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Oracle Enterprise Session Border Controller and CUCM 10.5 with TELUS Enterprise IP Trunking R2 for Dedicated and Registration Connection

Technical Application Note

Disclaimer

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Intended Audience

This document is intended for use by Oracle personnel, 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 – Acme Packet 4600 / Acme Packet 3820.

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 TELUS IP Trunking services are configured in the optimal manner.

Introduction

Audience

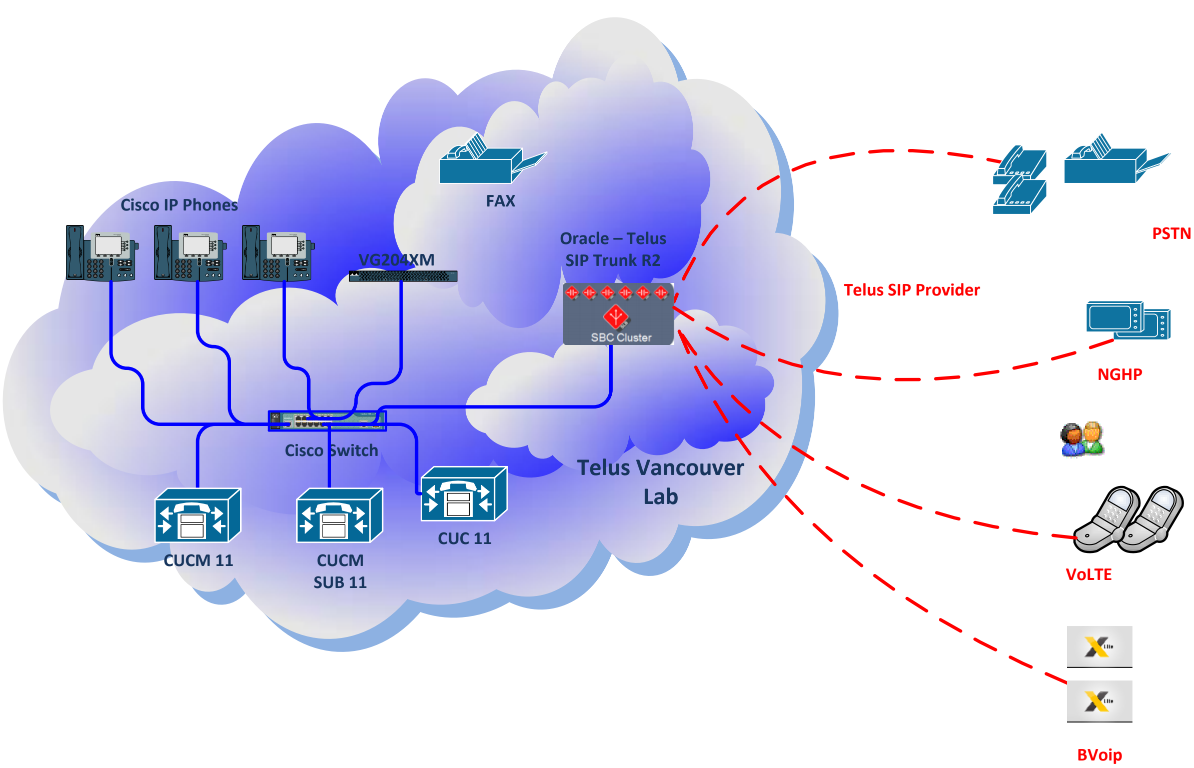
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise Session Border Controller and CUCM. There will be steps that require navigating the Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, SIP/RTP, 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 running ECZ7.3.0 m1. Note: the configuration running on the E-SBC is backward/forward compatible with any release in the 7.3.0 stream.
- TELUS IP trunk based customers with dedicated data connectivity to TELUS.

Architecture

- The following reference architecture shows a logical view of the connectivity between CUCM, E-SBC and the TELUS trunk.



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 refer to 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

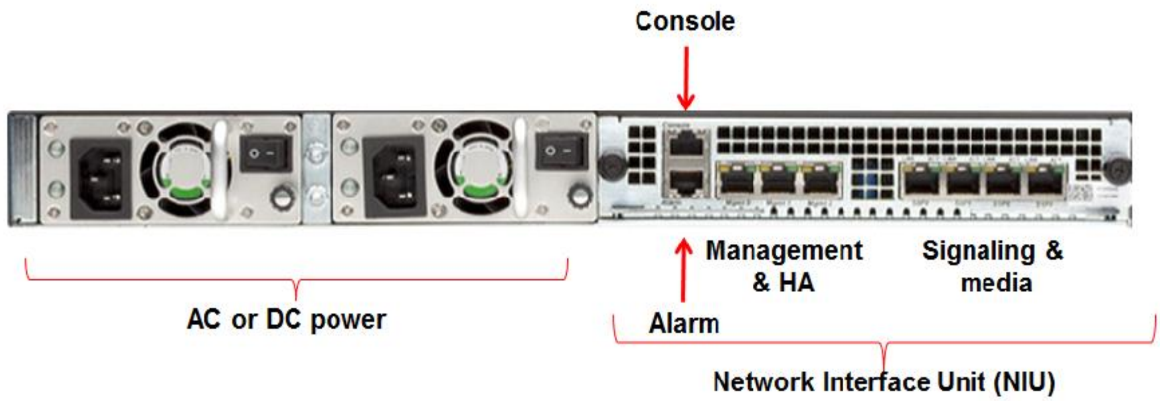
- Configuration of Network management including SNMP and RADIUS

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 (TELUS next-hop facing) network and the slot 1 port 0 (s1p0) interface into your inside (CUCM server-facing) network. Once connected, you are ready to power on and perform the following steps.

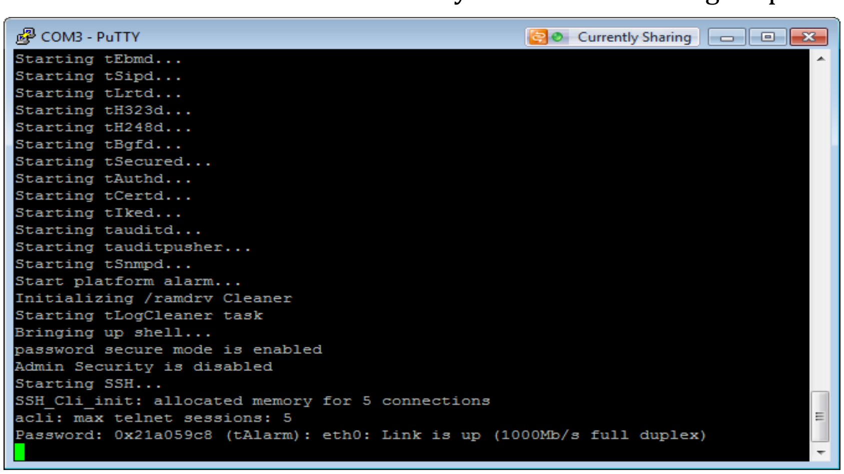
All commands are in bold, such as configure terminal; parameters in bold red such as SBC1 are parameters which are specific to an individual deployment. Note: The ACLI is case sensitive.

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.



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: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)         :  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.

Header manipulation rule required for TELUS

The header manipulation rule towardstrunk deletes the Require:100rel header from all SIP messaging which is going towards the TELUS trunk. The sip-feature 100rel is added to the config for SIP PRACK interworking, and 100rel-interworking is added as OPTIONS on the trunk and core side sip-interface.

```
sip-manipulation
        name                                    towardstrunk
        header-rule
                name                                    DelReq
                header-name                             Require
                action                                  delete
                match-value                             100rel
sip-feature
        name                                    100rel
        realm                                   Peer
        support-mode-inbound                    Pass
        require-mode-inbound                    Pass
        proxy-require-mode-inbound              Pass
        support-mode-outbound                   Pass
        require-mode-outbound                   Pass
        proxy-require-mode-outbound             Pass
        last-modified-by                        admin@192.168.20.101
        last-modified-date                      2016-05-05 17:18:13
```

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
```

TELUS Trunk Authentication

TELUS offers two types of connections: direct connections over dedicated data circuits or MPLS and SIP trunks over public networks. Trunk authentication and surrogate registration are only required for publicly connected connections. If the connection is a private connection, the following section is NOT required. TELUS IP Trunking Release 2 requires both Registration of the trunk and Authentication challenges on SIP INVITE Methods. TELUS will provide the information similar to the following:

- SIP User Name: user123456
- SIP Domain: ipnet4.com
- SIP Password: pass123456
- DID: 2223334444

There are 3 parts to the configuration. A surrogate agent is needed to register the trunk on behalf of the IPPBX. Surrogate registration requires registration-caching to be set to enabled on the sip-interface of PBX realm. Auth challenges to INVITEs are handled on the session-agent to the IP-PBX via auth-attributes.

```
surrogate-agent
        register-host                           ipinet4.com
        register-user                           user123456
        description
        realm-id                                core
        state                                   enabled
        customer-host                           172.16.154.35
        customer-next-hop                       10.27.56.7
        register-contact-host                   ipinet4.com
        register-contact-user                   user123456
        password                                pass123456
        register-expires                        3600
        replace-contact                         disabled
        options                                 auth-info=refresh
                                                auth-
method="INVITE,CANCEL,ACK,BYE"
        route-to-registrar                      enabled
        aor-count                               1
        auth-user                               user123456
        max-register-attempts                   10
        register-retry-time                     300
        count-start                             1
        register-mode                           automatic
        triggered-inactivity-interval           30
        triggered-oos-response                  503
```

Reg-cache on the IPPBX sip-interface

```
sip-interface
        state                                   enabled
        realm-id                                core
        description
        sip-port
                address                                 172.16.153.34
                port                                    5060
                transport-protocol                      UDP
                tls-profile
                allow-anonymous                         all
```

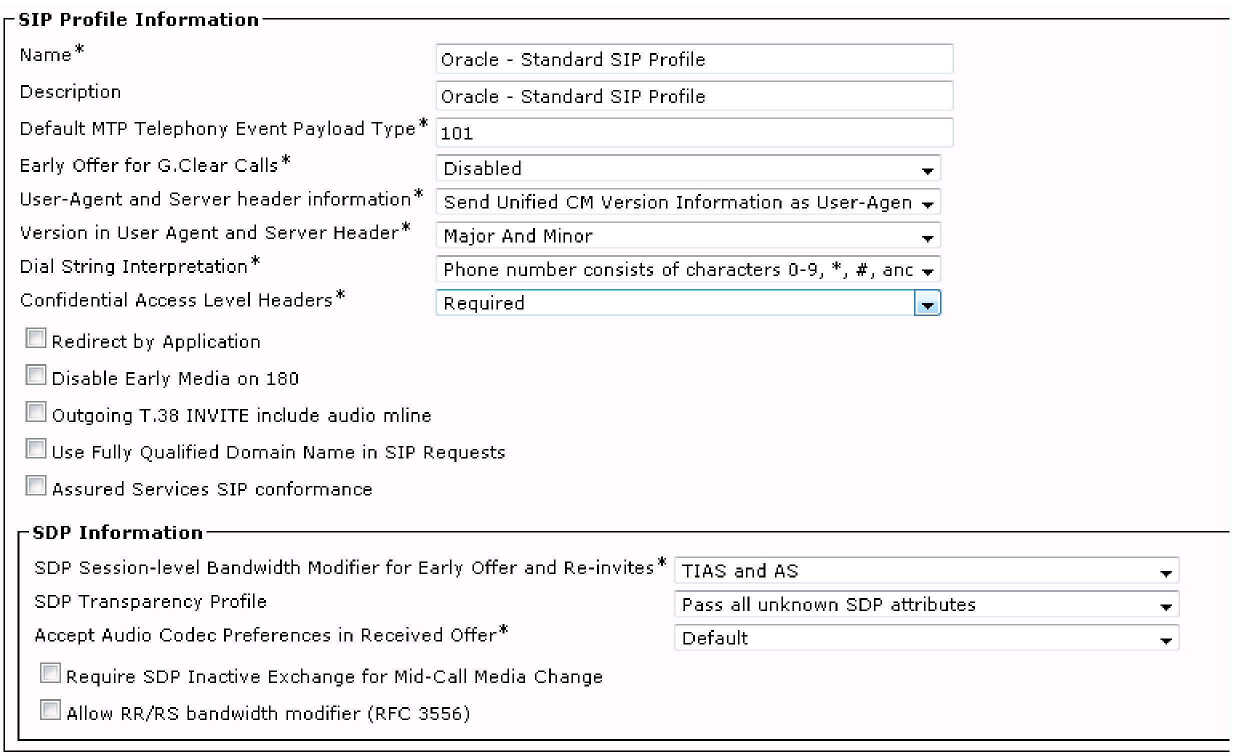
```
                multi-home-addrs
                ims-aka-profile
        carriers
…
        tcp-nat-interval                        90
        registration-caching                    enabled
```

IP-PBX session-agent configuration

```
session-agent
        hostname                                172.16.149.38
        ip-address                              172.16.149.38
        port                                    5060
        state                                   enabled
        app-protocol                            SIP
…
        sip-isup-profile
        kpml-interworking                       inherit
        monitoring-filters
        auth-attributes
                auth-realm                              ipnet4.com
                username                                user123456
                password                                ********
                in-dialog-methods                          INVITE BYE ACK CANCEL
OPTIONS SUBSCRIBE PRACK NOTIFY UPDATE REFER
```

Cisco Unified Communication Manager configuration

CUCM SIP Profile Configuration:



Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	(•) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	On
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN:

Caller Name:

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK for all 1xx Messages
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Disabled (Default value)

- [ ] Enable ANAT
- [ ] Deliver Conference Bridge Identifier
- [ ] Allow Passthrough of Configured Line Device Caller Information
- [ ] Reject Anonymous Incoming Calls
- [ ] Reject Anonymous Outgoing Calls
- [ ] Send ILS Learned Destination Route String

SIP OPTIONS Ping

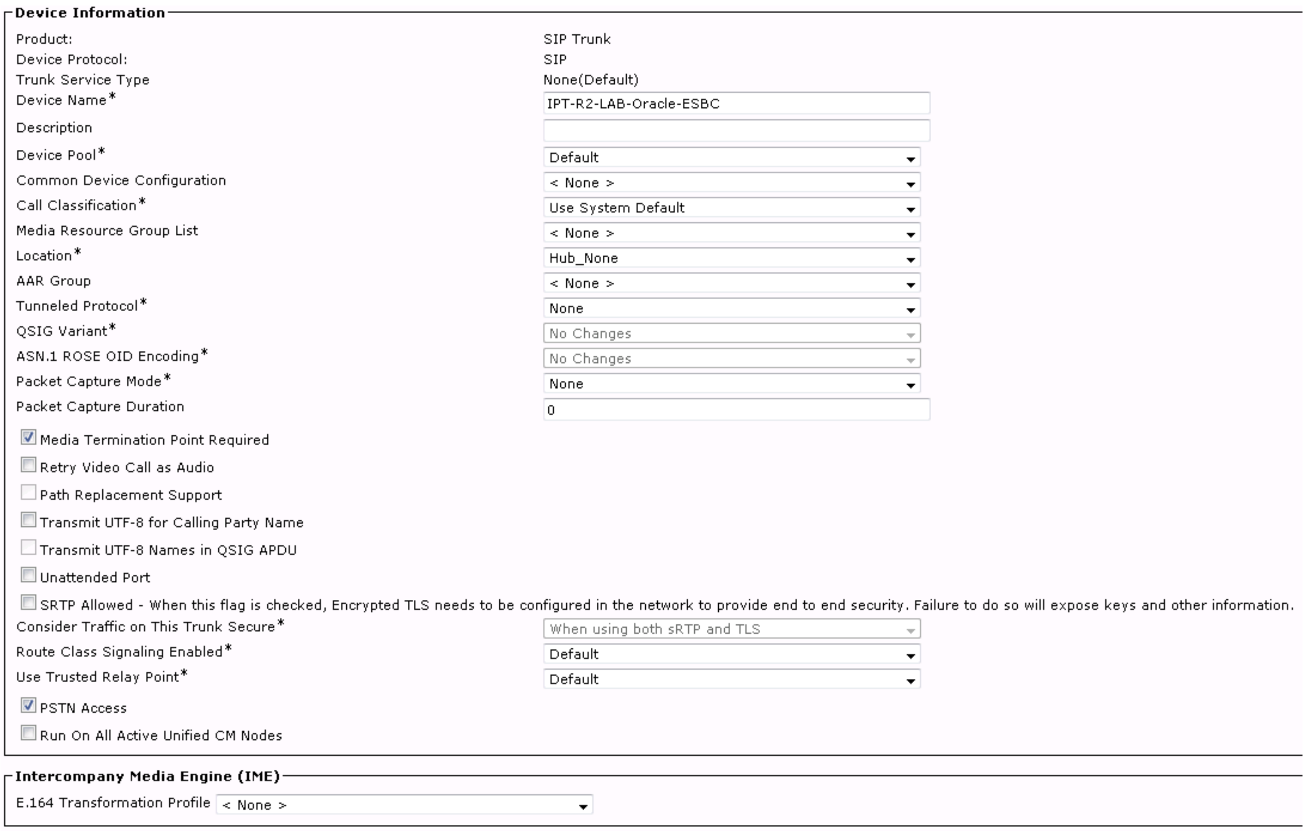
- [ ] Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

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

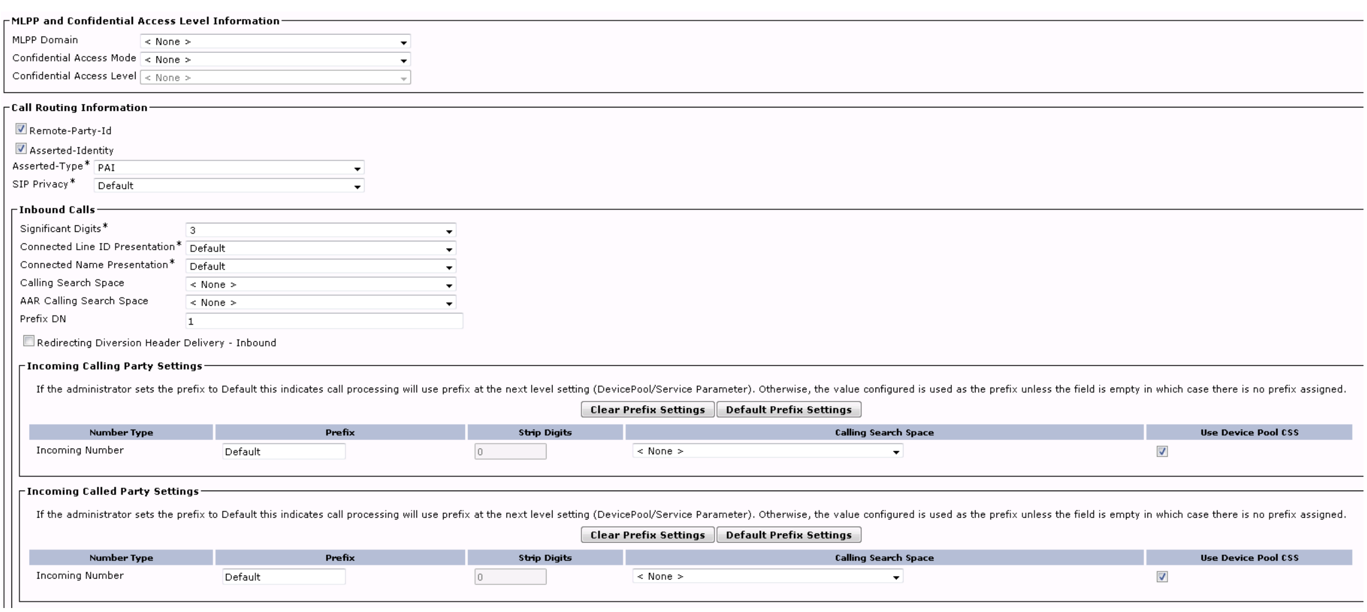
SDP Information

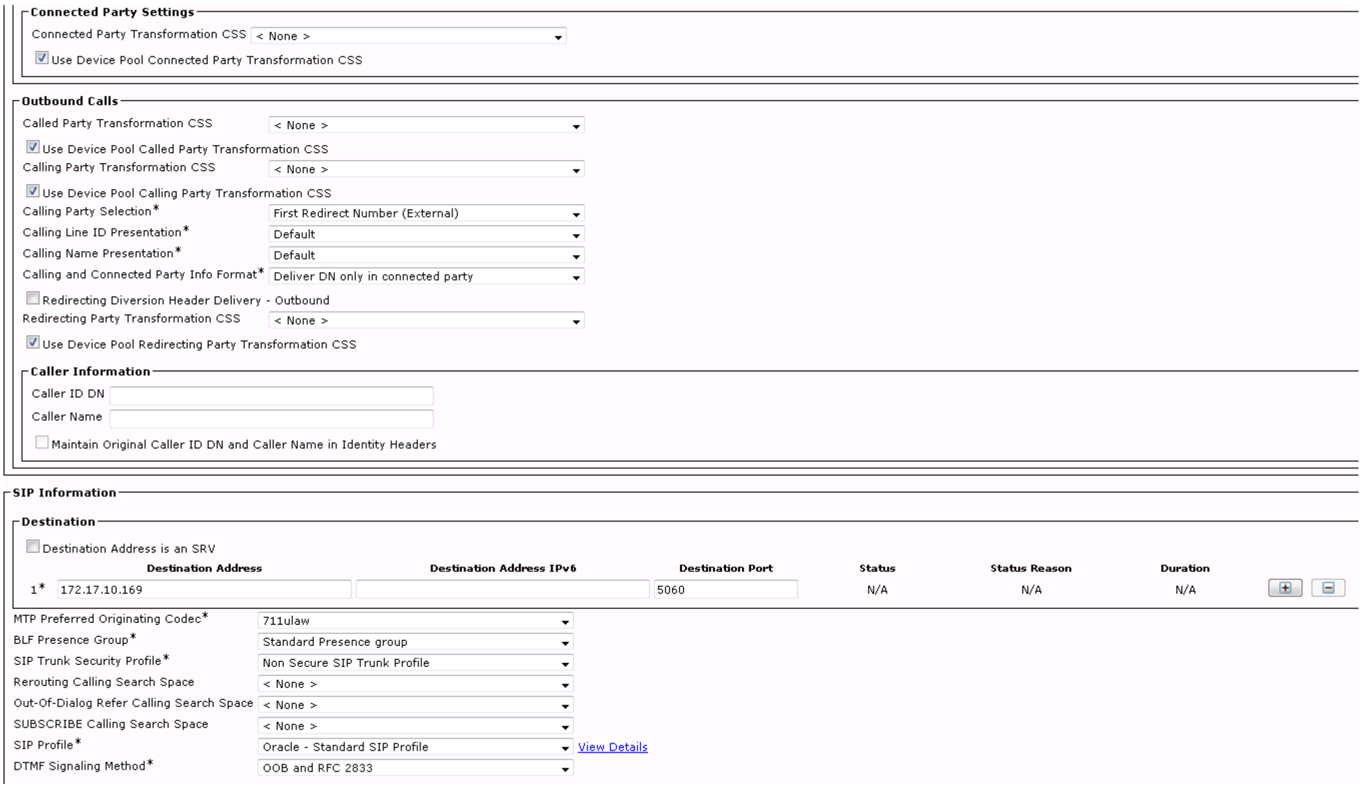
- [ ] Send send-receive SDP in mid-call INVITE
- [ ] Allow Presentation Sharing using BFCP
- [ ] Allow iX Application Media
- [ ] Allow multiple codecs in answer SDP

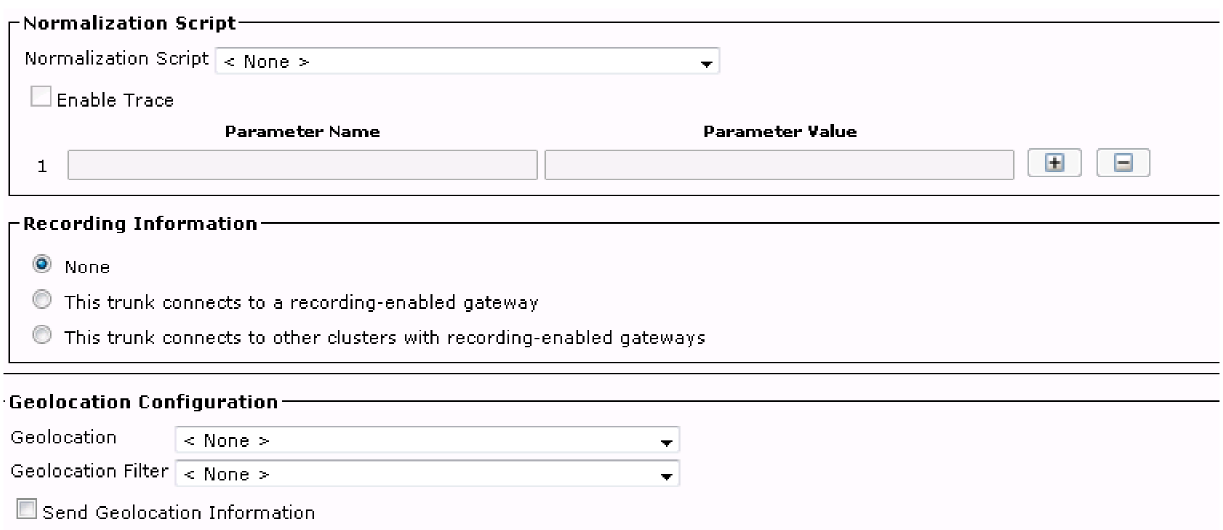
Configure SIP Trunk to Oracle ESBC:



Enable Trunk specific features:




Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS < None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection* First Redirect Number (External)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
Status
Status Reason
Duration
1* 172.17.10.169 5060 N/A N/A N/A
MTP Preferred Originating Codec* 711ulaw
BLF Presence Group* Standard Presence group
SIP Trunk Security Profile* Non Secure SIP Trunk Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Oracle - Standard SIP Profile View Details
DTMF Signaling Method* OOB and RFC 2833


Normalization Script
Normalization Script < None >
Enable Trace
Parameter Name
Parameter Value
1
Recording Information
None
This trunk connects to a recording-enabled gateway
This trunk connects to other clusters with recording-enabled gateways
Geolocation Configuration
Geolocation < None >
Geolocation Filter < None >
Send Geolocation Information

Test Plan

PSTN test cases

Test Number	Test Details	Pass/Fail/NA - Not Applicable
1. Test with PSTN line		
Basic inbound/outbound call		
TELUS_TC1.1	Call from PSTN phone to IP PBX phone 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
TELUS_TC1.2	Call from IP PBX phone to PSTN phone 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
Basic inbound/outbound call with privacy		
TELUS_TC1.3	Call from PSTN phone to IP PBX phone, prefix the IP PBX phone number with *63 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
TELUS_TC1.4	Call from IP PBX phone to PSTN phone, when dialling from the IP PBX phone, use the prefix if applicable to temporary suppress the call display 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
Hold and resume		
TELUS_TC1.5	Call from PSTN to IP PBX - after the call setup the PBX phone puts the call on-hold or (MOH), waits 30 seconds, resumes. Confirm audio both way after resume.	Pass
TELUS_TC1.6	Call from IP PBX to PSTN - after the call setup, use PSTN phone to put the call on-hold, wait 30 seconds, resume. Confirm audio both way after resume.	Pass
Call Transfer (Blind transfer)		
TELUS_TC1.7	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a blind transfer to PSTN phone Confirm audio both way after the transfer	Pass
TELUS_TC1.8	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to IP PBX phone 2 Confirm audio both way after the transfer	Pass

TELUS_TC1.9	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to another PSTN Confirm both way audio.	Pass
TELUS_TC1.10	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to another PSTN Confirm both way audio. Repeat the same test using SIP REFER	Pass
Call Transfer (Consult transfer)		
TELUS_TC1.11	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a consult transfer to PSTN phone Confirm audio both way after the transfer	Pass
TELUS_TC1.12	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to IP PBX phone 2 Confirm audio both way after the transfer	Pass
TELUS_TC1.13	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another PSTN Confirm both way audio.	Pass
TELUS_TC1.14	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another PSTN Confirm both way audio. Repeat the same test using SIP REFER	Pass
Call Forwarding Unconditional		
TELUS_TC1.15	Configure IP PBX phone 1 to CFU to PSTN phone IP PBX phone 2 calls phone 1 and should CFU to PSTN phone 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN phone	Pass
Call Forwarding Busy		
TELUS_TC1.17	Configure IP PBX phone 1 to CFB to PSTN phone IP PBX phone 2 calls phone 1 and should CFB toPSTN phone 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN phone	Pass
TELUS_TC1.19	Configure IP PBX phone 1 to CFDA to PSTN phone IP PBX phone 2 calls phone 1 and should CFDA toPSTN phone 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN phone	Pass
TELUS_TC1.20	Configure IP PBX phone 1 to CFDA to PSTN phone from PSTN calls phone 1 and should CFDA to PSTN phone 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN phone	Pass
Conference call		
TELUS_TC1.22	PSTN phone calls IP PBX phone 1 IP PBX phone 1 performs a conference call with IP PBX phone 2 Confirm audio among the parties	Pass

DTMF		
TELUS_TC1.24	From PBX dial 4036929600 (conference bridge) When hearing the prompt, enter valid Telus conference code 6913642. Follow prompts and verify connected to conference bridge. Verify that pressed keys are recognized and successfully accessed conference bridge. Verify by calling to conference bridge from PSTN. Test Inband DTMF by programming PBX end point	Pass
TELUS_TC1.25	From PBX dial 4036929600 (conference bridge) When hearing the prompt, enter valid Telus conference code 6913642. Verify that pressed keys are recognized and successfully accessed conference bridge. Verify by calling to conference bridge from PSTN. Test RFC2833 by programming PBX endpoint	Pass
Long calls - minimum recommendation		
TELUS_TC1.27	long duration call: 10 mins - to PSTN phone	Pass
TELUS_TC1.28	long duration call on hold: Call to PSTN, PBX places call on hold for 10 min, resume call, verify 2 way audio	Pass

VoIP Test cases

Test Number	Test Details	Pass/Fail/NA - Not Applicable
2. Test with TELUS VoIP Account		
Basic inbound/outbound call		
TELUS_TC2.1	Test by G.729. Call from TELUS VoIP client to IP PBX phone, 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
TELUS_TC2.2	Test by setup the call with G.729. Call from IP PBX phone to TELUS VoIP client, 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
Basic inbound/outbound call with privacy		
TELUS_TC2.3	Call from TELUS VoIP client with G.711 to IP PBX phone with privacy 1. CConfirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
TELUS_TC2.4	Call from IP PBX phone to TELUS VoIP client G.711, when dialling from the IP PBX phone, use the prefix if applicable to temporary suppress the call display 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
Hold and resume		
TELUS_TC2.5	Call from TELUS VoIP to IP PBX - after the call setup the PBX phone puts the call on-hold or (MOH), waits 30 seconds, resumes. Confirm audio both way after resume.	Pass
TELUS_TC2.6	Call from IP PBX to TELUS VoIP - after the call setup, use TELUS VoIP to put the call on-hold or (MOH), waits 30 seconds, resumes. Confirm 2-way voice after resume.	Pass
Call Transfer (Blind transfer)		
TELUS_TC2.7	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a blind transfer to TELUS VoIP client Confirm 2-way voice after the transfer	Pass
TELUS_TC2.8	TELUS VoIP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to PSTN Confirm 2-way voice after the transfer	Pass

Call Transfer (Consult transfer)		
TELUS_TC2.9	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a consult transfer to TELUS VoIP client Confirm 2-way voice after the transfer	Pass
TELUS_TC2.10	TELUS VoIP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to PSTN Confirm 2-way voice after the transfer	Pass
TELUS_TC2.11	TELUS VoIP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to PSTN Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Forwarding Unconditional		
TELUS_TC2.12	Configure IP PBX phone 1 to CFU to TELUS VoIP client IP PBX phone 2 calls phone 1 and should CFU toTELUS VoIP client 1. Confirm 2-way voice 2. Confirm phone 1 number and display at TELUS VoIP client	Pass
TELUS_TC2.13	Configure IP PBX phone 1 to CFU to 647-837-0597 TELUS VoIP client calls phone 1 to trigger the call forwarding 1. Confirm 2-way voice 2. Press 1234# to interrupt the prompt	Pass
Voicemail		
TELUS_TC2.14	Test with G.711. IP PBX phone 1 calls TELUS VoIP client, Don't answer the call in the TELUS VoIP client; after 4 ring, voicemail kick in Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass
Conference call		
TELUS_TC2.15	TELUS VoIP client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with IP PBX phone 2 Confirm audio among the parties	Pass

Mobile Test cases

Test Number	**Test Details**	**Pass/Fail/NA - Not Applicable**
3. Test with TELUS mobile		
Basic inbound/outbound call		
TELUS_TC3.1	Call from TELUS mobile client to IP PBX phone 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
TELUS_TC3.2	Repeat the test by setup the call with G.729. Call from IP PBX phone to TELUS mobile client 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
Basic inbound/outbound call with privacy		
TELUS_TC3.3	Call from TELUS mobile client to IP PBX phone G.711 with privacy enabled. 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
TELUS_TC3.4	Call from IP PBX phone G.711 to TELUS mobile client, when dialling from the IP PBX phone, use the prefix if applicable to temporary suppress the call display 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
Hold and resume		
TELUS_TC3.5	Call from TELUS mobile to IP PBX - after the call setup the PBX phone puts the call on-hold or (MOH), waits 30 seconds, resumes. Confirm audio both way after resume.	Pass
TELUS_TC3.6	Call from IP PBX to TELUS mobile - after the call setup, use TELUS mobile to put the call on-hold or (MOH), waits 30 seconds, resumes. Confirm 2-way voice after resume.	Pass
Call Transfer (Blind transfer)		
TELUS_TC3.7	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a blind transfer to telus mobile client Confirm 2-way voice after the transfer	Pass
TELUS_TC3.8	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass

TELUS_TC3.9	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to PSTN Confirm 2-way voice after the transfer	Pass
TELUS_TC3.10	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to PSTN Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC3.11	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus mobile client Confirm 2-way voice after the transfer	Pass
TELUS_TC3.12	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus mobile client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Transfer (Consult transfer)		
TELUS_TC3.13	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a consult transfer to Telus mobile client Confirm 2-way voice after the transfer	Pass
TELUS_TC3.14	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass
TELUS_TC3.15	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to BVoIP Confirm 2-way voice after the transfer	Pass
TELUS_TC3.16	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to BVoIP Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC3.17	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus Mobile client Confirm 2-way voice after the transfer	Pass
TELUS_TC3.18	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus Mobile client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Forwarding Don't Answer		

TELUS_TC3.19	Configure a Mobile Phone to Forward calls to a PSTN when Dont Answer. Test G711 Mobile Phone to CFNA to PSTN Number IP PBX phone 1 calls Mobile Phone and should CFNA to PSTN Number 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN number	Pass
Call Forwarding Unconditional		
TELUS_TC3.22	Configure IP PBX phone 1 to CFU to TELUS mobile client IP PBX phone 2 calls phone 1 and should CFU toTELUS mobile client 1. Confirm 2-way voice 2. Confirm phone 1 number and display at TELUS mobile client	Pass
Voicemail		
TELUS_TC3.25	From UMTS call PBX phone, CFB to VM or CFDA to PBX VM Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass
Conference call		
TELUS_TC3.26	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with IP PBX phone 2 Confirm audio among the parties	Pass
TELUS_TC3.27	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with Telus VOIP Confirm audio with mobile client and IP PBX phone	Pass
TELUS_TC3.28	TELUS mobile client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with PSTN Confirm audio with mobile client and IP PBX phone	Pass
TELUS_TC3.29	IP PBX phone 1 calls TELUS mobile client client a conference call to Telus VOIP Confirm audio with mobile client and IP PBX phone	Pass

VoLTE Test cases

Test Number	Test Details	Pass/Fail/NA - Not Applicable
4. Test with TELUS VoLTE/IMS		
Basic inbound/outbound call		
TELUS_TC4.1	Repeat the test by setup the call with G.711. Call from IP PBX phone to TELUS VoLTE client 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
Basic inbound/outbound call with privacy		
TELUS_TC4.3	Call from TELUS VoLTE client to IP PBX phone with privacy enabled. 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
TELUS_TC4.4	Call from IP PBX phone to TELUS VoLTE client, when dialling from the IP PBX phone, use the prefix if applicable to temporary suppress the call display 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
Hold and resume		
TELUS_TC4.5	Call from TELUS VoLTE to IP PBX - after the call setup the PBX phone puts the call on-hold or (MOH), waits 30 seconds, resumes. Confirm audio both way after resume.	Pass
TELUS_TC4.6	Call from IP PBX to TELUS VoLTE - after the call setup, use TELUS VoLTE to put the call on-hold or (MOH), waits 30 seconds, resumes. Confirm 2-way voice after resume.	Pass
Call Transfer (Blind transfer)		
TELUS_TC4.7	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a blind transfer to telus VoLTE client Confirm 2-way voice after the transfer	Pass
TELUS_TC4.8	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass

TELUS_TC4.9	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to PSTN Confirm 2-way voice after the transfer	Pass
TELUS_TC4.10	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to PSTN Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC4.11	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to TELUS Mobile Confirm 2-way voice after the transfer	Pass
TELUS_TC4.12	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to TELUS Mobile Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC4.13	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus VoLTE client Confirm 2-way voice after the transfer	Pass
TELUS_TC4.14	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus VoLTE client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Transfer (Consult transfer)		
TELUS_TC4.15	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a consult transfer to Telus VoLTE client Confirm 2-way voice after the transfer	Pass
TELUS_TC4.16	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass
TELUS_TC4.17	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to BVoIP Confirm 2-way voice after the transfer	Pass
TELUS_TC4.18	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to BVoIP Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC4.19	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus VoLTE client Confirm 2-way voice after the transfer	Pass

TELUS_TC4.20	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus VoLTE client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Forwarding Don't Answer		
TELUS_TC4.21	Configure a VoLTE Phone to Forward calls to a PSTN when Dont Answer. Test G711 IP PBX phone 1 calls VoLTE Phone and should CFNA to PSTN Number 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN number	Pass
Voicemail		
TELUS_TC4.24	Repeat the test by setup the call with G.711.Test with Movius VM platforms. IP PBX phone 1 calls TELUS VoLTE client Don't answer the call in the TELUS VoLTE client; after 4 ring, voicemail kick in Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass
TELUS_TC4.25	Repeat the test by setup the call with G.729.Test with Movius VM platforms. IP PBX phone 1 calls TELUS VoLTE client Don't answer the call in the TELUS VoLTE client; after 4 ring, voicemail kick in Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass
Conference call		
TELUS_TC4.26	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with IP PBX phone 2 Confirm audio among the parties	Pass
TELUS_TC4.27	TELUS VoLTE client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with Telus VOIP Confirm audio with VoLTE client and IP PBX phone	Pass
TELUS_TC4.28	IP PBX phone 1 calls TELUS VoLTE client client a conference call to Telus VOIP Confirm audio with VoLTE client and IP PBX phone	Pass
TELUS_TC4.29	IP PBX phone 1 calls TELUS VoLTE client client a conference call to Telus Mobile Confirm audio with VoLTE client and IP PBX phone	Pass

NGHP Test cases

Test Number	Test Details	Pass/Fail/NA - Not Applicable
5. Test with TELUS NGHP		
Basic inbound/outbound call		
TELUS_TC5.1	Repeat the test by setup the call with G.729. Call from IP PBX phone to TELUS NGHP client 1. Confirm 2-way voice 2. Confirm the proper calling number is shown 3. Confirm the proper call display name is shown	Pass
Basic inbound/outbound call with privacy		
TELUS_TC5.2	Call from TELUS NGHP client to IP PBX phone with privacy enabled. 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
TELUS_TC5.3	Call from IP PBX phone to TELUS NGHP client, when dialling from the IP PBX phone, use the prefix if applicable to temporary suppress the call display 1. Confirm 2-way voice 2. Confirm the proper calling number is not shown 3. Confirm the proper call display name is not shown	Pass
Hold and resume		
TELUS_TC5.4	Call from TELUS NGHP to IP PBX - after the call setup the PBX phone puts the call on-hold or (MOH), waits 30 seconds, resumes. Confirm audio both way after resume.	Pass
TELUS_TC5.5	Call from IP PBX to TELUS NGHP - after the call setup, use TELUS NGHP to put the call on-hold or (MOH), waits 30 seconds, resumes. Confirm 2-way voice after resume.	Pass
Call Transfer (Blind transfer)		
TELUS_TC5.6	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a blind transfer to telus NGHP client Confirm 2-way voice after the transfer	Pass
TELUS_TC5.7	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass

TELUS_TC5.8	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to BVoIP Confirm 2-way voice after the transfer	Pass
TELUS_TC5.9	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to BVoIP Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC5.10	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to TELUS Mobile Confirm 2-way voice after the transfer	Pass
TELUS_TC5.11	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to TELUS Mobile Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC5.12	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to VoLTE Confirm 2-way voice after the transfer	Pass
TELUS_TC5.13	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to VoLTE Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC5.14	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus NGHP client Confirm 2-way voice after the transfer	Pass
TELUS_TC5.15	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a blind transfer to Telus NGHP client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Transfer (Consult transfer)		
TELUS_TC5.16	IP PBX phone 1 calls IP PBX phone 2 IP PBX phone 2 performs a consult transfer to Telus NGHP client Confirm 2-way voice after the transfer	Pass
TELUS_TC5.17	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to IP PBX phone 2 Confirm 2-way voice after the transfer	Pass
TELUS_TC5.19	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to BVoIP Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC5.20	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to TELUS Mobile Confirm 2-way voice after the transfer	Pass

TELUS_TC5.21	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to TELUS Mobile Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
TELUS_TC5.22	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to VoLTE Confirm 2-way voice after the transfer	Pass
TELUS_TC5.24	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus NGHP client Confirm 2-way voice after the transfer	Pass
TELUS_TC5.25	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a consult transfer to another Telus NGHP client Confirm 2-way voice after the transfer Repeat the same test using SIP REFER	Pass
Call Forwarding Unconditional		
TELUS_TC5.26	Configure a NGHP Phone to Forward calls to a PSTN unconditional IP PBX phone 1 calls NGHP Phone and should CFU to PSTN Number 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN number	Pass
TELUS_TC5.27	Configure a NGHP Phone to Forward calls to a VoLTE unconditional IP PBX phone 1 calls NGHP Phone and should CFU toVoLTE Number 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN number	Pass
TELUS_TC5.28	Configure a NGHP Phone to Forward calls to a TELUS Mobile unconditional. IP PBX phone 1 calls NGHP Phone and should CFU to TELUS Mobile Number 1. Confirm 2-way voice 2. Confirm phone 1 number and display at PSTN number	Pass
Voicemail		
TELUS_TC5.29	Repeat the test by setup the call with G.711.Test with Movius VM platforms. IP PBX phone 1 calls TELUS NGHP client Don't answer the call in the TELUS NGHP client; after 4 ring, voicemail kick in Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass

TELUS_TC5.30	Repeat the test by setup the call with G.729.Test with Movius VM platforms. IP PBX phone 1 calls TELUS NGHP client Don't answer the call in the TELUS NGHP client; after 4 ring, voicemail kick in Record a message Follow the prompt to play back the message Follow the prompt to cancel the recording then hang up	Pass
Conference call		
TELUS_TC5.32	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with IP PBX phone 2 Confirm audio among the parties	Pass
TELUS_TC5.33	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with Telus VOIP Confirm audio with NGHP client and IP PBX phone	Pass
TELUS_TC5.34	TELUS NGHP client calls IP PBX phone 1 IP PBX phone 1 performs a conference call with VoLTE Confirm audio with NGHP client and IP PBX phone	Pass
TELUS_TC5.35	IP PBX phone 1 calls TELUS NGHP client client a conference call to Telus Mobile Confirm audio with NGHP client and IP PBX phone	Pass

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

```
filter-config
        name                                    all
        user                                    *
host-route
        dest-network                            172.25.128.64
        netmask                                 255.255.255.192
        gateway                                 198.162.151.38
        description                             IPT-Core-SBC
local-policy
        from-address                            *
        to-address                              *
        source-realm                            Core
        policy-attribute
                next-hop                                172.25.128.75
                realm                                   Peer
                action                                  replace-uri
local-policy
        from-address                            *
        to-address                              *
        source-realm                            Peer
        policy-attribute
                next-hop                                172.17.10.120
                realm                                   Core
                action                                  replace-uri
media-manager
        media-policing                          disabled
network-interface
        name                                    M00
        ip-address                              198.162.151.37
        netmask                                 255.255.255.252
        gateway                                 198.162.151.38
        hip-ip-list                             198.162.151.37
        icmp-address                            198.162.151.37
network-interface
        name                                    M10
        ip-address                              172.17.10.169
        netmask                                 255.255.255.0
        gateway                                 172.17.10.254
        dns-ip-primary                          172.17.99.41
        dns-domain                              npsi.lab
        hip-ip-list                             172.17.10.169
        icmp-address                            172.17.10.169
ntp-config
        server                                  172.17.99.12
phy-interface
        name                                    M00
        operation-type                          Media
        speed                                   1000
phy-interface
        name                                    M10
        operation-type                          Media
        slot                                    1
        speed                                   1000
realm-config
        identifier                              Core
        network-interfaces                      M10:0.4
realm-config
        identifier                              Peer
        network-interfaces                      M00:0.4
session-agent
        hostname                                172.17.10.120
        ip-address                              172.17.10.120
        transport-method                        StaticTCP
```

```
        realm-id                                Core
        description                             CUCM 10.5 IPT
        ping-method                             OPTIONS;hops=0
        ping-interval                           60
session-agent
        hostname                                172.25.128.75
        ip-address                              172.25.128.75
        realm-id                                Peer
        description                             TELUS SIP Trunk
        ping-interval                           30
        in-manipulationid                       towardstrunk
session-translation
        id                                      del9
        rules-calling                           del9
        rules-called                            del9
sip-config
        options                                 max-udp-length=0
sip-feature
        name                                    100rel
        realm                                   Peer
        require-mode-inbound                    Pass
        require-mode-outbound                   Pass
sip-interface
        realm-id                                Core
        sip-port
                address                                 172.17.10.169
                transport-protocol                      TCP
        sip-port
                address                                 172.17.10.169
        registration-caching                    enabled
        options                                 100rel-interworking
sip-interface
        realm-id                                Peer
        sip-port
                address                                 198.162.151.37
        options                                 100rel-interworking
sip-manipulation
        name                                    towardstrunk
        header-rule
                name                                    DelReq
                header-name                             Require
                action                                  delete
                match-value                             100rel
sip-monitoring
        monitoring-filters                      *
snmp-community
        community-name                          public
        ip-addresses                            172.17.71.8
                                                172.18.129.145
                                                172.18.129.141
                                                172.18.129.144
                                                172.18.129.154
steering-pool
        ip-address                              172.17.10.169
        start-port                              50000
        end-port                                60000
        realm-id                                Core
steering-pool
        ip-address                              198.162.151.37
        start-port                              50000
        end-port                                60000
        realm-id                                Peer
system-config
        comm-monitor
                state                                   enabled
                monitor-collector
                        address                                 172.17.100.90
        call-trace                              enabled
        default-gateway                         172.17.100.254
```

```
        source-routing                                enabled
translation-rules
        id                                            del9
        type                                          delete
        delete-string                                 9
        delete-index                                  9
web-server-config
        inactivity-timeout                            0
```

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.

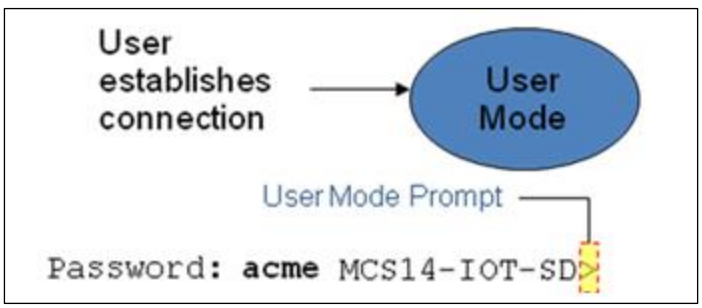
  Initial connectivity will be through the serial console port. At a minimum, this is how to configure the management (eth0) interface on the SBC.

ACLI Basics

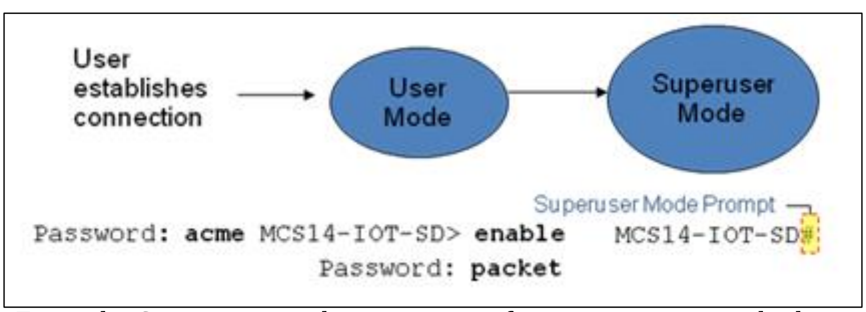
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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.



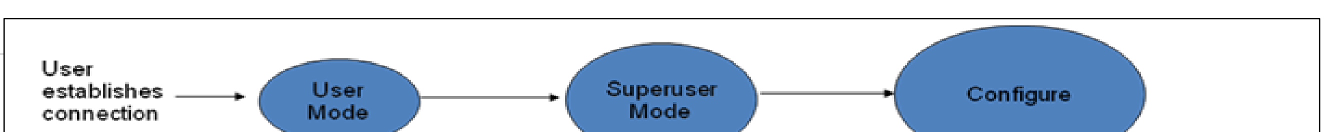
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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.

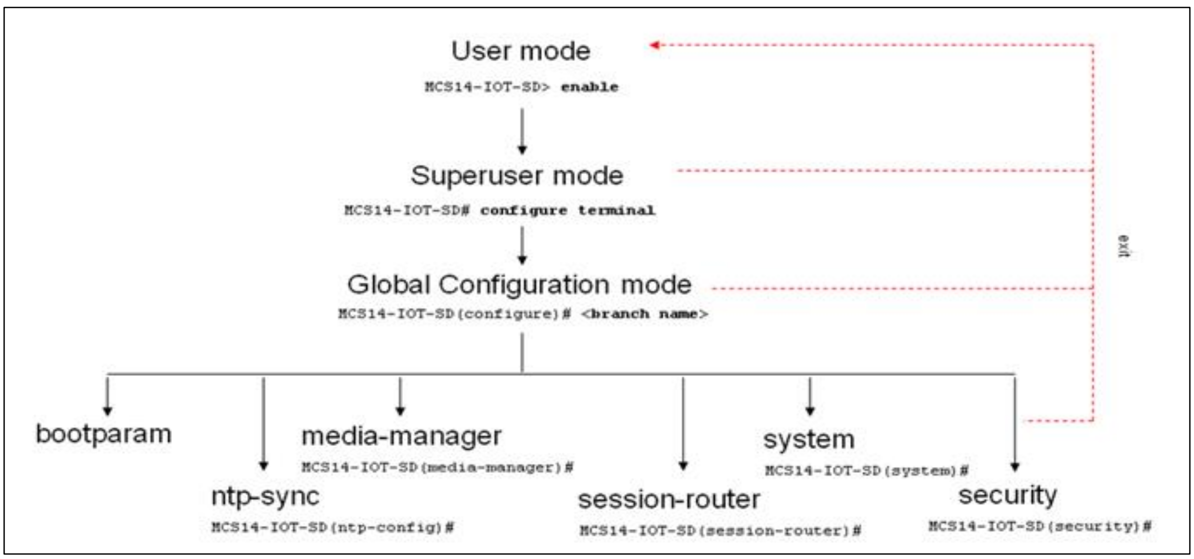
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the configure terminal command in the Superuser mode.

Configuration mode is identified by the word configure in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, oraclesbc1(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.

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.

```
oraclesbc1 # 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'.
oraclesbc1 #
```

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.

```
oraclesbc1# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
oraclesbc1#
```


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

Worldwide Inquiries
Phone: +1.650.506.7000
Fax: +1.650.506.7200

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