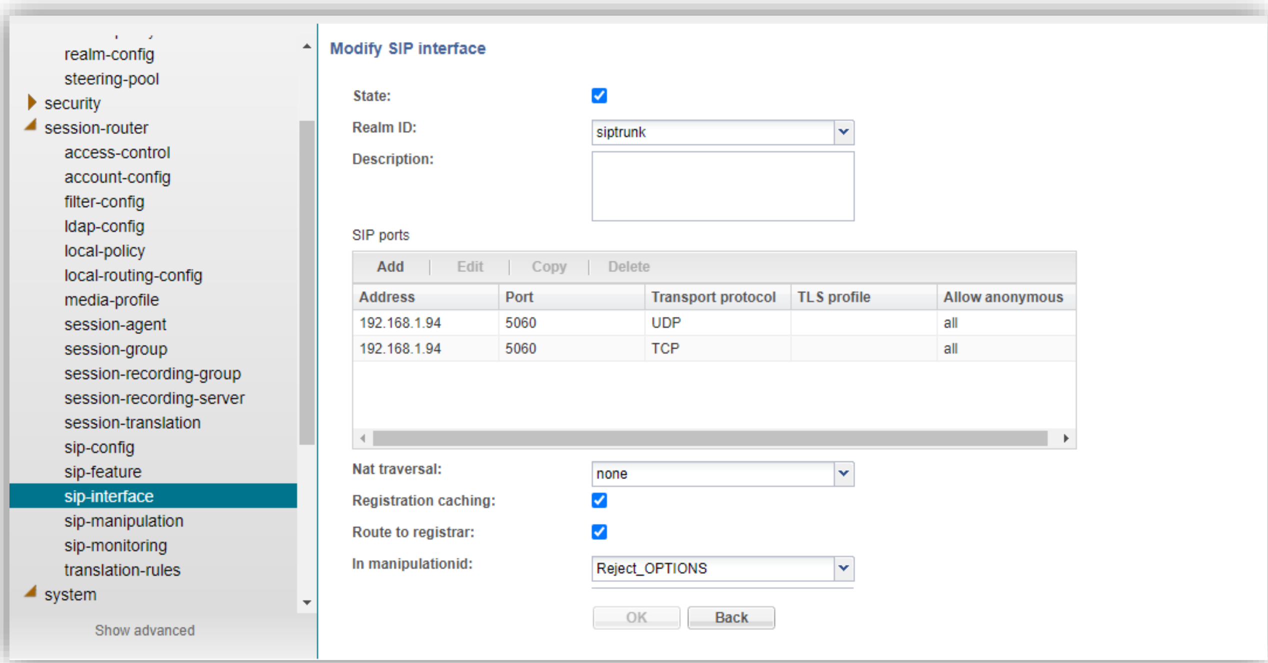
Three sip-interface are configured on the SBC where-

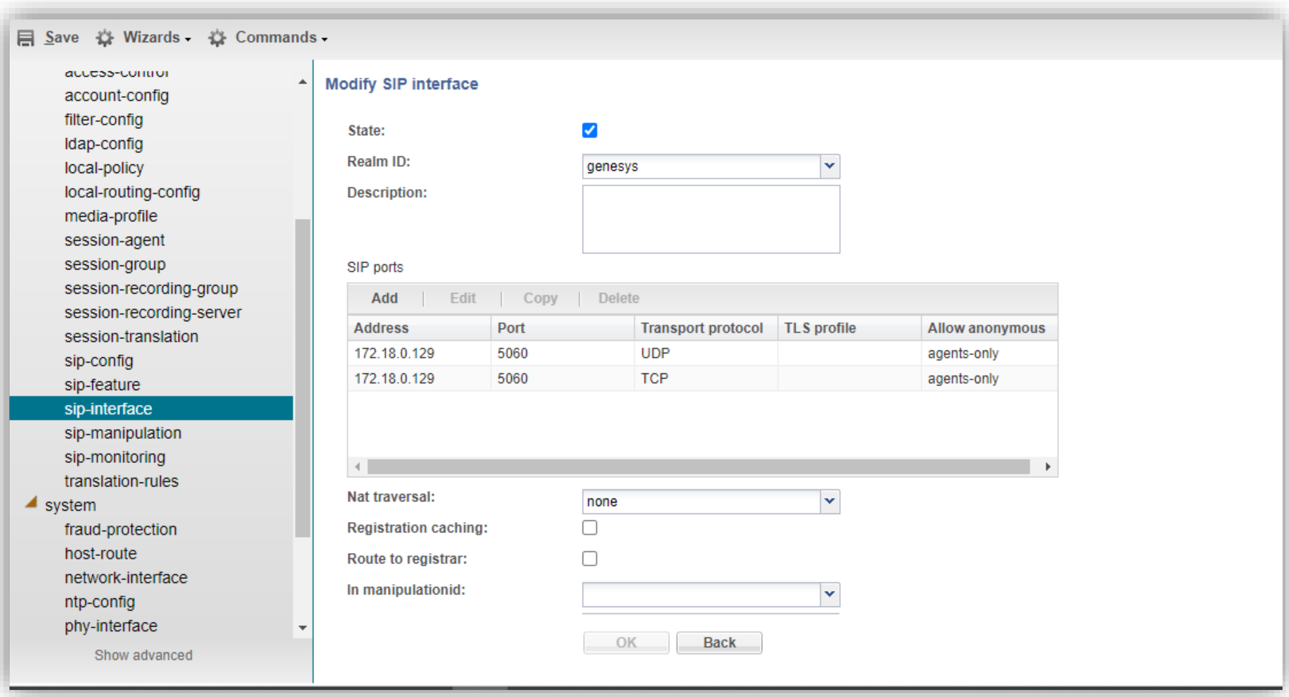
1. sip-interface 192.168.1.94 is configured with Realm siptrunk is to route inbound traffic from Trunk to the Genesys SIP Server. Registration caching is enabled in order for SBC to cache the registration data and Route to registrar parameter is enabled to send all requests that match cached registration to the destination defined for the registrar host.

2. sip-interface 172.18.0.129 is configured with Realm genesys to route the outbound traffic from Genesys SIP Server to the SIP Trunk.

3. 3. sip-interface **172.18.0.255** is configured with Realm **remoteworker** to route the registration from remote endpoint which register onto the SIP Server via the SBC. This is covered in detail another section of the document




realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
sip-interface
sip-manipulation
sip-monitoring
translation-rules
system
Show advanced
Modify SIP interface
State:
Realm ID:
siptrunk
Description:
SIP ports
Add
Edit
Copy
Delete
Address
Port
Transport protocol
TLS profile
Allow anonymous
192.168.1.94
5060
UDP
all
192.168.1.94
5060
TCP
all
Nat traversal:
none
Registration caching:
Route to registrar:
In manipulationid:
Reject_OPTIONS
OK
Back


Save
Wizards
Commands
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
sip-interface
sip-manipulation
sip-monitoring
translation-rules
system
fraud-protection
host-route
network-interface
ntp-config
phy-interface
Show advanced
Modify SIP interface
State:
Realm ID:
genesys
Description:
SIP ports
Add
Edit
Copy
Delete
Address
Port
Transport protocol
TLS profile
Allow anonymous
172.18.0.129
5060
UDP
agents-only
172.18.0.129
5060
TCP
agents-only
Nat traversal:
none
Registration caching:
Route to registrar:
In manipulationid:
OK
Back

5.10 Configure Session-agents

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

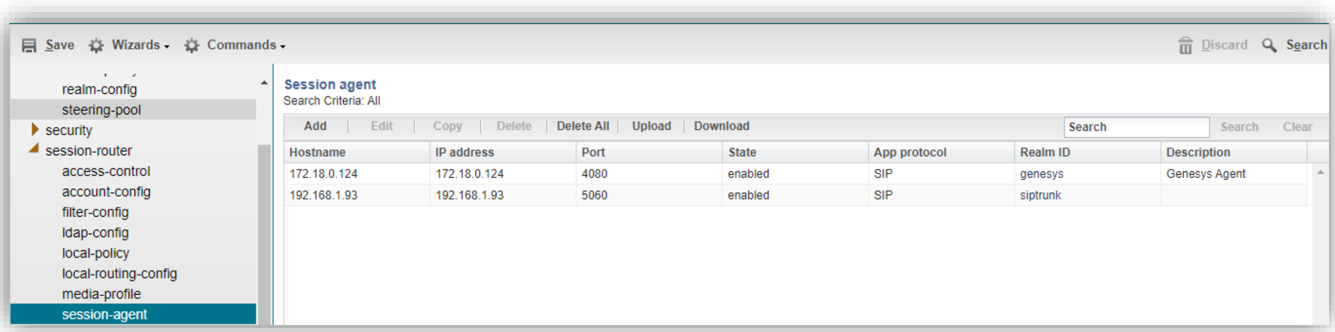
To Configure the session-agent for with the following parameters. Navigate to **session-router->Session-Agent.**

Below is the example from test bed for the session-agent configuration.

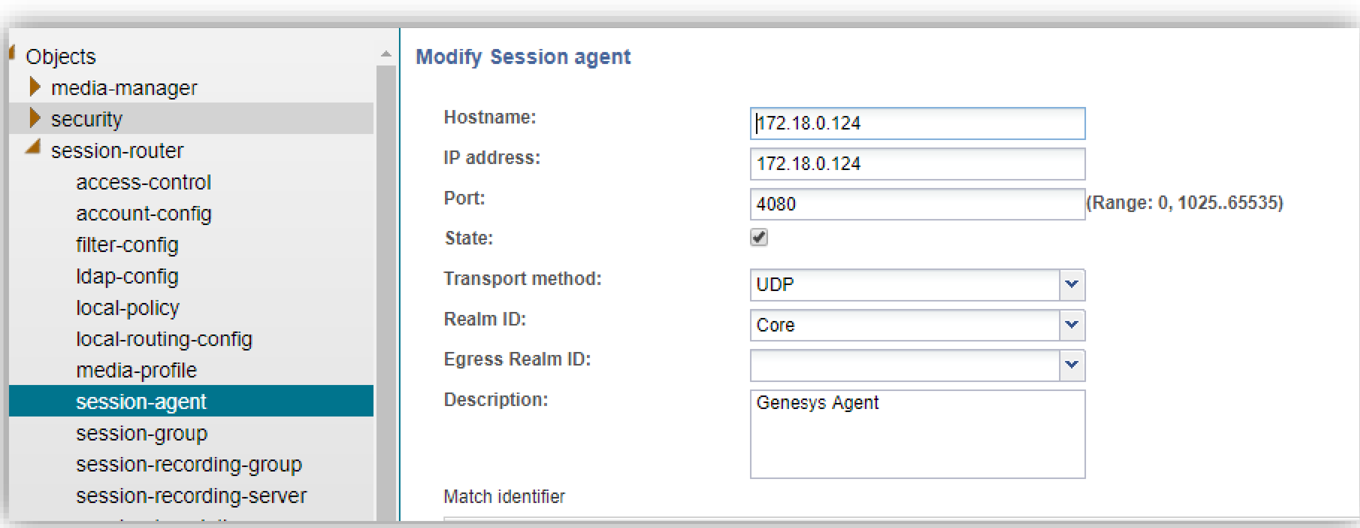
Here two session-agents are configured on the SBC for the trunk Side connection and other is for the Genesys SIP Server.

172.18.0.124 ----------- Genesys SIP Server

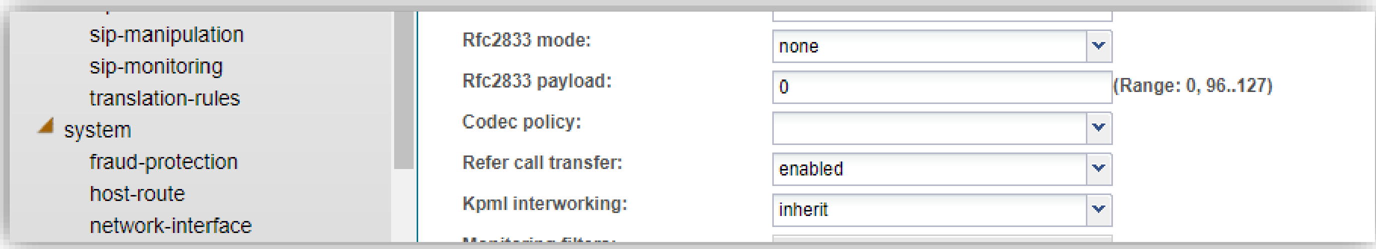
192.168.1.93 --------- PSTN SIP Trunk



Below is the example for the session-agent configured for Genesys SIP-server.



In the setup refer-call-transfer parameter is enabled on the SBC to locally handle the refer message for call transfer scenarios.



Certain test scenarios require handling of SIP Refer with replaces header. In order to complete those scenarios we also enabled option "**refer-reinvite**" on the session-agent to enable sip refer handling that contains replaces header.

The parameter should only be enabled when it is required by Oracle SBC to handle the 'refer with replaces' header and must not be configured for normal refer scenarios.

If, after the conclusion of static or dynamic REFER handling, the REFER is terminated and a new INVITE issued, users now specify a policy lookup behavior based upon either the source realm of the calling party (the INVITE originator), or the source realm of the referring party (the REFER originator).

Behavior is controlled by a 'refer-src-routing' parameter in the sip-config configuration element.

disabled, the default value, specifies that the Oracle SBC performs a policy lookup based on the source realm of the calling party.

enabled specifies that the Oracle Communications Session Border Controller performs a policy lookup based on the source realm of the referring party.

5.11 Configure Local-policy

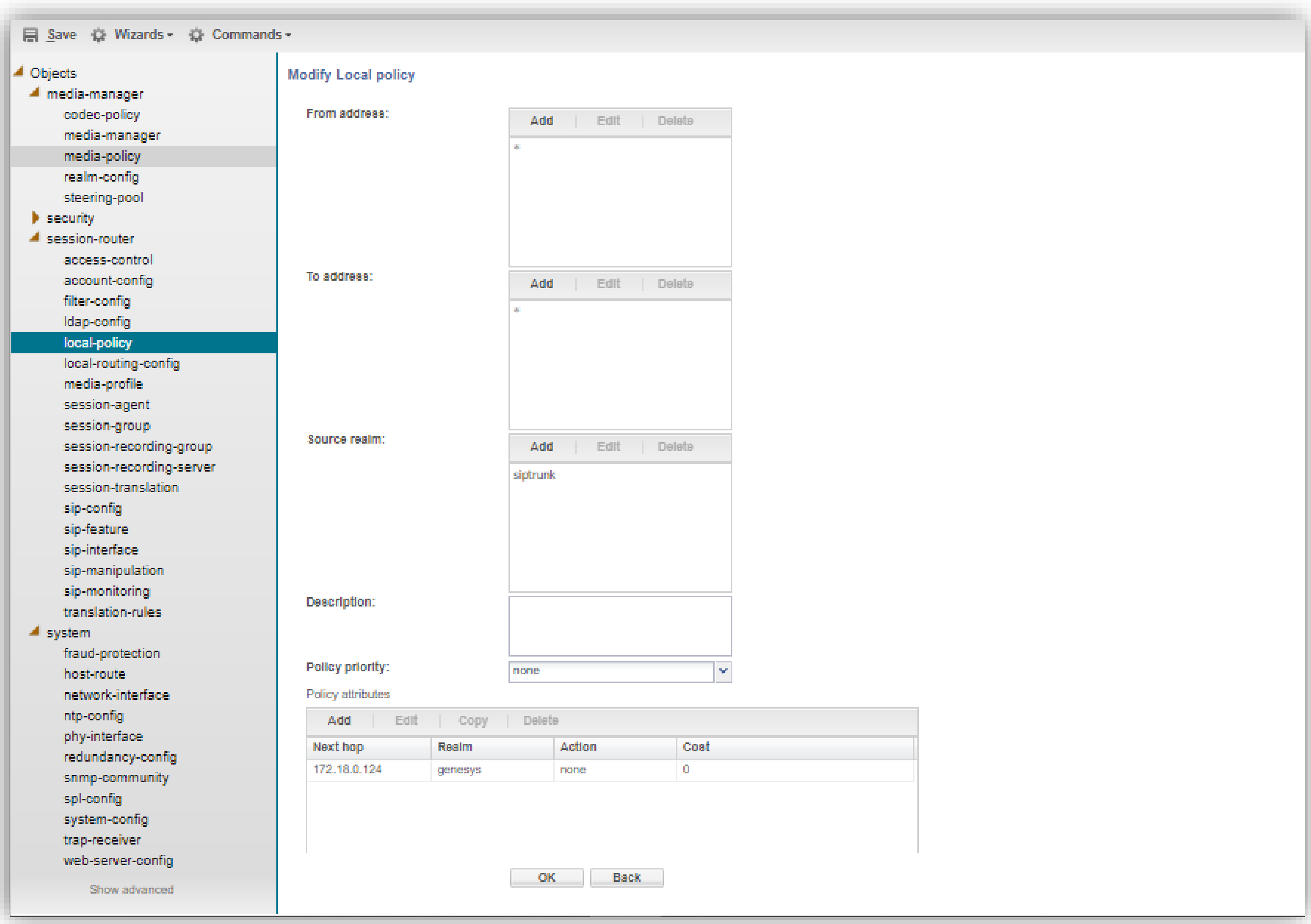
- The Local Policy mechanism provides SIP signaling routing based on:

  Ingress realm
  Calling and/or called number pattern
  Route priority (cost and availability time)
- Multiple local policies can be (and typically are) created.
- The Local Policy configuration element contains:

  Matching criteria
- Zero or more "policy-attributes" sub elements, each of which defines a "route"

To configure local-policy, navigate to **session-router->local-policy**. Configure the required local policy to route the calls.

Below is an example from the test bed for the local-policy configuration. Here From address and To address * denotes calls coming from any number to any called number should be forwarded to the mentioned destination in the next hop parameter.

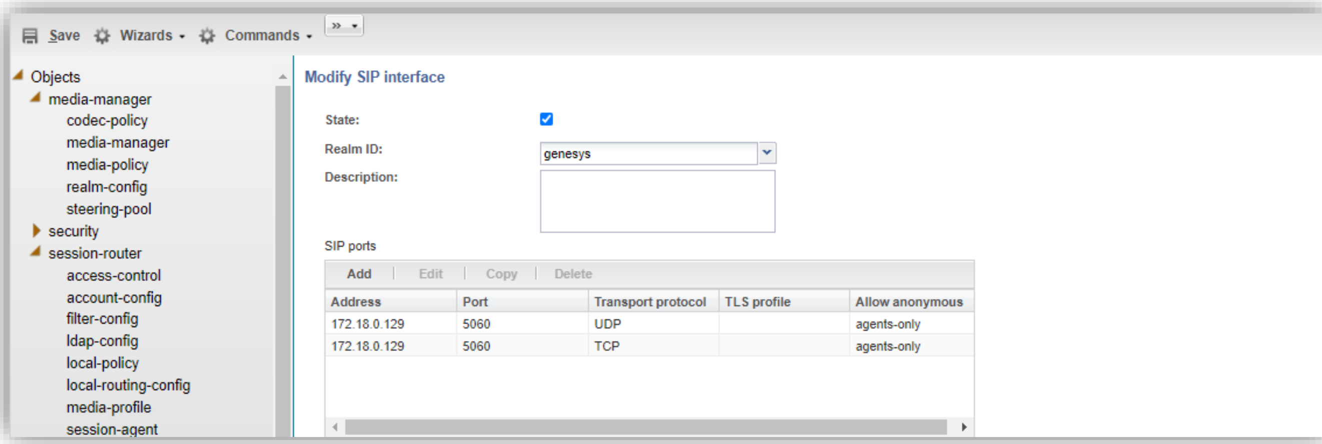


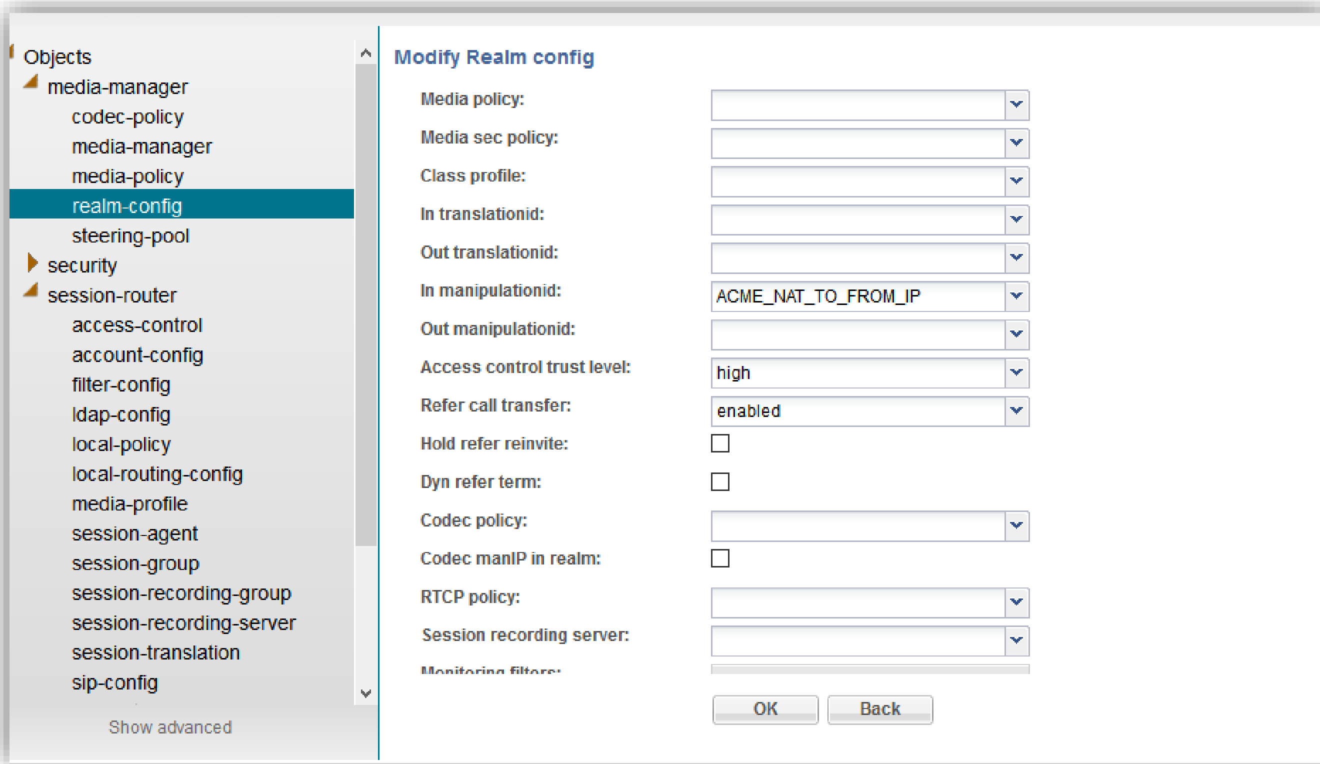
5.12. Header manipulation rule.

The following system-default Header manipulation rule is automatically applied on Genesys sip-interface involved in the test bed as an out-manipulationid.

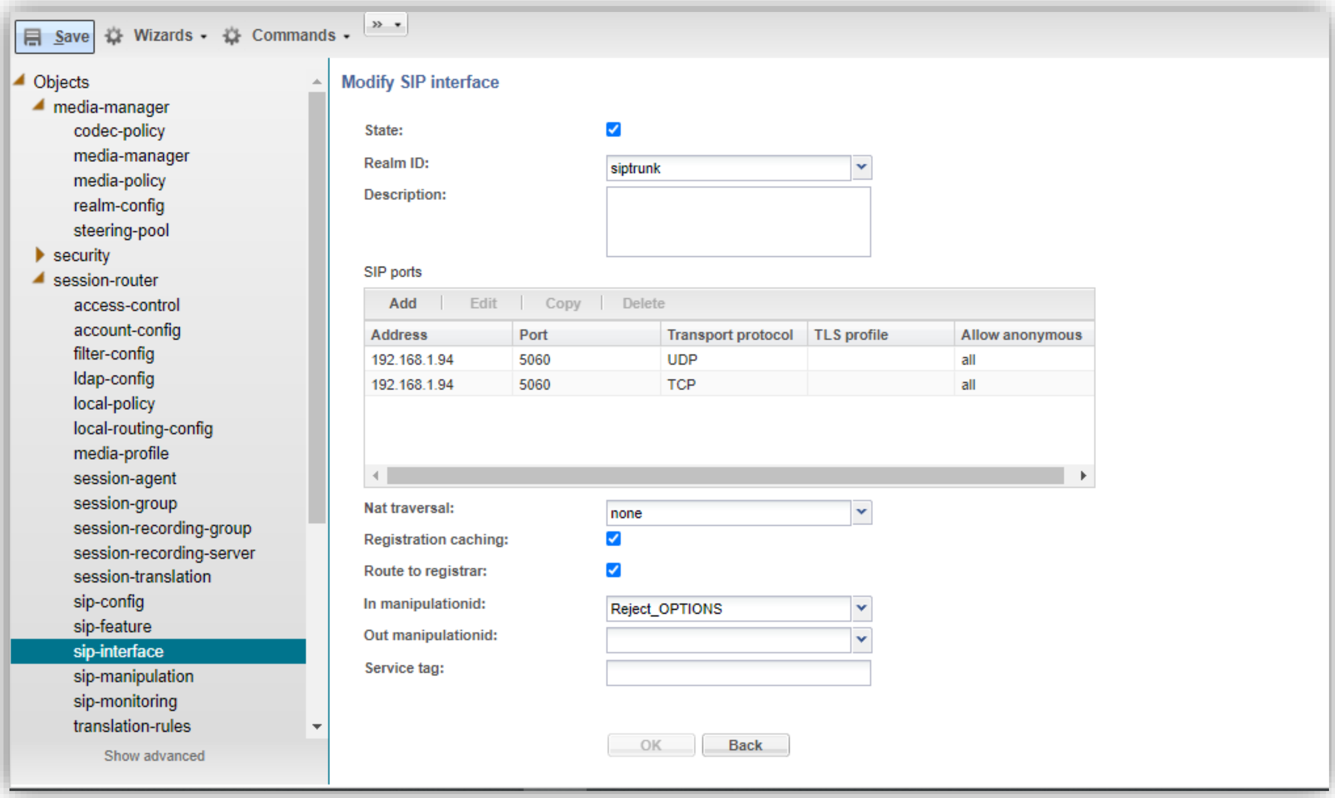
This HMR is used for topology hiding onto the SBC and it updates Contact and From host portion with SBC outside sip-interface IP address.

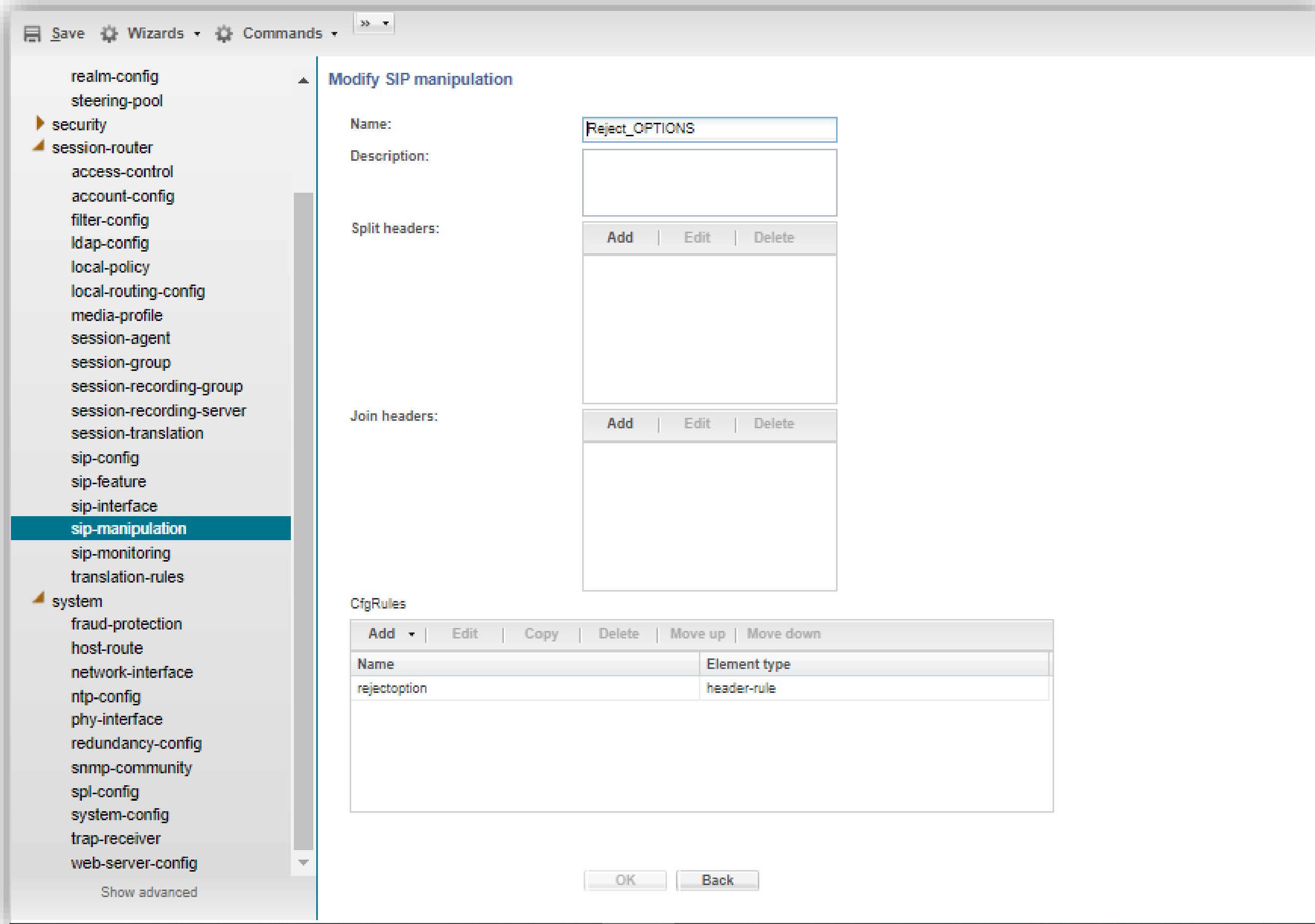
ACME_NAT_TO_FROM_IP

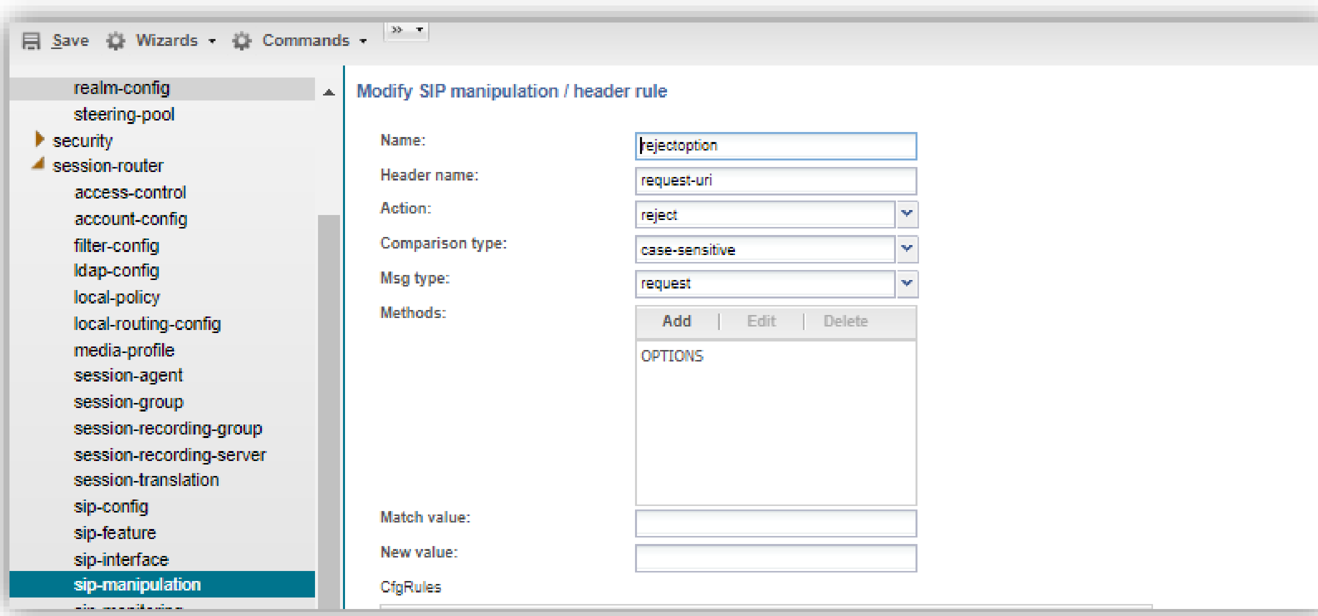




Another HMR Reject_OPTIONS is created and applied on the siptrunk sip-interface to locally respond to the SIP OPTIONS message with a 200 OK by the SBC rather than forwarding them to the Genesys SIP Server.



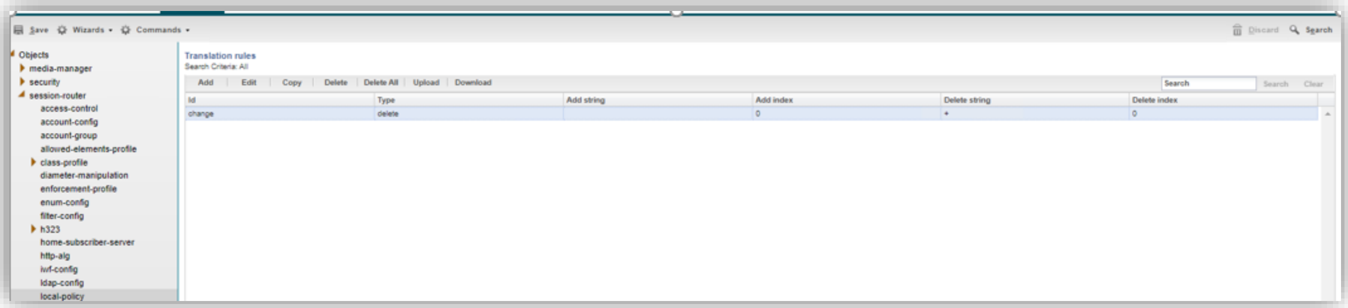

Save Wizards Commands
realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
sip-interface
sip-manipulation
sip-monitoring
translation-rules
system
fraud-protection
host-route
network-interface
ntp-config
phy-interface
redundancy-config
snmp-community
spl-config
system-config
trap-receiver
web-server-config
Show advanced
Modify SIP manipulation
Name:
Reject_OPTIONS
Description:
Split headers:
Add Edit Delete
Join headers:
Add Edit Delete
CfgRules
Add Edit Copy Delete Move up Move down
Name
Element type
rejectoption
header-rule
OK
Back

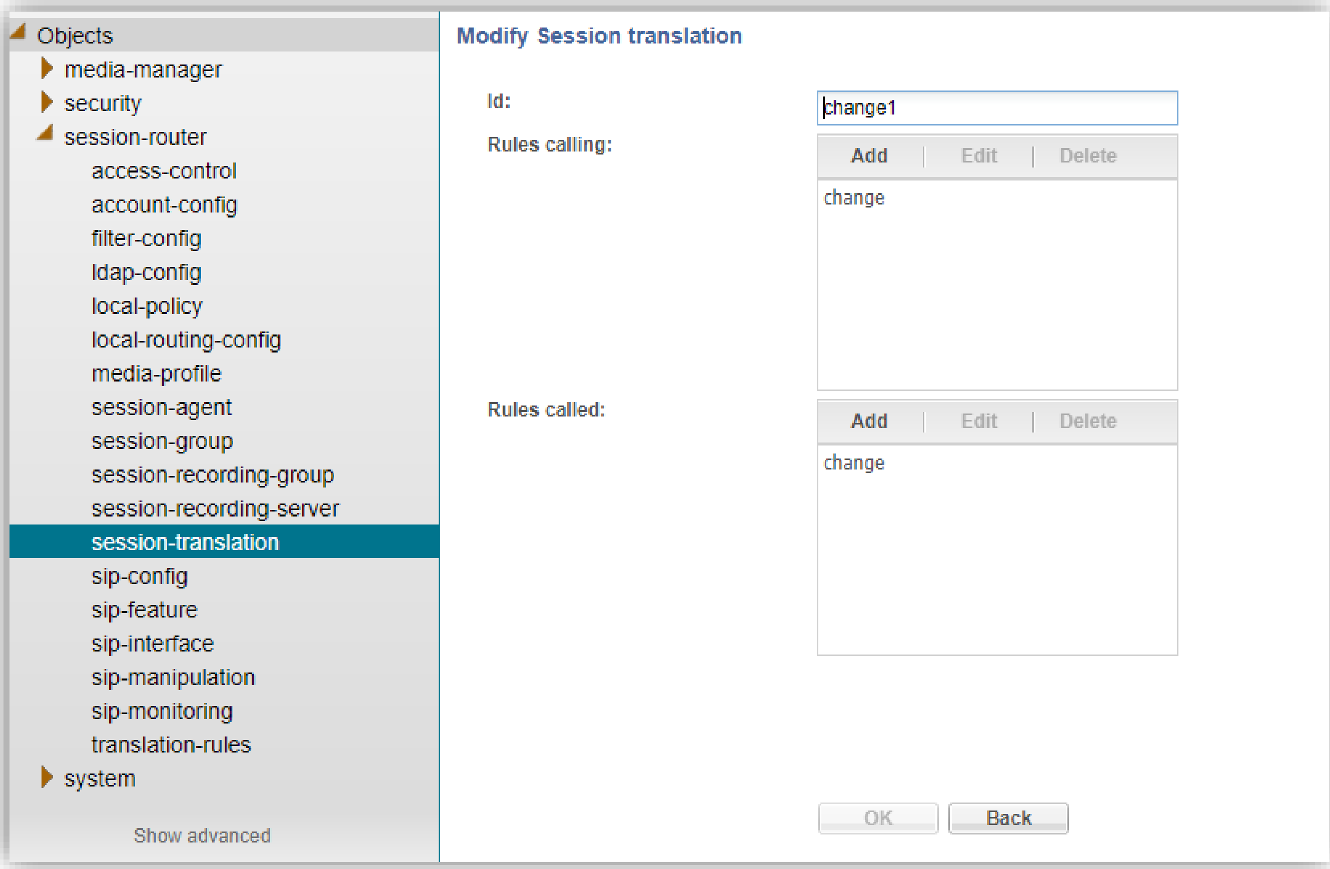

Save Wizards Commands
realm-config
steering-pool
security
session-router
access-control
account-config
filter-config
ldap-config
local-policy
local-routing-config
media-profile
session-agent
session-group
session-recording-group
session-recording-server
session-translation
sip-config
sip-feature
sip-interface
sip-manipulation
Modify SIP manipulation / header rule
Name:
rejectoption
Header name:
request-uri
Action:
reject
Comparison type:
case-sensitive
Msg type:
request
Methods:
Add Edit Delete
OPTIONS
Match value:
New value:
CfgRules

5.13 Session translation Rule

The following session-translation rule is configured on the SBC which strips the '+' from the called number of the request-uri as the numbers are defined without + on the SIP Server.

The session translation rule is applied as out-translationid on the genesys Realm.





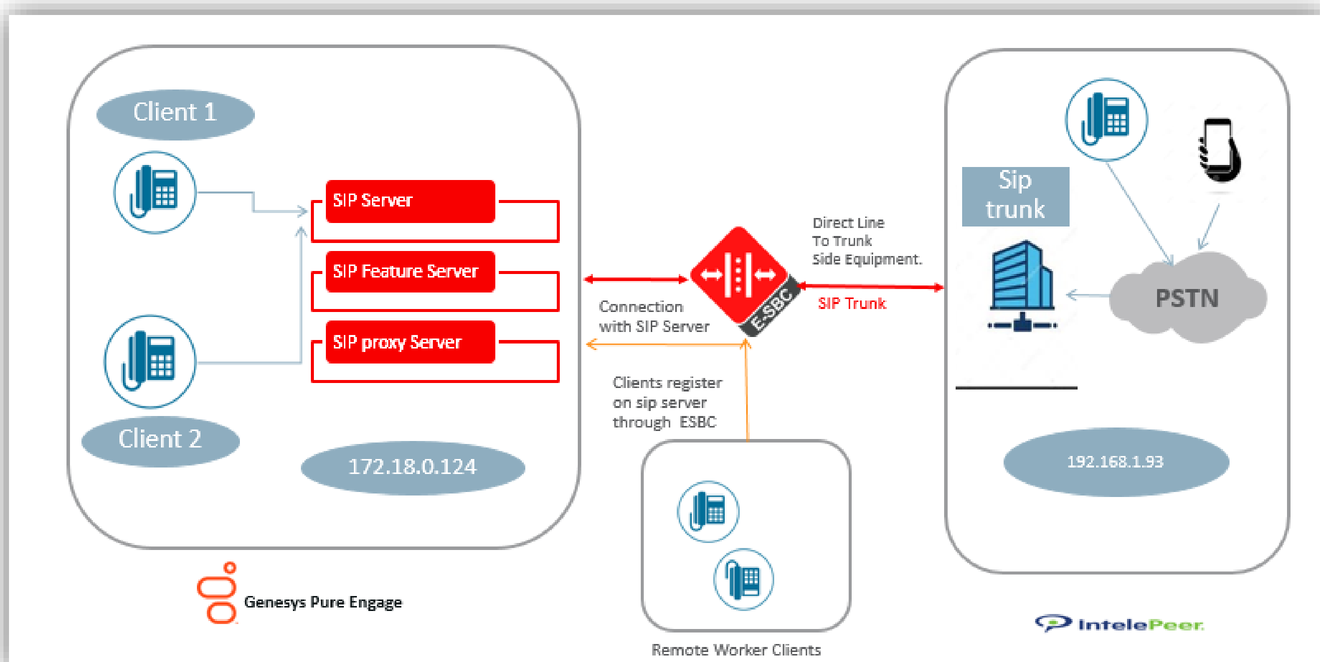
A basic configuration on the Oracle SBC to route calls to and from Genesys server environment is now complete. The following sections highlight some of the useful tips to configure the Oracle SBC in order to successfully resolve and overcome interoperability challenges in a SIP trunking environment between the Genesys SIP Server and Service provider network. It is outside the scope of this document to include all the interoperability working information as it will differ in every deployment.

6. Enabling Remote worker (for remote workers registering into Genesys SIP server via the Oracle SBC)

A section of the testing also included remote endpoints that register through the Oracle SBC to the SIP server. 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 endpoints which register onto the SIP Server through the SBC.SBC terminates the call to the number based on the registration information present in the cache.

Below figure illustrates how remote workers register onto the SIP Server via the SBC



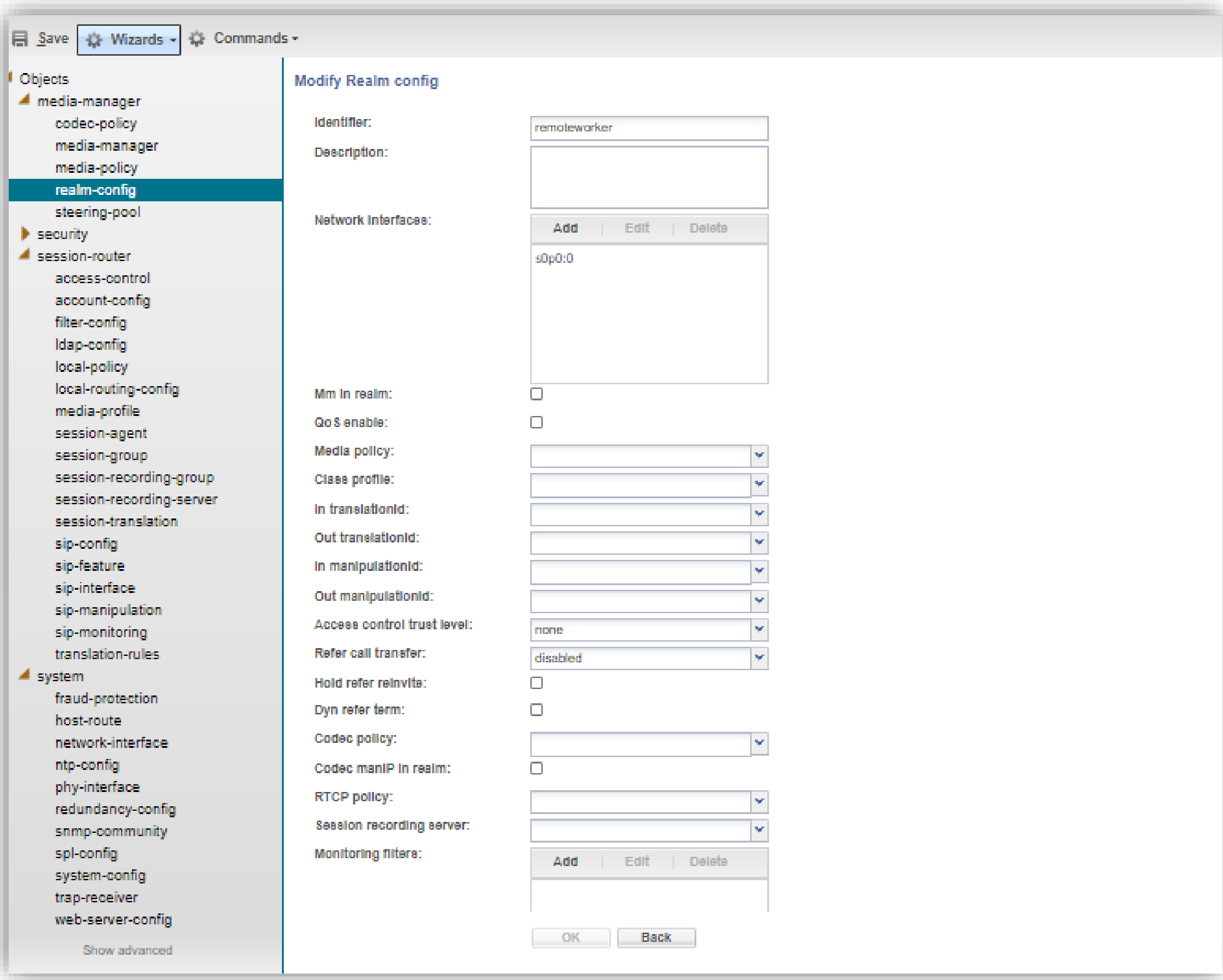
In order to achieve the requirement we have made below configuration on the Oracle SBC

Realm – remoteworker
Steering Pool associated with the Realm remoteworker
Sip-interface associated with the Realm remoteworker
(Optional) A local-policy to route the registration requests from this Realm to the SIP Server.

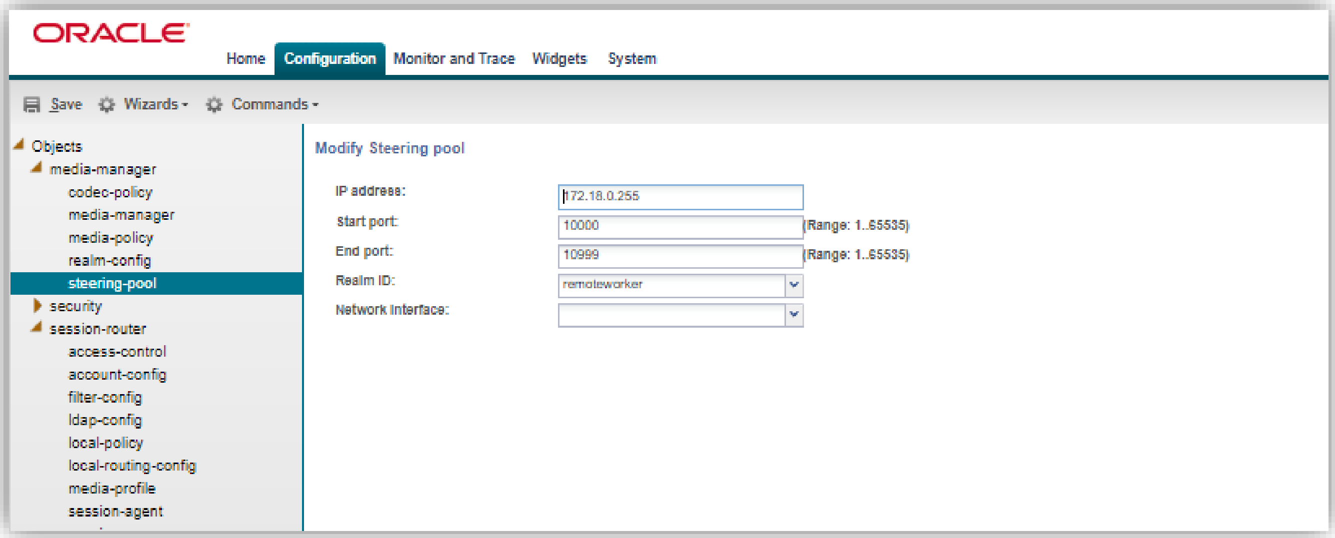
Note - The local-policy element is optional as we can enable the Route to registrar parameter on the sip-interface config to route the requests to the Registrar. The registrar host and port is configured in the sip-config element on the SBC.

The remote endpoint sends register requests from Genesys Realm onto the SBC and then SBC registers these endpoints onto the SIP Server maintaining the registration cache in its database to route inbound calls to these endpoint. Below are the snippets from the Oracle SBC WebGUI for the remote worker configuration.

6.1 Realm ‘remoteworker’

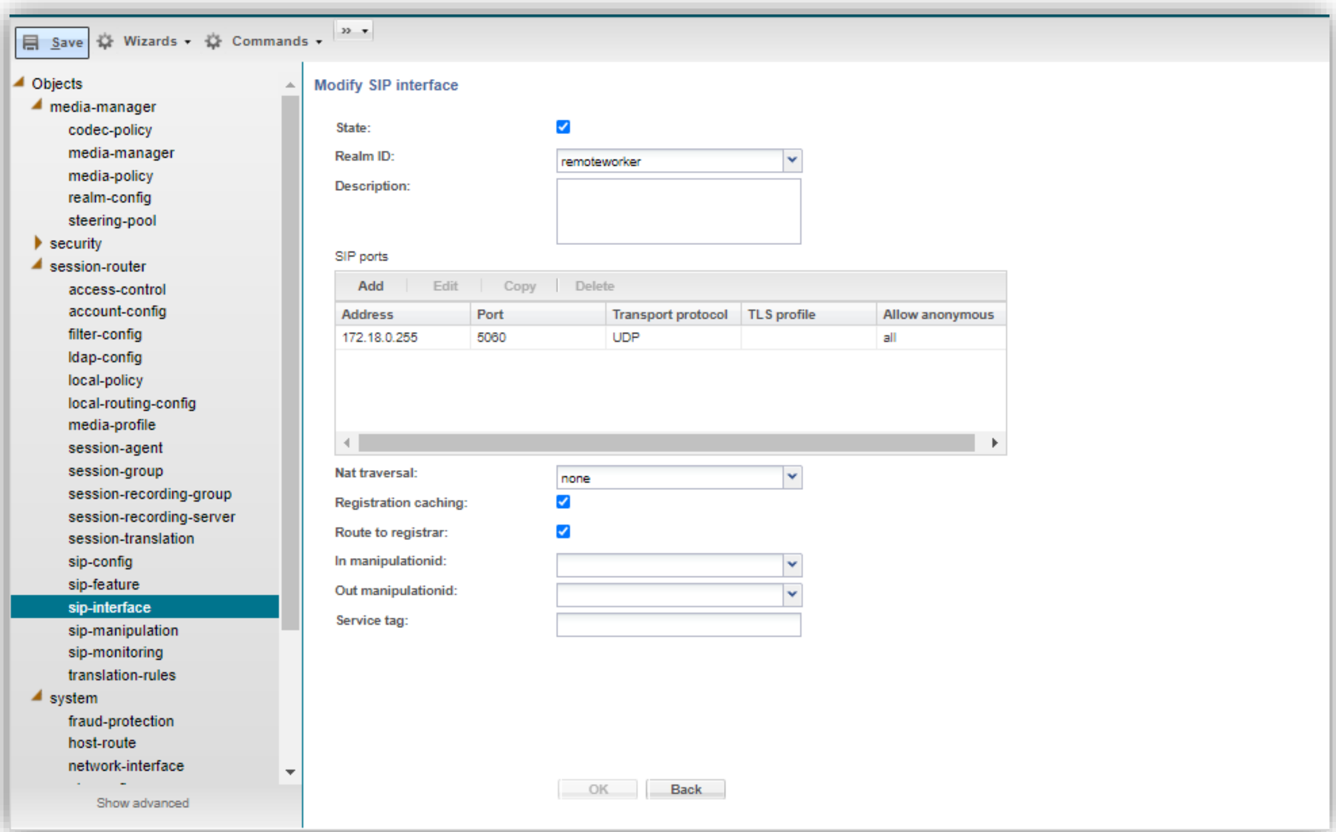


6.2. Steering Pool associated with realm remoteworker.

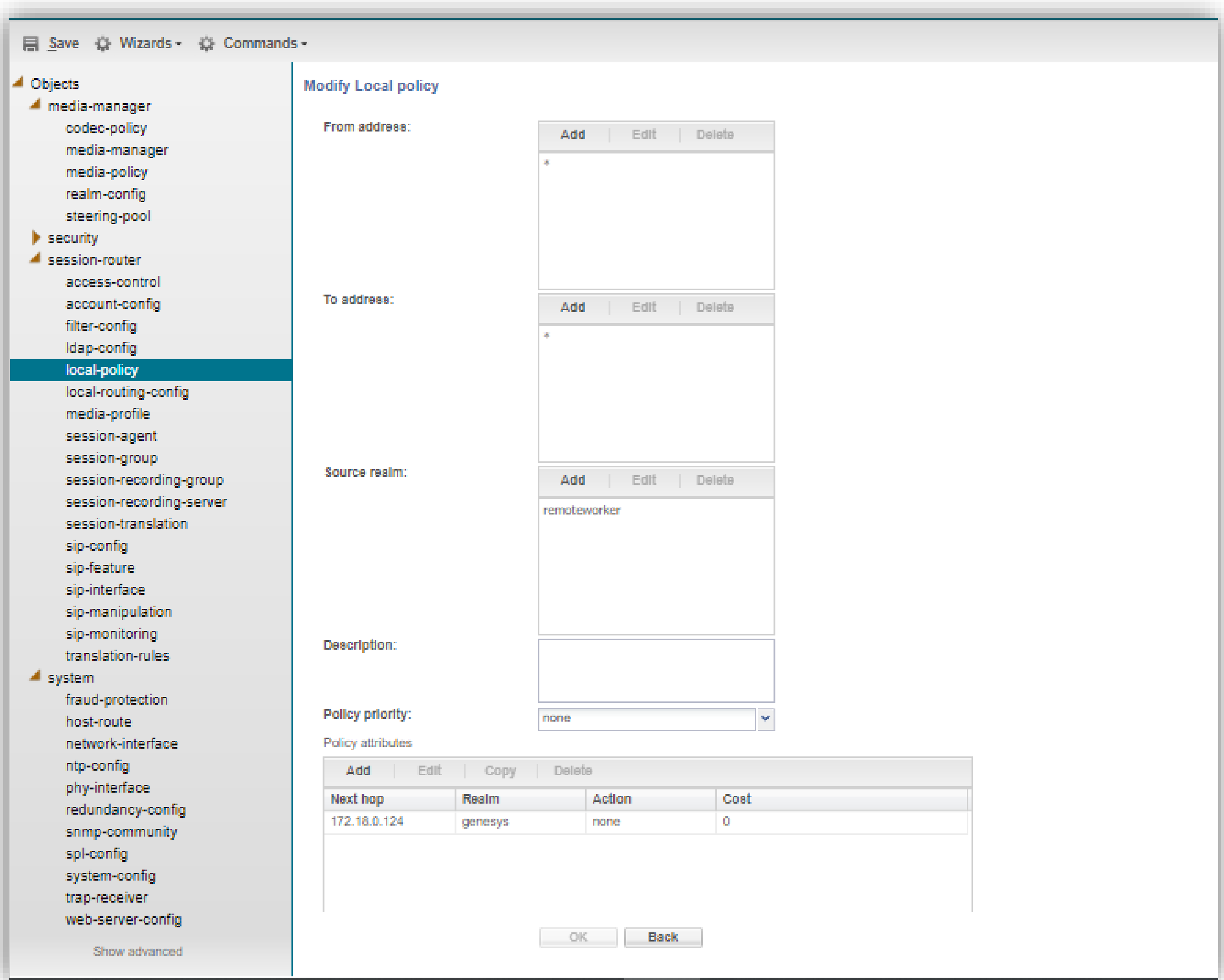


6.3 Sip-interface associated with realm remoteworker.

Registration caching must be enabled on this sip-interface so that SBC caches the registration of the subscriber which register through this sip-interface.



6.4 Local-policy



7. Test cases requiring authentication.

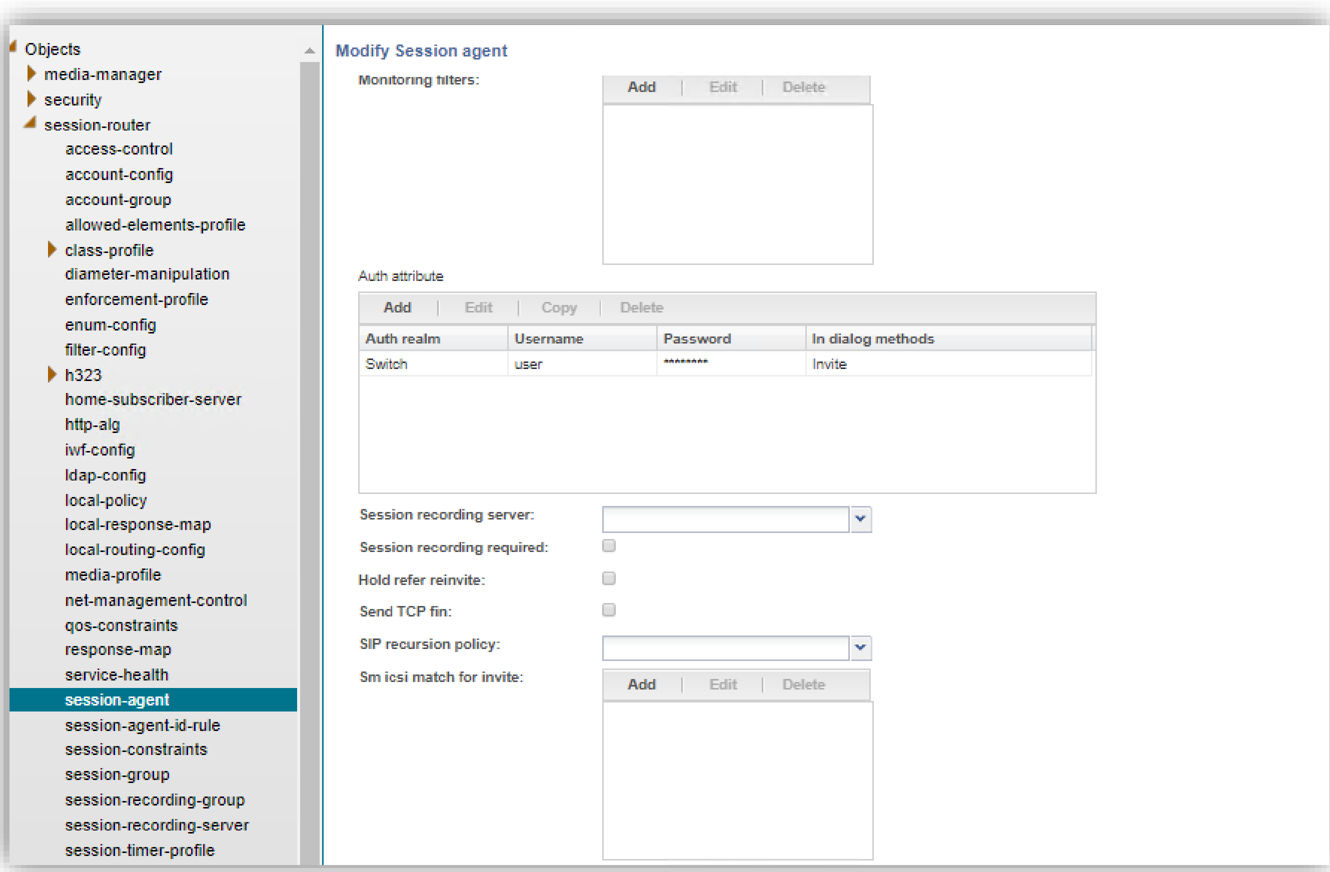
There are two test cases that require SIP Digest authentication.

[SIP Authentication for outbound calls](#)
[SIP Authentication for incoming calls](#)

The SIP Server is configured to challenge the identity of SBC when SBC sends a SIP INVITE to the SIP Server DN configured to demand authentication.

The inbound call made from PSTN to that DN. SIP Server send challenges to SBC by sending a 401 unauthorized message to the SBC.SBC further responds with a new INVITE based on the authentication attributes configured on the Session-agent. There is no configuration required for Outbound calls from Genesys SIP server.

In order to achieve the required configuration and pass the test scenarios we have configured below parameters onto the SBC for the SIP Trunk Session-agent.



8. Test Plan Executed.

8.1 Equipment Requirements

Table below identifies equipment used for testing

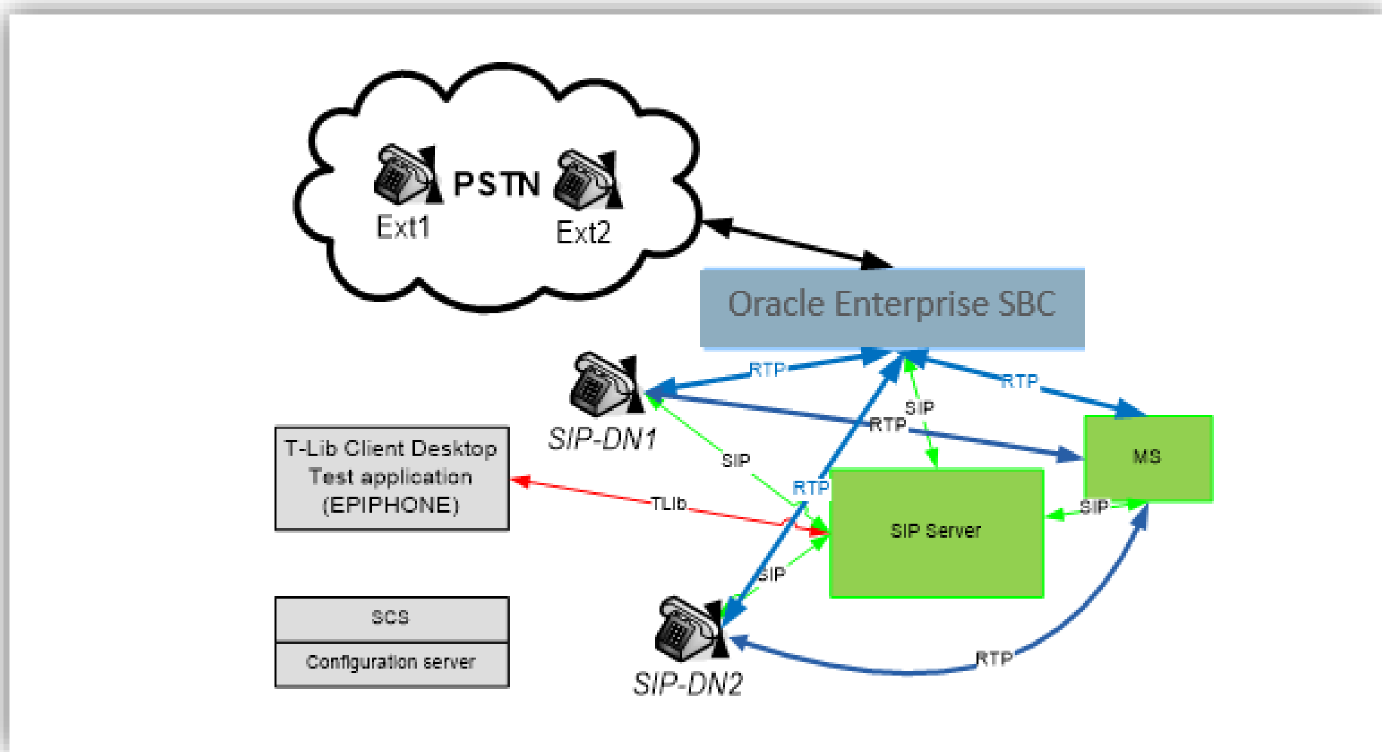
Product	Version	Units	Notes
SIP Server	8.1.1	1	Standalone deployment
Resourse manager (RM)	9.0	1	
Media Control Platform (MCP)	9.0	1	
Genesys Management Framework	8.1	1	
T-Lib Client Desktop Test application		1	EpiPhone application
Oracle SBC SCZ830m1p2 or above		1	
SIP End Point		3	
PSTN phone		2	

Execution of test plan requires having PSTN phones and Oracle SBC to be configured with Genesys. Below figure illustrates the setup required for test bed.

Two PSTN phones representing external numbers has to be configured for accessing Genesys SIP Server through Oracle SBC, and be accessible as outbound destination from SIP Server.

2 SIP phones are configured with Sip Server as local Sip Endpoints. Genesys EpiPhone test application is configured to simulate Agent Desktops for 2 local Sip Endpoints.

Genesys EpiPhone is used to issue 3PCC Apply Treatment and Route requests. Internal SIP endpoints can be registered at SIP Server or provisioned. EpiPhone is a test tool for testing Genesys SIP Server. It provides functionality of Genesys T-Library GUI client with call/parties visualization and 3rd party call control. From EpiPhone GUI it is possible to perform all 3PCC requests required for execution of current test plan. This include Treatment request and Route requests, thus we don’t need to include URS in the testing environment.



8.2 Default Sip Server Options

The default Sip Server Options configuration is as below. Configuration changes will be required on the Genesys SIP Server Trunk, DN objects as per the test case requirement in order to pass the test scenarios.

SIP Server Application Options TServer section
SIP-hold-rfc3264=true router-timeout=30 default-dn= blind-transfer-enabled=true resource-management-by-rm=true msml-support=true sip-enable-moh=true

8.3 Sample EPIphone configuration

Below is the sample EPIphone configuration from the Test Bed. Here the DN's and Route Points are configured for the SBC Trunk. Please note the below configuration is just for reference as it will change with respect to each environment.

```
[HOME]
server = (host="${loc_host_ip}", port=${loc_tserv_port})
sip-proxy = ${loc_host_ip}:${loc_sip_port};transport=udp
sip-register = true
dn1 = 100001, sip-port = ${sip_port_dn1} , sip=simple, play=DN1, [AA] on-invite = 486
dn2 = 100011, sip-port = ${sip_port_dn2} , sip=simple, play=DN2
dn3 = 100021, sip-port = ${sip_port_dn3} , sip=simple, play=DN3
dn4 = 17814437266, sip-port = ${sip_port_dn4} , sip=simple, play=DN4
dn5 = 100041, sip-port = ${sip_port_dn5} , sip=simple, play=DN5
dn10 = 17814437285, pool="shared"
dn11 = 9001, pool="shared",script="annc=(PROMPT=(\"1\"=(INTERRUPTABLE=1,ID=1)))"
dn12 =  9002,
pool="shared",script="collect=(MAX_DIGITS=4,RESET_DIGITS=11,BACKSPACE_DIGITS=22,TOTA
L_TIMEOUT=1000) annc=(PROMPT=(ID=1))"
```

8.3 Test Plan executed

The following Test Plan has been executed against this setup and results are documented below.

Scenario	Supported
Inbound Call to Agent released by caller	Yes
Inbound Call to Agent released by agent	Yes
Inbound Calls rejected	Yes
Inbound Call abandoned	Yes
Inbound Call to Route Point with Treatment	Yes
Interruptible Treatment	Yes
IVR (Collect Digit) Treatment	Yes
Inbound Call routed by using 302 out of SIP Server signaling path	Yes
1PCC Outbound Call from SIP Endpoint to external destination	Yes
3PCC Outbound Call to external destination	Yes
1PCC Outbound Call Abandoned	Yes
Caller is put on hold and retrieved by using RFC 2543 method	Yes
T-Lib-Initiated Hold/Retrieve Call with MOH using RFC 3264 method	Yes
3PCC 2 Step Transfer to internal destination by using re-INVITE method	Yes
3PCC Alternate from consult call to main call	Yes
1PCC Unattended (Blind) transfer using REFER	Yes
1PCC Attended Transfer to external destination	Yes
3PCC Two Step Conference to external party	Yes
3PCC (same as 1PCC) Single-Step Transfer to another agent	Yes
3PCC Single Step Transfer to external destination using REFER	Yes
3PCC Single Step Transfer to internal busy destination using REFER	Yes
Early Media for Inbound Call to Route Point with Treatment	Yes
Early Media for Inbound Call with Early Media for Routed to Agent	Yes
Inbound call routed outbound (Remote Agent) using INVITE without SDP	Yes
Call Progress Detection	Yes
Out of Service detection Checking MGW live status	Yes
SIP Authentication for outbound calls	Yes
SIP Authentication for incoming calls	Yes
T-Lib-Initiated Answer/Hold/Retrieve Call for Remote SIP endpoint which supports the BroadSoft SIP Extension Event Package	Yes
3PCC Outbound Call from Remote SIP endpoint to external destination	Yes
3PCC 2 Step Transfer from Remote SIP endpoint to internal destination	Yes
1PCC Attended Transfer from Remote SIP endpoint to external destination	Yes

9. Caveats

Oracle SBC does not support CPD Call Progress Detection, The functionality is available on the Genesys SIP server where Media Server (Genesys) detects the CPD and sends the result to SIP Server.

ORACLE

Oracle Corporation, World Headquarters
500 Oracle Parkway
Redwood Shores, CA 94065, USA

Worldwide Inquiries
Phone: +1.650.506.7000
Fax: +1.650.506.7200

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