



Oracle Enterprise Session Border Controller and CUCM 10.5 with Bell Canada Enterprise SIP Trunk

Technical Application Note

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Intended Audience

This document is intended for use by Oracle Systems Engineers, third party Systems Integrators, 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.

Document Overview

Cisco Call Manager offers the ability to connect to Internet telephony service providers (ITSP) using an IP-based SIP trunk. This reduces the cost and complexity of extending an enterprise's telephony system outside its network borders. Oracle Enterprise Session Border Controllers (E-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 SIP trunking between Cisco Call Manager, Oracle E-SBCs and Bell Canada IP Trunking services are configured in the optimal manner.

Introduction

Audience

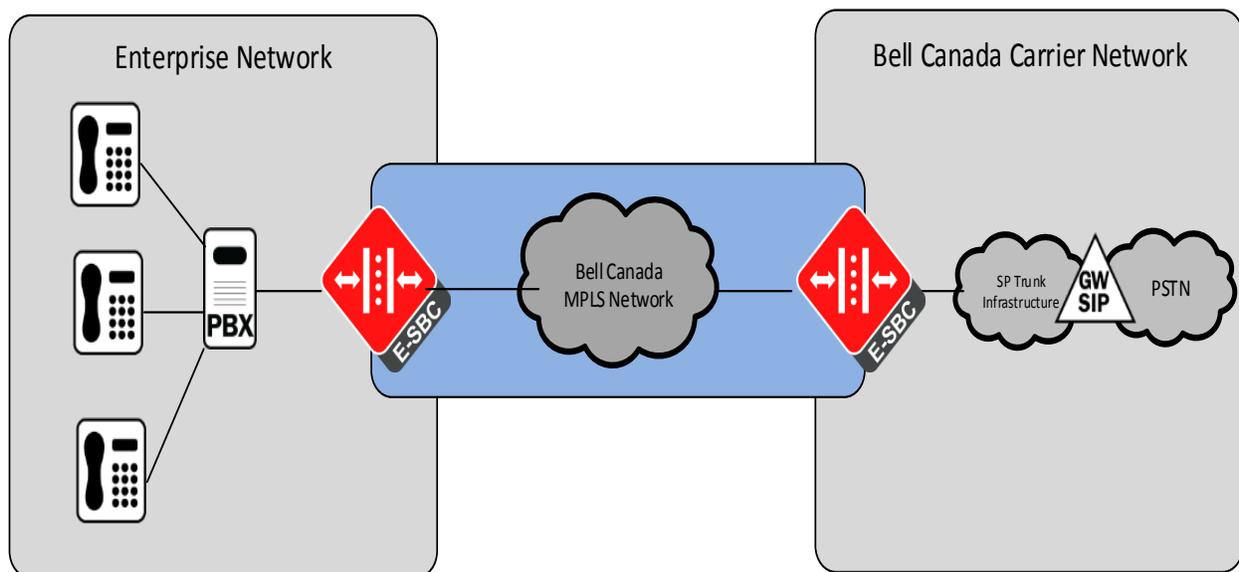
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise Session Border Controller and CUCM. There will be steps that require navigating the Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, SIP/RTP, TLS and SRTP are also necessary to complete the configuration and for troubleshooting, if necessary.

Requirements

- Fully installed and configured Cisco Call Manager 10.5
- Oracle Enterprise Session Border Controller is running ECZ7.3.0 Patch 2 (Build 75)
 - Note: the configuration running on the E-SBC is backward/forward compatible with any release in the 7.3.0 stream.
- Bell Canada trunk based customers with dedicated data connectivity to Bell Canada.

Architecture

The following reference architecture shows a logical view of the connectivity between CM and the E-SBC.



Phase 1: Configuring the Oracle Enterprise Session Border Controller

In this section we describe the steps for configuring an Oracle Enterprise Session Border Controller, formally known as an Acme Packet Net-Net Enterprise Session Director, for use with CUCM Server in a SIP trunking scenario.

In Scope

The following guide configuring the Oracle E-SBC assumes that this is a newly deployed device dedicated to a single customer. If a service provider currently has the E-SBC deployed then please see the ACLI Configuration Guide on 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 E-SBC. This document covers the setup for the E-SBC platform running ECZ7.3.0 or later. If instructions are needed for other Oracle E-SBC models, please contact your Oracle representative.

Out of Scope

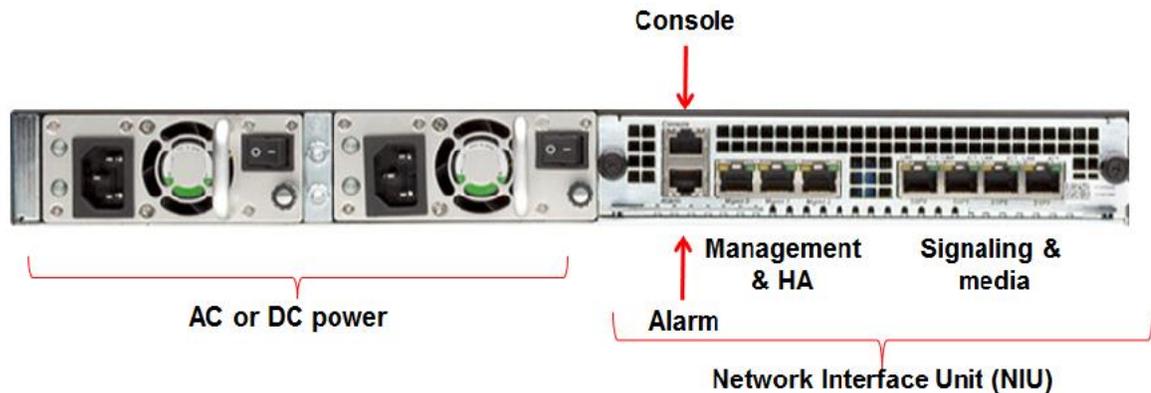
- Configuration of Network management including SNMP and RADIUS

What will you need

- Hypervisor with console connectivity through the hypervisor
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Super user modes on the Oracle E-SBC
- IP address to be assigned to management interface (Wancom0) of the E-SBC - the Wancom0 management interface must be connected and configured to a management network separate from the service interfaces. Otherwise the E-SBC is subject to ARP overlap issues, loss of system access when the network is down, and compromising DDoS protection. Oracle does not support E-SBC configurations with management and media/service interfaces on the same subnet.
- IP address of CUCM external facing NIC
- IP addresses to be used for the E-SBC internal and external facing ports (Service Interfaces)
- IP address of the next hop gateway in the service provider network

Configuring the E-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 (Bell Canada next-hop facing) network and the slot 1 port 1 (s1p1) interface into your inside (CUCM 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 **SBC1** 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 tEbmd...
Starting tSipd...
Starting tLrtd...
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
accli: max telnet sessions: 5
Password: 0x21a059c8 (tAlarm): eth0: Link is up (1000Mb/s full duplex)
```

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

```
Password: acme
SBC1> enable
Password: packet
SBC1# configure terminal
SBC1 (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 E-SBC by going to

```
SBC1#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
SBC1#(configure)bootparam
'.' = clear field;          '-' = go to previous field;      q = quit boot
device                      : eth0
processor number            : 0
host name                   : acmesystem
file name                   : /boot/nnECZ730m1p1.32.bz --- >location where the
                           software is loaded on the SBC
inet on ethernet (e)       : 192.168.1.22:fffff80 --- > 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)          : 192.168.1.1 -> gateway address here
user (u)                   : vxftp
ftp password (pw) (blank = use rsh): vxftp flags (f)   :
target name (tn)          : SBC1 -> ACLI prompt name & HA peer name
startup script (s)        :
other (o)                  :
```

Configuring the E-SBC

The following section walks you through configuring the Oracle E-SBC. It is outside the scope of this document to include all of the configuration elements as it will differ in every deployment.

High Availability

For additional information on High Availability please see the enterprise SBC documentation for more information (<http://www.oracle.com/technetwork/indexes/documentation/oracle-comms-acme-packet-2046907.html>). Interfaces wancom1 and 2 need to be added to facilitate HA communication between the two HA pairs.

```
phy-interface
  name          wancom1
  operation-type Control
  port          1
  slot          0
  virtual-mac
  admin-state   enabled
  auto-negotiation enabled
  duplex-mode   FULL
  speed         100
  wancom-health-score 8
  overload-protection disabled
  mac-filtering disabled
  last-modified-by admin@172.18.0.139
  last-modified-date 2016-07-21 18:12:08
phy-interface
  name          wancom2
  operation-type Control
  port          2
  slot          0
  virtual-mac
  admin-state   enabled
  auto-negotiation enabled
  duplex-mode   FULL
  speed         100
  wancom-health-score 9
  overload-protection disabled
  mac-filtering disabled
  last-modified-by admin@172.18.0.139
  last-modified-date 2016-07-21 18:12:15
network-interface
  name          wancom1
  sub-port-id   0
  description   HA_HEARTBEAT1
```

```
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
hip-ip-list
ftp-address
icmp-address
snmp-address
telnet-address
ssh-address
network-interface
  name               wancom2
  sub-port-id        0
  description        HA_HEARTBEAT2
  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
hip-ip-list
ftp-address
icmp-address
snmp-address
telnet-address
ssh-address

```

```

redundancy-config
  becoming-standby-time    360000
  peer
    name                    SBC1
    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                    SBC2
    type                    Secondary
    destination
      address                169.254.1.2:9090
      network-interface      wancom1:0
    destination
      address                169.254.2.2:9090
      network-interface      wancom2:0

```

Bell Canada Trunk Authentication Handling

Bell Canada forces authentication challenges on INVITE's. The Oracle Communications Enterprise Session Boarder Controller supports auth challenges. The SBC will respond to any auth challenges for SIP methods that are configured. The auth configuration need to be configured on the inside realm session-agent(s).

```

session-agent
  hostname              10.232.50.89
  ip-address            10.232.50.89

```

```

port                5060
state               enabled
app-protocol        SIP
app-type
transport-method    StaticTCP
realm-id            cisco-inside
...
monitoring-filters
auth-attributes
  auth-realm      lab.ca
  username        abc_123456_ca
  password        *****
  in-dialog-methods  INVITE
session-recording-server
session-recording-required  disabled

```

Header manipulation rules required for the Bell Canada Trunk

The header rules Update_Request, Update_To, Update_From and Update_Contact update the host portion of the URI to the fqdn for Request-URI, To, From and Contact headers according to the Bell Canada Spec. Some other parameters like otg.user=phone and tgrp are also added to the URI portion of the To and From headers according to the Bell Canada trunk specification.

```

sip-manipulation
  name                To_Bell
  description
  split-headers
  join-headers
  header-rule
    name              Update_Request
    header-name        request-uri
    action             manipulate
    comparison-type    case-sensitive
    msg-type           any
    methods
    match-value
    new-value
    element-rule
      name              Update_URI_Host
      parameter-name
      type              uri-host
      action            replace
      match-val-type    any
      comparison-type    case-sensitive
      match-value

```

new-value	lab.ca
element-rule	
name	Rmv_User
parameter-name	user
type	uri-param
action	delete-element
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	
element-rule	
name	Rmv_Port
parameter-name	
type	uri-port
action	delete-element
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	
header-rule	
name	Update_To
header-name	To
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	Update_URI_Host
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	lab.ca
element-rule	
name	Rmv_User
parameter-name	user
type	uri-param

action	delete-element
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	
header-rule	
name	Update_From
header-name	from
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	Update_URI_Host
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	lab.ca
element-rule	
name	Add_OTG_URI_Param
parameter-name	otg
type	uri-param
action	add
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	abc_ca
element-rule	
name	Rmv UriParam_User
parameter-name	user
type	uri-param
action	add
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	phone

```

header-rule
  name          Update_Contact
  header-name   Contact
  action        manipulate
  comparison-type case-sensitive
  msg-type      any
  methods       INVITE
  match-value
  new-value
  element-rule
    name        Add_tgrp
    parameter-name tgrp
    type         uri-user-param
    action       add
    match-val-type any
    comparison-type case-sensitive
    match-value
    new-value    abc_ca
  element-rule
    name        Add_trunk_context
    parameter-name trunk-context
    type         uri-user-param
    action       add
    match-val-type any
    comparison-type case-sensitive
    match-value
    new-value    lab.ca

```

The header rule Max-Forwards_0 changes the Max-forwards header in the OPTIONS message to 0, and the

```

header-rule
  name          Max_Forward_0
  header-name   Max-Forwards
  action        manipulate
  comparison-type pattern-rule
  msg-type      request
  methods       OPTIONS
  match-value
  new-value     0
  header-rule
    name        Rmv_UserAgent_Hdr
    header-name user-agent

```

```

action                delete
comparison-type      case-sensitive
msg-type              any
methods
match-value
new-value

```

The below set of header rules store the Diversion header in case it is present, if not then a new one is added. This Diversion header contains the BTN/Pilot number and is the responsibility of the administrator to configure. The Update_Diversion header rule adds the uri-params otg and user to the Diversion header according to the Bell spec.

```

header-rule
  name                save_Diversion
  header-name         Diversion
  action              store
  comparison-type     case-sensitive
  msg-type            any
  methods
  match-value
  new-value
header-rule
  name                Chk_Add_Diversion
  header-name         Diversion
  action              manipulate
  comparison-type     boolean
  msg-type            any
  methods             INVITE
  match-value         !$save_Diversion
  new-value           <sip:<613xxxxxxx>@domain-name;user=phone>
header-rule
  name                Update_Diversion
  header-name         Diversion
  action              manipulate
  comparison-type     case-sensitive
  msg-type            any
  methods
  match-value         !$save_Diversion
  new-value
element-rule
  name                Update_URI_Host

```

```

parameter-name
type          uri-host
action        replace
match-val-type    any
comparison-type    case-sensitive
match-value
new-value     lab.ca
element-rule
name          Add_OTG_URI_Param
parameter-name    otg
type          uri-param
action        add
match-val-type    any
comparison-type    case-sensitive
match-value
new-value     abc_ca
element-rule
name          Del_User_Param
parameter-name    user
type          uri-param
action        add
match-val-type    any
comparison-type    case-sensitive
match-value
new-value

```

The below set of header rules update the host portion of the PAI header to a FQDN specified by Bell, also adding the uri-param otg and user=phone.

```

header-rule
name          Update_PAI
header-name    P-Asserted-Identity
action        manipulate
comparison-type    case-sensitive
msg-type      any
methods
match-value
new-value
element-rule
name          Update_URI_Host

```

```

parameter-name
type          uri-host
action        replace
match-val-type    any
comparison-type    case-sensitive
match-value
new-value     lab.ca
element-rule
name          Add_User_UriParam
parameter-name    user
type          uri-param
action        add
match-val-type    any
comparison-type    case-sensitive
match-value
new-value     phone
element-rule
name          Add_OTG_URI_Param
parameter-name    otg
type          uri-param
action        add
match-val-type    any
comparison-type    case-sensitive
match-value
new-value     abc_ca

```

The below set of header-rules store the Referred-By header in case of call transfers using the REFER method. The Referred-By header is stored and then added back as the Diversion header on the INVITE sent out to the trunk, and then deleted so that it's not passed on to the trunk.

```

header-rule
name          save_Referred_By
header-name    Referred-by
action        store
comparison-type    case-sensitive
msg-type      request
methods       INVITE
match-value
new-value
element-rule

```

```

name                               Fix_URI_Host
parameter-name
type                               uri-host
action                             replace
match-val-type                     any
comparison-type                    case-sensitive
match-value
new-value                           lab.ca
header-rule
name                               Referred_By_2_Div
header-name                        Diversion
action                             add
comparison-type                    boolean
msg-type                           any
methods                            INVITE
match-value                        $save_Referred_By
new-value                          $save_Referred_By.$0
element-rule
name                               Update_URI_Host
parameter-name
type                               uri-host
action                             replace
match-val-type                     any
comparison-type                    case-sensitive
match-value
new-value                           lab.ca
header-rule
name                               RmvReferredBy
header-name                        Referred-by
action                             delete
comparison-type                    case-sensitive
msg-type                           any
methods
match-value
new-value

```

The below two header rules delete the Call-Info and Cisco-GUID headers sent by CUCM which are not required on the trunk side.

```

header-rule
name                               Rmv_CallInfo

```

header-name	Call-Info
action	delete
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
header-rule	
name	Rmv_CiscoGUID
header-name	Cisco-Guid
action	delete
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	

Webserver Configuration

A webserver is available on all Enterprise versions of Oracle E-SBCs. The Webserver can be used to provide tracing, configuration and dashboard info. For tracing info, 2 parts must be configured.

- The webserver must be enabled.
- Tracing filters must be applied.

web-server-config	
state	enabled
inactivity-timeout	5
http-state	enabled
http-port	80
https-state	disabled
https-port	443
tls-profile	

sip-monitoring	
match-any-filter	disabled
state	enabled
short-session-duration	0
monitoring-filters	*
trigger-window	30

Phase 2 – Configuring the Cisco Unified Call Manager v10.5

The enterprise will have a fully functioning CUCM v10.5 installed and deployed for this certification.

There are a few parts for configuring CUCM v10.5 to be configured and connected to operate with the Oracle E-SBC:

- Creating a SIP profile in CUCM and enabling OPTIONS ping to pro-actively monitor the SIP connectivity with the SBC
- Adding the SBC as a trunk to the CUCM infrastructure
- Creating a route pattern in the CUCM configuration to utilize the configured SBC trunk and route calls from CUCM to the SBC
- Additional configuration to add Directory Numbers, Phones to register to the CUCM and enabling a DHCP server for assigning IP addresses to Cisco phones

Creating a SIP Profile in CUCM

To add a new SIP Profile in CUCM, login into the CUCM console, use the Device --- > Device settings --- > SIP Profile menu path in CUCM. Click on Add new and following are the settings, rest can be default:

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Profile. The page is titled "SIP Profile Configuration" and includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. Below the navigation menu, there are icons for Save, Delete, Copy, Reset, Apply Config, and Add New.

SIP Profile Information

- Name*: SIP Profile - SIP Trunk to Bell Canada
- Description: [Empty field]
- Default MTP Telephony Event Payload Type*: 101
- Early Offer for G.Clear Calls*: Disabled
- User-Agent and Server header information*: Send Unified CM Version Information as User-Agent
- Version in User Agent and Server Header*: Major And Minor
- Dial String Interpretation*: Phone number consists of characters 0-9, *, #, an
- Confidential Access Level Headers*: Disabled
- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance

SDP Information

- SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*: TIAS and AS
- SDP Transparency Profile: Pass all unknown SDP attributes
- Accept Audio Codec Preferences for Re-invites*: [Empty field]

SIP Profile Configuration

- Reject Anonymous Outgoing Calls
- Send ILS Learned Destination Route String

SIP OPTIONS Ping

- Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
- Ping Interval for In-service and Partially In-service Trunks (seconds)*: 60
- Ping Interval for Out-of-service Trunks (seconds)*: 120
- Ping Retry Timer (milliseconds)*: 500
- Ping Retry Count*: 6

SDP Information

- Send send-receive SDP in mid-call INVITE
- Allow Presentation Sharing using BFCP
- Allow iX Application Media
- Allow multiple codecs in answer SDP

At the bottom of the page, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New.

Adding the E-SBC as a trunk in CUCM

The following process details the steps to add the SBC as a trunk in CUCM Web UI

1. On the CUCM administration console (UI), maneuver to **Device --- > Trunk**. Click on New
2. Select SIP Trunk from the Trunk Type drop down menu and protocol will also be SIP
3. Let default of none be selected on the Trunk service type
4. Following 2 screenshots are the other settings to be configured on the Trunk, all other parameters set to default

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Save Delete Reset Add New

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

Media Termination Point Required
 Retry Video Call as Audio

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration
admin | Search Documentation | About

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links: [Back To Find/List](#)

Save Delete Reset Add New

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason
1*	10.232.50.20		5060	down	local=2

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1		

Creating a route Pattern in CUCM

Route pattern in CUCM take the form of regular expressions to define specific routes and give flexibility in network design for dialing outbound calls from CUCM users to the PSTN via the E-SBC. A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that route calls to a route list or a gateway/trunk. In CUCM administration console, use the Call Routing --- >Route/Hunt --- >Route Pattern menu path to configure route patterns. Follow the fields in the screenshots below:



Route Pattern Configuration

Save Delete Copy Add New

Route Pattern*

Route Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern

Call Classification*

External Call Control Profile

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*



Route Pattern Configuration

Save Delete Copy Add New

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

Adding DHCP server and subnet in CUCM

In CUCM administration console, use the System --- >DHCP --- >DHCP Server menu path to define/add a new DHCP server. Use the IP address of the CUCM as the DHCP server/primary/secondary TFTP server address for the phones. Phones will use DHCP option 150 to discover the address of CUCM and request an IP address. Below is the screenshot for the same:

The screenshot shows the Cisco Unified CM Administration interface for DHCP Server Configuration. The page title is "DHCP Server Configuration" and it includes a navigation menu with options like System, Call Routing, Media Resources, etc. Below the title, there are icons for Save, Delete, Copy, and Add New. The main section is titled "DHCP Server Information" and contains a form with the following fields:

Field Name	Value
Host Server*	CUCM-Cisco
Primary DNS IPv4 Address	8.8.8.8
Secondary DNS IPv4 Address	4.2.2.1
Primary TFTP Server IPv4 Address (Option 150)	10.232.50.89
Secondary TFTP Server IPv4 Address (Option 150)	10.232.50.89
Bootstrap Server IPv4 Address	
Domain Name	home.local
TFTP Server Name (Option 66)	
ARP Cache Timeout(sec)*	0
IP Address Lease Time(sec)*	3600
Renewal(T1) Time(sec)*	0
Rebinding(T2) Time(sec)*	0

At the bottom of the form, there are buttons for Save, Delete, Copy, and Add New.

Add a DHCP subnet from the same menu path: System --- >DHCP --- > DHCP subnet

 **Cisco Unified CM Administration**
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

DHCP Subnet Configuration

 Save  Delete  Copy  Add New

Status

 Status: Ready

DHCP Subnet Information

DHCP Server*	<input type="text" value="CUCM-Cisco"/>
Subnet IPv4 Address*	<input type="text" value="10.232.50.0"/>
Primary Start IPv4 Address*	<input type="text" value="10.232.50.70"/>
Primary End IPv4 Address*	<input type="text" value="10.232.50.79"/>
Secondary Start IPv4 Address	<input type="text"/>
Secondary End IPv4 Address	<input type="text"/>
Primary Router IPv4 Address	<input type="text" value="10.232.50.86"/>
Secondary Router IPv4 Address	<input type="text"/>
IPv4 Subnet Mask*	<input type="text" value="255.255.255.0"/>
Domain Name	<input type="text"/>
Primary DNS IPv4 Address	<input type="text"/>
Secondary DNS IPv4 Address	<input type="text"/>

Adding Devices/Phones and configuring Directory numbers

Cisco phones need to be added in CUCM by way of their MAC address and assigned to a specific user and then when powered on, they obtain an IP address in the CUCM topology with the subnet defined in CUCM administration console. Use the Device --- > Phone menu path to add new devices. One will need to define the template based on the device being configured, for example Cisco 9971 phone template as in the screenshots below. Also, some highlights of the configuration to add a user and configure a directory number (DN) to it in CUCM are shown below:

The screenshot displays the Cisco Unified CM Administration console. The main navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Phone Configuration' for a 'Cisco 7821' device. The interface is divided into two main sections: a left-hand configuration list and a right-hand details panel.

Left-hand Configuration List:

- 1. Line [1] - 7814437248 (no partition) (highlighted with a red circle)
- 2. Line [2] - Add a new DN
- 3-9. Add a new SD (multiple instances)
- 10. Add a new BLF Directed Call Park
- 11-16. Call Park, Call Pickup, CallBack, Conference List, Do Not Disturb, End Call

Right-hand Details Panel:

- Product Type:** Cisco 7821
- Device Protocol:** SIP
- Real-time Device Status:**
 - Registration: Registered with Cisco Unified Communications Manager CUCM-Cisco
 - IPv4 Address: 10.232.50.11
 - Active Load ID: sip78xx.10-2-1-12
 - Inactive Load ID: None
 - Download Status: None
- Device Information:**
 - Device is Active (checked)
 - Device is trusted (checked)
 - MAC Address*: 00E16DDB6C2E (highlighted with a red circle)
 - Description: SEP00E16DDB6C2E for Andy
 - Device Pool*: Default
 - Common Device Configuration: <None>
 - Phone Button Template*: Standard 7821 SIP (highlighted with a red circle)
 - Softkey Template: <None >
 - Common Phone Profile*: Standard Common Phone Profile

The CUCM is now ready to send/receive calls and establish SIP connectivity with the Oracle E-SBC.

Test Plan

Caveats and out of scope items: Fax was not tested because the Lab CM did not have an analog card to test these capability there for Fax is considered out of scope for this testing.

Following is the test plan executed against this setup and results have been documented below.

ID	Test Case Title	Status
1000	Section 1	-
1100	SIP Connectivity	
1101	Validate syntax of OPTIONS messages sent to service provider	p
1102	Validate syntax of OPTIONS messages sent from service provider	p
1103	Validate in service reponse codes to OPTIONS messages from provider	p
1104	Validate in service reponse codes to OPTIONS messages to provider	p
1105	Validate OPTIONS messages are not sent more than once every 10 seconds to provider	p
2000	Section 2	-
2100	Initial Calls To/From External Phones	
2101	Inbound call from an external phone to an enterprise extension. Hang-up at called party (enterprise extension). Wait for calling party to disconnect. Validate proper SIP header syntax, ringback tone, two-way audio and proper call clearance	p
2102	Inbound call from an external phone to an enterprise extension. Hang-up at calling party (PSTN phone). Wait for called party to disconnect. Validate proper SIP header syntax, ringback tone, two-way audio and proper call clearance	p
2103	Outbound call from an enterprise extension to an external phone. Hang-up at called party (PSTN phone). Wait for calling party to disconnect. Make sure originating party is properly identified (Diversion/History-Info, PAI or From- in that order), using exactly 10 digits for the user part and the domain matching this TN's "PBX" (to which its TG is assigned). Also validate "tgrp/trunk-context" in Contact, if doing explicit TG selection (usually for Toll-bypass). Validate ringback tone, two-way audio and proper call clearance	p
2104	Outbound call from an enterprise extension to an external phone. Hang-up at calling party (enterprise extension). Wait for called party to disconnect. Make sure originating party is properly identified (Diversion/History-Info, PAI or From- in that order), using exactly 10 digits for the user part and the domain matching this TN's "PBX" (to which its TG is assigned). Also validate "tgrp/trunk-context" in Contact, if doing explicit TG selection (usually for Toll-bypass).	p

	Validate ringback tone, two-way audio and proper call clearance	
2105	Trunk Group Selection: test absense of explicit trunk group selection	p
2106	Trunk Group Selection: testtrunk group selection with tgrp tag	p
2107	Trunk Group Selection: testtrunk group selection with otg tag	p
3000	Section 3	-
3100	Incomplete Call Attempts	
3101	Inbound call from an external phone to an enterprise extension. Hang-up before far-end answers.	p
3102	Outbound call from an enterprise extension to an external phone. Hang-up before far-end answers.	p
3103	No Answer of inbound call from an external phone to an enterprise extension. (No explicit rules on CPE. Let extension ring.)	p
3104	No Answer of outbound call from an enterprise extension to an external phone.	p
3105	Inbound call from an external phone to an enterprise extension that is "Busy".	p
3107	Inbound call from an external phone to an unassigned enterprise extension.	p
3108	Outbound call from an enterprise extension to an invalid external number (Note that this also happens to test CPE support for early media)	p
3109	Validation of explicit treatments/terminating responses to basic conditions (busy, no circuit avail, bldn etc)	p
4000	Section 4	-
4100	Codec Support and Negotiation with Hard Phones	
4101	Whenever the CPE sends out SDP, the Content-Type must be "application/sdp"	p
4102	Validate inbound G.729 calls	p
4103	Validate outbound G.729 calls (annexb=no is required)	p
5000	Section 5	-
5100	Voicemail and DTMF Tone Support	
5101	Inbound call from an external phone to an enterprise extension, transfer to voicemail. Leave a message.	p
5102	Inbound call from an external phone to an enterprise extension, let ring for close to 2 minutes, then transfer to voicemail. Leave a message.	p
5103	Login to enterprise voicemail and retrieve message from 5102.	p
5104	Outbound call to an external number, transfer to voicemail. (Ex. Call office or cell phone with voicemail). Leave a message.	p
5105	Login to external voicemail and retrieve message from 5104.	p
5108	RFC2833 DTMF sent from the CPE outbound to an external device are recognised by the recieving equipment	p

5109	RFC2833 DTMF sent from an external device inbound to the CPE are recognised by the receiving equipment	p
5111	Inband (Q.24) DTMF sent from an external device inbound to the CPE are recognised by the receiving equipment	p
6000	Section 6	-
6100	PSTN Numbering Plans	
6101	Inbound Call	p
6102	Outbound Toll-Free Call	p
6103	Outbound Local Call	p
6104	Outbound International Calls (011)961-865-0650	p
6105	Operator call (0)	p
6106	Operator Assisted Calls (e.g. 0+10 digits in US)	
6107	Validation of e.164 handling on DID	p
6108	Validation number plan format is correct across all headers according to interop spec	p
6109	Operator Assisted International Call (e.g. 0+1 8 to 35 digits)	p
6110	Casual Dial: 101+xxxx+NDC call (from 13 to 40 digits)	p
6111	n11 call (e.g. 211)	p
6112	911 call	p
6113	1-xxx-555-1212 call	p
6114	310-xxxx call	p
6115	1-700-xxx-xxxx call	p
6118	Operator-assisted long-distance call (00)	p
7000	Section 7 - Calling Name and Number Presentation	-
7100	Static ONND	
7101	Outbound call with Static ONND - using only the From header and a pre-provisioned number (with user=phone)	p
7102	Outbound call with Static ONND - using the P-Asserted-Identify header and a pre-provisioned number (with user=phone)	p
7103	Outbound call with Static ONND - using explicit trunk group selection (with user=phone)	p
7104	Outbound call with Static ONND - using the Diversion header without PAI (with user=phone)	p
7105	Outbound call with Static ONND - using the Diversion header (valid Bell number) with PAI (with user=phone)	p
7106	Outbound call with Static ONND - using the Diversion header (external number) with PAI (with user=phone and implicit trunk group selection)	p
7107	Outbound call with Static ONND - using the Diversion header (external number) with PAI (with user=phone and explicit trunk group selection)	p
7108	Validate proper syntax used in PAI, PPI, From and Diversion for CNAM/CLID display on outbound calls	p
7200	Dynamic ONND	

7201	Outbound call with Dynamic ONND - using the From header (without user=phone)	p
7202	Outbound call with Dynamic ONND - using the P-Asserted-Identify header (without user=phone)	p
7203	Outbound call with Dynamic ONND - using the Diversion header (with user=phone) without PAI and using a valid Bell SIP Trunking number in both the Diversion and From	p
7204	Outbound call with Dynamic ONND - using the Diversion header (with user=phone) without PAI and using an external number in either the Diversion or From	p
7205	Outbound call with Dynamic ONND - using the Diversion header (with user=phone) with PAI and using a valid Bell SIP Trunking number in both the Diversion and PAI	p
7206	Outbound call with Dynamic ONND - using the Diversion header (with user=phone) with PAI and using an external number in the Diversion	p
7207	Outbound call with Dynamic ONND to party A, transfer via tromboning to party B	p
7209	Validate proper syntax used in PAI, PPI, From and Diversion for CNAM/CLID display on outbound calls	p
7300	Private and Unknown Calls	
7301	Place an outbound private call. Validate privacy header syntax and interworking on outbound private call against Bell spec and document differences.	p
7302	Place an inbound private call. Validate privacy header syntax and interworking on inbound private call against Bell spec and document differences. CPE must respect the privacy header.	p
7303	Validate handling of incoming unknown calls	p
7304	Validate handling of incoming calls when not subscribed to Calling Line ID Delivery	p
8000	Section 8	-
8100	Supplementary Features – Call Hold	
8101	Inbound Call – PBX Hold and Resume (No music) – Short Hold Duration	p
8102	Inbound Call – PBX Hold and Resume (With music) – Short Hold Duration	p
8103	Outbound Call – PBX Hold and Resume (No music) – Short Hold Duration	p
8104	Outbound Call – PBX Hold and Resume (With music) – Short Hold Duration	p
8105	Inbound Call – PSTN Hold and Resume (No music) – Short Hold Duration	p
8106	Inbound Call – PSTN Hold and Resume (With music) – Short Hold Duration	p
8107	Outbound Call – PSTN Hold and Resume (No music) – Short Hold Duration	p
8108	Outbound Call – PSTN Hold and Resume (With music) – Short Hold Duration	p

8109	Inbound Call - PBX Hold and Resume (No music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8110	Inbound Call - PBX Hold and Resume (With music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8111	Outbound Call - PBX Hold and Resume (No music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8112	Outbound Call - PBX Hold and Resume (With music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8113	Inbound Call - PSTN Hold and Resume (No music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8114	Inbound Call - PSTN Hold and Resume (With music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8115	Outbound Call - PSTN Hold and Resume (No music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8116	Outbound Call - PSTN Hold and Resume (With music) – Long Hold Duration that exceeds the SIP session timers (~10 min)	p
8200	Supplementary Features – Call Forward	
8203	Call Forwarding (All) to External Number (Off-net) - Tromboning	p
8206	Call Forwarding (No Answer) to External Number (Off-net) – Tromboning	p
8209	Call Forwarding (Busy) to External Number (Off-net) – Tromboning	p
8300	Supplementary Features – Call Transfer, Conference	
8302	Blind Call Transfer of inbound call: Transfer to External Number (Tromboning)	p
8304	Blind Call Transfer of inbound call: Transfer to Internal Number (Tromboning)	p
8306	Blind Call Transfer of outbound call: Transfer to External Number (Tromboning)	p
8308	Blind Call Transfer of outbound call: Transfer to Internal Number (Tromboning)	p
8309	Attended Transfer of inbound call: Transfer to External Number (Tromboning)	p
8310	Attended Transfer of inbound call: Transfer to Internal Number (Tromboning)	p
8311	Attended Transfer of outbound call: Transfer to External Number (Tromboning)	p
8312	Attended Transfer of outbound call: Transfer to Internal Number (Tromboning)	p
8313	Validate call park and unpark	p
9000	Section 9	
9100	Failover	
9101	Validate handling of ICMP unreachable messages on a new call, by pointing CPE primary IP to unreachable IP	p
9102	Validate handling of bell SBC silently discarding packets on a new call, by pointing to 207.236.202.114:50505	p

9103	Validate handling of SIP 503 responses on a new call, by pointing to 207.236.202.114:50503	p
9104	Validate Handling of out service response codes to OPTIONS pings, out of service codes are anything other than 200 and 483 by pointing to 207.236.202.114:50504	p
9105	Validate traffic to CPE from multiple Bell IPs in order to simulate SBC failover. Requires Bell participation.	p
11000	Section 11	
11100	Miscellaneous	
11101	Validate handling of multiple concurrent calls for the same number	p
11102	Long Duration Calls - Inbound	p
11103	Long Duration Calls - Outbound	p
11104	Outgoing call with wrong DID number or wrong PBX domain.	p
11105	(Optional) Validate handling of outbound call to full TG (403 Forbidden)	p
11106	Validate handling of session audits every 5 or 10 min (UPDATE or re-INVITE)	p
11107	Validate handling of CPE-initiated session audits	p

Caveats: For call transfers, the INVITE sent from the CUCM to the transferred phone does not have the transferee phone number in the SIP messaging, hence the CLID on the call transfer target shows as the transferor and not the transferee.

Troubleshooting Tools

Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from www.wireshark.org.

On the Oracle E-SBC

The Oracle E-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 E-SBC Console:

```
SBC1# reset sipd
SBC1# notify sipd debug
SBC1#
enabled SIP Debugging
SBC1# notify all rotate-logs
```

Examining the log files

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

```
C:\Documents and Settings\user>ftp 192.168.1.22
Connected to 192.168.85.55.
220 SBC1 server (VxWorks 6.4) ready. User (192.168.1.22:(none)): user
331 Password required for user. Password: acme
230 User user logged in.
ftp> cd /opt/logs
250 CWD command successful. ftp> get sipmsg.log
200 PORT command successful.
150 Opening ASCII mode data connection for '/opt/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 '/opt/logs/log.sipd' (204681 bytes).
226 Transfer complete.
ftp: 206823 bytes received in 0.11Seconds 1897.46Kbytes/sec
```

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

Full E-SBC Configuration

```
codec-policy
  name          prefer_729
  allow-codecs  telephone-event G729 PCMU
  order-codecs  G729 PCMU telephone-event
local-policy
  from-address  *
  to-address    *
  source-realm  cisco-inside
  policy-attribute
    next-hop    SAG:BelISIPTrunkGRP1
    realm       outside
    action      replace-uri
local-policy
  from-address  *
  to-address    *
  source-realm  outside
  policy-attribute
    next-hop    10.232.50.89
    action      replace-uri
media-manager
  initial-guard-timer  86400
  subsq-guard-timer   86400
media-profile
  name          G729
  payload-type  18
  parameters    annexb=no
media-profile
  name          PCMA
  payload-type  8
media-profile
  name          PCMU
  payload-type  0
network-interface
  name          s0p0
  description   outside
  ip-address    150.200.200.160
  netmask       255.255.255.0
  gateway       150.200.200.1
  hip-ip-list   150.200.200.160
  icmp-address  150.200.200.160
  ssh-address   150.200.200.160
network-interface
  name          s1p1
  ip-address    10.232.50.20
  netmask       255.255.255.0
  hip-ip-list   10.232.50.20
```

```

ftp-address          10.232.50.20
icmp-address         10.232.50.20
ssh-address          10.232.50.20
network-interface
  name               wancom1
  description        HA_HEARTBEAT1
  pri-utility-addr   169.254.1.1
  sec-utility-addr   169.254.1.2
  netmask             255.255.255.252
network-interface
  name               wancom2
  description        HA_HEARTBEAT2
  pri-utility-addr   169.254.2.1
  sec-utility-addr   169.254.2.2
  netmask             255.255.255.252
phy-interface
  name               s0p0
  operation-type     Media
phy-interface
  name               s1p1
  operation-type     Media
  port               1
  slot               1
phy-interface
  name               wancom1
  port               1
  wancom-health-score 8
phy-interface
  name               wancom2
  port               2
  wancom-health-score 9
realm-config
  identifier          cisco-inside
  network-interfaces s1p1:0
realm-config
  identifier          outside
  network-interfaces s0p0:0
  codec-policy        prefer_729
redundancy-config
  becoming-standby-time 360000
  peer
    name              SBC1
    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          SBC2
  type          Secondary
  destination
    address     169.254.1.2:9090
    network-interface wancom1:0
  destination
    address     169.254.2.2:9090
    network-interface wancom2:0
session-agent
  hostname      10.232.50.89
  ip-address    10.232.50.89
  transport-method StaticTCP
  realm-id      cisco-inside
  ping-method   OPTIONS
  ping-interval 90
  auth-attributes
    auth-realm   lab.ca
    username     abc_123456_ca
    password     *****
    in-dialog-methods INVITE
  auth-attributes
    auth-realm   test
session-agent
  hostname      207.236.202.114
  ip-address    207.236.202.114
  port          50504
  realm-id      outside
  ping-interval 30
  out-service-response-codes 503
session-agent
  hostname      200.236.200.170
  ip-address    200.236.200.170
  ping-method   OPTIONS
  ping-interval 90
  out-manipulationid To_Bell
session-agent
  hostname      60.150.190.70
  ip-address    60.150.190.70
  realm-id      outside
  ping-interval 90
  ping-in-service-response-codes 200,483
  out-manipulationid To_Bell
session-group
  group-name    BellsIPTrunkGRP1
  dest          60.150.190.70
               200.236.200.170
  sag-recursion enabled
session-timer-profile

```

```

name test
session-expires 400
force-reinvite enabled
session-translation
  id addplus1
  rules-called addplus1
sip-config
  options max-udp-length=0
            session-timer-support
sip-interface
  realm-id cisco-inside
  sip-port
    address 10.232.50.20
    transport-protocol TCP
    allow-anonymous agents-only
  session-timer-profile test
sip-interface
  realm-id outside
  sip-port
    address 150.200.200.160
    allow-anonymous agents-only
  initial-inv-trans-expire 6
  add-sdp-invite invite
  add-sdp-profiles PCMU
                    PCMA
                    G729
  session-timer-profile test
sip-manipulation
  name To_Bell
  header-rule
    name Update_Request
    header-name request-uri
    action manipulate
  element-rule
    name Update_URI_Host
    type uri-host
    action replace
    new-value lab.ca
  element-rule
    name Rmv_User
    parameter-name user
    type uri-param
    action delete-element
  element-rule
    name Rmv_Port
    type uri-port
    action delete-element
  element-rule
    name Update_URI_User

```

```

type uri-user
comparison-type pattern-rule
match-value \+?(\d+)
new-value \++$1
header-rule
name Update_To
header-name To
action manipulate
element-rule
name Update_URI_Host
type uri-host
action replace
new-value lab.ca
element-rule
name Rmv_User
parameter-name user
type uri-param
action delete-element
element-rule
name Update_URI_User
type uri-user
comparison-type pattern-rule
match-value \+?(\d+)
new-value \++$1
header-rule
name Update_From
header-name from
action manipulate
element-rule
name Update_URI_Host
type uri-host
action replace
new-value lab.voice.ca
element-rule
name Add_OTG_URI_Param
parameter-name otg
type uri-param
new-value abc_123456_ca
element-rule
name Rmv_UriParam_User
parameter-name user
type uri-param
action add
new-value phone
header-rule
name Update_Contact
header-name Contact
action manipulate
methods INVITE

```

```

element-rule
  name          Add_User
  type          uri-user
  new-value     613xxxxxxx
element-rule
  name          Add_tgrp
  parameter-name tgrp
  type          uri-user-param
  new-value     abc_123456_ca
element-rule
  name          Add_trunk_context
  parameter-name trunk-context
  type          uri-user-param
  new-value     lab.ca
header-rule
  name          Max_Forward_0
  header-name   Max-Forwards
  action        manipulate
  comparison-type pattern-rule
  msg-type      request
  methods       OPTIONS
  new-value     0
header-rule
  name          Rmv_UserAgent_Hdr
  header-name   user-agent
  action        delete
header-rule
  name          save_Diversion
  header-name   Diversion
  action        store
header-rule
  name          Chk_Add_Diversion
  header-name   Diversion
  action        manipulate
  comparison-type boolean
  methods       INVITE
  match-value   $save_Diversion
  new-value     <sip:613xxxxxxx@lab.voice.ca;user=phone>
header-rule
  name          Update_Diversion
  header-name   Diversion
  match-value   !$save_Diversion
  element-rule
    name          Update_URI_Host
    type          uri-host
    action        replace
    new-value     lab.voice.ca
  element-rule
    name          Add_OTG_URI_Param

```

parameter-name	otg
type	uri-param
new-value	abc_123456_ca
element-rule	
name	Chg_Tmp_UriUser
type	uri-user
action	replace
new-value	7207759641
element-rule	
name	Del_User_Param
parameter-name	user
type	uri-param
header-rule	
name	Rmv_Priv_Hdr
header-name	Privacy
header-rule	
name	Update_PAi
header-name	P-Asserted-Identity
action	manipulate
element-rule	
name	Update_URI_Host
type	uri-host
action	replace
new-value	lab.voice.ca
element-rule	
name	Add_User_UriParam
parameter-name	user
type	uri-param
action	add
new-value	phone
element-rule	
name	Add_OTG_URI_Param
parameter-name	otg
type	uri-param
new-value	abc_123456_ca
element-rule	
name	Add_User
type	uri-user
new-value	7207759641
header-rule	
name	save_Referred_By
header-name	Referred-by
action	store
msg-type	request
methods	INVITE
element-rule	
name	Fix_URI_Host
type	uri-host
action	replace

```

new-value lab.voice.ca
header-rule
  name Referred_By_2_Div
  header-name Diversion
  action add
  comparison-type boolean
  methods INVITE
  match-value $save_Referred_By
  new-value $save_Referred_By.$0
  element-rule
    name Update_URI_Host
    type uri-host
    action replace
    new-value lab.voice.ca
header-rule
  name RmvReferredBy
  header-name Referred-by
  action delete
header-rule
  name Rmv_CallInfo
  header-name Call-Info
  action delete
header-rule2
  name Rmv_CiscoGUID
  header-name Cisco-Guid
  action delete
sip-monitoring
  monitoring-filters *
steering-pool
  ip-address 10.232.50.20
  start-port 40000
  end-port 60000
  realm-id cisco-inside
steering-pool
  ip-address 150.200.200.160
  start-port 49152
  end-port 49200
  realm-id outside
system-config
  process-log-level DEBUG
  default-gateway 172.18.0.1
translation-rules
  id addplus1
  type add
  add-string 01
  delete-string +1
web-server-config

```

Appendix B

Accessing the ACLI

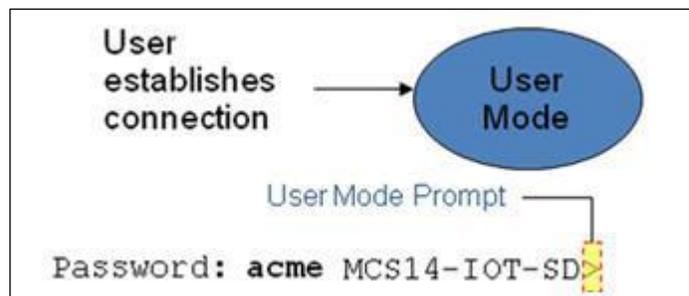
Access to the ACLI is provided by:

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

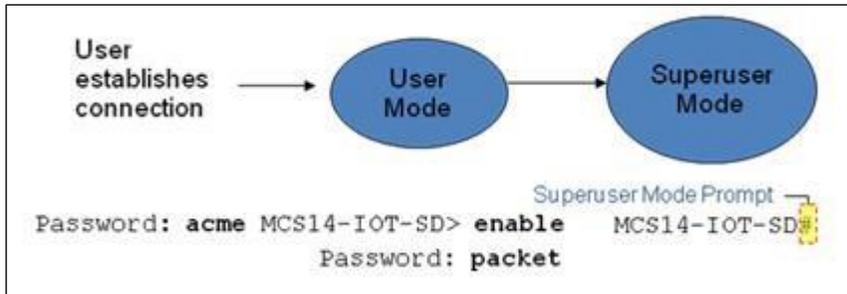
Initial connectivity will be through the serial console port. At a minimum, this is how to configure the management (eth0) interface on the E-SBC.

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 E-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.



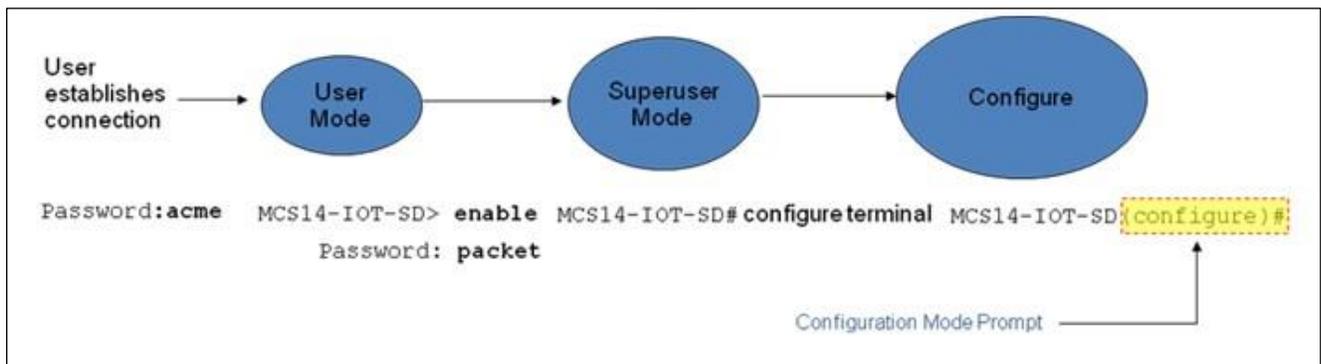
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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.

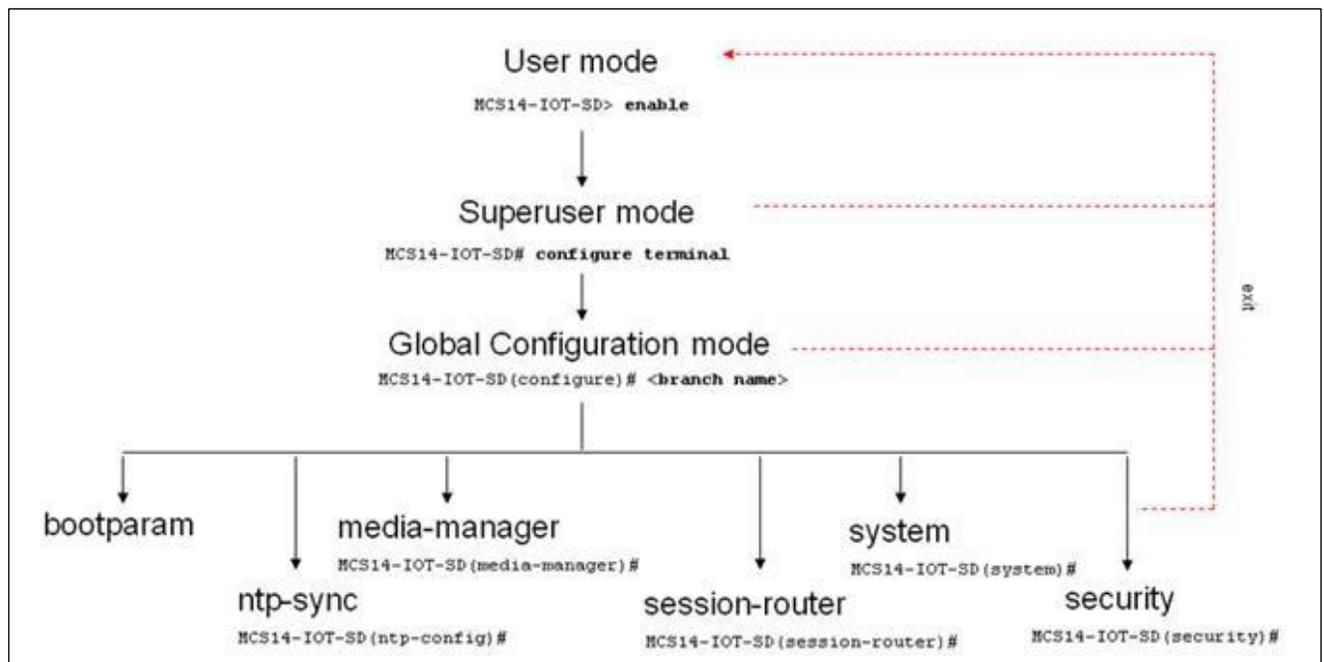
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the **configure terminal** command in the Superuser mode.

Configuration mode is identified by the word configure in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, **SBC1 (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 E-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 E-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

- 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.
- When creating a multiple-instance element, you must specify a unique identifier for each instance of the element.
- 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.
- 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.
- Issue the **exit** command to exit the selected element.
- Note that the configurations at this point are not permanently saved yet. If the E-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.

- Enter the element that you will edit at the correct level of the ACLI path.

- 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.
- 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.
- 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.
- On completion, you must issue the **done** command.
- Issue the **exit** command to exit the selected element.

Note that the configurations at this point are not permanently saved yet. If the E-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,

- Enter the **no** command from within the path for that specific element
- Issue the **exit** command. To delete a multiple instance element,

Enter the **no** command from within the path for that particular element.

The key field prompt, such as <name>:<sub-port-id>, appears.

Use the <Enter> key to display a list of the existing configured elements.

Enter the number corresponding to the element you wish to delete.

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 E-SBC reboots, your configurations will be lost.

Configuration Versions

At any time, three versions of the configuration can exist on the E-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 E-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 E-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 E-SBC displays a reminder on screen stating that you must use the **activate-config** command if you want the configurations to be updated.

```
SBC1 # 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'.
SBC1
```

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 E-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.

```
SBC1# activate-config Activate-Config received,  
processing. waiting 120000 for request to finish Request  
to 'ACTIVATE-CONFIG' has Finished, Activate Complete  
SBC1#
```



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