

ATT IP Flexible Reach Service Including
MIS/PNT/AVPN Transports with Avaya
Session Manager 6.3.15 & Oracle Session
Border Controller 6300

Technical Application Note




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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 (E-SBC) as well as service provider based session border controller. It assumes that the reader is familiar with basic operations of the Oracle Session Border Controller 3820/4500 and 6300 platforms.

Document Overview

This Oracle technical application note outlines the recommended configurations for the Oracle enterprise session border controller 6300 series for connecting AT&T's IP Flexible Reach service to Avaya Session Manager (SM) version 6.3.15 customers. The solution contained within this document has been certified on Oracle's Acme Packet OS ECZ 7.2p6

Avaya Aura Session Manager is at the core of Avaya's SIP based architecture. Oracle Enterprise Session Border Controllers (SBCs) play an important role in SIP trunking as they are used by many ITSPs and some enterprises as part of their SIP trunking infrastructure.

This application note has been prepared as a means of ensuring that ATT's IP Flexible Reach SIP trunking between Avaya Aura Session Manager 6.3.15 and Oracle E-SBC and SBCs are configured in the optimal manner.

It should be noted that while this application note focuses on the optimal configurations for the Oracle SBC in an enterprise Avaya SM 6.3.15 environment, the same SBC configuration model can also be used for other enterprise SIP trunking 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 Avaya Aura SM 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.



Introduction

Audience

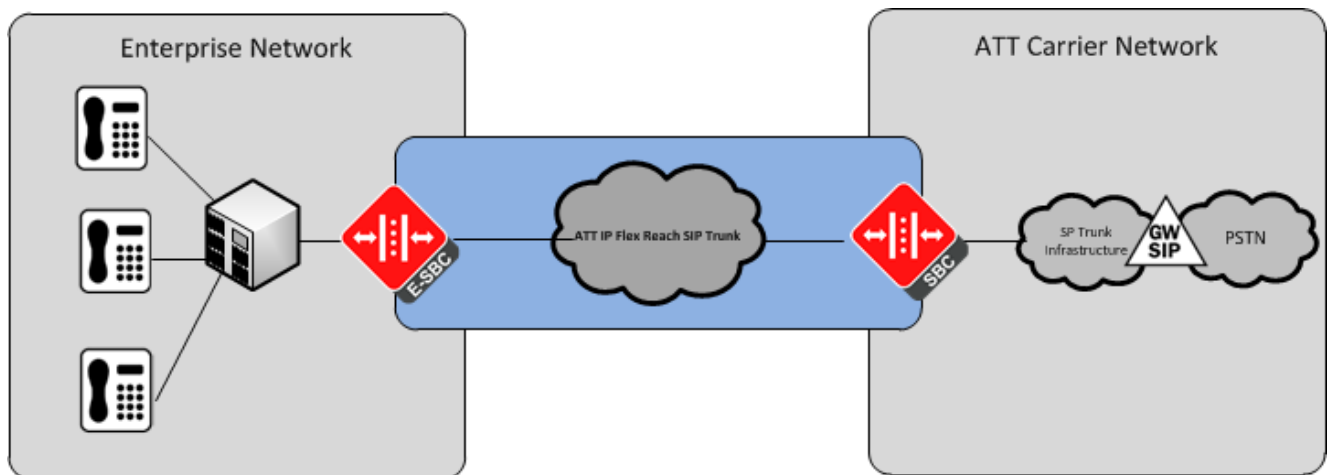
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise SBC and Avaya Aura SM 6.3.15. There will be steps that require navigating the Avaya SM and Communication manager server configuration as well as the Acme Packet Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

Requirements

- Working configuration and functioning Avaya SM 6.3.15
- Avaya communication manager configuration
- Asterisk server configuration for Voicemail and Auto Attendant based enterprise functions
- Avaya One-X communicator soft phone and hard phones connected/registered to the Session manager server
- Oracle Session Border Controller (hereafter SBC) 6300 series running ECZ7.2.0p6. Note: the configuration running on the SBC is backward/forward compatible with any release in the 7.2.0 stream
- Oracle SBC in SIP Peering configuration having established SIP connectivity with Avaya SM on CPE side and ATT IP FR SIP trunk on PSTN side.

Architecture

The following reference architecture shows a logical view of the connectivity between Avaya SM and the SBC.

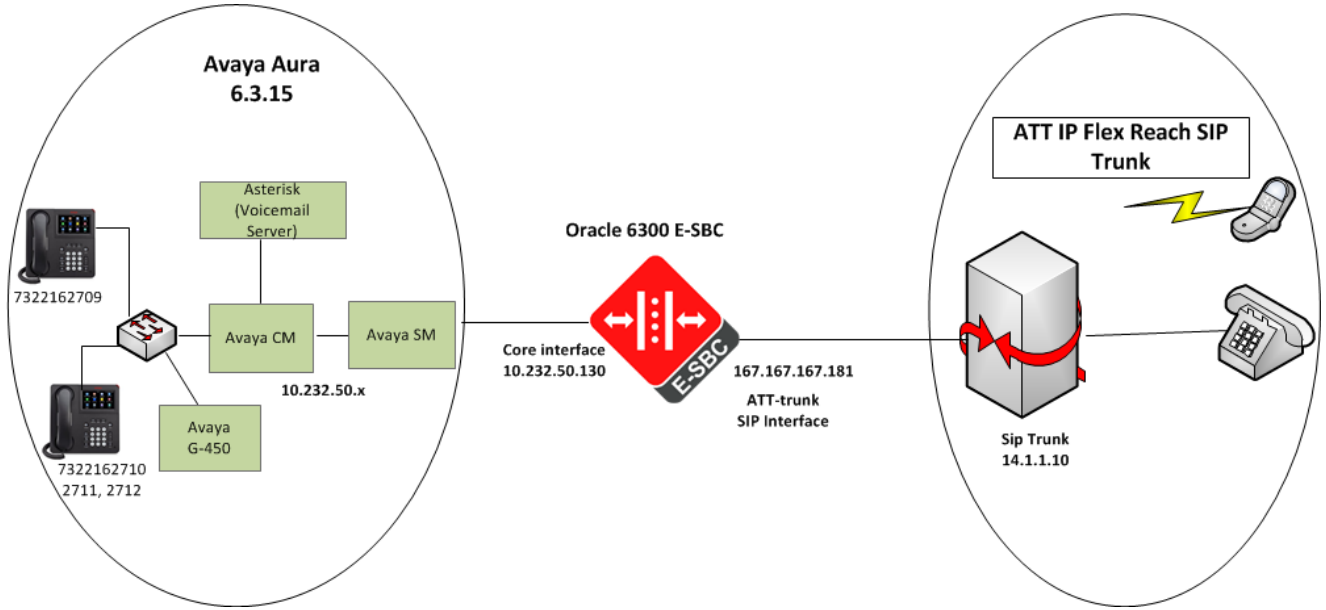


Area on left of the Oracle SBC brown box is the customer's on premise infrastructure, which includes the Avaya Aura SM, PBX/CM with the enterprise phones systems. Area on right of the SBC represents the service provider infrastructure which provides PSTN service via the SIP trunk. The SBC provides integration of these two environments over an IP network and provides security, service reachability, interoperability/normalization of SIP messages over the IP network. The Avaya SM and SBC are the edge components that form the boundary of the SIP trunk. The configuration, validation and troubleshooting of these two is the focus of this document and will be described in three phases:

- Phase 1 – Configuring the Oracle Enterprise SBC
- Phase 2 – Configuring the Avaya Aura Session Manager and adaptations
- Phase 3 - Configuring the Avaya Aura Communication Manager

Lab Configuration

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





Phase 1 – Configuring the Oracle Enterprise SBC

In this section we describe the steps for configuring an Oracle Enterprise SBC, formally known as an Acme Packet Net-Net Session Director (“SBC”), for use with Avaya Aura Session Manager 6.3.15 in an ATT IP Flex Reach SIP Trunk service.

In Scope

The following guide configuring the Oracle E-SBC assumes that this is a newly deployed Avaya Aura session manager 6.3.15 topology in an enterprise dedicated to a single customer.

Note that Oracle offers several models of SBC. This document covers the setup for the 6300 platform series running Net-Net OS ECZ7.2.0p6 or newer. If instructions are needed for other Oracle SBC models, please contact your Oracle representative.

Out of Scope

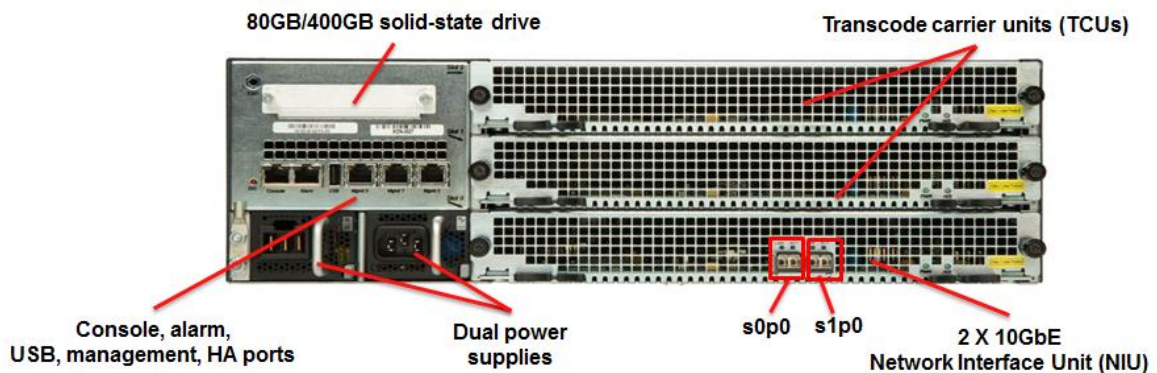
- Configuration of Network management including SNMP and RADIUS

What you will need

- Serial Console cross over cable with RJ-45 connector
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the Oracle E-SBC
- IP address to be assigned to management interface (Wancom0) of the SBC - the Wancom0 management interface must be connected and configured to a management network separate from the service interfaces. Otherwise the SBC is subject to ARP overlap issues, loss of system access when the network is down, and compromising DDoS protection. Oracle does not support SBC configurations with management and media/service interfaces on the same subnet.
- IP address of the Avaya Aura SM SIP interface facing the SBC
- IP addresses to be used for the SBC core (Avaya facing) and SIP trunk/ATT network facing SIP interfaces
- IP address of the next hop gateway in the ATT IP Flex Reach network
- IP address of the enterprise DNS server

Configuring the E-SBC

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



As seen in the above picture, the 6300 platform has a field replaceable 2 x 10 Gb/sec NIU that contains two 10 Gb/sec Ethernet fiber ports with enhanced Small form factor pluggable (SFP+) for short and long reach options. Plug the slot 0 port 0 (s0p0) 10G interface into your outside (ATT next-hop facing) network and the slot 1 port 0 (s0p1) interface into your inside (Avaya) network. Once connected, you are ready to power on and perform the following steps.

All commands are in bold, such as **configure terminal**; parameters in bold red such as **Avaya-SBC-ATT** are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

Establish the serial connection and logging in the E-SBC

Confirm the SBC is powered off and 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 bootup sequence.

```
COM3 - PuTTY
Starting tEbmd...
Starting tSipd...
Starting tLtd...
Starting tH323d...
Starting tH248d...
Starting tBgfd...
Starting tSecured...
Starting tAuthd...
Starting tCertd...
Starting tIked...
Starting tauditd...
Starting tauditpusher...
Starting tSnmpd...
Start platform alarm...
Initializing /ramdrv Cleaner
Starting tLogCleaner task
Bringing up shell...
password secure mode is enabled
Admin Security is disabled
Starting SSH...
SSH_Cli_init: allocated memory for 5 connections
acl: max telnet sessions: 5
Password: 0x21a059c8 (tAlarm): eth0: Link is up (1000Mb/s full duplex)
```

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

```
Password: acme
Avaya-SBC-ATT> enable
Password: packet
Avaya-SBC-ATT# configure terminal
Avaya-SBC-ATT(configuration)#
```

You are now in the global configuration mode.

Initial Configuration – Assigning the management Interface an IP address

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

Avaya-SBC-ATT#configure terminal --- >bootparams

- Once you type “bootparam” you have to use “carriage return” key to navigate down
- A reboot is required if changes are made to the existing bootparams

```
Avaya-SBC-ATT#(configuration)bootparam
'.' = clear field; '-' = go to previous field; q = quit
boot device          : eth0
processor number     : 0
host name            : acmesystem
file name            : /boot/EZ720p6.64.bz --- >location where the
software is loaded on the SBC
inet on ethernet (e) : 172.18.255.104:ffffff80 --- > This is the ip
address of the management interface of the SBC, type the IP address and
mask in hex
```

```

inet on backplane (b)      :
host inet (h)              :
gateway inet (g)           : 172.18.0.1 --- > gateway address here
user (u)                   : vxftp
ftp password (pw) (blank = use rsh) : vxftp
flags (f)                  :
target name (tn)           : Avaya-SBC-ATT
startup script (s)         :
other (o)                  :

```

The following section walks you through configuring the Oracle Communications Enterprise SBC configuration required to work with Avaya Aura Session Manager v6.3.15 and ATT's IP Flex Reach SIP Trunk service. In the configuration, the transport protocol used between the SBC and Avaya SM server is TCP and the SIP trunk is configured for UDP. The test plan requires G.729 codec as first offered and calls on trunk unless where specified, therefore the configuration describing that is included as well.

It is outside the scope of this document to include all the interoperability working information as it will differ in every deployment.

High Availability

The wancom1 and wancom 2 port which is on the rear panel of the SBC is used for the purpose of High Availability in the 6300. Please refer to the Oracle Enterprise Session Border Controller ECZ7.2.0 ACLI Configuration guide for more detailed update on High availability configuration. (http://docs.oracle.com/cd/E61547_01/index.htm)

The following section entails notable configuration highlights that pertain to an enterprise environment that deploys an Oracle E-SBC and Avaya Aura v6.3.15 to work with ATT IP Flex Reach SIP trunk service. A full copy of the configuration that was used for this certification follows the section as well.

Configuration Highlights

The SBC configuration in general follows enterprise SIP trunk configuration, with a few additional elements specific to interworking with Avaya SM. These are outlined below. We have also configured some options and sip-manipulations which are specific to any Avaya deployment with the Oracle Enterprise SBC. They are explained below

Avaya Session Manager Session Group

We define a session agent group containing both the session managers in the SBC and enable recursion to ensure that the call would complete if either of the SMs is online.

```

session-group
  group-name          Avaya-SM-SAG
  description         Avaya SMs
  state               enabled
  app-protocol        SIP

```

strategy	Hunt
dest	10.232.50.102
	10.232.50.112
trunk-group	
sag-recursion	enabled
stop-sag-recurse	401,407

To give a glimpse of how the routing would work, here is an example of the local-policy (routing configuration) from the ATT trunk side to the Avaya SM.

local-policy	
from-address	*
to-address	*
source-realm	ATT-Trunk
description	
activate-time	
deactivate-time	
state	enabled
policy-priority	none
policy-attribute	
next-hop	SAG:Avaya-SM-SAG
realm	Core
action	replace-uri
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
state	enabled
app-protocol	SIP
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	

SIP Manipulations

The Oracle E-SBC helps resolve certain SIP interoperability issues and preferences on ATT trunk side by invoking one of its most strongest and robust features SIP Header Manipulation Rules (HMR). Below is a summary of the SIP manipulations used and their use cases in this project.

SIP HMR	Description
ChangePAI	Fixes P-Asserted-Identity header towards ATT Trunk
NATting	Provides topology hiding
fordelavayaheaders	Strips Avaya proprietary headers towards ATT trunk
fordiv	Adds Diversion header for Call forwarding Unconditional scenario
ForREFER	Changes Refer-To header and RURI in Refer-To to have appropriate call transferee information in case of Call transfer test cases
ChangeforPAIandNAT	Consolidated one outbound HMR applied on ATT trunk facing SIP interface of SBC - PAI HMR, adds diversion header for CFU, provides topology hiding, deletes Avaya/enterprise proprietary headers and fixes REFERs
changedisplay	Modify From header and add Privacy header for Anonymous calls
ACME_NAT_TO_FROM_IP	Provides topology hiding, applied as outbound HMR on Avaya facing sip-interface of SBC

The header manipulation rules listed in the above table are further elaborated:

Setting the correct P-Asserted-Identity and ensuring topology hiding in place

This sip-manipulation is applied as out-manipulationid on the ATT trunk facing sip-interface in E-SBC. ATT IP Flex Reach SIP trunk service requires the CPE to set P-Asserted-Identity and Remote-party-ID headers correctly to reflect SBC IP.

```
sip-manipulation
  name          ChangePAI
  description
  split-headers
  join-headers
```

```
header-rule
    name Storecontacthost
    header-name Contact
    action store
    comparison-type pattern-rule
    msg-type any
    methods INVITE
    match-value
    new-value
    element-rule
        name storehost
        parameter-name
        type uri-host
        action store
        match-val-type any
        comparison-type pattern-rule
        match-value
        new-value

header-rule
    name ModPAI
    header-name P-Asserted-Identity
    action manipulate
    comparison-type boolean
    msg-type any
    methods INVITE
    match-value $Storecontacthost.$storehost.$0
    new-value
    element-rule
        name modhost
        parameter-name
        type uri-host
        action replace
        match-val-type any
        comparison-type pattern-rule
        match-value
        new-value $Storecontacthost.$storehost.$0
```

Topology Hiding

Since the SBC is a B2BUA and border element, one of the standard procedures in SIP trunk applications is to protect enterprise IP topology as well as for SIP trunk provider network as well. The below rule provides topology hiding by replacing host-portions in SIP URIs of From and To headers or outbound messages from the CPE.

```

sip-manipulation
  name                               NATting
  description
  split-headers
  join-headers
  header-rule
    name                               From
    header-name                         From
    action                               manipulate
    comparison-type                     case-sensitive
    msg-type                             any
    methods
    match-value
    new-value
    element-rule
      name
From_header
      parameter-name
      type                               uri-host
      action                             replace
      match-val-type                     any
      comparison-type                   case-
sensitive
      match-value
      new-value                          $LOCAL_IP
    header-rule
      name                               To
      header-name                         To
      action                               manipulate
      comparison-type                     case-sensitive
      msg-type                             any
      methods
      match-value
      new-value
      element-rule

```


	name	To
	parameter-name	
	type	uri-host
	action	replace
	match-val-type	any
	comparison-type	case-
sensitive	match-value	
	new-value	\$REMOTE_IP

Delete Avaya headers when sending messages to ATT network

Certain headers that are avaya related are not required to be sent to the ATT IPFR network, the E-SBC in this case uses sip manipulation rules to strip the headers

sip-manipulation		
	name	RemoveAvayaheaders
	description	remove avaya specific non-
important headers towards ATT	split-headers	
	join-headers	
	header-rule	
	name	delPAVMessageID
	header-name	P-AV-Message-Id
	action	delete
	comparison-type	case-sensitive
	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	header-rule	
	name	
delAVGlobalsessionId	header-name	AV-Global-Session-
ID	action	delete
	comparison-type	case-sensitive
	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	header-rule	

name	delPlocation
header-name	P-Location
action	delete
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	

Add Diversion Header for CFU scenario

For a particular test case where inbound call to enterprise requires to call forwarded out unconditionally (Call Forwarding unconditional), the SBC adds a Diversion header to inform ATT IPFR network of call diversion and inserting Inbound DID/CPE information as shown below

sip-manipulation	
name	AddDiversion
description	
split-headers	
join-headers	
header-rule	
name	checkfor800
header-name	To
action	manipulate
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	
element-rule	
name	
checuriuser	
parameter-name	
type	uri-user
action	store
match-val-type	any
comparison-type	pattern-
rule	
match-value	
18772427372	
new-value	
header-rule	

```

name addDiv
header-name Diversion
action add
comparison-type boolean
msg-type request
methods INVITE
match-value
$checkfor800.$checuriuser
new-value
<sip:7322162709@167.167.167.181>

```

Note: This test case has been marked as Conditional Passed as ATT IPFR trunk 1-800 test services were unavailable at the time of testing. The SIP manipulation rule will be applicable when testing and has been verified tested.

Refer message manipulation for Call transfer scenarios

For Network based call transfer scenarios, (Attended Consultative transfer and unattended consultative transfer), the Avaya SM sends Refer message with blank Refer-To header user portion as well as blank Request URI in the REFER. The E-SBC uses To header user information in the REFER message which indicates the call transferee, copies that and inserts it into the Refer-To header as well as the Request-URI of the REFER message. The complete SIP manipulation is given below:

```

sip-manipulation
name changeRefer
description
split-headers
join-headers
header-rule
name ModReferto
header-name Refer-To
action manipulate
comparison-type case-sensitive
msg-type any
methods REFER
match-value
new-value
element-rule
name
ChangeURIhost
parameter-name
type uri-host
action replace
match-val-type any

```

	comparison-type	case-
sensitive	match-value	
	new-value	\$LOCAL_IP
header-rule		
name		ModReferredBY
header-name		Referred-By
action		manipulate
comparison-type		case-sensitive
msg-type		any
methods		REFER
match-value		
new-value		
element-rule		
name		
ChangeURIhost		
	parameter-name	
	type	uri-host
	action	replace
	match-val-type	any
	comparison-type	case-
sensitive	match-value	
	new-value	\$LOCAL_IP
header-rule		
name		StoreTouser
header-name		To
action		store
comparison-type		pattern-rule
msg-type		request
methods		REFER
match-value		
new-value		
element-rule		
name		
CheckReferTo		
	parameter-name	
	type	uri-user
	action	store
	match-val-type	any
	comparison-type	pattern-
rule		

```

        match-value
        new-value
    header-rule
        name                ModifyReferto
        header-name          Refer-To
        action                manipulate
        comparison-type       boolean
        msg-type              request
        methods                REFER
        match-value
    $StoreTouser.$CheckReferTo
        new-value
        element-rule
            name
    ChangeReferTo
        parameter-name
        type                  uri-user
        action                add
        match-val-type        any
        comparison-type       pattern-
    rule
        match-value
        new-value
    $StoreTouser.$CheckReferTo.$0
        header-rule
            name                ChangeReferRURI
            header-name          Request-URI
            action                manipulate
            comparison-type       boolean
            msg-type              request
            methods                REFER
            match-value
    $StoreTouser.$CheckReferTo
        new-value
        element-rule
            name                ModRURI
            parameter-name
            type                  uri-user
            action                add
            match-val-type        any
            comparison-type       pattern-
    rule

```

```

match-value
new-value
$StoreTouser.$CheckReferTo.$0

```

All the above SIP manipulations are then consolidated into one single HMR by calling individual ones within a header-rule

```

sip-manipulation
  name ChangeforPAIandNAT
  description
  split-headers
  join-headers
  header-rule
    name changePAI
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value ChangePAI
  header-rule
    name forprivacy
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value NATting
  header-rule
    name fordelayayaheaders
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value RemoveAvayaheaders
  header-rule
    name forddiv
    header-name From

```

```

        action sip-manip
        comparison-type case-sensitive
        msg-type any
        methods
        match-value
        new-value AddDiversion
    header-rule
        name ForREFER
        header-name From
        action sip-manip
        comparison-type case-sensitive
        msg-type any
        methods
        match-value
        new-value changeRefer

```

Once configured, this sip-manipulation needs to be applied as an out-manipulationid on the ATT trunk facing sip-interface of E-SBC

```

sip-interface
    state enabled
    realm-id ATT-Trunk
    description
    sip-port
        address 167.167.167.181
        port 5060
        transport-protocol UDP
        tls-profile
        allow-anonymous agents-only
        multi-home-addr
        ims-aka-profile
    carriers
    trans-expire 0
...
out-manipulationid ChangeforPAIandNAT

```

Change display and add Privacy header for Anonymous calls

For anonymous calls, the From header needs to have display and user portion changed to Anonymous and needs to add Privacy header as shown below

```

sip-manipulation
  name                                changedisplayname
  description
  split-headers
  join-headers
  header-rule
    name                                changedisplay
    header-name                          From
    action                                manipulate
    comparison-type                       case-sensitive
    msg-type                              request
    methods                               INVITE
    match-value
    new-value
    element-rule
      name
ChngFromuser
      parameter-name
      type                                uri-user
      action                              replace
      match-val-type                     any
      comparison-type                    case-
sensitive
      match-value
      new-value                          Anonymous
      element-rule
        name
Changefromdisplay
      parameter-name
      type                                uri-
display
      action                              replace
      match-val-type                     any
      comparison-type                    case-
sensitive
      match-value
      new-value
\"Anonymous\"
```



```
header-rule
    name                Addprivacyheader
    header-name         Privacy
    action              add
    comparison-type     case-sensitive
    msg-type            request
    methods              INVITE
    match-value
    new-value           id
```

SIP Manipulation on Core sip interface of E-SBC towards Avaya for topology hiding

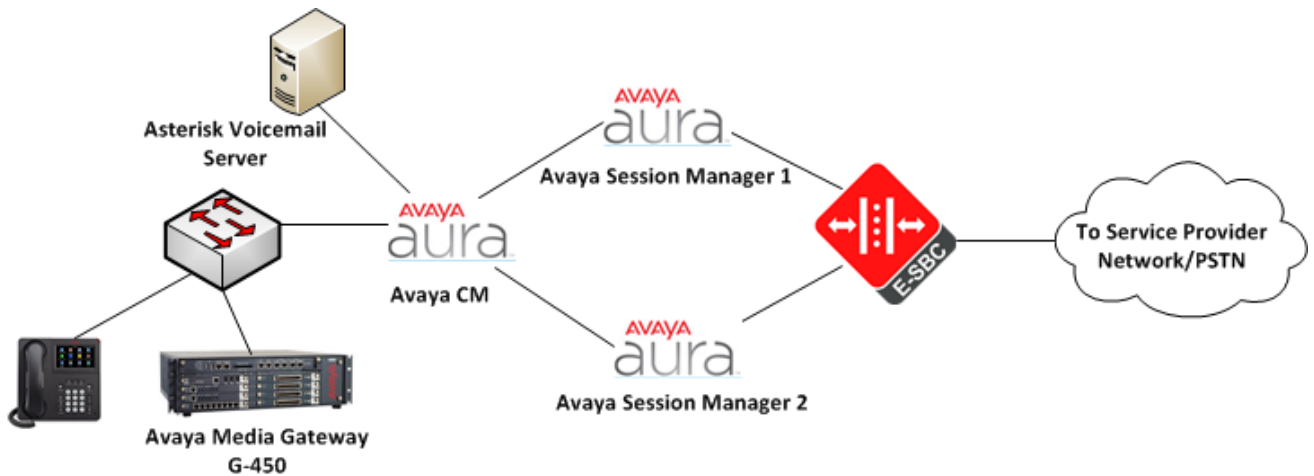
The E-SBC in-built variable ACME_TO_FROM_NAT_IP is applied as an out-manipulation on the sip-interface facing Avaya SM.

This completes the major configuration highlights from the testing. A fully copy of the E-SBC configuration is elaborated in the Appendix Section of this document.

Phase 2 – Configuring the Avaya Aura Session Manager v6.3.15

The enterprise will have a fully functioning Avaya SM installed and deployed for this certification. The Avaya Aura setup consists of Avaya session manager, system manager, Communication manager and the media gateway for any TDM based trunk termination.

The following diagram gives an overview on the Avaya Aura System architecture.



The Avaya Aura session manager is a SIP routing and integration tool and the core component within the Avaya Aura Enterprise edition solution. It connects the Avaya Aura Communication Manager as a SIP feature server and enterprise PBX for SIP and no-SIP phones, and routes SIP sessions across the network. In addition to the above, a free-pbx/Asterisk was used to provide voicemail services for two test cases. This will also be described later in this document.

In this section we describe the steps for configuring Avaya SM server, system manager and integrate it with the E-SBC.

In Scope

- Configuration in System manager, routing policy, SIP entity definition to establish connectivity with E-SBC.

Out of Scope

- Installing the Avaya SM and network management to connect to E-SBC and CM topology

What you will need

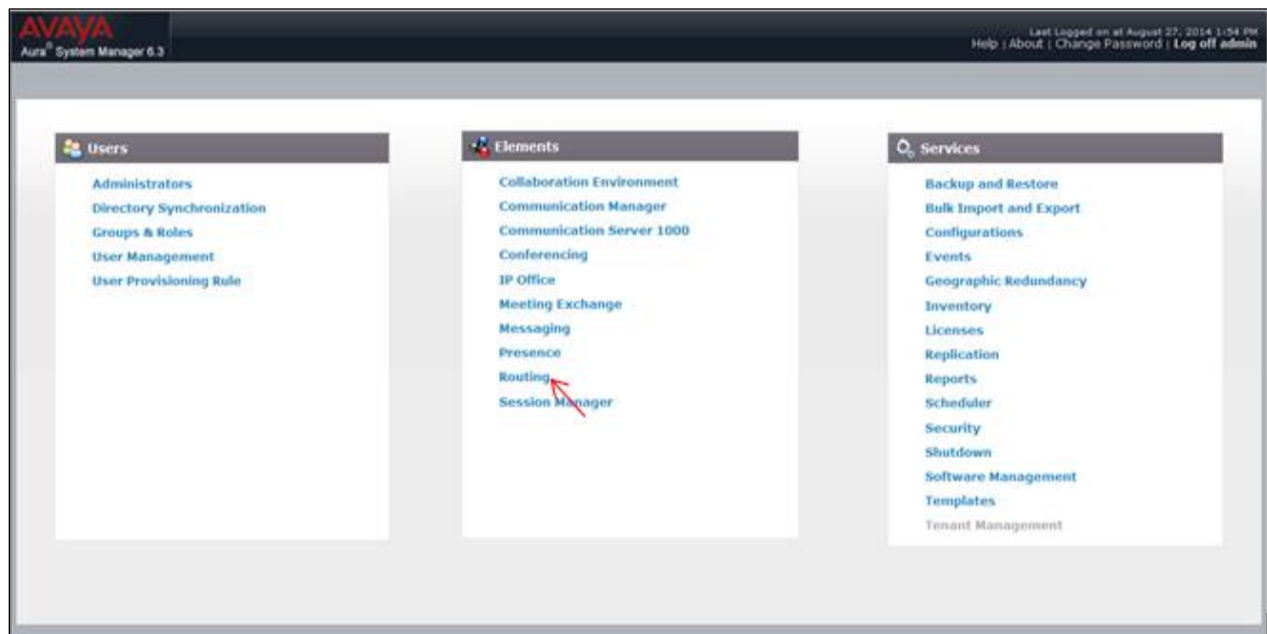
Avaya SM installed and base configuration

To configure the session manager, one the System manager portal is used. The Avaya aura system manager is a central management system that delivers a set of shared management services and common console for Avaya Aura applications and systems like Session manager. The system manager has synchronization tool to provide replication across the session manager clusters. The configuration to operate and connect to the E-SBC consists of the following steps:

- Adding the E-SBC as a SIP Entity
- Configuring an Entity link between E-SBC and Session Manager
- Creating a Routing policy to assign the appropriate routing destination

Adding the E-SBC as a SIP Entity

Login to the Aura system manager. Click on **Routing**, under the **Elements** section



On the **Routing** tab, select **SIP Entities** from the menu on the left side of the screen. Click **New** to add E-SBC as a SIP entity as shown below and click **Commit**

Home / Elements / Routing / SIP Entities

SIP Entity Details Commit Cancel

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

CommProfile Type Preference:

Loop Detection

Loop Detection Mode:

SIP Link Monitoring

SIP Link Monitoring:

* Proactive Monitoring Interval (in seconds):

* Reactive Monitoring Interval (in seconds):

* Number of Retries:

Supports Call Admission Control:

Shared Bandwidth Manager:

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

Override Port & Transport with DNS SRV:

Add Remove

2 Items Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* SM1_to_Acme6300	acme-sm	TCP	* 5060	SBC_6300	* 5060	trusted	<input type="checkbox"/>
* SM2_to_Acme6300	acme-sm2	TCP	* 5060	SBC_6300	* 5060	trusted	<input type="checkbox"/>

Select : All, None

Configure Entity Link between Session manager and E-SBC

Select **Entity Links** from the menu and click on **New** to add an Entity Link between E-SBC and SM (for SM1 and SM2) with the following settings and click **Commit**.

The screenshot shows the AVAYA Aura System Manager 6.3 interface. The left sidebar menu is expanded to 'Entity Links'. The main content area displays the 'Entity Links' configuration page. At the top right, there are 'Commit' and 'Cancel' buttons. Below the title, there is a table with one item. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The row contains the following values: Name: *SM1_to_Acme6300, SIP Entity 1: acme-sm, Protocol: TCP, Port: *5060, SIP Entity 2: *SBC_6300, DNS Override: , Port: *5060, Connection Policy: trusted. Below the table, there is a 'Select : All, None' dropdown.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
<input type="checkbox"/>	*SM1_to_Acme6300	* acme-sm	TCP	* 5060	* SBC_6300	<input type="checkbox"/>	* 5060	trusted

The screenshot shows the AVAYA Aura System Manager 6.3 interface. The left sidebar menu is expanded to 'Entity Links'. The main content area displays the 'Entity Links' configuration page. At the top right, there are 'Commit' and 'Cancel' buttons. Below the title, there is a table with one item. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The row contains the following values: Name: *SM2_to_Acme6300, SIP Entity 1: acme-sm2, Protocol: TCP, Port: *5060, SIP Entity 2: *SBC_6300, DNS Override: , Port: *5060, Connection Policy: trusted. Below the table, there is a 'Select : All, None' dropdown.

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
<input type="checkbox"/>	*SM2_to_Acme6300	* acme-sm2	TCP	* 5060	* SBC_6300	<input type="checkbox"/>	* 5060	trusted

Creating a Routing policy to assign the appropriate routing destination

Select **Routing policies** from the menu and click on **New** to add a routing policy between E-SBC interface (10.232.50.130) and SM with the following settings and click **Commit**.

Home / Elements / Routing / Routing Policies

Routing Policy Details Commit Cancel

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
SBC_6300	10.232.50.130	Other	SBC_6300

Time of Day

1 Item

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Another routing policy destination is added for route from SM to CM (Communication manager) as show in below screenshot:

The screenshot shows a web-based configuration interface for routing policies. On the left is a navigation menu with 'Routing Policies' selected. The main area is titled 'Routing Policy Details' and includes a 'Commit' button and a 'Cancel' button. The 'General' section contains fields for 'Name' (toAvayaCM), 'Disabled' (checkbox), 'Retries' (0), and 'Notes' (toAvayaCM). The 'SIP Entity as Destination' section has a 'Select' button and a table listing destinations.

Name	FQDN or IP Address	Type	Notes
acme-cm	10.232.50.103	CM	

The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. It shows a table with 1 item:

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Below the table is a 'Select' dropdown menu with options 'All, None'.

The configuration to establish SIP connectivity with the SBC is now complete. Following next sections talk about Dial patterns, Feature code based tests for which additional configuration is required on the system manager and that is described in greater detail.

Dial Patterns

The above steps have configured the SM to establish connectivity with the SBC. For dialing outbound, regular outbound calls, and certain tests require 911 as well as international call dialing. The system manager has Dial patterns section under **Routing** tab where one can configure these. This is shown below in the screen shots:

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 6.3", and a "Last Logged on at Marc" indicator. Below the navigation bar, there are tabs for "Home" and "Routing". The "Routing" tab is active, and a sub-menu is open showing "Dial Patterns".

The "Dial Patterns" section includes a breadcrumb trail: "Home / Elements / Routing / Dial Patterns". Below this, there are action buttons: "New", "Edit", "Delete", "Duplicate", and "More Actions".

The main content area displays a table with 7 items. The table has the following columns: "Pattern", "Min", "Max", "Emergency Call", "Emergency Type", "Emergency Priority", "SIP Domain", and "Notes".

Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
011xxxxxxx	13	14	<input type="checkbox"/>			-ALL-	International Dialing
1xxxxxxx	11	12	<input type="checkbox"/>			-ALL-	
+1xxxxxxx	12	12	<input type="checkbox"/>			-ALL-	
555555555	10	36	<input type="checkbox"/>			-ALL-	ATTfeaturecodeTest
732216xxxx	10	10	<input type="checkbox"/>			-ALL-	10digit
732320xxxx	10	10	<input type="checkbox"/>			-ALL-	10digit for IPTC
911	3	3	<input type="checkbox"/>			-ALL-	911 test

At the bottom of the table, there is a "Select : All, None" option.

Clicking on one of the assigned DID patterns

Home Routing x

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

General

* Pattern:

* Min:

* Max:

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

1 Item Filter: E

<input type="checkbox"/>	Originating Location Name ^	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy
<input type="checkbox"/>	ACME IP ADDRESS	ACME IP ADDRESS	toAvayaCM	0	<input type="checkbox"/>	acme-cm	toAvayaCM

Select : All, None

Adaptations for feature code based tests

In ATT IP Flex Reach service, certain tests like Network-based call forwarding scenarios are based on ATT's enhanced network portal. This requires activation and deactivation of the feature per test case by dialing special digits from CPE phone to the trunk. Since Avaya phones use the character "*" for invoking its own feature codes (local to its PBX), an adaptation in system manager is necessary to invoke the feature and send this dialing towards the trunk. In the system manager, maneuver to Routing tab and click on **Adaptations** link. Click on **New** to add adaptation to dial 5555555555 to be converted to *72 (for TC 20421) and then hit **Commit**. For each individual test case, this adaptation needed to be changed. A system administrator can set different digits to dial different patterns so that all network-based call forwarding scenarios are covered in the enterprise. This is shown below in the screenshots:

AVAYA
Aura[®] System Manager 6.3

Home Routing

Home / Elements / Routing / Adaptations

Adaptation Details Commit Cancel

General

* Adaptation Name:

Module Name:

Module Parameter Type:

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	fromto	true

Select : All, None

Egress URI Parameters:

Notes:

Digit Conversion for Incoming Calls to SM

0 Items Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
--------------------------	------------------	-----	-----	---------------	---------------	---------------	-------------------	-----------------	-------

Digit Conversion for Outgoing Calls from SM

1 Item Filter: Enable

<input type="checkbox"/>	Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Adaptation Data	Notes
<input type="checkbox"/>	*5555555555	*10	*36		*10	*72	both		test

Select : All, None

Phase 3 - Configuring the Avaya Communication Manager

Avaya Communication manager provides centralized call control for a distributed network of IP-communication devices, gateways. It consolidates several components into a holistic package that enterprises need for Unified communications. To define configuration, one can use the Avaya Site Administration tool (ASA) to login/connect to the communication manager and execute different commands.

In this section we describe the steps for configuring Avaya CM server to support Voicemail, Auto Attendant scenarios and also integrate it with the existing SM topology to E-SBC to be fully functional.

In Scope

ASA tool, configuring and assigning trunk group, hunt group, call vector, creating announcement, codec set, and establishing connectivity with SM.

Out of Scope

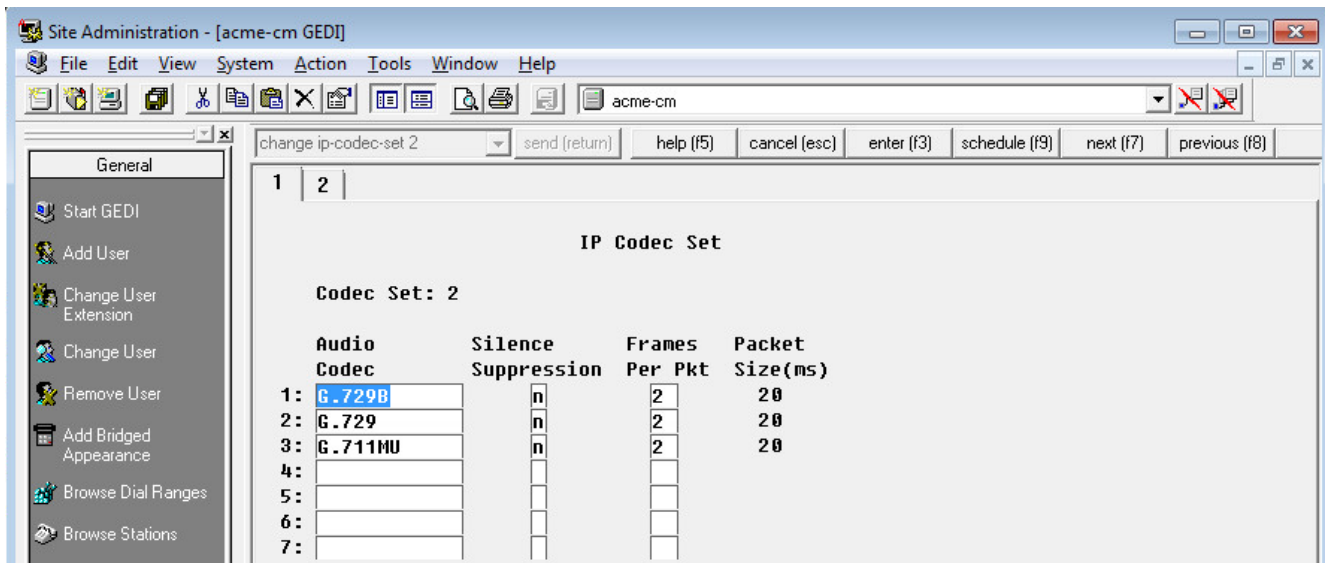
- Installing the Avaya CM on a VM and network management to connect to SM topology

What you will need

- Avaya CM installed and base configuration

Define codec set and reference in network region

ATT IP Flex Reach service requires G.729 annex B offered as first codec for outbound calls, hence the codec set is defined and referenced in ip-network-region (see codec set 2 referenced) as shown below:



The screenshot shows the Avaya Site Administration tool interface. The window title is "Site Administration - [acme-cm GEDI]". The menu bar includes File, Edit, View, System, Action, Tools, Window, and Help. The toolbar contains various icons for file operations and navigation. The main window displays the "change ip-codec-set 2" command in the command line, with buttons for "send (return)", "help (f5)", "cancel (esc)", "enter (f3)", "schedule (f9)", "next (f7)", and "previous (f8)". The main content area shows the "IP Codec Set" configuration for "Codec Set: 2".

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1: G.729B	n	2	20
2: G.729	n	2	20
3: G.711MU	n	2	20
4:			
5:			
6:			
7:			

Site Administration - [acme-cm GEDI]

File Edit View System Action Tools Window Help

change ip-network-region 1 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8)

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20

IP NETWORK REGION

Region: 1
 Location: 1 Authoritative Domain: aura.com
 Name: SIP Trunks Stub Network Region: n
MEDIA PARAMETERS
 Codec Set: 2 Intra-region IP-IP Direct Audio: yes
 UDP Port Min: 2048 Inter-region IP-IP Direct Audio: yes
 UDP Port Max: 3329 IP Audio Hairpinning? n
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
 Audio PHB Value: 46
 Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
 Audio 802.1p Priority: 6
 Video 802.1p Priority: 5
AUDIO RESOURCE RESERVATION PARAMETERS
 RSUP Enabled? n
H.323 IP ENDPOINTS
 H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
 Keep-Alive Interval (sec): 5
 Keep-Alive Count: 5

General

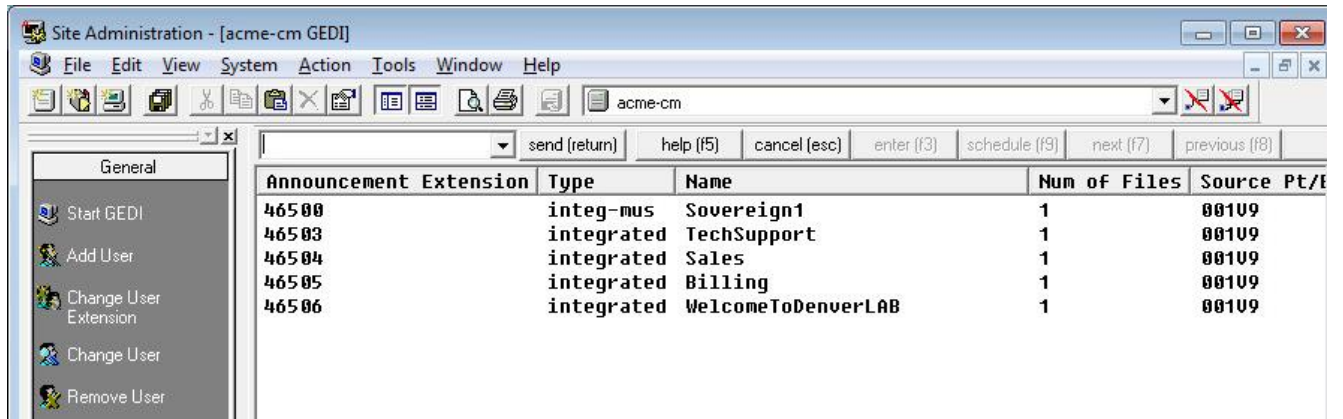
- Start GEDI
- Add User
- Change User Extension
- Change User
- Remove User
- Add Bridged Appearance
- Browse Dial Ranges
- Browse Stations
- Browse Unused Ports
- Find Unused Extension(s)
- Print Button Labels
- Advanced
- Fault & Performance
- Announcements

Configuration for Auto attendant test

For auto attendant test (TC 20201), the Avaya communication manger Vector directory number feature is used. Vector design handles incoming calls into the Avaya environment. Incoming call on a particular trunk is handled differently and transposed to a VDN and played an announcements. The following steps need to be accomplished:

- Create announcement
- Create VDN
- Create Call Vector
- Change incoming call handling on trunk

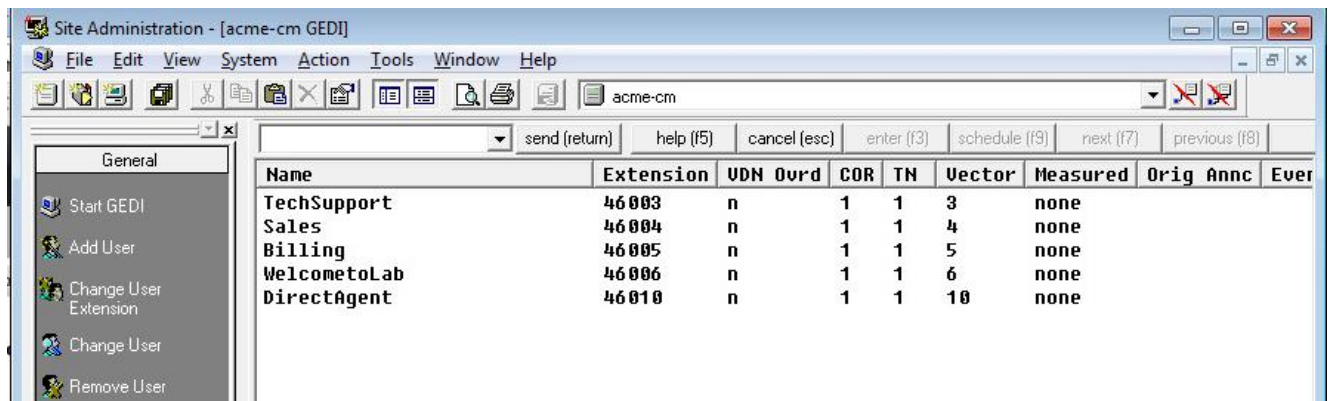
Create Announcement



The screenshot shows the 'Site Administration - [acme-cm GEDI]' interface. The 'General' sidebar is visible on the left. The main window displays a table of announcements with the following data:

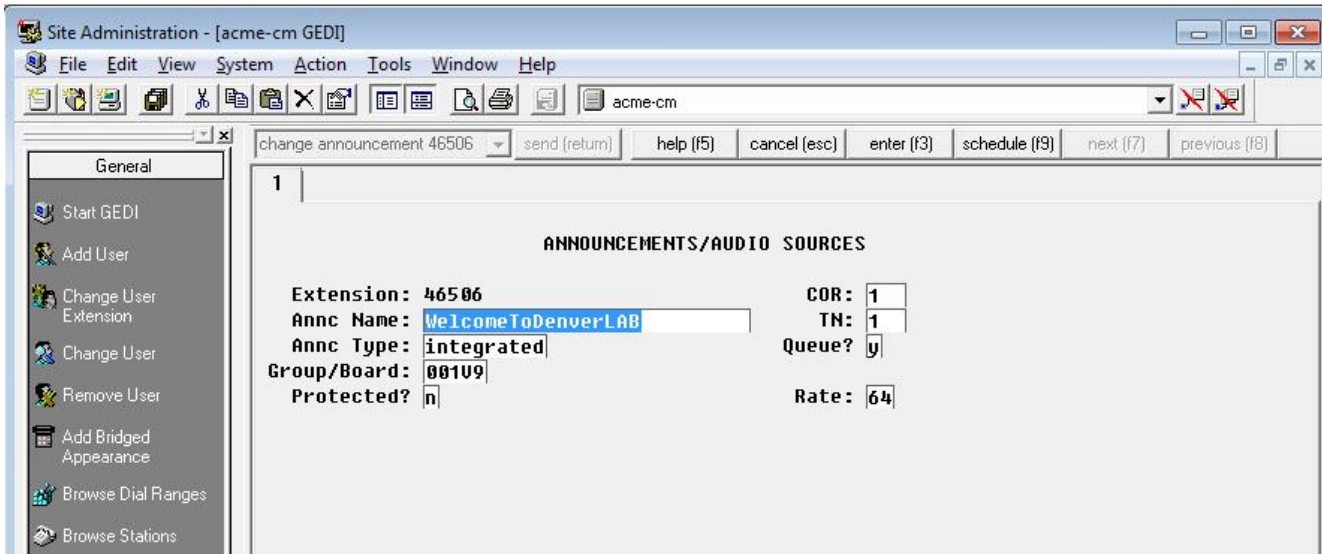
Announcement Extension	Type	Name	Num of Files	Source Pt/
46500	integ-mus	Sovereign1	1	00109
46503	integrated	TechSupport	1	00109
46504	integrated	Sales	1	00109
46505	integrated	Billing	1	00109
46506	integrated	WelcomeToDenverLAB	1	00109

Create Vector directory numbers

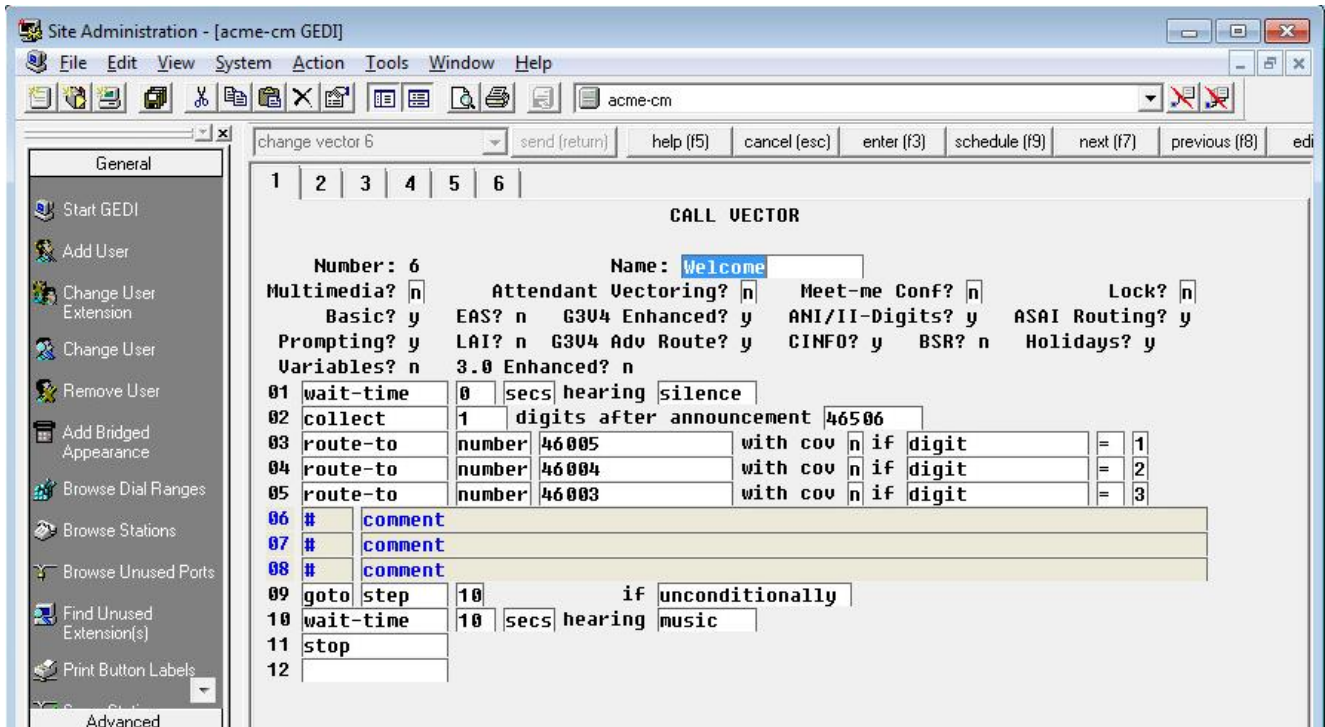


The screenshot shows the 'Site Administration - [acme-cm GEDI]' interface. The 'General' sidebar is visible on the left. The main window displays a table of vector directory numbers with the following data:

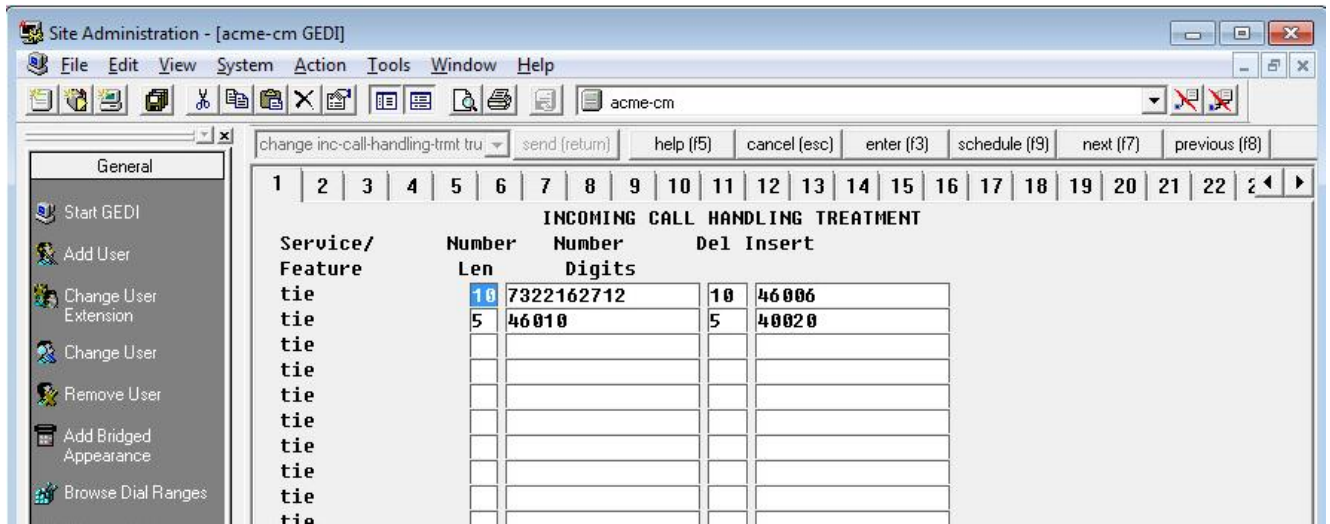
Name	Extension	VDN Ovr	COR	TN	Vector	Measured	Orig Annc	Ever
TechSupport	46003	n	1	1	3	none		
Sales	46004	n	1	1	4	none		
Billing	46005	n	1	1	5	none		
WelcomeToLab	46006	n	1	1	6	none		
DirectAgent	46010	n	1	1	10	none		



Create Call Vector



Change incoming call handling on trunk



Configuration for Voicemail Test

The ATT IPFR test required a PSTN caller to call into the enterprise and deposit voicemail and then place a second call to retrieve the voice message (TC 20151 and TC 20152). One of the Avaya Aura session manager system components – Avaya Messaging server was not available during the testing, therefore the team decided to use an Asterisk server (FreePBX) as an alternative. A trunk and coverage path established for a particular DID in Avaya to bounce call to voicemail if not answered within a few rings. The following steps need to be accomplished to have voicemail working:

- Install and create trunk in asterisk (out of scope of this document)
- Create coverage path with hunt group and assign to station (DID)
- Map hunt group to Asterisk trunk in Avaya CM
- Ensure AAR routing is mapped to asterisk trunk

Create Coverage path and assign to station

Site Administration - [acme-cm GEDI]

File Edit View System Action Tools Window Help

change coverage path 1 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1

COVERAGE PATH

Coverage Path Number: 1
 Cvg Enabled for UDN Route-To Party? Hunt after Coverage?
 Next Path Number: Linkage

COVERAGE CRITERIA

Station/Group Status	Inside Call	Outside Call	
Active?	<input type="text" value="n"/>	<input type="text" value="n"/>	
Busy?	<input type="text" value="y"/>	<input type="text" value="y"/>	
Don't Answer?	<input type="text" value="y"/>	<input type="text" value="y"/>	Number of Rings: <input type="text" value="3"/>
All?	<input type="text" value="n"/>	<input type="text" value="n"/>	
DND/SAC/Goto Cover?	<input type="text" value="y"/>	<input type="text" value="y"/>	
Holiday Coverage?	<input type="text" value="n"/>	<input type="text" value="n"/>	

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances?

Point1: Rng: Point2:
 Point3: Point4:
 Point5: Point6:

General

- Start GEDI
- Add User
- Change User Extension
- Change User
- Remove User
- Add Bridged Appearance
- Browse Dial Ranges
- Browse Stations
- Browse Unused Ports
- Find Unused Extension(s)
- Print Button Labels
- Advanced
- Fault & Performance
- Announcements

Assign it to station

Site Administration - [acme-cm GEDI]

change station 7322162709 | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8)

1 | 2 | 3 | 4 | 5

STATION

Extension: 732-216-2709 | Lock Messages? n | BCC: 0
Type: 4620 | Security Code: * | TN: 1
Port: S00009 | Coverage Path 1: 1 | COR: 1
Name: 7322162709 | Coverage Path 2: | COS: 1
Hunt-to Station: | Tests? y

STATION OPTIONS

Loss Group: 19 | Time of Day Lock Table: |
Personalized Ringing Pattern: 1
Speakerphone: 2-way | Message Lamp Ext: 732-216-2709
Display Language: english | Mute Button Enabled? y
Survivable GK Node Name: | Expansion Module? n
Survivable COR: internal | Media Complex Ext: |
Survivable Trunk Dest? y | IP SoftPhone? y
IP Video Softphone? n
Short/Prefixed Registration Allowed: default
Customizable Labels? y

Map hunt group to Asterisk

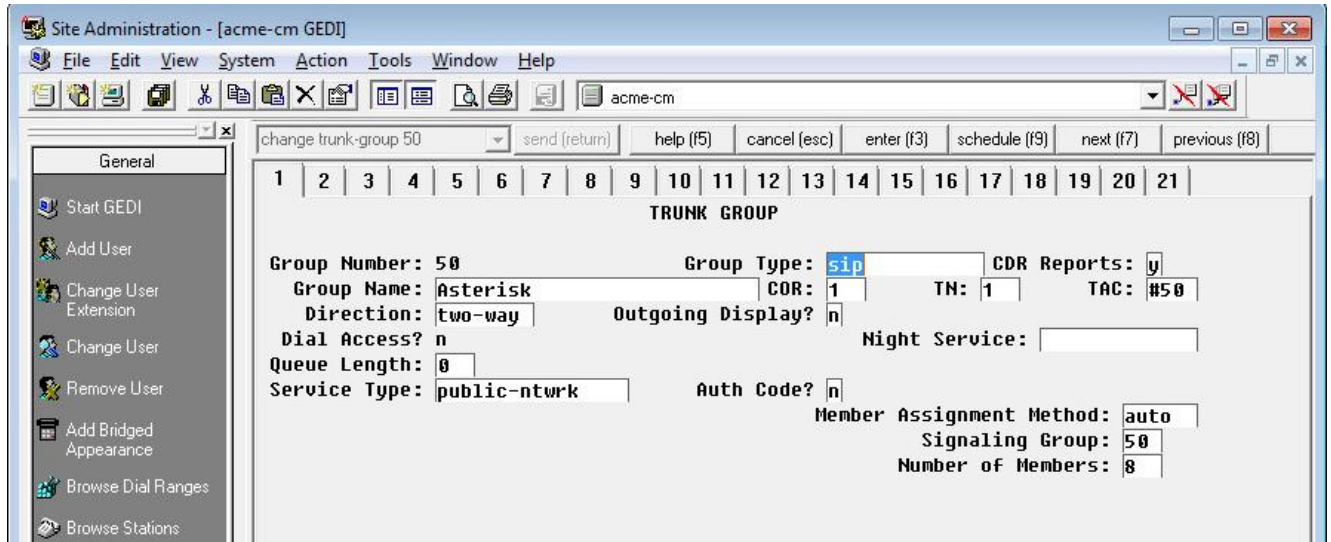
Site Administration - [acme-cm GEDI]

change hunt-group 1 | send (return) | help (f5) | cancel (esc) | enter (f3) | schedule (f9) | next (f7) | previous (f8)

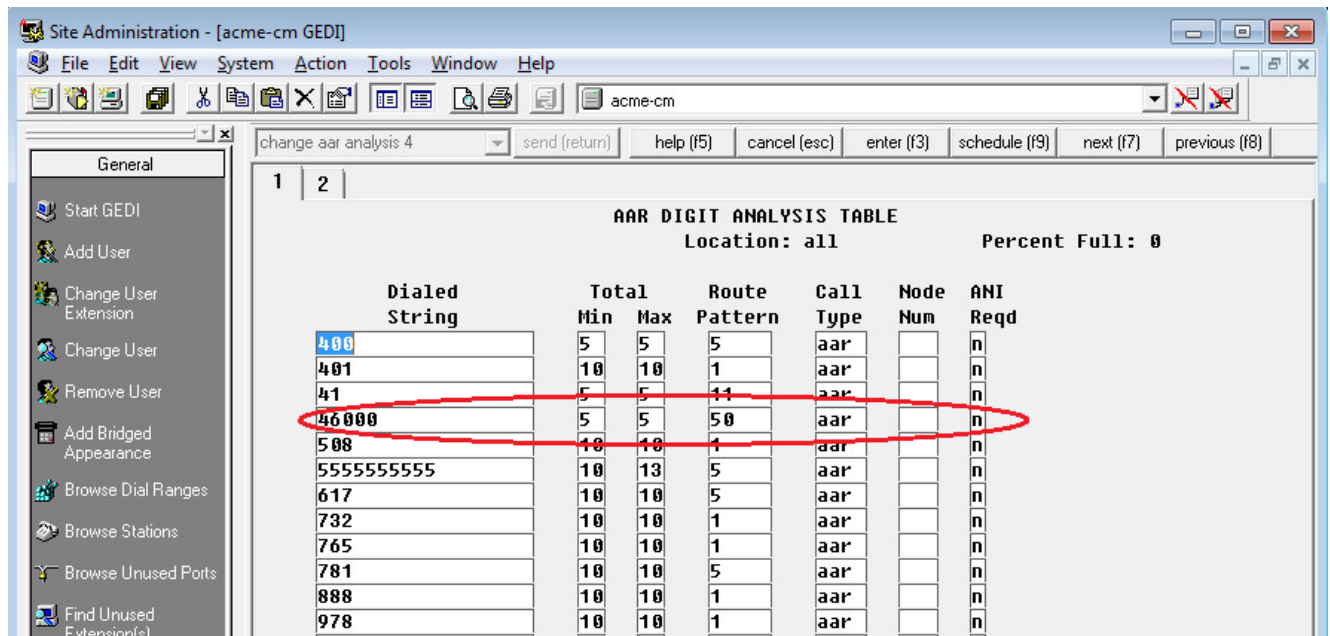
1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 | < | >

HUNT GROUP

Group Number: 1 | ACD? n
Group Name: AsteriskUH | Queue? n
Group Extension: 46000 | Vector? n
Group Type: ucd-mia | Coverage Path: |
TN: 1 | Night Service Destination: |
COR: 1 | MM Early Answer? n
Security Code: | Local Agent Preference? n
ISDN/SIP Caller Display: mbr-name



Alternate Abbreviated Routing – AAR (internal routing) to map to trunk





Test Summary

A comprehensive test plan was executed per ATT test specifications and call flows. For a copy of full test report, please contact your Oracle Sales account team.

Troubleshooting Tools

If you find that you are not able to complete calls or run into issues when going through the test plan, there are a few tools and methodologies available in Avaya SM and CM, Oracle SBC logging and tracing which may be of assistance. In this section we will provide a list of tools which you can use to aid in troubleshooting any issues you may encounter.

A good area to start troubleshooting when calls are not working or having issues is to look at signaling traces for SIP messages during call establishment through traces between Avaya Aura SM and SBC.

Avaya Aura Session Manager and Communication Manager

The Avaya Aura session manager can be accessed via a SSH session. One can run a command traceSM to start tracing messages between Avaya SM and CM.

Configuration checklist when outbound calls are failing:

- Check for Dial plan/route issues
- Check for codec mismatch or signaling complete as ATT IPFR requires G.729 as first choice codec on the offer
- Check Gateway and trunk configuration for TCP connections
- SIP OPTIONS message connectivity between SBC and Avaya Aura environment

The Avaya Communication manager site administration tool console gives ability to list and view the configuration, trunk, signaling groups, stations assigned, etc. The documentation available at the following link will help in navigating through different menus and commands:

https://downloads.avaya.com/elmodocs2/comm_mgr/r2_0/245801_1_1/233506_7/233506_7.pdf

Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from www.wireshark.org. Wireshark could be installed on the server hosting Avaya softphone/one-x communicator and have the SBC send packet trace to this remote location.

Oracle E-SBC 6300

The Oracle SBC provides a rich set of statistical counters available from the ACLI, as well as log file output with configurable detail. The follow sections detail enabling, adjusting and accessing those interfaces.

Resetting the statistical counters, enabling logging and restarting the log files.

At the SBC Console:

```
Avaya-SBC-ATT# reset sipd
Avaya-SBC-ATT# notify sipd debug
Avaya-SBC-ATT#
enabled SIP Debugging
Avaya-SBC-ATT# notify all rotate-logs
```

Examining the log files

Note: You will FTP to the management interface of the SBC with the username user and user mode password (the default is “acme”).

```
C:\Documents and Settings\user>ftp 192.168.5.24
Connected to 192.168.85.55.
220 Avaya-SBC-ATTFTP server (VxWorks 6.4) ready.
User (192.168.85.55:(none)): user
331 Password required for user.
Password: acme
230 User user logged in.
ftp> cd /ramdrv/logs
250 CWD command successful.
ftp> get sipmsg.log
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/sipmsg.log' (3353
bytes).
226 Transfer complete.
ftp: 3447 bytes received in 0.00Seconds 3447000.00Kbytes/sec.
ftp> get log.sipd
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/log.sipd' (204681
bytes).
226 Transfer complete.
ftp: 206823 bytes received in 0.11Seconds 1897.46Kbytes/sec.
ftp> bye
221 Goodbye.
```

You may now examine the log files with the text editor of your choice.

Through the Web GUI

You can also check the display results of filtered SIP session data from the Oracle Enterprise Session Border Controller, and provides traces in a common log format for local viewing or for exporting to your PC. Please check the “Monitor and Trace” section (page 145) of the Web GUI User Guide available at http://docs.oracle.com/cd/E56581_01/index.htm

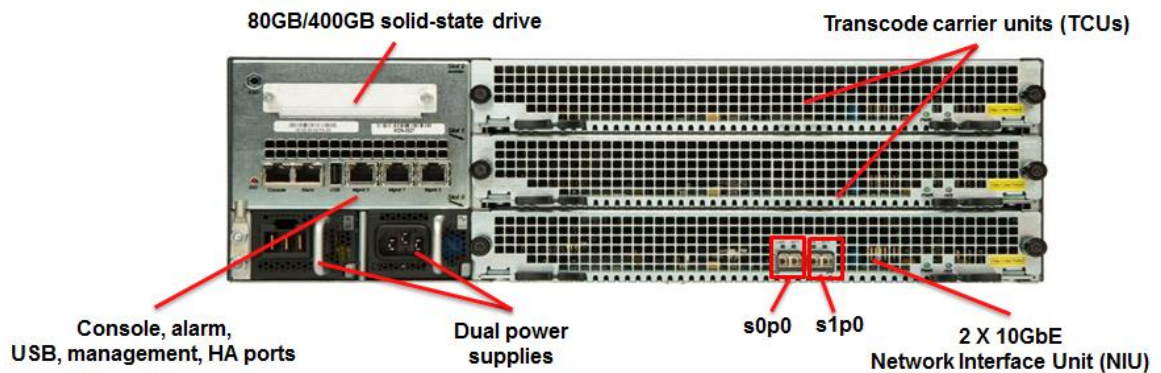
Appendix A

Accessing the ACLI

Access to the ACLI is provided by:

- The serial console connection;
- TELNET, which is enabled by default but may be disabled; and
- SSH, this must be explicitly configured.

Initial connectivity will be through the serial console port. At a minimum, this is how to configure the management (eth0) interface on the SBC.

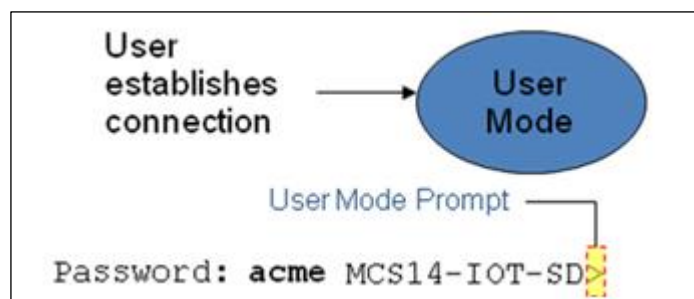


ACLI Basics

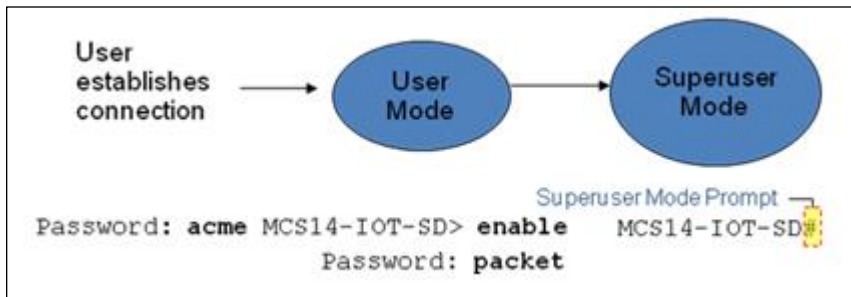
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

When you establish a connection to the SBC, the prompt for the User mode password appears. The default password is acme.

User mode consists of a restricted set of basic monitoring commands and is identified by the greater than sign (>) in the system prompt after the target name. You cannot perform configuration and maintenance from this mode.



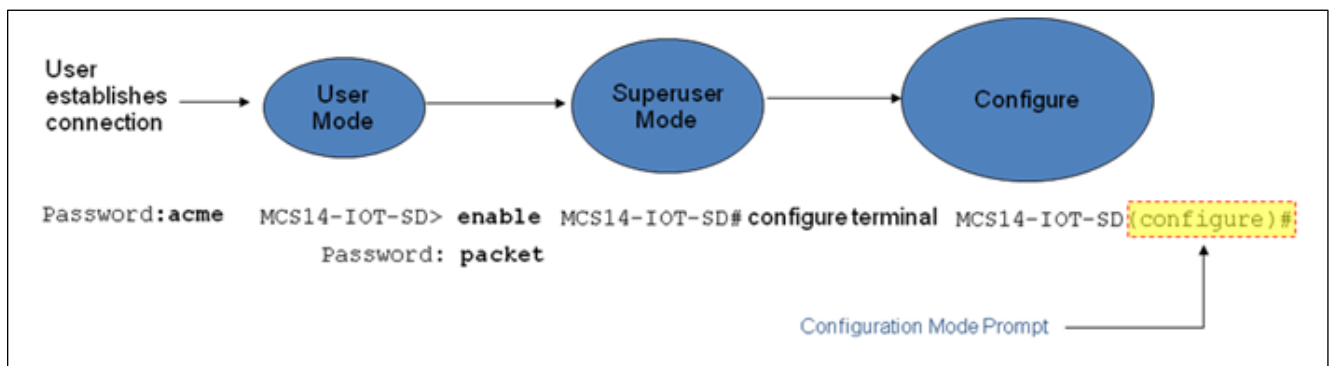
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



From the Superuser mode, you can perform monitoring and administrative tasks; however you cannot configure any elements. To return to User mode, issue the exit command.

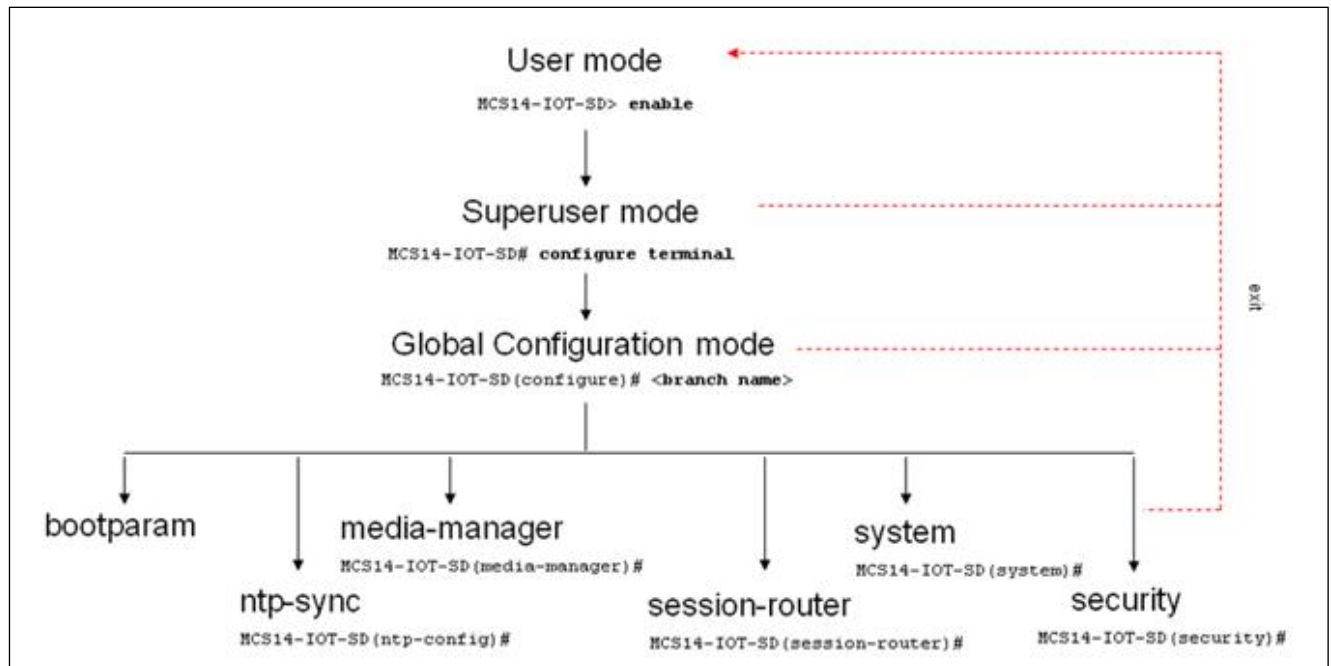
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the **configure terminal** command in the Superuser mode.

Configuration mode is identified by the word configure in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, **Avaya-SBC-ATT(configure)#**. To return to the Superuser mode, issue the **exit** command.



In the configuration mode, there are six configuration branches:

- bootparam;
- ntp-sync;
- media-manager;
- session-router;
- system; and
- security.




The ntp-sync and bootparams branches are flat branches (i.e., they do not have elements inside the branches). The rest of the branches have several elements under each of the branches.

The bootparam branch provides access to SBC boot parameters. Key boot parameters include:

- boot device – The global management port, usually eth0
- file name – The boot path and the image file.
- inet on ethernet – The IP address and subnet mask (in hex) of the management port of the SD.

- host inet –The IP address of external server where image file resides.
- user and ftp password – Used to boot from the external FTP server.
- gateway inet – The gateway IP address for reaching the external server, if the server is located in a different network.

```
Avaya-SBC-ATT#(configure)bootparam
'.' = clear field; '-' = go to previous field; q = quit
boot device          : eth0
processor number     : 0
host name            : acmesystem
file name            : /boot/EZ720p6.64.bz --- >location where the
software is loaded on the SBC
inet on ethernet (e) : 172.18.255.104:ffffff80 --- > This is the ip
address of the management interface of the SBC, type the IP address and
mask in hex
inet on backplane (b) :
host inet (h)         :
gateway inet (g)      : 172.18.0.1 --- > gateway address here
user (u)              : vxftp
ftp password (pw) (blank = use rsh) : vxftp
flags (f)             :
target name (tn)      : Avaya-SBC-ATT
startup script (s)    :
other (o)             :
```



The ntp-sync branch provides access to ntp server configuration commands for synchronizing the SBC time and date.

The security branch provides access to security configuration.

The system branch provides access to basic configuration elements as system-config, snmp-community, redundancy, physical interfaces, network interfaces, etc.

The session-router branch provides access to signaling and routing related elements, including H323-config, sip-config, iwf-config, local-policy, sip-manipulation, session-agent, etc.

The media-manager branch provides access to media-related elements, including realms, steering pools, dns-config, media-manager, and so forth.

You will use media-manager, session-router, and system branches for most of your working configuration.

Configuration Elements

The configuration branches contain the configuration elements. Each configurable object is referred to as an element. Each element consists of a number of configurable parameters.

Some elements are single-instance elements, meaning that there is only one of that type of the element - for example, the global system configuration and redundancy configuration.


Some elements are multiple-instance elements. There may be one or more of the elements of any given type. For example, physical and network interfaces.

Some elements (both single and multiple instance) have sub-elements. For example:

- SIP-ports - are children of the sip-interface element
- peers – are children of the redundancy element
- destinations – are children of the peer element

Creating an Element

1. To create a single-instance element, you go to the appropriate level in the ACLI path and enter its parameters. There is no need to specify a unique identifier property because a single-instance element is a global element and there is only one instance of this element.
2. When creating a multiple-instance element, you must specify a unique identifier for each instance of the element.
3. It is important to check the parameters of the element you are configuring before committing the changes. You do this by issuing the **show** command before issuing the **done** command. The parameters that you did not configure are filled with either default values or left empty.

- 
4. On completion, you must issue the **done** command. The done command causes the configuration to be echoed to the screen and commits the changes to the volatile memory. It is a good idea to review this output to ensure that your configurations are correct.
 5. Issue the **exit** command to exit the selected element.

Note that the configurations at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

Editing an Element

The procedure of editing an element is similar to creating an element, except that you must select the element that you will edit before editing it.

1. Enter the element that you will edit at the correct level of the ACLI path.
2. Select the element that you will edit, and view it before editing it.
The **select** command loads the element to the volatile memory for editing. The **show** command allows you to view the element to ensure that it is the right one that you want to edit.
3. Once you are sure that the element you selected is the right one for editing, edit the parameter one by one. The new value you provide will overwrite the old value.
4. It is important to check the properties of the element you are configuring before committing it to the volatile memory. You do this by issuing the **show** command before issuing the **done** command.
5. On completion, you must issue the **done** command.
6. Issue the **exit** command to exit the selected element.

Note that the configurations at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

Deleting an Element

The **no** command deletes an element from the configuration in editing.

To delete a single-instance element,

1. Enter the **no** command from within the path for that specific element
2. Issue the **exit** command.

To delete a multiple-instance element,

1. Enter the **no** command from within the path for that particular element.
The key field prompt, such as <name>:<sub-port-id>, appears.
2. Use the <Enter> key to display a list of the existing configured elements.
3. Enter the number corresponding to the element you wish to delete.

4. Issue the `select` command to view the list of elements to confirm that the element was removed.

Note that the configuration changes at this point are not permanently saved yet. If the SBC reboots, your configurations will be lost.

Configuration Versions

At any time, three versions of the configuration can exist on the SBC: the edited configuration, the saved configuration, and the running configuration.

- The **edited configuration** – this is the version that you are making changes to. This version of the configuration is stored in the SBC's volatile memory and will be lost on a reboot.
To view the editing configuration, issue the `show configuration` command.
- The **saved configuration** – on issuing the `save-config` command, the edited configuration is copied into the non-volatile memory on the SBC and becomes the saved configuration. Because the saved configuration has not been activated yet, the changes in the configuration will not take effect. On reboot, the last activated configuration (i.e., the last running configuration) will be loaded, not the saved configuration.
- The **running configuration** is the saved then activated configuration. On issuing the `activate-config` command, the saved configuration is copied from the non-volatile memory to the volatile memory. The saved configuration is activated and becomes the running configuration. Although most of the configurations can take effect once being activated without reboot, some configurations require a reboot for the changes to take effect.
To view the running configuration, issue command `show running-config`.

Saving the Configuration


The `save-config` command stores the edited configuration persistently.

Because the saved configuration has not been activated yet, changes in configuration will not take effect. On reboot, the last activated configuration (i.e., the last running configuration) will be loaded. At this stage, the saved configuration is different from the running configuration.

Because the saved configuration is stored in non-volatile memory, it can be accessed and activated at later time.

Upon issuing the `save-config` command, the SBC displays a reminder on screen stating that you must use the `activate-config` command if you want the configurations to be updated.

```
Avaya-SBC-ATT# save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
```



```
To sync & activate, run 'activate-config' or 'reboot activate'.  
Avaya-SBC-ATT#
```

Activating the Configuration

On issuing the **activate-config** command, the saved configuration is copied from the non-volatile memory to the volatile memory. The saved configuration is activated and becomes the running configuration.

Some configuration changes are service affecting when activated. For these configurations, the SBC warns that the change could have an impact on service with the configuration elements that will potentially be service affecting. You may decide whether or not to continue with applying these changes immediately or to apply them at a later time.

```
Avaya-SBC-ATT# activate-config  
Activate-Config received, processing.  
waiting 120000 for request to finish  
Request to 'ACTIVATE-CONFIG' has Finished,  
Activate Complete  
Avaya-SBC-ATT#
```

Appendix B: E-SBC Configuration

```
Avaya-SBC-ATT# show running-config
capture-receiver
  state          enabled
  address        10.232.50.78
  network-interface M01:0
  last-modified-by admin@172.18.0.141
  last-modified-date 2015-12-04 14:00:37
local-policy
  from-address   *
  to-address     *
  source-realm   ATT-Trunk
  description
  activate-time
  deactivate-time
  state          enabled
  policy-priority none
  policy-attribute
    next-hop     SAG:Avaya-SM-SAG
    realm        Core
    action       replace-uri
    terminate-recursion disabled
    carrier
    start-time   0000
    end-time     2400
    days-of-week U-S
    cost         0
    state        enabled
    app-protocol SIP
    methods
    media-profiles
    lookup       single
    next-key
    eloc-str-lkup disabled
    eloc-str-match
  last-modified-by admin@172.18.0.115
  last-modified-date 2015-10-14 10:51:33
local-policy
  from-address   *
```

```

to-address *
source-realm Core
description
activate-time
deactivate-time
state enabled
policy-priority none
policy-attribute
    next-hop 14.1.1.10
    realm ATT-Trunk
    action replace-uri
    terminate-recursion disabled
    carrier
    start-time 0000
    end-time 2400
    days-of-week U-S
    cost 0
    state enabled
    app-protocol SIP
    methods
    media-profiles
    lookup single
    next-key
    eloc-str-lkup disabled
    eloc-str-match
last-modified-by admin@172.18.0.115
last-modified-date 2015-10-14 11:17:21
media-manager
state enabled
latching enabled
flow-time-limit 86400
initial-guard-timer 300
subsq-guard-timer 300
tcp-flow-time-limit 86400
tcp-initial-guard-timer 300
tcp-subsq-guard-timer 300
tcp-number-of-ports-per-flow 2
hnt-rtcp disabled
algd-log-level NOTICE
mbcd-log-level NOTICE
options

```

red-flow-port	1985
red-mgcp-port	1986
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
media-policing	enabled
max-signaling-bandwidth	10000000
max-untrusted-signaling	100
min-untrusted-signaling	30
tolerance-window	30
trap-on-demote-to-deny	disabled
trap-on-demote-to-untrusted	disabled
syslog-on-demote-to-deny	disabled
syslog-on-demote-to-untrusted	disabled
rtcp-rate-limit	0
anonymous-sdp	disabled
arp-msg-bandwidth	32000
rfc2833-timestamp	disabled
default-2833-duration	100
rfc2833-end-pkts-only-for-non-sig	enabled
translate-non-rfc2833-event	disabled
media-supervision-traps	disabled
dnssalg-server-failover	disabled
syslog-on-call-reject	disabled
last-modified-by	admin@172.18.0.198
last-modified-date	2015-03-25 16:48:52
network-interface	
name	M00
sub-port-id	0
description	
hostname	
ip-address	167.167.167.181
pri-utility-addr	167.167.167.182
sec-utility-addr	167.167.167.183
netmask	255.255.255.0
gateway	167.167.167.1
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0


```
        retry-timeout          1
        health-score           0
    dns-ip-primary
    dns-ip-backup1
    dns-ip-backup2
    dns-domain
    dns-timeout                 11
    signaling-mtu               0
    hip-ip-list                 167.167.167.181
                                167.167.167.182
                                167.167.167.183
    ftp-address
    icmp-address                167.167.167.181
                                167.167.167.182
                                167.167.167.183
    snmp-address
    telnet-address
    ssh-address
    last-modified-by            admin@172.18.0.136
    last-modified-date          2015-11-10 15:39:35
network-interface
    name                        M01
    sub-port-id                 0
    description
    hostname
    ip-address                  10.232.50.130
    pri-utility-addr            10.232.50.131
    sec-utility-addr            10.232.50.132
    netmask                     255.255.255.0
    gateway                     10.232.50.1
    sec-gateway
    gw-heartbeat
        state                   disabled
        heartbeat                0
        retry-count              0
        retry-timeout            1
        health-score             0
    dns-ip-primary
    dns-ip-backup1
    dns-ip-backup2
    dns-domain
```

```
dns-timeout 11
signaling-mtu 0
hip-ip-list 10.232.50.130
ftp-address 10.232.50.130
icmp-address 10.232.50.130
snmp-address
telnet-address
ssh-address 10.232.50.130
last-modified-by admin@172.18.0.152
last-modified-date 2015-12-18 16:40:45
network-interface
name wancom1
sub-port-id 0
description
hostname
ip-address
pri-utility-addr 169.254.1.1
sec-utility-addr 169.254.1.2
netmask 255.255.255.252
gateway
sec-gateway
gw-heartbeat
state disabled
heartbeat 0
retry-count 0
retry-timeout 1
health-score 0
dns-ip-primary
dns-ip-backup1
dns-ip-backup2
dns-domain
dns-timeout 11
signaling-mtu 0
hip-ip-list
ftp-address
icmp-address
snmp-address
telnet-address
ssh-address
last-modified-by admin@172.18.0.115
last-modified-date 2015-10-23 14:52:02
```

```
network-interface
  name wancom2
  sub-port-id 0
  description
  hostname
  ip-address
  pri-utility-addr 169.254.2.1
  sec-utility-addr 169.254.2.2
  netmask 255.255.255.252
  gateway
  sec-gateway
  gw-heartbeat
    state disabled
    heartbeat 0
    retry-count 0
    retry-timeout 1
    health-score 0
  dns-ip-primary
  dns-ip-backup1
  dns-ip-backup2
  dns-domain
  dns-timeout 11
  signaling-mtu 0
  hip-ip-list
  ftp-address
  icmp-address
  snmp-address
  telnet-address
  ssh-address
  last-modified-by admin@172.18.0.115
  last-modified-date 2015-10-23 14:52:44
phy-interface
  name M00
  operation-type Media
  port 0
  slot 0
  virtual-mac 00:08:25:a3:36:ee
  admin-state enabled
  auto-negotiation enabled
  duplex-mode
  speed 100
```

```

wancom-health-score      50
overload-protection      disabled
last-modified-by        admin@172.18.0.115
last-modified-date      2015-10-23 14:50:01
phy-interface
  name                    M01
  operation-type          Media
  port                    1
  slot                    0
  virtual-mac             00:08:25:a3:36:ef
  admin-state             enabled
  auto-negotiation        enabled
  duplex-mode
  speed
  wancom-health-score    50
  overload-protection    disabled
  last-modified-by      admin@172.18.0.115
  last-modified-date    2015-10-23 14:50:20
phy-interface
  name                    wancom1
  operation-type          Control
  port                    1
  slot                    0
  virtual-mac
  admin-state            enabled
  auto-negotiation        enabled
  duplex-mode
  speed
  wancom-health-score    8
  overload-protection    disabled
  last-modified-by      admin@172.18.0.115
  last-modified-date    2015-10-23 14:50:57
phy-interface
  name                    wancom2
  operation-type          Control
  port                    2
  slot                    0
  virtual-mac
  admin-state            enabled
  auto-negotiation        enabled
  duplex-mode

```

```

speed
wancom-health-score          9
overload-protection         disabled
last-modified-by            admin@172.18.0.115
last-modified-date          2015-10-23 14:51:12
realm-config
  identifier                  ATT-Trunk
  description
  addr-prefix                 0.0.0.0
  network-interfaces          M00:0
  mm-in-realm                 enabled
  mm-in-network               enabled
  mm-same-ip                   enabled
  mm-in-system                 enabled
  bw-cac-non-mm               disabled
  msm-release                  disabled
  qos-enable                   disabled
  max-bandwidth                0
  fallback-bandwidth          0
  max-priority-bandwidth      0
  max-latency                  0
  max-jitter                   0
  max-packet-loss             0
  observ-window-size          0
  parent-realm
  dns-realm
  media-policy
  media-sec-policy
  srtp-msm-passthrough        disabled
  class-profile
  in-translationid
  out-translationid
  in-manipulationid
  out-manipulationid
  average-rate-limit          0
  access-control-trust-level  none
  invalid-signal-threshold    0
  maximum-signal-threshold    0
  untrusted-signal-threshold  0
  nat-trust-threshold          0
  max-endpoints-per-nat       0

```

nat-invalid-message-threshold	0
wait-time-for-invalid-register	0
deny-period	30
cac-failure-threshold	0
untrust-cac-failure-threshold	0
ext-policy-svr	
diam-e2-address-realm	
subscription-id-type	END_USER_NONE
symmetric-latching	disabled
pai-strip	disabled
trunk-context	
device-id	
early-media-allow	
enforcement-profile	
additional-prefixes	
restricted-latching	none
restriction-mask	32
user-cac-mode	none
user-cac-bandwidth	0
user-cac-sessions	0
icmp-detect-multiplier	0
icmp-advertisement-interval	0
icmp-target-ip	
monthly-minutes	0
options	
spl-options	
accounting-enable	enabled
net-management-control	enabled
delay-media-update	disabled
refer-call-transfer	disabled
refer-notify-provisional	none
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
codec-manip-in-network	enabled
rtcp-policy	
constraint-name	
call-recording-server-id	
session-recording-server	
session-recording-required	disabled
manipulation-string	

```
manipulation-pattern
stun-enable                disabled
stun-server-ip            0.0.0.0
stun-server-port          3478
stun-changed-ip           0.0.0.0
stun-changed-port         3479
sip-profile
sip-isup-profile
match-media-profiles
qos-constraint
block-rtcp                 disabled
hide-egress-media-update  disabled
tcp-media-profile
monitoring-filters
node-functionality
default-location-string
alt-family-realm
pref-addr-type             none
last-modified-by          admin@172.18.0.177
last-modified-date        2015-09-03 13:56:27
realm-config
  identifier               Core
  description
  addr-prefix              0.0.0.0
  network-interfaces       M01:0
  mm-in-realm              enabled
  mm-in-network            enabled
  mm-same-ip               enabled
  mm-in-system             enabled
  bw-cac-non-mm            disabled
  msm-release              disabled
  qos-enable               disabled
  max-bandwidth            0
  fallback-bandwidth       0
  max-priority-bandwidth   0
  max-latency              0
  max-jitter               0
  max-packet-loss          0
  observ-window-size       0
  parent-realm
  dns-realm
```

```
media-policy
media-sec-policy
srtp-msm-passthrough          disabled
class-profile
in-translationid
out-translationid
in-manipulationid
out-manipulationid
average-rate-limit           0
access-control-trust-level    none
invalid-signal-threshold     0
maximum-signal-threshold     0
untrusted-signal-threshold   0
nat-trust-threshold          0
max-endpoints-per-nat        0
nat-invalid-message-threshold 0
wait-time-for-invalid-register 0
deny-period                   30
cac-failure-threshold         0
untrust-cac-failure-threshold 0
ext-policy-svr
diam-e2-address-realm
subscription-id-type          END_USER_NONE
symmetric-latching           disabled
pai-strip                     disabled
trunk-context
device-id
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching          none
restriction-mask              32
user-cac-mode                 none
user-cac-bandwidth           0
user-cac-sessions             0
icmp-detect-multiplier        0
icmp-advertisement-interval   0
icmp-target-ip
monthly-minutes               0
options
spl-options
```


accounting-enable	enabled
net-management-control	disabled
delay-media-update	disabled
refer-call-transfer	disabled
refer-notify-provisional	none
dyn-refer-term	disabled
codec-policy	
codec-manip-in-realm	disabled
codec-manip-in-network	enabled
rtcp-policy	
constraint-name	
call-recording-server-id	
session-recording-server	
session-recording-required	disabled
manipulation-string	
manipulation-pattern	
stun-enable	disabled
stun-server-ip	0.0.0.0
stun-server-port	3478
stun-changed-ip	0.0.0.0
stun-changed-port	3479
sip-profile	
sip-isup-profile	
match-media-profiles	
qos-constraint	
block-rtcp	disabled
hide-egress-media-update	disabled
tcp-media-profile	
monitoring-filters	
node-functionality	
default-location-string	
alt-family-realm	
pref-addr-type	none
last-modified-by	admin@172.18.0.182
last-modified-date	2015-10-16 14:07:15
redundancy-config	
state	enabled
log-level	INFO
health-threshold	75
emergency-threshold	50
port	9090

```

advertisement-time      500
percent-drift            210
initial-time            1250
becoming-standby-time  180000
becoming-active-time    100
cfg-port                1987
cfg-max-trans           10000
cfg-sync-start-time     5000
cfg-sync-comp-time      1000
gateway-heartbeat-interval 0
gateway-heartbeat-retry 0
gateway-heartbeat-timeout 1
gateway-heartbeat-health 0
media-if-peercheck-time 0
peer
    name                PE-6300-1
    state               enabled
    type               Primary
    destination
        address
169.254.1.1:9090
        network-interface wancom1:0
    destination
        address
169.254.2.1:9090
        network-interface wancom2:0
    peer
        name            PE-6300-2
        state           enabled
        type            Secondary
        destination
            address
169.254.1.2:9090
            network-interface wancom1:0
        destination
            address
169.254.2.2:9090
            network-interface wancom2:0
    last-modified-by    admin@172.18.0.115
    last-modified-date  2015-10-23 15:07:19
response-map
    name                change486to603

```

entries	
recv-code	486
xmit-code	603
reason	Decline
method	
register-response-expires	
last-modified-by	admin@172.18.0.115
last-modified-date	2015-12-03 10:45:21
session-agent	
hostname	10.232.50.102
ip-address	
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	StaticTCP
realm-id	Core
egress-realm-id	
description	Avaya SM1
carriers	
allow-next-hop-lp	enabled
constraints	disabled
max-sessions	0
max-inbound-sessions	0
max-outbound-sessions	0
max-burst-rate	0
max-inbound-burst-rate	0
max-outbound-burst-rate	0
max-sustain-rate	0
max-inbound-sustain-rate	0
max-outbound-sustain-rate	0
min-seizures	5
min-asr	0
time-to-resume	0
ttr-no-response	0
in-service-period	0
burst-rate-window	0
sustain-rate-window	0
req-uri-carrier-mode	None
proxy-mode	
redirect-action	

loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS
ping-interval	60
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
load-balance-dns-query	hunt
options	
spl-options	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
refer-notify-provisional	none
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0

```

sip-profile
sip-isup-profile
kpml-interworking                inherit
monitoring-filters
session-recording-server
session-recording-required        disabled
last-modified-by                  admin@172.18.0.115
last-modified-date                2015-10-15 15:57:50
session-agent
hostname                          10.232.50.103
ip-address                        10.232.50.103
port                              5060
state                             enabled
app-protocol                      SIP
app-type                          StaticTCP
transport-method                  Core
realm-id                          enabled
egress-realm-id                   disabled
description
carriers
allow-next-hop-lp                 enabled
constraints                       disabled
max-sessions                      0
max-inbound-sessions              0
max-outbound-sessions             0
max-burst-rate                   0
max-inbound-burst-rate           0
max-outbound-burst-rate          0
max-sustain-rate                 0
max-inbound-sustain-rate         0
max-outbound-sustain-rate        0
min-seizures                      5
min-asr                          0
time-to-resume                   0
ttr-no-response                  0
in-service-period                0
burst-rate-window                0
sustain-rate-window              0
req-uri-carrier-mode              None
proxy-mode
redirect-action

```

loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS;hops=0
ping-interval	60
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
load-balance-dns-query	hunt
options	
spl-options	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
refer-notify-provisional	none
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0

```

sip-profile
sip-isup-profile
kpml-interworking                inherit
monitoring-filters
session-recording-server
session-recording-required       disabled
last-modified-by                 admin@172.18.0.177
last-modified-date               2015-09-03 13:47:19
session-agent
hostname                          10.232.50.112
ip-address
port                              5060
state                             enabled
app-protocol                      SIP
app-type
transport-method                 StaticTCP
realm-id                          Core
egress-realm-id
description                       Avaya SM 2
carriers
allow-next-hop-lp                enabled
constraints                       disabled
max-sessions                      0
max-inbound-sessions              0
max-outbound-sessions             0
max-burst-rate                    0
max-inbound-burst-rate            0
max-outbound-burst-rate           0
max-sustain-rate                  0
max-inbound-sustain-rate          0
max-outbound-sustain-rate         0
min-seizures                      5
min-asr                           0
time-to-resume                    0
ttr-no-response                   0
in-service-period                 0
burst-rate-window                 0
sustain-rate-window               0
req-uri-carrier-mode              None
proxy-mode
redirect-action
```

loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS
ping-interval	60
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
load-balance-dns-query	hunt
options	
spl-options	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
refer-notify-provisional	none
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0


```

sip-profile
sip-isup-profile
kpml-interworking                inherit
monitoring-filters
session-recording-server
session-recording-required       disabled
last-modified-by                 admin@172.18.0.115
last-modified-date               2015-10-15 15:58:02
session-agent
hostname                          14.1.1.10
ip-address                        14.1.1.10
port                              5060
state                             enabled
app-protocol                      SIP
app-type
transport-method                 UDP
realm-id                         ATT-Trunk
egress-realm-id
description                       ATT
carriers
allow-next-hop-lp                 enabled
constraints                       disabled
max-sessions                      0
max-inbound-sessions              0
max-outbound-sessions             0
max-burst-rate                    0
max-inbound-burst-rate            0
max-outbound-burst-rate           0
max-sustain-rate                  0
max-inbound-sustain-rate          0
max-outbound-sustain-rate         0
min-seizures                      5
min-asr                           0
time-to-resume                    0
ttr-no-response                   0
in-service-period                 0
burst-rate-window                 0
sustain-rate-window               0
req-uri-carrier-mode              None
proxy-mode
redirect-action
```

loose-routing	enabled
send-media-session	enabled
response-map	
ping-method	OPTIONS
ping-interval	30
ping-send-mode	keep-alive
ping-all-addresses	disabled
ping-in-service-response-codes	
out-service-response-codes	
load-balance-dns-query	hunt
options	
spl-options	
media-profiles	
in-translationid	
out-translationid	
trust-me	disabled
request-uri-headers	
stop-recurse	
local-response-map	
ping-to-user-part	
ping-from-user-part	
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
p-asserted-id	
trunk-group	
max-register-sustain-rate	0
early-media-allow	
invalidate-registrations	disabled
rfc2833-mode	none
rfc2833-payload	0
codec-policy	
enforcement-profile	
refer-call-transfer	disabled
refer-notify-provisional	none
reuse-connections	NONE
tcp-keepalive	none
tcp-reconn-interval	0
max-register-burst-rate	0
register-burst-window	0

```

sip-profile
sip-isup-profile
kpml-interworking                inherit
monitoring-filters
session-recording-server
session-recording-required        disabled
last-modified-by                  admin@172.18.0.177
last-modified-date                2015-09-03 13:57:13
session-group
  group-name                       Avaya-SM-SAG
  description                       Avaya SMs
  state                             enabled
  app-protocol                       SIP
  strategy                           Hunt
  dest                               10.232.50.102
                                   10.232.50.112
  trunk-group
    sag-recursion                   enabled
    stop-sag-recurse                401,407
    last-modified-by                 admin@172.18.0.115
    last-modified-date               2015-10-14 10:30:40
sip-config
  state                             enabled
  operation-mode                     dialog
  dialog-transparency                 enabled
  home-realm-id                       Core
  egress-realm-id
  auto-realm-id
  nat-mode                             None
  registrar-domain                     *
  registrar-host                       *
  registrar-port                       5060
  register-service-route                always
  init-timer                           5000
  max-timer                             4000
  trans-expire                          128
  initial-inv-trans-expire              0
  invite-expire                         180
  inactive-dynamic-conn                 32
  enforcement-profile
  pac-method

```

```
pac-interval 10
pac-strategy PropDist
pac-load-weight 1
pac-session-weight 1
pac-route-weight 1
pac-callid-lifetime 600
pac-user-lifetime 3600
red-sip-port 1988
red-max-trans 10000
red-sync-start-time 5000
red-sync-comp-time 1000
options max-udp-length=0
add-reason-header disabled
sip-message-len 4096
enum-sag-match disabled
extra-method-stats disabled
extra-enum-stats disabled
rph-feature disabled
nsep-user-sessions-rate 0
nsep-sa-sessions-rate 0
registration-cache-limit 0
register-use-to-for-lp disabled
refer-src-routing disabled
add-ucid-header disabled
proxy-sub-events
allow-pani-for-trusted-only disabled
atcf-stn-sr
atcf-psi-dn
atcf-route-to-sccas enabled
eatf-stn-sr
pass-gruu-contact disabled
sag-lookup-on-redirect disabled
set-disconnect-time-on-bye disabled
msrp-delayed-bye-timer 15
transcoding-realm
transcoding-agents
create-dynamic-sa disabled
node-functionality P-CSCF
last-modified-by admin@172.18.0.177
last-modified-date 2015-09-03 13:59:33
sip-interface
```

state	enabled
realm-id	ATT-Trunk
description	
sip-port	
address	167.167.167.181
port	5060
transport-protocol	UDP
tls-profile	
allow-anonymous	agents-only
multi-home-addr	
ims-aka-profile	
carriers	
trans-expire	0
initial-inv-trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	
spl-options	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	ChangeforPAIandNAT

sip-ims-feature	disabled
sip-atcf-feature	disabled
subscribe-reg-event	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
ldap-policy-server	
default-location-string	
term-tgrp-mode	none
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
manipulation-string	
manipulation-pattern	
sip-profile	
sip-isup-profile	
tcp-conn-dereg	0
tunnel-name	
register-keep-alive	none
kpml-interworking	disabled
msrp-delay-egress-bye	disabled
send-380-response	
pcscf-restoration	

```

session-timer-profile
session-recording-server
session-recording-required          disabled
service-tag
last-modified-by                    admin@172.18.0.141
last-modified-date                  2015-12-04 14:44:00
sip-interface
state                                enabled
realm-id                             Core
description
sip-port
    address                          10.232.50.130
    port                              5060
    transport-protocol                TCP
    tls-profile
    allow-anonymous                   agents-only
    multi-home-addr
    ims-aka-profile
carriers
trans-expire                          0
initial-inv-trans-expire              0
invite-expire                         0
max-redirect-contacts                0
proxy-mode
redirect-action
contact-mode                          none
nat-traversal                         none
nat-interval                          30
tcp-nat-interval                     90
registration-caching                 disabled
min-reg-expire                       300
registration-interval                 3600
route-to-registrar                   disabled
secured-network                       disabled
teluri-scheme                         disabled
uri-fqdn-domain
options
spl-options
trust-mode                            all
max-nat-interval                     3600
nat-int-increment                    10

```

```
nat-test-increment 30
sip-dynamic-hnt disabled
stop-recurse 401,407
port-map-start 0
port-map-end 0
in-manipulationid
out-manipulationid ACME_NAT_TO_FROM_IP
sip-ims-feature disabled
sip-atcf-feature disabled
subscribe-reg-event disabled
operator-identifier
anonymous-priority none
max-incoming-conns 0
per-src-ip-max-incoming-conns 0
inactive-conn-timeout 0
untrusted-conn-timeout 0
network-id
ext-policy-server
ldap-policy-server
default-location-string
term-tgrp-mode none
charging-vector-mode pass
charging-function-address-mode pass
ccf-address
ecf-address
implicit-service-route disabled
rfc2833-payload 101
rfc2833-mode transparent
constraint-name
response-map
local-response-map
ims-aka-feature disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive none
add-sdp-invite disabled
add-sdp-profiles
manipulation-string
manipulation-pattern
sip-profile
sip-isup-profile
```



```

tcp-conn-dereg          0
tunnel-name
register-keep-alive     none
kpml-interworking       disabled
msrp-delay-egress-bye  disabled
send-380-response
pcscf-restoration
session-timer-profile
session-recording-server
session-recording-required disabled
service-tag
last-modified-by       admin@172.18.0.115
last-modified-date     2015-10-14 10:52:56
sip-manipulation
  name                  AddDiversion
  description
  split-headers
  join-headers
  header-rule
    name                checkfor800
    header-name          To
    action               manipulate
    comparison-type      case-sensitive
    msg-type             request
    methods              INVITE
    match-value
    new-value
    element-rule
      name
checuriuser
  parameter-name
  type                  uri-user
  action               store
  match-val-type       any
  comparison-type      pattern-
rule
  match-value
18772427372
  new-value
  header-rule
    name                addDiv
    header-name          Diversion

```

```

        action                add
        comparison-type       boolean
        msg-type              request
        methods               INVITE
        match-value           $checkfor800.$schecuriuser
        new-value              <sip:7322162709@167.167.167.181>
        last-modified-by      admin@172.18.0.115
        last-modified-date    2015-12-09 13:48:38
sip-manipulation
    name                      ChangePAI
    description
    split-headers
    join-headers
    header-rule
        name                  Storecontacthost
        header-name           Contact
        action                store
        comparison-type       pattern-rule
        msg-type              any
        methods               INVITE
        match-value
        new-value
        element-rule
            name               storehost
            parameter-name
            type                uri-host
            action             store
            match-val-type     any
            comparison-type    pattern-
rule
        match-value
        new-value
    header-rule
        name                  ModPAI
        header-name           P-Asserted-
Identity
    action                manipulate
    comparison-type       boolean
    msg-type              any
    methods               INVITE

```

```

    match-value
$Storecontacthost.$storehost.$0
    new-value
    element-rule
        name                                modhost
        parameter-name
        type                                uri-host
        action                              replace
        match-val-type                      any
        comparison-type                     pattern-
rule
    match-value
    new-value
$Storecontacthost.$storehost.$0
    last-modified-by                       admin@172.18.0.182
    last-modified-date                      2015-10-16 13:08:23
sip-manipulation
    name                                    ChangeforPAIandNAT
    description
    split-headers
    join-headers
    header-rule
        name                                changePAI
        header-name                         From
        action                              sip-manip
        comparison-type                     case-sensitive
        msg-type                            any
        methods
        match-value
        new-value                           ChangePAI
    header-rule
        name                                forprivacy
        header-name                         From
        action                              sip-manip
        comparison-type                     case-sensitive
        msg-type                            any
        methods
        match-value
        new-value                           NATting
    header-rule
        name                                fordelayayaheaders
        header-name                         From

```

```

        action sip-manip
        comparison-type case-sensitive
        msg-type any
        methods
        match-value
        new-value RemoveAvayaheaders
header-rule
    name forddiv
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value AddDiversion
header-rule
    name ForREFER
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value changeRefer
last-modified-by admin@172.18.0.189
last-modified-date 2016-01-28 12:08:37
sip-manipulation
    name NATting
    description
    split-headers
    join-headers
header-rule
    name From
    header-name From
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule

```

```

From_header      name
                 parameter-name
                 type                uri-host
                 action              replace
                 match-val-type     any
                 comparison-type     case-
sensitive
                 match-value
                 new-value           $LOCAL_IP
  header-rule
    name          To
    header-name   To
    action        manipulate
    comparison-type case-sensitive
    msg-type      any
    methods
    match-value
    new-value
    element-rule
      name        To
      parameter-name
      type        uri-host
      action      replace
      match-val-type any
      comparison-type case-
sensitive
                 match-value
                 new-value           $REMOTE_IP
  last-modified-by      admin@172.18.0.182
  last-modified-date    2015-10-16 14:41:55
sip-manipulation
  name                  RemoveAvayaheaders
  description           remove avaya specific non-
important headers towards ATT
  split-headers
  join-headers
  header-rule
    name                delPAVMessageID
    header-name         P-AV-Message-Id
    action              delete
    comparison-type     case-sensitive

```

	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	header-rule	
	name	
delAVGlobalSessionID	header-name	AV-Global-Session-
ID	action	delete
	comparison-type	case-sensitive
	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	header-rule	
	name	delPlocation
	header-name	P-Location
	action	delete
	comparison-type	case-sensitive
	msg-type	any
	methods	INVITE
	match-value	
	new-value	
	last-modified-by	admin@172.18.0.177
	last-modified-date	2015-11-18 16:25:58
sip-manipulation	name	changeRefer
	description	
	split-headers	
	join-headers	
	header-rule	
	name	ModReferto
	header-name	Refer-To
	action	manipulate
	comparison-type	case-sensitive
	msg-type	any
	methods	REFER
	match-value	
	new-value	
	element-rule	
	name	

```

ChangeURIhost
    parameter-name
    type                uri-host
    action              replace
    match-val-type     any
    comparison-type    case-
sensitive
    match-value
    new-value          $LOCAL_IP
    header-rule
        name                ModReferredBY
        header-name         Referred-By
        action              manipulate
        comparison-type     case-sensitive
        msg-type            any
        methods             REFER
        match-value
        new-value
        element-rule
            name
ChangeURIhost
    parameter-name
    type                uri-host
    action              replace
    match-val-type     any
    comparison-type    case-
sensitive
    match-value
    new-value          $LOCAL_IP
    header-rule
        name                StoreTouser
        header-name         To
        action              store
        comparison-type     pattern-rule
        msg-type            request
        methods             REFER
        match-value
        new-value
        element-rule
            name
CheckReferTo
    parameter-name

```

```

type uri-user
action store
match-val-type any
comparison-type pattern-
rule
match-value
new-value
header-rule
name ModifyReferto
header-name Refer-To
action manipulate
comparison-type boolean
msg-type request
methods REFER
match-value
$StoreTouser.$CheckReferTo
new-value
element-rule
name
ChangeReferTo
parameter-name
type uri-user
action add
match-val-type any
comparison-type pattern-
rule
match-value
new-value
$StoreTouser.$CheckReferTo.$0
header-rule
name ChangeReferRURI
header-name Request-URI
action manipulate
comparison-type boolean
msg-type request
methods REFER
match-value
$StoreTouser.$CheckReferTo
new-value
element-rule
name ModRURI
parameter-name

```



```

type uri-user
action add
match-val-type any
comparison-type pattern-
rule
match-value
new-value
$StoreTouser.$CheckReferTo.$0
last-modified-by admin@172.18.0.158
last-modified-date 2016-02-04 23:14:55
sip-manipulation
name changedisplayname
description
split-headers
join-headers
header-rule
name changedisplay
header-name From
action manipulate
comparison-type case-sensitive
msg-type request
methods INVITE
match-value
new-value
element-rule
name
ChngFromuser
parameter-name
type uri-user
action replace
match-val-type any
comparison-type case-
sensitive
match-value
new-value Anonymous
element-rule
name
Changefromdisplay
parameter-name
type uri-
display
action replace

```

```

                                match-val-type      any
                                comparison-type     case-
sensitive
                                match-value
                                new-value
\"Anonymous\"
  header-rule
    name                        Addprivacyheader
    header-name                 Privacy
    action                      add
    comparison-type             case-sensitive
    msg-type                    request
    methods                     INVITE
    match-value
    new-value                   id
  last-modified-by             admin@172.18.0.149
  last-modified-date           2015-12-16 15:05:21
steering-pool
  ip-address                   10.232.50.130
  start-port                   20000
  end-port                     30000
  realm-id                     Core
  network-interface
  last-modified-by             admin@172.18.0.182
  last-modified-date           2015-10-16 15:10:40
steering-pool
  ip-address                   167.167.167.181
  start-port                   16384
  end-port                     32767
  realm-id                     ATT-Trunk
  network-interface
  last-modified-by             admin@172.18.0.182
  last-modified-date           2015-10-16 15:10:16
system-config
  hostname
  description
  location
  mib-system-contact
  mib-system-name
  mib-system-location
  snmp-enabled                 enabled





```



```
trap-event-lifetime      0
ids-syslog-facility     -1
options
default-v6-gateway      ::
ipv6-signaling-mtu      1500
ipv4-signaling-mtu      1500
cleanup-time-of-day     00:00
snmp-engine-id-suffix
snmp-agent-mode          v1v2
last-modified-by        admin@172.18.0.115
last-modified-date      2015-11-30 15:30:06
task done
Avaya-SBC-ATT#
```



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