

# ATT IP Toll Free Service with Avaya SM/CM 6.3 & Oracle Enterprise Session Border Controller

Technical Application Note



## Disclaimer

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

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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). It assumes that the reader is familiar with basic operations of the Oracle Enterprise Session Border Controller 3820/4500 platforms.

## Document Overview

This Oracle technical application note outlines the recommended configurations for the Oracle enterprise session border controller 3820 series for connecting AT&T's IP Toll Free service to Avaya Aura 6.3 customers. The solution contained within this document has been certified on Oracle's Acme Packet OS SCZ 7.3p2.

Avaya Aura® Session Manager 6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Communication Manager 6.3 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. The Oracle ESBC is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability. The AT&T IP Toll Free service, (referred to in the remainder of this document as IPTF), is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing AVPN or MIS/PNT transport.

It should be noted that while this application note focuses on the optimal configurations for the Oracle ESBC in an enterprise Avaya 6.3 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 Server associated parameters. Many SBC applications may have additional configuration requirements that are specific to individual customer requirements. These configuration items are not covered in this guide. Please contact your Oracle representative with any questions pertaining to this topic.

For additional information on Avaya Aura Communication Manager 6.3, please visit

<https://downloads.avaya.com/css/P8/documents/100171719>



# Introduction

## Audience

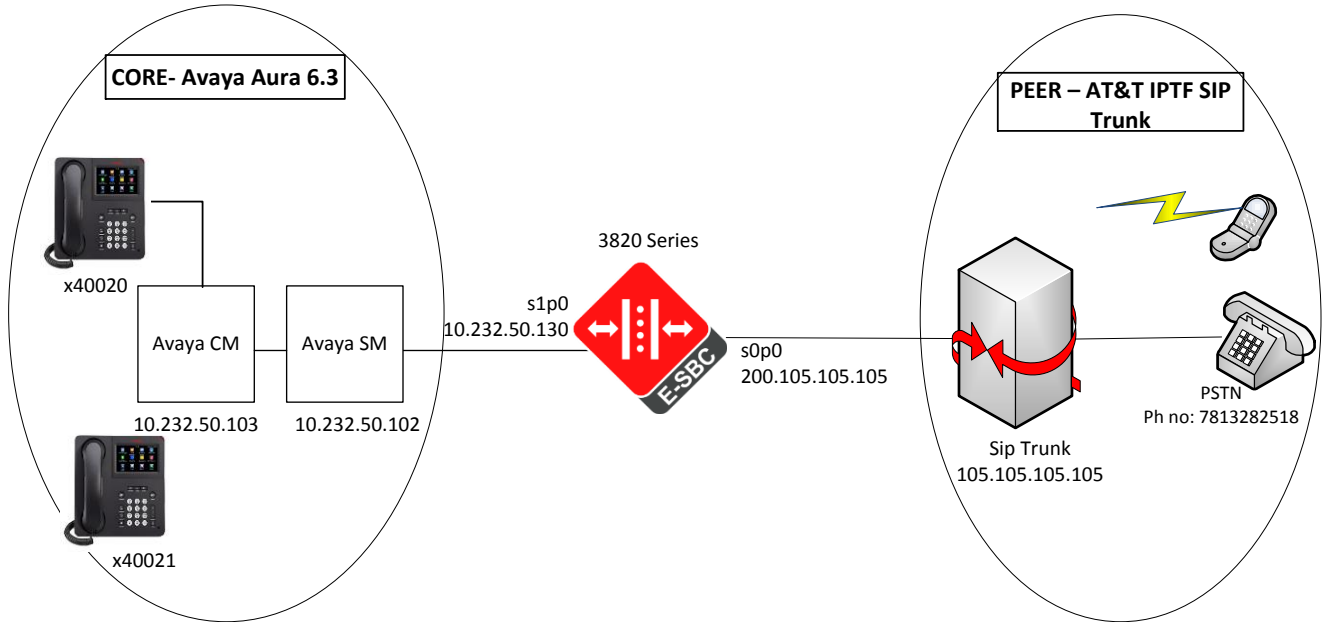
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise SBC and Avaya Aura 6.3. There will be steps that require navigating the Avaya System Manager configuration as well as the Acme Packet Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

## Requirements

- Fully functional Avaya Aura setup consisting of the Avaya CM, SM and System Manager
- Avaya soft phone and hard phones connected/registered to the Avaya CM server
- Oracle Enterprise Session Border Controller (hereafter Oracle E-SBC) 3820 or any other platforms (VME, 1100, 4500, 4600, 6300) series running ECZ7.3.0p2. Note: the configuration running on the SBC is backward/forward compatible with any release in the 7.2.0 or greater stream
- Oracle E-SBC having established SIP connectivity with Avaya SM on CPE side and ATT IPTFSIP trunk on PSTN side.

## Lab Configuration

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



# 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 SM in an ATT IPTF SIP Trunk service.

## In Scope

The following guide configuring the Oracle E-SBC assumes that this is a newly deployed Avaya topology in an enterprise dedicated to a single customer. If a service provider currently has the SBC deployed and is adding Avaya CM customers, then please see the ACLI Configuration Guide on [http://docs.oracle.com/cd/E56581\\_01/index.htm](http://docs.oracle.com/cd/E56581_01/index.htm) for a better understanding of the Command Line Interface (CLI).

Note that Oracle offers several models of SBC. This document covers the setup for the 3820 platform series running Net-Net OS ECZ7.3.0 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

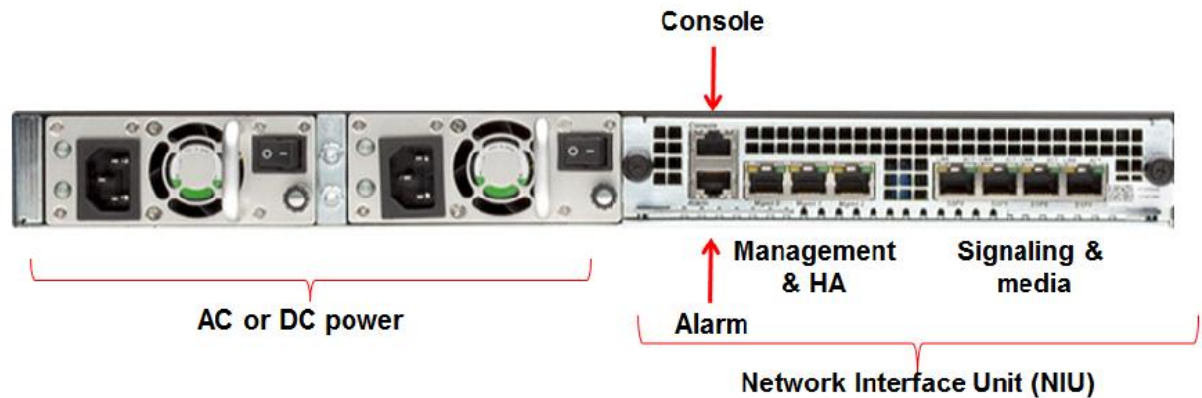
## What you will need

- 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 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 internal (Avaya facing) and external facing ports (Service Interfaces)
- IP address of the next hop gateway in the ATT IPTF network



## Configuring the SBC

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



Plug the slot 0 port 0 (s0p0) interface into your outside (ATT next-hop facing) network and the slot 1 port 0 (s0p1) interface into your inside (Avaya server-facing) network. Once connected, you are ready to power on and perform the following steps.

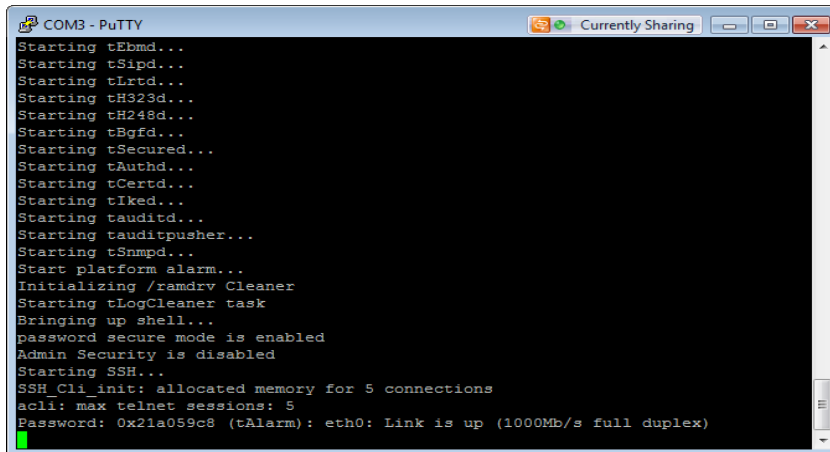
All commands are in bold, such as **configure terminal**; parameters in bold red such as **ATT-IPTF** are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

### Establish the serial connection and logging in the 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 tEcmd...
Starting tSipd...
Starting tLtd...
Starting tH323d...
Starting tH248d...
Starting tBgfd...
Starting tSecured...
Starting tAuthd...
Starting tCertd...
Starting tKed...
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
accli: 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
ATT-IPTF> enable
Password: packet
ATT-IPTF# configure terminal
ATT-IPTF(configure)#
```

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

ATT-IPTF#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

```
ATT-IPTF#(configure)bootparam
'.' = clear field; '-' = go to previous field; q = quit
boot device      : eth0
processor number : 0
host name        : acmesystem
file name        : /boot/EZ730p2.32.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)    : ATT-IPTF
startup script (s)  :
other (o)           :
```

## Configuring the SBC

The following section shows the Oracle Communications Enterprise SBC configuration required to work with Avaya Aura 6.3 and ATT's IPTF SIP Trunk server. In the configuration, the transport protocol used between the SBC and the Avaya server is TCP and the SIP trunk is configured for UDP in this certification testing.

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 3820. Please refer to the Oracle Enterprise Session Border Controller ECZ7.3.0 ACLI Configuration guide for more detailed update on High availability configuration. ([http://docs.oracle.com/cd/E61547\\_01/index.htm](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 6.3 to work with ATT IPTF 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. These are outlined below.

#### SIP Manipulations

SIP Manipulation for Setting the correct ptime and ensuring topology hiding in place

This sip-manipulation is applied as in-manipulationid facing the core interface of the SBC. ATT IPTF SIP trunk service requires the CPE to set ptime header in the SDP of the SIP requests to 30, which the SBC sets.

```

sip-manipulation
  name                               Changeptime
  description
  split-headers
  join-headers
  header-rule
    name                               Checkptimeexists
    header-name                         Content-type
    action                               store
    comparison-type                     pattern-rule
    msg-type                             any
    methods
    match-value
    new-value
    element-rule
      name                               ptimexists
      parameter-name                     application/sdp
      type                                mime
      action                              store
      match-val-type                     any
      comparison-type                    pattern-rule
      match-value                         (a=ptime:30)
      new-value
  header-rule
    name                               Addptime
    header-name                         Content-type
    action                              manipulate
```

```

comparison-type          boolean
msg-type                 any
methods
match-value              !($Checkptimeexists.$ptimeexists)
new-value
element-rule
    name                  Changesdp
    parameter-name        application/sdp
    type                  mime
    action                 find-replace-all
    match-val-type        any
    comparison-type        pattern-rule
    match-value            (m=audio\,*)
    new-value              $1+$CRLF+a=ptime:30
last-modified-by         admin@172.18.0.119
last-modified-date       2015-04-07 14:03:21

```

The below SIP manipulation NAT\_IP is used for topology hiding, as well as for adding the P-Asserted-ID header if not already present in the SIP requests. It also includes the manipulation for deleting a b line which is present in the SDP of the SIP requests sent from the Avaya to the SIP trunk through the SBC. Hence this manipulation is applied on the out-manipulationid of the SIP interface facing the ATT trunk.

```

sip-manipulation
    name                  NAT_IP
    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
        element-rule
            name            SourcePort
            parameter-name
            type              uri-port
            action            replace
            match-val-type    any

```

```

        comparison-type      case-sensitive
        match-value
        new-value            $LOCAL_PORT
header-rule
    name                    To
    header-name             To
    action                  manipulate
    comparison-type        case-sensitive
    msg-type                request
    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          $REMOTE_IP
header-rule
    name                    CheckPAIexists
    header-name             P-Asserted-Identity
    action                  store
    comparison-type        case-sensitive
    msg-type                request
    methods
    match-value
    new-value
header-rule
    name                    Add_P_Asserted_ID_new
    header-name             P-Asserted-Identity
    action                  add
    comparison-type        boolean
    msg-type                request
    methods                INVITE
    match-value            !$CheckPAIexists
    new-value              "<sip:"+$FROM_USER.$0+"@"+$LOCAL_IP+>"
header-rule
    name                    Changetransport
    header-name             From
    action                  manipulate
    comparison-type        case-sensitive
    msg-type                any
    methods
    match-value
    new-value
    element-rule
        name                Changetransport
        parameter-name      transport
        type                uri-param
        action              find-replace-all

```

```

        match-val-type          any
        comparison-type         case-sensitive
        match-value             Tcp
        new-value               udp

header-rule
  name                         StripBlIne
  header-name                  Content-type
  action                        manipulate
  comparison-type              pattern-rule
  msg-type                     any
  methods
  match-value
  new-value
  element-rule
    name                       deleteblIne
    parameter-name             application/sdp
    type                       mime
    action                      find-replace-all
    match-val-type             any
    comparison-type            case-sensitive
    match-value                 (b=AS.*)\r\n
    new-value
  last-modified-by            admin@172.18.0.119
  last-modified-date          2015-04-15 14:45:53

```

The SIP manipulation NATtingavaya is used to change the host portion of the To, From, Request-URI and PAI headers to aura.com. This is applied on the out-manipulationid of the SIP interface facing the Avaya.

```

sip-manipulation
  name                         NATtingavaya
  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                 aura.com
  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 aura.com
header-rule
    name Ruri_hr
    header-name Request-URI
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule
        name Ruri_er
        parameter-name
        type uri-host
        action find-replace-all
        match-val-type any
        comparison-type case-sensitive
        match-value
        new-value aura.com
header-rule
    name Pai
    header-name P-Asserted-Identity
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule
        name Pai_header
        parameter-name
        type uri-host
        action replace
        match-val-type any
        comparison-type case-sensitive
        match-value
        new-value aura.com

```

last-modified-by	admin@172.18.0.109
last-modified-date	2015-02-13 15:26:31

The ATT IPTF SIP trunk converts the 888XXXXXXX number into a DNIS which is of 5 leading 0's followed by 5 previously configured digits. When a call is made to the 888 number, the INVITE coming from the SIP trunk has the 888 number in the To header and the 10 digit DNIS in the RequestURI. The SIP manipulation RemoveExtraDigits checks how many digits are received in the Request URI header in the INVITE from the ATT SIP trunk, and if it receives extra leading 0's, it strips the Request URI user part to exact 10 digits with only 5 leading 0's.

```

sip-manipulation
  name                               RemoveExtraDigits
  description
  split-headers
  join-headers
  header-rule
    name                               RemovefromReqURI
    header-name                         Request-URI
    action                               manipulate
    comparison-type                     case-sensitive
    msg-type                             request
    methods
    match-value
    new-value
    element-rule
      name                               Stripdigits
      parameter-name
      type                               uri-user
      action                             find-replace-all
      match-val-type                     any
      comparison-type                   case-sensitive
      match-value                       [0-9]{6,}([0-9]{5,})
      new-value                           "00000"+$1
  last-modified-by                     admin@172.18.0.119
  last-modified-date                   2015-05-19 15:25:43

```

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.



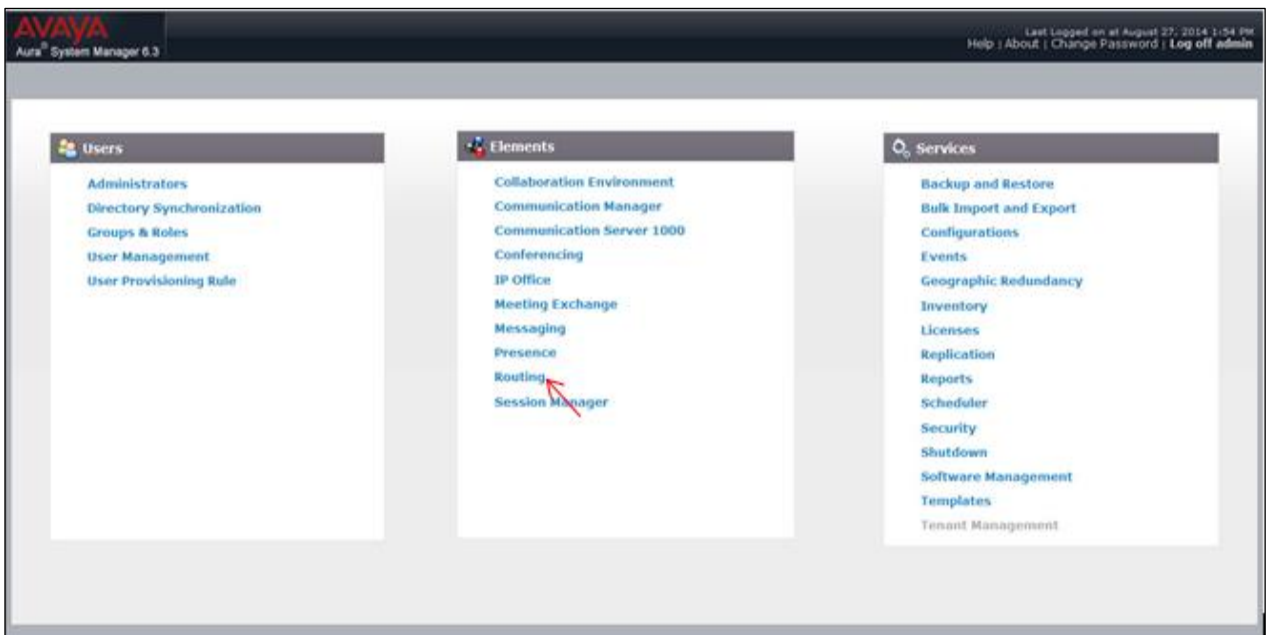
## Configuring the Avaya System Manager

The enterprise has a fully functional Avaya Aura System Manager. Configuring the System Manager to operate with the E-SBC consists of three 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

Log in 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 Help ?

**General**

\* Name: acme-ecb

\* FQDN or IP Address: 192.168.1.90

Type: SIP Trunk

Notes: acme-ecb

Adaptation: [dropdown]

Location: acme- Ent Comm Broker [dropdown]

Time Zone: America/Fortaleza [dropdown]

\* SIP Timer B/F (in seconds): 4

Credential name: [text field]

Call Detail Recording: none [dropdown]

Call Detail Recording: none ▼

### Loop Detection

Loop Detection Mode: Off ▼

### SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration ▼

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:

2 Items 							Filter: <a href="#">Enable</a>	
<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service	
<input type="checkbox"/>	acme-sm2 ▼	TCP ▼	* 5060	acme-ecb ▼	* 5068	trusted ▼	<input type="checkbox"/>	
<input type="checkbox"/>	acme-sm ▼	TCP ▼	* 5060	acme-ecb ▼	* 5068	trusted ▼	<input type="checkbox"/>	

Select : All, None

## Configuring an Entity link between E-SBC and Session Manager

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

Home / Elements / Routing / Entity Links

Entity Links Commit Cancel

1 Item

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
<input type="checkbox"/>	*acme-sm_acme-ecb	*acme-sm	TCP	*5060	*acme-ecb	<input type="checkbox"/>	*5068	trusted

Select : All, None

## 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 and SM with the following settings and click **Commit**.

Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details Commit Cancel

**General**

\* Name: Calls to ECB from SM1

Disabled:

\* Retries: 0

Notes: Calls to ECB from SM1

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
acme-ecb	192.168.1.90	SIP Trunk	acme-ecb

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

<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

**Time of Day**

1 Item Filter: Enable

<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

**Dial Patterns**

3 Items Filter: Enable

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
<input type="checkbox"/>	765	3	36	<input type="checkbox"/>	-ALL-	-ALL-	
<input type="checkbox"/>	978	3	11	<input type="checkbox"/>	aura.com	acme	Assigned Originating Locat
<input type="checkbox"/>	x	1	36	<input type="checkbox"/>	-ALL-	-ALL-	

Select : All, None

**Regular Expressions**

0 Items Filter: Enable

<input type="checkbox"/>	Pattern	Rank Order	Deny	Notes
--------------------------	---------	------------	------	-------

The Avaya System Manager is now configured to operate with E-SBC.

## Configuring the Avaya Communication Manager

For the IPTF testing, hunt groups are configured on the Avaya CM to emulate Call Center Application. There are 3 hunt groups with 1 agent skilled for each. The three hunt groups created are Tech Support, Billing and Sales. For the purpose of this testing, the incoming call from the AT&T trunk would reach a welcome message and be prompted for each of the hunt groups through the SBC. Once one was selected the caller would hear another announcement and then ring the agent's phone.

On the direct agent testing, "change in-call-handling-trmt trunk" command is executed to take the ATT IPTF numbers and convert them to a direct agent VDN. On the other test calls we used the digit manipulation on the Digit Conversion adapter ADAPTATION pointed at the Avaya Communication Manager.

Steps for configuring on CM:

### 1. Create the Agent ID and password

The screenshot shows the 'Site Administration' window for 'acme-cm GEDI'. The main area displays a table of user configurations. The table has columns for Login ID, Extn, Name, Dir, Agt, AAS/AUD, COR, Agt Prf, and seven levels of SKI (Sk1 1 to Sk1 7) and seven levels of Lvl (Lvl 1 to Lvl 7). Two users are listed: Agent1 (Login ID 40001, Extn unstaffed) and Agent2 (Login ID 44444, Extn unstaffed). Both have a COR of 1 and Agt Prf of 1. The SKI and Lvl columns contain numerical values representing different levels of service or quality of service.

Login ID	Extn	Name	Dir	Agt	AAS/AUD	COR	Agt Prf	Sk1 1	Lvl 1	Sk1 2	Lvl 2	Sk1 3	Lvl 3	Sk1 4	Lvl 4	Sk1 5	Lvl 5	Sk1 6	Lvl 6	Sk1 7	Lvl 7
40001	unstaffed	Agent1				1	1v1	2	01	3	02	4	03								
44444	unstaffed	Agent2				1	1v1	2	01	3	02	4	03								

Below the table, a status bar indicates 'Command successfully completed'. At the bottom, a log window shows several 'Info' messages from 'acme-cm' regarding 'change ip-codec-set 1' at various times on 6/3/2015 and 5/29/2015.

## 2. Create announcements

The screenshot displays the 'Site Administration' interface for 'acme-cm GED1'. The main window shows a table of announcements with the following data:

Announcement	Extension	Type	Name	Num of Files	Source Pt/Bd/Grp
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

Below the table, a status bar indicates 'Command successfully completed'. At the bottom, a log window shows several 'Info' messages from the 'acme-cm' system, all describing 'change ip-codec-set 1' operations. The log entries are as follows:

Severity	Date/Time	System	Description
Info	6/3/2015 4:53:11 PM	acme-cm	change ip-codec-set 1
Info	6/3/2015 3:01:42 PM	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:30:06 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:17 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:01 ...	acme-cm	change ip-codec-set 1

The interface also includes a left-hand navigation menu with options like 'Start GEDI', 'Add User', 'Change User', and 'Remove User'. The bottom status bar shows 'Ready' and a 'NUM' indicator.

### 3. Create the Vector Directory Numbers (VDN)

The screenshot displays the 'Site Administration' interface for 'acme-cm GED'. The main window shows a table of Vector Directory Numbers (VDN) with the following data:

Name	Extension	UDN	Ovrd	CDR	TN	Vector	Measured	Orig	Ann	Event	Notif	Skill 1	Skill 2	Skill 3	VEC/PRT
TechSupport	46003	n		1	1	3	none								U
Sales	46004	n		1	1	4	none								U
Billing	46005	n		1	1	5	none								U
WelcomeToLab	46006	n		1	1	6	none								U
DirectAgent	46010	n		1	1	10	none								U

Below the table, a status bar indicates 'Command successfully completed'. At the bottom, a log window shows several 'Info' messages from 'acme-cm' regarding 'change ip-codec-set 1' at various times on 6/3/2015 and 5/29/2015.

#### 4. Create the HUNT GROUPS and associated extensions

The screenshot shows the 'Site Administration - [acme-cm GEDI]' interface. The main window displays a table of hunt groups and extensions. The table has the following columns: Number, Name, Extension, Type, ACD/HEARS, Vector?, MCH, Queue, Members, Cover, Path, Notif/Ctg Adj, Dom Ctr1, Msg Center, and Msg Center AUDIX N. The data rows are:

Number	Name	Extension	Type	ACD/HEARS	Vector?	MCH	Queue	Members	Cover	Path	Notif/Ctg Adj	Dom Ctr1	Msg Center	Msg Center AUDIX N
1	MM	46000	ucd-mia	n/-	n	none	n	0			n		S	
2	TechSupport	46001	ucd-mia	y/I	SK	none	y	0			n		n	
3	Billing	46002	ucd-mia	y/I	SK	none	y	0			n		n	
4	Sales	46007	ucd-mia	y/I	SK	none	y	0			n		n	

Below the table, a status bar indicates 'Command successfully completed'. At the bottom, there is a log window showing several 'Info' messages from 'acme-cm' regarding 'change ip-codec-set 1' at various times on 6/3/2015 and 5/29/2015. The interface also includes a left-hand navigation menu with options like 'Start GEDI', 'Add User', 'Change User', etc., and a top menu bar with 'File', 'Edit', 'View', 'System', 'Action', 'Tools', 'Window', and 'Help'.



Site Administration - [acme-cm GED]

File Edit View System Action Tools Window Help

acme-cm

send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8) Restore Column order

Ext.	Type	Port	Name	Data Ext.	Cover 1	Cover 2	COS	COR	TN	Room	Jack	Cable	Survivable	GK	Node	Name
40000	4620	S00090	H323StationSoftPhone				1	1	1							
40010	9641SIP	S00076	test5, avaya				1	1	1							
40011	9641SIP	S00075	test4, avaya				1	1	1							
40020	4620	S00097	H323Station1				1	1	1							
40021	4620	S00100	H323Station2				1	1	1							
555-555-0001	4620	S00004	charles 10 digits				1	1	1							
617-555-1000	9640SIP	S00071	test0, avaya				1	1	1							
617-555-1001	9621SIP	S00072	test1, avaya				1	1	1							
617-555-1002	9621SIP	S00073	test2, avaya				1	1	1							
617-555-1003	9621SIP	S00074	test3, avaya				1	1	1							
617-555-1032	9621SIP	S00077	test6, avaya				1	1	1							
781-443-7246	9640SIP	S00000	test, avaya				1	1	1							
781-443-7247	9620SIP	S00003	test8, avaya				1	1	1							

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
- Swap Stations
- MultiLingual Name Wizard

Advanced

Fault & Performance

Announcements

Tasks Tree

Command successfully completed

Severity	Date/Time	System	Description
Info	6/3/2015 4:53:11 PM	acme-cm	change ip-codec-set 1
Info	6/3/2015 3:01:42 PM	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:30:06 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:17 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:01 ...	acme-cm	change ip-codec-set 1

History Schedule Connection Status

Ready NUM

## 5. Create the Vectors needed for conditional routing

The screenshot shows the 'Site Administration' interface for 'acme-cm GEDI'. The main window displays a table of vectors under the 'General' tab. The table has two columns: 'Number' and 'Name'. The data is as follows:

Number	Name
3	TechSupport
4	Sales
5	Billing
6	Welcome
10	DirectAgent

Below the table, a status bar indicates 'Command successfully completed'. At the bottom of the window, a log window shows several entries:

Severity	Date/Time	System	Description
Info	6/3/2015 4:53:11 PM	acme-cm	change ip-codec-set 1
Info	6/3/2015 3:01:42 PM	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:30:06 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:17 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:01 ...	acme-cm	change ip-codec-set 1

The log window also includes navigation buttons for 'History', 'Schedule', and 'Connection Status'.

Site Administration - [acme-cm GEDI]

File Edit View System Action Tools Window Help

acme-cm

display vector 3 send (return) help (F5) cancel (esc) enter (F3) schedule (F8) next (F7) previous (F8) Restore Column order

1 2 3 4 5 6

**CALL VECTOR**

Number: 3 Name: TechSupport

Multimedia? n Attendant Vectoring? n Meet-me Conf? n Lock? n  
 Basic? y EAS? y G3U4 Enhanced? y ANI/11-Digits? y ASAI Routing? y  
 Prompting? y LAI? y G3U4 Adv Route? y CINFD? y BSR? y Holidays? y  
 Variables? y 3.0 Enhanced? y

01 wait-time 0 secs hearing ringback  
 02 announcement 46503  
 03 queue-to skill 2 pri m  
 04 wait-time 10 secs hearing music  
 05 goto step 3 if unconditionally  
 06 stop  
 07  
 08  
 09  
 10  
 11  
 12

Advanced  
 Fault & Performance  
 Announcements

Tasks Tree

Severity	Date/Time	System	Description
Info	6/3/2015 4:53:11 PM	acme-cm	change ip-codec-set1
Info	6/3/2015 3:01:42 PM	acme-cm	change ip-codec-set1
Info	5/29/2015 3:30:06 ...	acme-cm	change ip-codec-set1
Info	5/29/2015 3:28:17 ...	acme-cm	change ip-codec-set1
Info	5/29/2015 3:28:01 ...	acme-cm	change ip-codec-set1

History Schedule Connection Status

Ready NUM

Site Administration - [acme-cm GED]

File Edit View System Action Tools Window Help

display inc-call-handling-tim tur send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8) Restore Column order

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30

**INCOMING CALL HANDLING TREATMENT**

Service/Feature	Number Len	Number Digits	Del	Insert
tie	5	46010	5	40020

Start GEDI  
Add User  
Change User Extension  
Change User  
Remove User  
Add Bridged Appearance  
Browse Dial Planes  
Browse Stations  
Browse Unused Ports  
Find Unused Extension(s)  
Print Button Labels  
Swap Stations  
MultiLingual Name Wizard

Advanced  
Fault & Performance  
Announcements

Tasks Tree

Severity	Date/Time	System	Description
Info	6/3/2015 4:53:11 PM	acme-cm	change ip-codec-set 1
Info	6/3/2015 3:01:42 PM	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:30:06 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:17 ...	acme-cm	change ip-codec-set 1
Info	5/29/2015 3:28:01 ...	acme-cm	change ip-codec-set 1

History Schedule Connection Status

Ready NUM

## Test Results

Category	ID	Abstract	Test Result	Comments
<b>Call Center Application (SIP) - PSTN to CPE Termination - DNIS or CED Routing</b>  Origination (US) and Termination (US)	3101	<b>DNIS Based Routing (CC/ACD)</b> - From populated with CPN information - PAI populated with CPN - Routes call to desired dept/group/skill set,etc), - Verify Voice Cut Through - Agent is Available - Keep call up for 2 minutes - Agent/Called Party disconnects from the call first - Caller/Calling Party is automatically disconnected after network receives BYE from CPE	PASS	
	3102	<b>CED Based Routing (CC/ACD)</b> - From "Unavailable" <sip: Anonymous@ip.address> - PAI (excluded) - Prompts caller for digits - Based on CED, Routes call to desired dept/group/skill set,etc), - Verify Voice Cut Through - Agent is Available - Keep call up for 2 minutes - Caller/Calling Party disconnects from the call first - Agent/Called Party is automatically disconnected after network receives BYE from CPE	PASS	
	3103	<b>DNIS or CED Based Routing (CC/ACD)</b> - Routes call to desired dept/group/skill set,etc), - Verify Voice Cut Through - Agent is Unavailable - Call placed into queue - Out-of-Queue when agent becomes available - Keep call up for 2 minutes	PASS	
	3104	<b>DNIS or CED Based Routing (ACD/CC)</b> - Agent places caller on hold - Verify Music on Hold, Off Hold, Retrieve from Hold - Verify voice path is re-established between agent/caller	PASS	
	3107	<b>DNIS or CED Based Routing (ACD/CC) - Long Duration Call</b> - up for a minimum of 1 hour - Tester verifies that call remains up for a minimum of 1 hour and then reflect test as pass or fail accordingly	PASS	

<b>IP-PBX (SIP) - PSTN to CPE termination</b>  Origination (US) and Termination (US)	3108	<b>IP PBX (no ACD or CC)</b> - From populated with CPN information - PAI populated with CPN - Routes directly to agent - Agent is available and answers call - Verify Voice Cut Through - Keep call up for 2 minutes - Agent/Called Party drops from call first - Caller is disconnected automatically after network receives BYE from CPE	PASS	
	3109	IP PBX (no ACD/CC) - From "Unavailable" <sip: Anonymous@ip.address> - PAI (excluded) - Routes directly to agent - Verify Voice Cut Through - Keep call up for 2 minutes - Caller/Calling Party drops from call first - Agent/IP PBX is automatically disconnected after IP PBX receives BYE from the network	PASS	
	3110	IP PBX (no ACD/CC) - Agent places caller on hold - Verify Music on Hold, Off Hold, Retrieve from Hold - Verify voice path is re-established between agent/caller	PASS	
<b>DNIS Translations</b>	3115	Translate 9 or 10 digits DNIS - 9 or 10 digits will be set as default - Mark as "Pass" or "Fail" based on results observed on CC/ACD and IP PBX calls	PASS	
	3116	Translate 15 digits DNIS	PASS	
	3117	Translate 21 digits DNIS	PASS	
	3118	PPSP 1 (G.729A) - annex=no	PASS	
	3119	PPSP 2 (G.729AB) - without annex parameter or with annex=yes	PASS	
	3120	PPSP 3 (G.711) - mulaw only	PASS	
	3122	PPSP X (any PPSP populated with 3 codecs) Default - g.729 w/annex=no, g.711u, g.726	PASS	

<b>Codec Negotiation</b>	3124A	<b>G.729 Mismatch</b> - Network offers G.729A, CPE provisioned G.729B CPE negotiates G.729A (label test case 3124A)  (Describe CPE's response)	PASS	
	3124B	<b>G.729 Mismatch</b> - Network offers G.729AB, CPE provisioned G.729A CPE negotiates G.729A (label test case 3124B) (Describe CPE's response)	PASS	
<b>Dynamic Payload</b>	3125	PSTN to Call Center or IP PBX - Verify that CPE automatically updates Dynamic Payload during an IPTF call	PASS	
<b>Codec Negotiation - Failure</b>	3126	CPE must return of the SIP Error responses for unsupported codec - 415 Unsupported media type - 488 Not Acceptable Here - 606 Not Acceptable	PASS	
	3130	ADR/BUSY - 486 - Busy Here (User Busy) - Indicate which configuration (CC or IP-PBX sends back this SIP Error response	PASS	
	3131	ADR/BUSY - 503 - Service Unavailable - Indicate which configuration (CC or IP-PBX sends back this SIP Error response	CONDITIONAL PASS	Avaya CM sends 503 Service Unavailable, but the Session Manager changes the 503 error code to 500
	3132	ADR/BUSY - 500 - Server Internal Error	PASS	
	3135	Transfer to 8YY - If CC subscription, call is connected to TP. Verify RP and TP voice path - TP drops from call (simply hangs up the phone) - Confirm that CP and RP voice-path is still established - If BT or ST subscription, the network drops RP from call therefore dropping the TP would not apply	PASS	

<b>Legacy Transfer Connect (Inband) Human</b>	3136	Transfer to POTS - TP drops from call (simply hangs up the phone) - Confirm that CP and RP voice-path is re-established - If BT or ST subscription, the network drops RP from call therefore dropping the TP would not apply	PASS	
	3137	Request Help - **4/**H - First request enter **4/**H (hear announcement) - Second request enter **4**H (hear 2nd announcement) - Third attempt is blocked - no future assistance	PASS	
	3138	Delete Partial Digits - *3/*D - Enter a few wrong digits to start transfer and then enter *3/*D to delete digits entered - Immediately enter correct digits - Verify call connects to desired TP	PASS	
	3139	Place CP on Hold - *4/*H - IPTF call is established - Agent/RP decides to place CP on hold by entering *4/*H - Verify CP is on hold - Agent/RP removes CP from hold by entering *7/*R - Verify voice path between CP and RP is re-established	PASS	
	3140	RP drops TP - **9/**X - Establish a call to the TP - After verifying voice path, drop TP from the call by entering **9/**X - Verify voice path between CP and RP is re-established	PASS	
	3141	RP drops immediately after transfer to TP - TP and CP merged - RP Transfers call to the TP - RP does not wait for connection and simply drops from the call - Verify voice path between CP and TP is established	CONDITIONAL PASS	CP heard the announcement played after the RP dropped from the call and responded by entering the prompter digits. However no indication on trunk that the call routed back into SBC for the TP leg of the call



	3142	CP, RP, and TP call legs merged - <b>CC subscription only</b> - Establish call to TP - RP enters */*R to retrieve CP from hold - Verify all three parties are on the call (CP, RP, and TP) - TP drops from the call	PASS	
	3144	Error Response - TP Busy/Unavailable - Attempt to transfer the call to the TP - Verify that when the TP is unavailable, the RP hears appropriate network response	PASS	
	3148	Error Response - Invalid DN/SDC - RP attempts to transfer call with an invalid DN/SDC - Verify that the network responds with appropriate announcement - RP ends calls	PASS	
<b>Intra-site Unattended Transfer</b>	3158	PSTN to Phone 1/Agent 1. Phone 1 transfers to Phone 2/Agent 2 at same IP PBX site.	PASS	
	3159	Ring Back Unattended Transfer	PASS	
<b>Intra-site Attended Transfer</b>	3160	PSTN to Phone 1/Agent 1. Phone 1 transfers to Phone 2/Agent 2 at same IP PBX site.	PASS	
<b>Intra-site Conference</b>	3161	PSTN to Phone 1/Agent 1. Phone 1 conferences Phone 2/Agent 2 at same IP PBX site. Drop Transfer Call	PASS	
<b>Voice Quality</b>	3171	PSTN to CPE (CC or IP PBX)	PASS	
<b>Simultaneous Calls (Minimum of 2)</b>	3173	PSTN to CPE (CC or IP PBX)	PASS	
<b>MISC-SIP Options</b>	3177	CPE - Accepts SIP Options - SIP Options received - CPE responds with 200 OK or other message	PASS	
<b>PSTN to CPE - AVPN Transport</b>	3178	Confirm Voice Traffic (RTP & Signaling) is assigned to Class Of Service 1 (COS1)	PASS	

<b>PSTN to CPE - US, Canada, and MOW</b>	3182	PSTN to CPE - Contact Center/IP PBX CP (Canada) RP (US)	PASS	
	3183	PSTN to CPE - Contact Center/IP PBX CP (US) RP (MOW)	PASS	
	3184	PSTN to CPE - Contact Center/IP PBX CP (Canada) RP (MOW)	PASS	
	3187	ADR/BUSY - 486 - Busy Here (User Busy) - Indicate which configuration (CC or IP-PBX sends back this SIP Error response	PASS	
	3188	ADR/BUSY - 500 - Server Internal Error	PASS	
	3189	ADR/BUSY - 503 - Service Unavailable - Indicate which configuration (CC or IP-PBX sends back this SIP Error response	CONDITIONAL PASS	Avaya CM sends 503 Service Unavailable, but the Session Manager changes the 503 error code to 500
<b>INFOPACK - CP to RP</b>	3301	<b>CPN = no, BN = yes, OLI = yes, UUI = yes</b> - Verify BN, OLI, UUI (if available) populated in incoming INVITE of RP Note: UUI will only be present if it is sent from the PSTN to IP network	CONDITIONAL PASS	UUI header not present in FROM header hence marking conditionally pass
	3302	<b>CPN = yes, BN = yes, OLI = yes, UUI = yes</b> - Verify CPN, BN, OLI, UUI (if available) populated in incoming INVITE of RP Note: UUI will only be present if it is sent from the PSTN to IP network	CONDITIONAL PASS	UUI header not present in FROM header hence marking conditionally pass
	3303	<b>CPN = yes, BN = no, OLI = yes, UUI = no</b> - Verify OLI, CPN(if available) populated in incoming INVITE of RP	PASS	

The summary of the test plan is as follows

Total Test Cases	Pass	Fail	Conditional Pass
49	44	0	5

## 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, 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 from Avaya and SBC.

### Avaya Aura SM/CM

The Avaya SM can be accessed via SSH. The command used to trace the SIP messages in and out of the Avaya SM/CM is traceSM, this is particularly useful to troubleshoot any signaling between the Avaya SM and CM.

Configuration checklist when outbound calls are failing:

- Check for Dial plan/route issues
- Check Gateway and trunk configuration for TCP connections
- SIP OPTIONS message connectivity between SBC and Avaya

### Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from [www.wireshark.org](http://www.wireshark.org). Wireshark could be installed on the server hosting Avaya and have the SBC send packet trace to this remote location.

### Oracle E-SBC 3820

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:

```
ATT-IPTF# reset sipd
ATT-IPTF# notify sipd debug
ATT-IPTF#
enabled SIP Debugging
ATT-IPTF# 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 ATT-IPTFFTP 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](http://docs.oracle.com/cd/E56581_01/index.htm)

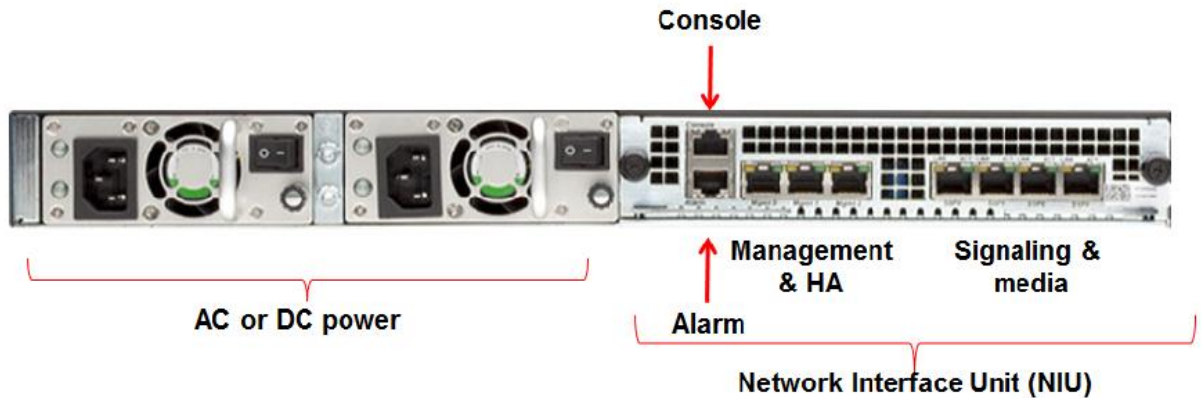
# Appendix A

## 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.

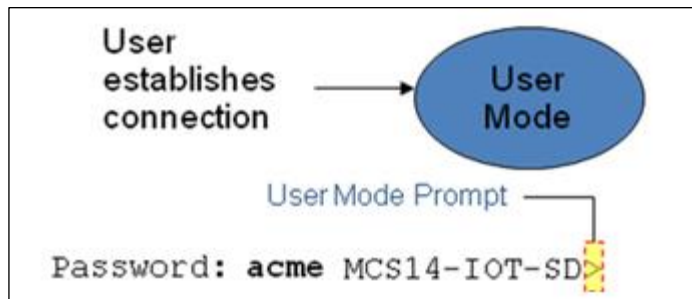


## ACLI Basics

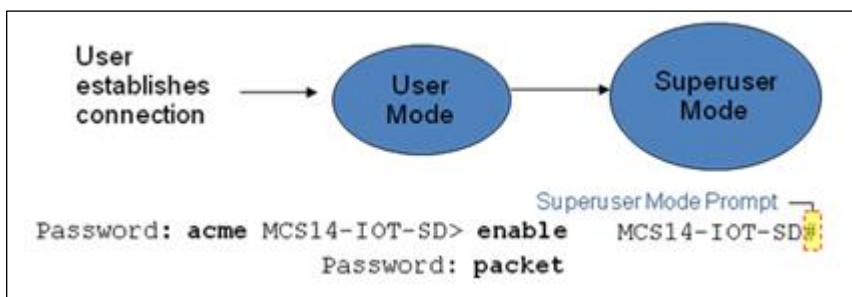
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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.



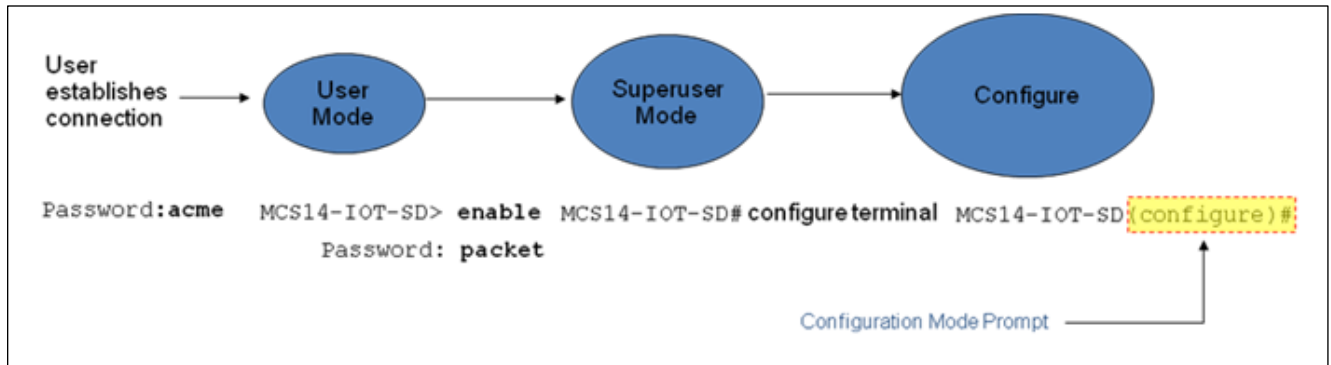
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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.

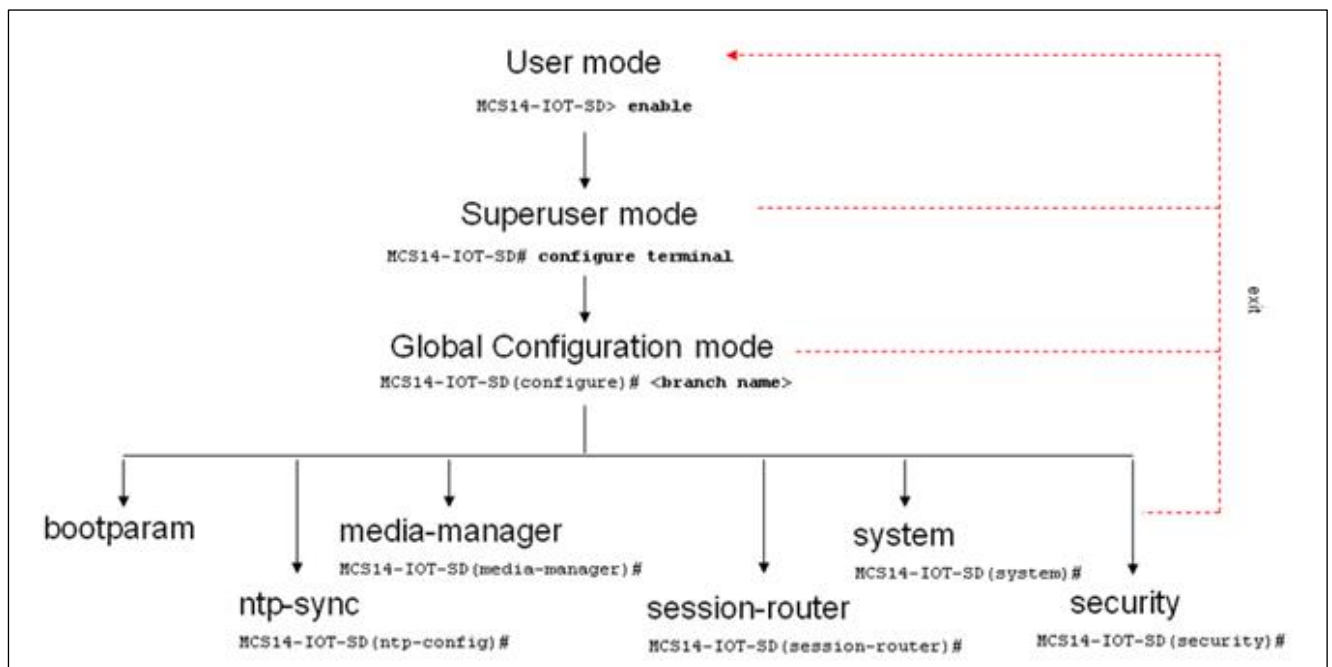
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the `configure terminal` command in the Superuser mode.

Configuration mode is identified by the word `configure` in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, `ATT-IPTF(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.

```
'.' = clear field; '-' = go to previous field; q = quit
boot device           : eth0
processor number      : 0
host name             :
file name             : /tffs0/nnSCX620.gz
inet on ethernet (e) : 10.0.3.11:ffff0000
inet on backplane (b) :
host inet (h)         : 10.0.3.100
gateway inet (g)      : 10.0.0.1
user (u)              : anonymous
ftp password (pw) (blank = rsh) : anonymous
flags (f)             : 0x8
target name (tn)      : MCS14-IOT-SD
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.

```
ATT-IPTF # 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'.
ATT-IPTF #
```

### 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.

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

## Appendix B – SBC configuration

```
PE11-ATT-Trunk# show run
capture-receiver
  state          enabled
  address        10.232.50.76
  network-interface  sip0:0
  last-modified-by  admin@172.18.0.119
  last-modified-date 2015-03-20 16:10:02
local-policy
  from-address   *
  to-address     *
  source-realm   core
  description
  activate-time
  deactivate-time
  state          enabled
  policy-priority none
  policy-attribute
    next-hop      105.105.105.105
    realm          trunk-side
    action         none
    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@172.18.0.119
  last-modified-date 2015-04-10 14:27:09
local-policy
  from-address   *
  to-address     *
  source-realm   trunk-side
  description
  activate-time
  deactivate-time
  state          enabled
  policy-priority none
  policy-attribute
    next-hop      10.232.50.102
    realm          core
    action         none
```

terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
state	enabled
app-protocol	
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	
last-modified-by	admin@172.18.0.109
last-modified-date	2015-02-10 20:46:01
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
app-signaling-bandwidth	0
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
fragment-msg-bandwidth	0
rfc2833-timestamp	disabled
default-2833-duration	100
rfc2833-end-pkts-only-for-non-sig	enabled
translate-non-rfc2833-event	disabled

```

media-supervision-traps      disabled
dnsmalg-server-failover    disabled
syslog-on-call-reject      disabled
last-modified-by           admin@172.18.0.119
last-modified-date         2014-06-02 15:58:04
network-interface
  name                       s0p0
  sub-port-id                0
  description
  hostname
  ip-address                 200.105.105.105
  pri-utility-addr
  sec-utility-addr
  netmask                   255.255.255.0
  gateway                   200.105.105.1
  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                200.105.105.105
  ftp-address
  icmp-address               200.105.105.105
  snmp-address
  telnet-address
  ssh-address
  last-modified-by          admin@172.18.0.193
  last-modified-date        2015-04-08 08:01:28
network-interface
  name                       slp0
  sub-port-id                0
  description
  hostname
  ip-address                 10.232.50.130
  pri-utility-addr
  sec-utility-addr
  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@172.18.0.109
last-modified-date   2015-02-16 13:19:35
phy-interface
  name                s0p0
  operation-type       Media
  port                 0
  slot                 0
  virtual-mac
  admin-state          enabled
  auto-negotiation     enabled
  duplex-mode          FULL
  speed                100
  wancom-health-score  50
  overload-protection  disabled
  mac-filtering        disabled
  last-modified-by     admin@172.18.0.158
  last-modified-date   2015-04-13 17:07:21
phy-interface
  name                slp0
  operation-type       Media
  port                 0
  slot                 1
  virtual-mac
  admin-state          enabled
  auto-negotiation     enabled
  duplex-mode          FULL
  speed                100
  wancom-health-score  50
  overload-protection  disabled
  mac-filtering        disabled
  last-modified-by     admin@172.18.0.170
  last-modified-date   2014-05-29 16:00:46
realm-config
  identifier           core
  description
  addr-prefix          0.0.0.0
  network-interfaces  slp0:0
  mm-in-realm          disabled
  mm-in-network        enabled
  mm-same-ip           enabled
  mm-in-system         enabled

```

bw-cac-non-mm	disabled
msm-release	disabled
qos-enable	disabled
generate-UDP-checksum	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
hold-refer-reinvite	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@172.18.0.119
last-modified-date	2014-06-02 15:55:09
realm-config	
identifier	trunk-side
description	
addr-prefix	0.0.0.0
network-interfaces	s0p0:0
mm-in-realm	disabled
mm-in-network	enabled
mm-same-ip	enabled
mm-in-system	enabled
bw-cac-non-mm	disabled
msm-release	disabled
qos-enable	disabled
generate-UDP-checksum	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
hold-refer-reinvite	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@172.18.0.109
last-modified-date	2015-02-11 20:50:16
session-agent	
hostname	10.232.50.102
ip-address	10.232.50.102
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	StaticTCP
realm-id	core
egress-realm-id	
description	Avaya SM
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	100
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
hold-refer-reinvite	disabled
send-tcp-fin	disabled
last-modified-by	web_admin@172.18.0.143
last-modified-date	2015-04-22 11:13:22
session-agent	
hostname	10.232.50.103
ip-address	10.232.50.103
port	5060
state	enabled
app-protocol	SIP
app-type	
transport-method	StaticTCP
realm-id	core
egress-realm-id	
description	Direct CM trunk
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
hold-refer-reinvite disabled
send-tcp-fin disabled
last-modified-by web_admin@172.18.0.143
last-modified-date 2015-04-14 23:01:12
session-agent
hostname 105.105.105.105
ip-address 105.105.105.105
port 5060
state enabled
app-protocol SIP
app-type
transport-method UDP
realm-id trunk-side
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	
ping-interval	0
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	strip1
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	100
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
hold-refer-reinvite	disabled
send-tcp-fin	disabled
last-modified-by	web_admin@172.18.0.143
last-modified-date	2015-04-22 11:13:07
session-translation	
id	strip1
rules-calling	strip1
rules-called	strip1
last-modified-by	admin@172.18.0.119
last-modified-date	2015-04-15 13:51:46
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	0
register-service-route	always
init-timer	500
max-timer	4000
trans-expire	32
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	6000
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	disabled
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
match-sip-instance	disabled
sa-routes-stats	disabled
sa-routes-traps	disabled
rx-sip-reason-mapping	disabled
add-ue-location-in-pani	disabled
hold-emergency-calls-for-loc-info	0
last-modified-by	admin@172.18.0.109
last-modified-date	2015-02-12 16:45:04
sip-interface	
state	enabled
realm-id	core
description	
sip-port	
address	10.232.50.130
port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	all
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                          Changeptime
out-manipulationid                         NATtingavaya
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
sec-agree-feature                          disabled
sec-agree-pref                             ipsec3gpp
enforcement-profile
route-unauthorized-calls

```

tcp-keepalive	none
add-sdp-invite	disabled
p-early-media-header	disabled
p-early-media-direction	
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	
reg-cache-route	disabled
last-modified-by	web_admin@172.18.0.143
last-modified-date	2015-04-14 16:42:50
sip-interface	
state	enabled
realm-id	trunk-side
description	
sip-port	
address	200.105.105.105
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         RemoveExtraDigits
out-manipulationid        NAT_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           100
rfc2833-mode              transparent
constraint-name
response-map
local-response-map
sec-agree-feature         disabled
sec-agree-pref            ipsec3gpp
enforcement-profile
route-unauthorized-calls
tcp-keepalive            none
add-sdp-invite            disabled
p-early-media-header      disabled
p-early-media-direction
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
reg-cache-route               disabled
last-modified-by              admin@172.18.0.119
last-modified-date            2015-05-19 15:30:42
sip-manipulation
  name                         Changeptime
  description
  split-headers
  join-headers
  header-rule
    name                       Checkptimeexists
    header-name                 Content-type
    action                      store
    comparison-type             pattern-rule
    msg-type                    any
    methods
    match-value
    new-value
    element-rule
      name                      ptimeexists
      parameter-name             application/sdp
      type                      mime
      action                    store
      match-val-type             any
      comparison-type            pattern-rule
      match-value                (a=ptime:30)
      new-value
  header-rule
    name                       Addptime
    header-name                 Content-type
    action                      manipulate
    comparison-type             boolean
    msg-type                    any
    methods
    match-value                 !($Checkptimeexists.$ptimeexists)
    new-value
    element-rule
      name                      Changesdp
      parameter-name             application/sdp
      type                      mime
      action                    find-replace-all
      match-val-type             any
      comparison-type            pattern-rule
      match-value                (m=audio\,*)
      new-value                  $1+$CRLF+a=ptime:30
last-modified-by              admin@172.18.0.119
last-modified-date            2015-04-07 14:03:21

```

```

sip-manipulation
  name NAT_IP
  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
    element-rule
      name SourcePort
      parameter-name
      type uri-port
      action replace
      match-val-type any
      comparison-type case-sensitive
      match-value
      new-value $LOCAL_PORT
  header-rule
    name To
    header-name To
    action manipulate
    comparison-type case-sensitive
    msg-type request
    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
  header-rule
    name CheckPAIexists
    header-name P-Asserted-Identity

```

action	store
comparison-type	case-sensitive
msg-type	request
methods	
match-value	
new-value	
header-rule	
name	Add_P_Asserted_ID_new
header-name	P-Asserted-Identity
action	add
comparison-type	boolean
msg-type	request
methods	INVITE
match-value	!\$CheckPAIexists
new-value	"<sip:"+\$FROM_USER.\$0+"@"+\$LOCAL_IP+>"
header-rule	
name	Changetransport
header-name	From
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	Changetransport
parameter-name	transport
type	uri-param
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	Tcp
new-value	udp
header-rule	
name	StripBlinc
header-name	Content-type
action	manipulate
comparison-type	pattern-rule
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	deleteblinc
parameter-name	application/sdp
type	mime
action	find-replace-all
match-val-type	any
comparison-type	case-sensitive
match-value	(b=AS.*)\r\n
new-value	
last-modified-by	admin@172.18.0.119
last-modified-date	2015-04-15 14:45:53

sip-manipulation

```
name NATtingavaya
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 aura.com
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 aura.com
header-rule
    name Ruri_hr
    header-name Request-URI
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule
        name er
        parameter-name
```

```

        type uri-host
        action find-replace-all
        match-val-type any
        comparison-type case-sensitive
        match-value
        new-value aura.com
header-rule
    name Pai
    header-name P-Asserted-Identity
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule
        name Pai_header
        parameter-name
        type uri-host
        action replace
        match-val-type any
        comparison-type case-sensitive
        match-value
        new-value aura.com
last-modified-by admin@172.18.0.109
last-modified-date 2015-02-13 15:26:31
sip-manipulation
    name RemoveExtraDigits
    description
    split-headers
    join-headers
    header-rule
        name RemovefromReqURI
        header-name Request-URI
        action manipulate
        comparison-type case-sensitive
        msg-type request
        methods
        match-value
        new-value
        element-rule
            name Stripdigits
            parameter-name
            type uri-user
            action find-replace-all
            match-val-type any
            comparison-type case-sensitive
            match-value [0-9]{6,}([0-9]{5,})
            new-value "00000"+$1
last-modified-by admin@172.18.0.119
last-modified-date 2015-05-19 15:25:43
steering-pool
    ip-address 10.232.50.130

```



start-port	10500
end-port	11000
realm-id	core
network-interface	
last-modified-by	admin@172.18.0.109
last-modified-date	2015-02-10 20:43:31
steering-pool	
ip-address	200.105.105.105
start-port	16384
end-port	20000
realm-id	trunk-side
network-interface	
last-modified-by	admin@172.18.0.170
last-modified-date	2014-07-10 14:12:53
system-config	
hostname	ATT-trunk-IOT
description	
location	
mib-system-contact	
mib-system-name	
mib-system-location	
snmp-enabled	enabled
enable-snmp-auth-traps	disabled
enable-snmp-syslog-notify	disabled
enable-snmp-monitor-traps	disabled
enable-env-monitor-traps	disabled
snmp-syslog-his-table-length	1
snmp-syslog-level	WARNING
system-log-level	WARNING
process-log-level	DEBUG
process-log-ip-address	0.0.0.0
process-log-port	0
collect	
sample-interval	5
push-interval	15
boot-state	disabled
start-time	now
end-time	never
red-collect-state	disabled
red-max-trans	1000
red-sync-start-time	5000
red-sync-comp-time	1000
push-success-trap-state	disabled
comm-monitor	
state	disabled
sbc-grp-id	0
tls-profile	
qos-enable	enabled
call-trace	disabled
internal-trace	disabled
log-filter	all
default-gateway	155.212.214.1
restart	enabled

exceptions	
telnet-timeout	0
console-timeout	0
remote-control	enabled
cli-audit-trail	enabled
link-redundancy-state	disabled
source-routing	disabled
cli-more	disabled
terminal-height	24
debug-timeout	0
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@172.18.0.119
last-modified-date	2014-06-04 11:03:23
translation-rules	
id	strip1
type	delete
add-string	
add-index	0
delete-string	1
delete-index	0
last-modified-by	admin@172.18.0.119
last-modified-date	2015-04-15 13:50:13



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