



Disclaimer

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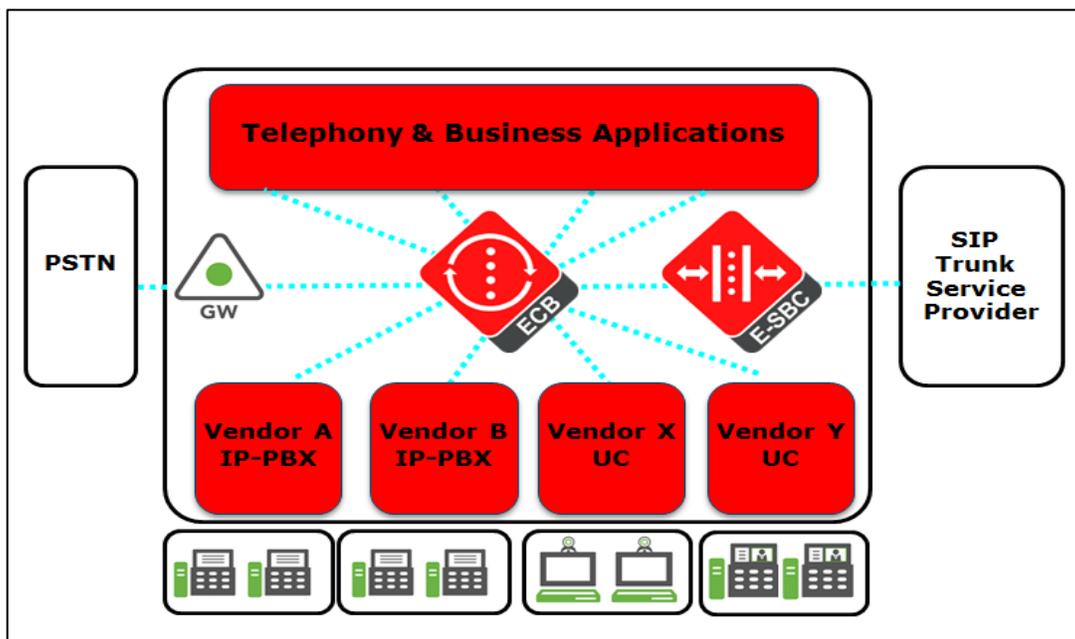
Intended Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring Oracle Enterprise Session Border Controller (E-SBC), Oracle Enterprise Communications Broker (EOM), Lync Mediation Server, Avaya Aura System Manager and Cisco Unified Communications Manager. There will be steps that require navigating Microsoft Windows Server as well as the Acme Packet Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

Introduction

Oracle Enterprise Communications Broker Overview

The Oracle Enterprise Communications Broker (ECB) is an enterprise-class, core signaling component designed to simplify communications networks. It combines innovative approaches toward dial plan management and SIP topology-aware routing with a purpose-built, intuitive GUI interface. While at its best in signaling environments comprised of products and solutions from multiple vendors, it is useful for consolidating policy enforcement decisions, integrating third-party applications, and managing a network-wide routing topology even in homogenous architectures.





The ECB is typically deployed in the core of a multi-vendor communications network where multiple UC, PBX and service provider trunk interfaces must be interconnected. It normalizes communications between disparate premise-based systems and connects them to service provider networks and hosted applications through E-SBCs.

Document Overview

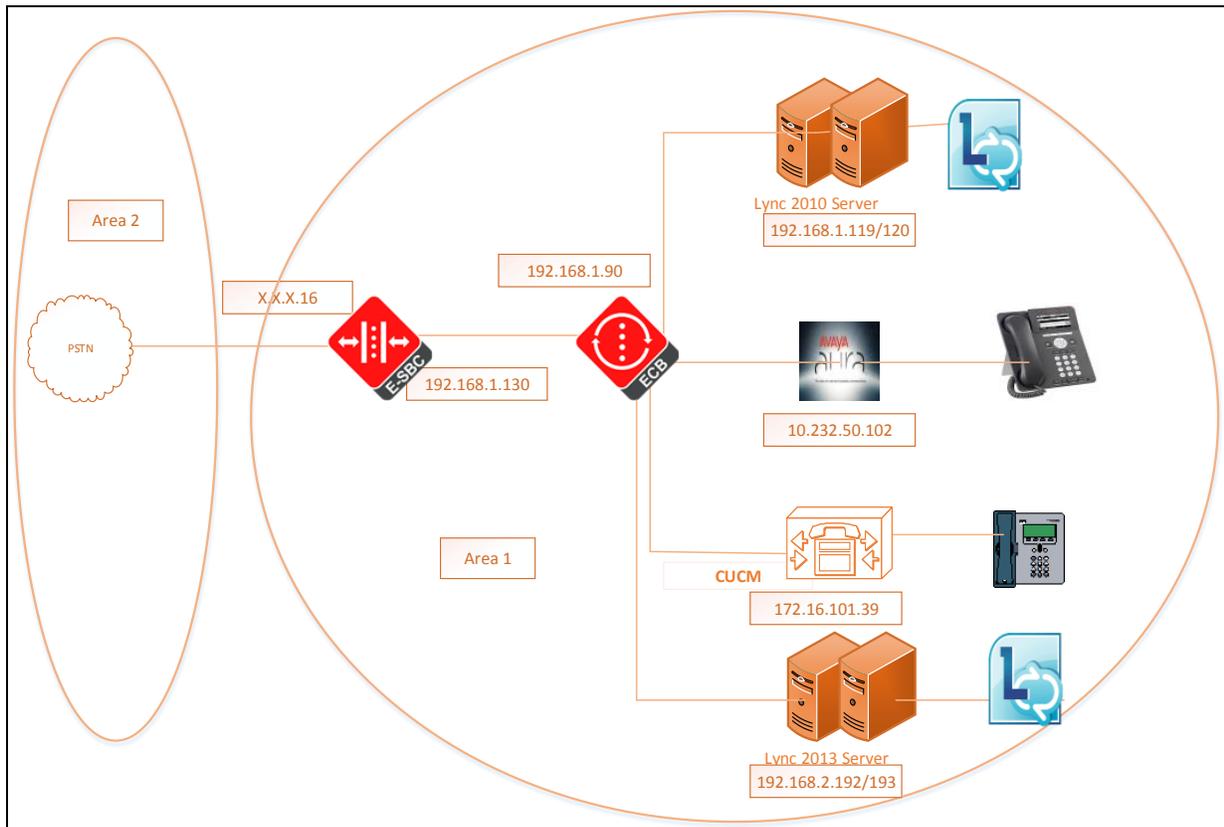
This technical application note documents the implementation of the Oracle Enterprise Communications Broker in an Enterprise network consisting of multi-vendor Unified Communications platforms - Microsoft Lync 2010/2013, Avaya Aura Session Manager and Cisco Unified Communications Manager - connecting to SIP trunk through an Enterprise Session Border Controller.

Requirements

- Oracle Enterprise Communications Broker
- Oracle Enterprise Session Border Controller
- Microsoft Lync 2010/2013
- Avaya Aura 6.3.4
- Cisco Unified Communications Manager 8.6

Lab Configuration

The following diagram illustrates the lab environment created to facilitate certification testing.



The network architecture consists of two areas. Area 1 represents the Enterprise network and Area 2 is the service provide network. The Enterprise network has ECB at its core connecting together multiple UC platforms. The ECB connects to the E-SBC which provides the enterprise network access to PSTN through the service provider network.

The configuration, validation and troubleshooting of the Area 1 is the focus of this document and will be described in five phases

- Phase 1 – Configure the ECB
- Phase 2 – Configure Lync 2010/2013 server
- Phase 3 – Configure the Avaya Aura System Manager
- Phase 4 – Configure the Cisco Unified Communications Manager
- Phase 5 – Configure the E-SBC.

Phase 1 – Configure the ECB

The Oracle Enterprise Communications Broker is available either as an appliance or as an application for operation on virtual machines. When running as an appliance, the ECB software is packaged with the Netra Server X3-2 for Oracle and delivered to the end customers. When running as a virtual application, the ECB software can be deployed on any third-party COTS hardware that meets the specified guidelines.

Once the ECB is deployed (in the appliance mode or the application mode) and connected, you can power on the ECB. Software installation of the ECB is required upon first startup. Although the ECB is primarily configured through the GUI, you need to perform the software installation and setup via the CLI.

Connecting to the ECB

The CLI can be accessed through the console connection. If the ECB is appliance based, you can connect to the ECB console using your laptop running a terminal emulator application like PuTTY and an RJ-5 cable. Start the terminal emulator application with the following settings:

- Baud Rate=115200
- Data Bits=8
- Parity=None
- Stop Bits=1
- Flow Control=None

Power the ECB on. Upon successful boot, the system prompts you to login. The default password for user mode is “acme” and super user mode is “packet”.

You can now use the installation wizard to setup your ECB. Using the wizard, you can enable the Web Server, set management access as well as configure high availability and service interface addressing.

```
Password: acme
ORACLE> enable
Password: packet
```

Running Setup

The following steps detail the process of using the installation wizard to configure the base setup of the ECB

1. Start the installation wizard by entering the command `run setup` in super user mode.

```
ORACLE# run setup
```

The following displays

```
-----  
Thank you for purchasing the Oracle ECB. The following short wizard  
will guide you through the initial set-up.  
-----  
'?' = Help; '.' = Clear; 'q' = Exit  
CONFIGURATION  
WARNING: Proceeding with wizard will result in existing configuration  
being erased.  
Erase config and proceed (yes/no) [no] : yes
```

2. Type yes and press Enter

```
Configuration will be backed up as  
bkup_setup_wizard_Apr_8_13_25_49_632.gz  
'-' = Previous; '?' = Help; '.' = Clear; 'q' = Exit  
HIGH AVAILABILITY  
This ECB may be a standalone or part of a highly available redundant  
pair.  
Oracle ECB mode  
1 - standalone  
2 - high availability  
Enter choice [1 - standalone] : 1
```

3. Our setup consists of a standalone server. Type 1 and hit Enter
4. You will then be asked to configure a unique target name, the ip address, subnet mask and gateway of the management interface of the ECB. Please note at any time during configuration if you would like to keep the default values (values mentioned in []), press Enter.

```
Unique target name of this ECB [primary] :ECB-Oracle
```

```
IP address on management interface [172.30.200.111] : 172.18.255.55
Subnet mask on management interface [255.255.0.0] :
Gateway IP address on management interface [172.18.0.1] :
```

5. You will then see a prompt to configure your sip-interface. This step is required; the system does not allow you to proceed without making a setting. When prompted enter the ip address, subnet mask and gateway ip address of the sip-interface.

```
IP address on SIP interface : 192.168.1.90
Subnet mask on SIP interface [255.255.255.0] :
Gateway IP address on SIP interface :192.168.1.1
```

6. The prompt to setup the system timezone will display

```
SETUP TIMEZONE Setup system timezone (yes/no) [yes] : yes
```

Type your response and press Enter.

7. You will then be asked to enter the number for sessions purchased for the ECB. Type your response and press Enter.

```
LICENSED SESSIONS
Number of licensed sessions : 400
```

You will see the following message prompting to save the settings before proceeding to the timezone setup.

```
Enter 1-20 to modify,'d' to display summary,'s' to save,'q' to
exit.[s]:
Saving changes and quitting wizard. Are you sure? [y/n]?:
```

8. Type your response and press Enter.

```
SETUP TIMEZONE Setup system timezone (yes/no) [yes] : yes
```

The following message displays

```
Deleting configuration
Erase-Cache received, processing.
waiting 1200 for request to finish
Request to 'ERASE-CACHE' has Finished,
Erase-Cache: Completed
Running timezone setup application
Calling tzselect. Use ^D to cancel without save
Please identify a location so that time zone rules can be set
```

```
correctly.
Please select a continent or ocean.
1) Africa
2) Americas
3) Antarctica
4) Arctic Ocean
5) Asia
6) Atlantic Ocean
7) Australia
8) Europe
9) Indian Ocean
10) Pacific Ocean
11) none - I want to specify the time zone using the Posix TZ format.
#?
```

Type your response, for example, 2 for Americas and press Enter. The system lists applicable countries in the Americas. Make your selection and press Enter. The system displays applicable time zones. Make your selection. The following message appears

```
The following information has been given:
United States
Eastern Time
Therefore TZ='America/New_York' will be used.
Local time is now: Thu Apr 11 10:13:38 EDT 2014.
Universal Time is now: Thu Apr 11 14:13:38 UTC 2014.
Is the above information OK?
1) Yes
2) No
#?
```

9. Type 1 and then hit Enter. You will be then shown a summary of your settings.

```
Saved configuration. -----
HIGH AVAILABILITY
 2 : ECB mode           : standalone
 3 : ECB role           : N/A
```

AUTOMATIC CONFIGURATION

6 : Acquire config from the Primary (yes/no) : N/A

ECB SETTINGS

7 : Unique target name of this ECB : ECB-Oracle

8 : Management interface IP address :
172.18.255.55

9 : Management interface subnet mask : 255.255.0.0

10: Management interface gateway IP address : 172.18.0.1

11: SIP interface VLAN id : 0

12: SIP interface IP address : 192.168.1.90

15: SIP interface subnet mask :
255.255.255.0

16: SIP interface gateway IP address : 192.168.1.1

PEER CONFIGURATION

18: Peer target name : N/A

SETUP TIMEZONE

19: Setup system timezone (yes/no) : yes

LICENSED SESSIONS

20: Number of licensed sessions : 400

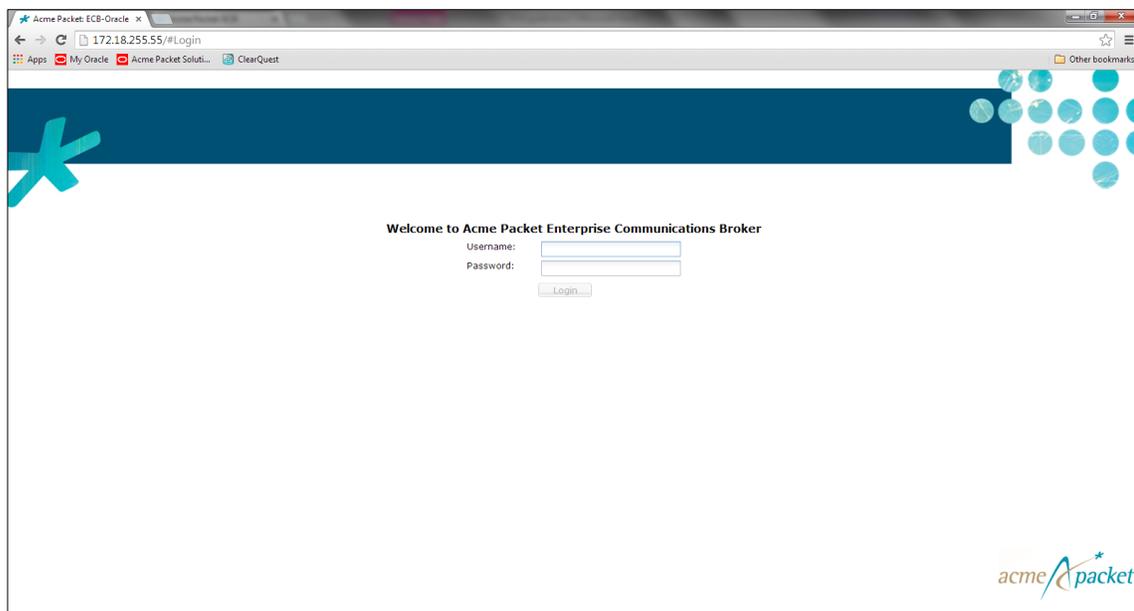
You may access the GUI via <http://172.18.255.55:80/> or continue using the acli after reboot.

Logging in the ECB

You can now access the ECB through the Web GUI. Start an Internet browser and start the GUI using the URL:

http://server ip address/.

The login screen will appear.

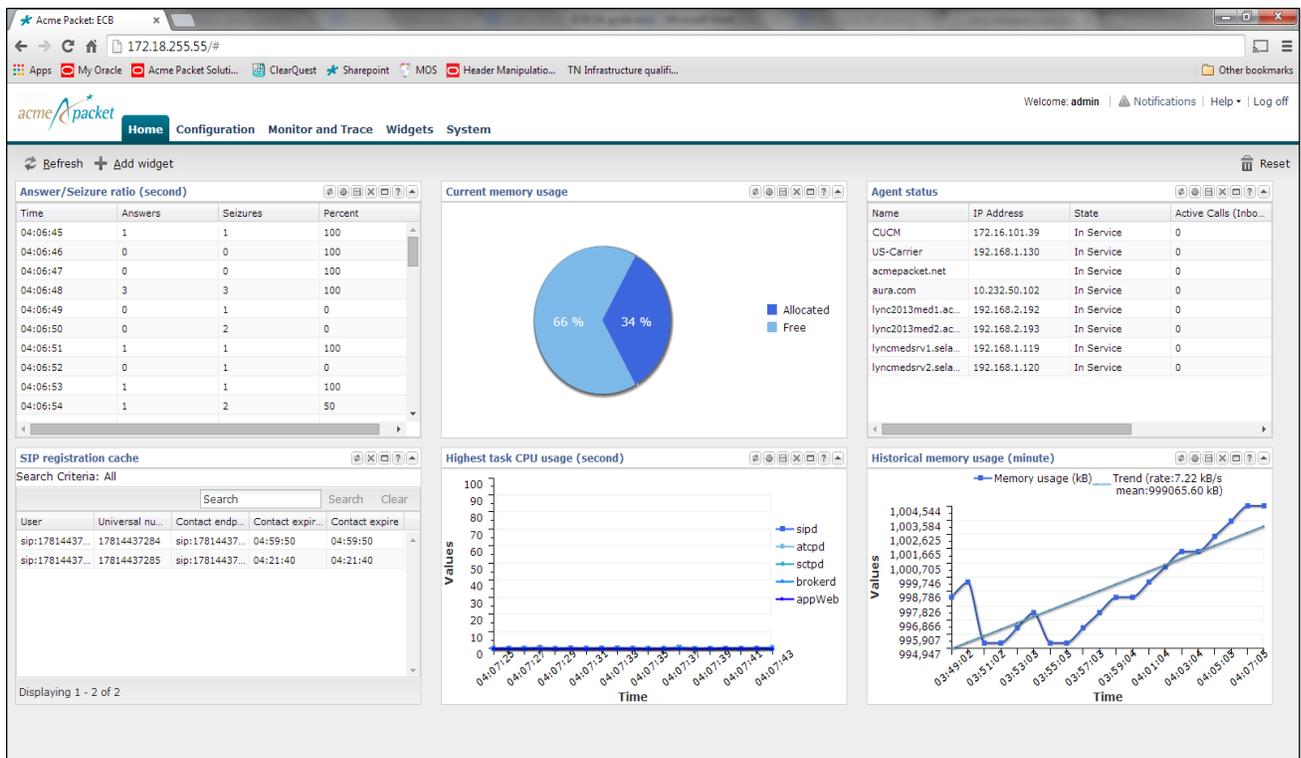


Enter your GUI username and password. The default username for the User level is "user" and the default password is "acme". The default username for an Administrator level is "admin", and the default password is "packet".

Configuring the ECB

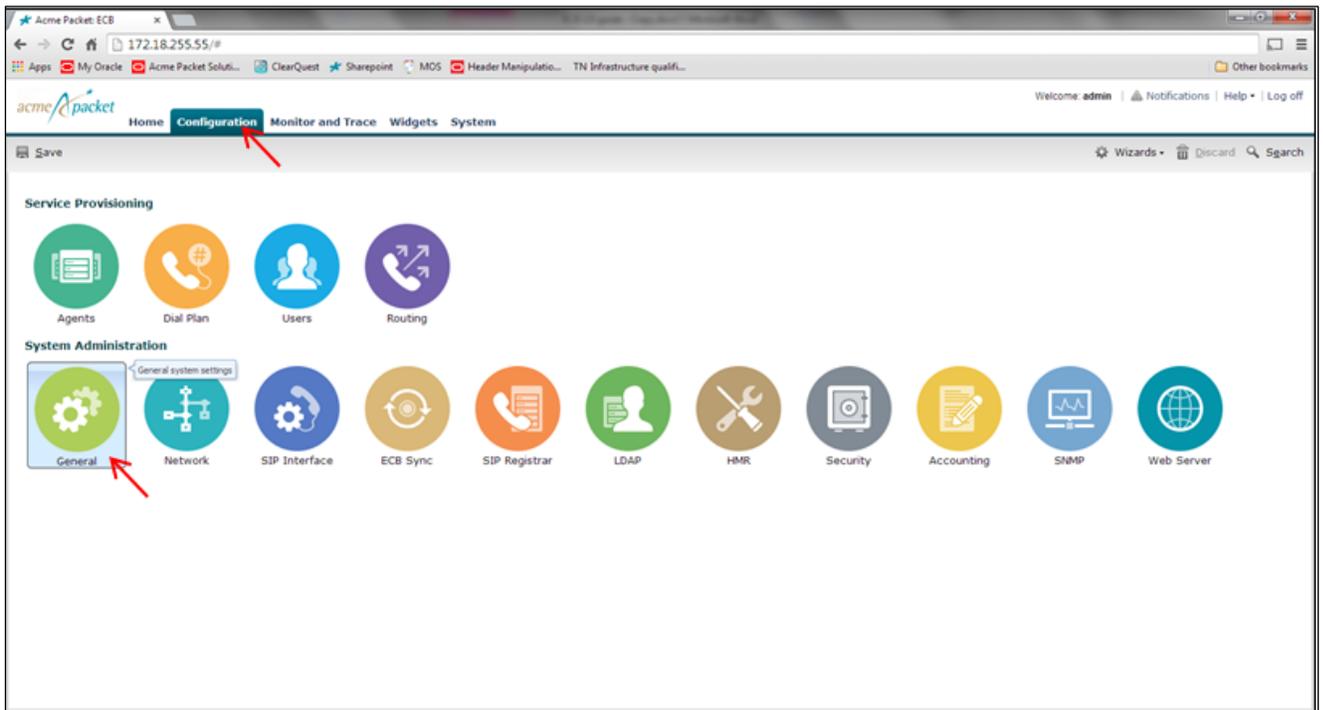
After logging into the ECB, the **Home** screen will be displayed. The ECB GUI has five tabs across the top – **Home**, **Configuration**, **Monitor and Trace**, **Widgets** and **System**.

The **Home** tab as shown below contains a configurable dashboard displaying the system statistics.



System Settings

Select the **Configuration** tab. This tab displays the configurable elements in the ECB in two sections – **Service Provisioning** and **System Administration**. Click on the **General** icon under **System Administration**.



Modify System Settings page is displayed.

Modify System settings

Hostname:

Description:

Location:

Default gateway IP address:

Enable restart on critical failure:

Enable SIP monitoring and tracing:

NTP servers:

Logging settings

SNMP settings

Denial of service settings

Communications monitoring probe settings

High availability settings

Expand the **Logging settings** section.

Modify System settings

Hostname:

Description:

Location:

Default gateway IP address:

Enable restart on critical failure:

Enable SIP monitoring and tracing:

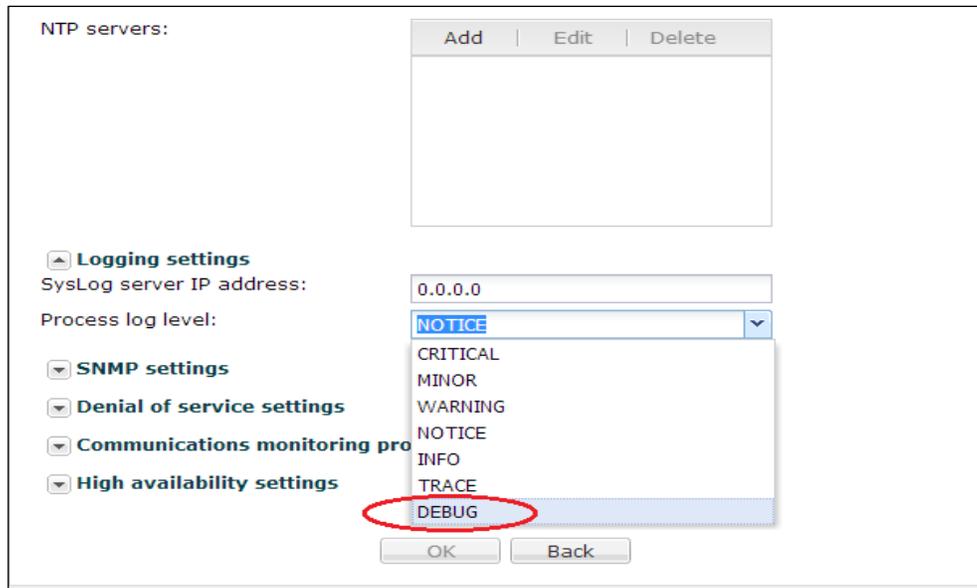
NTP servers:

Logging settings

SysLog server IP address:

Process log level:

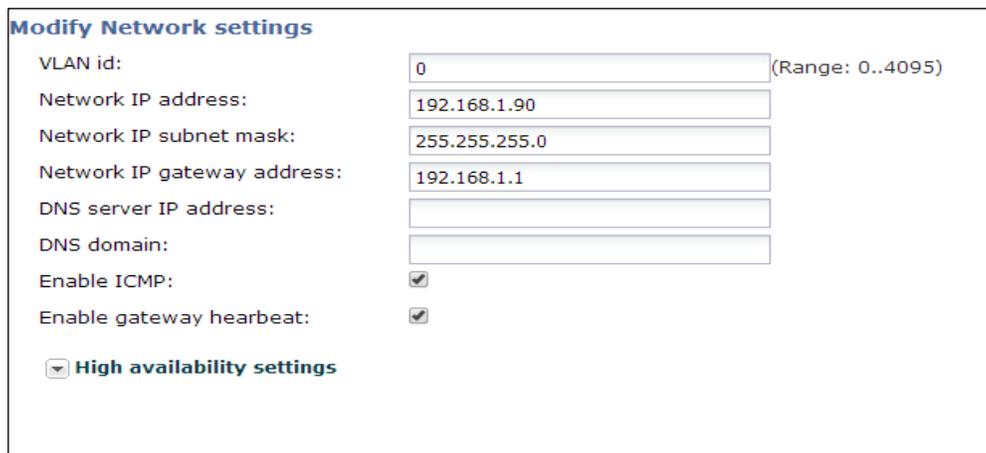
Process log level is set at **NOTICE**. Change the setting to **DEBUG** by selecting the option from the drop down menu and click **OK**.



The screenshot shows a configuration window with several sections. At the top, there is a section for 'NTP servers' with 'Add', 'Edit', and 'Delete' buttons. Below this is the 'Logging settings' section, which is expanded. It contains a text field for 'SysLog server IP address' with the value '0.0.0.0' and a dropdown menu for 'Process log level'. The dropdown menu is open, showing options: CRITICAL, MINOR, WARNING, NOTICE, INFO, TRACE, and DEBUG. The 'DEBUG' option is highlighted with a red circle. At the bottom of the window are 'OK' and 'Back' buttons.

Click the **Configuration** button at the top to go to the **Configuration** tab.

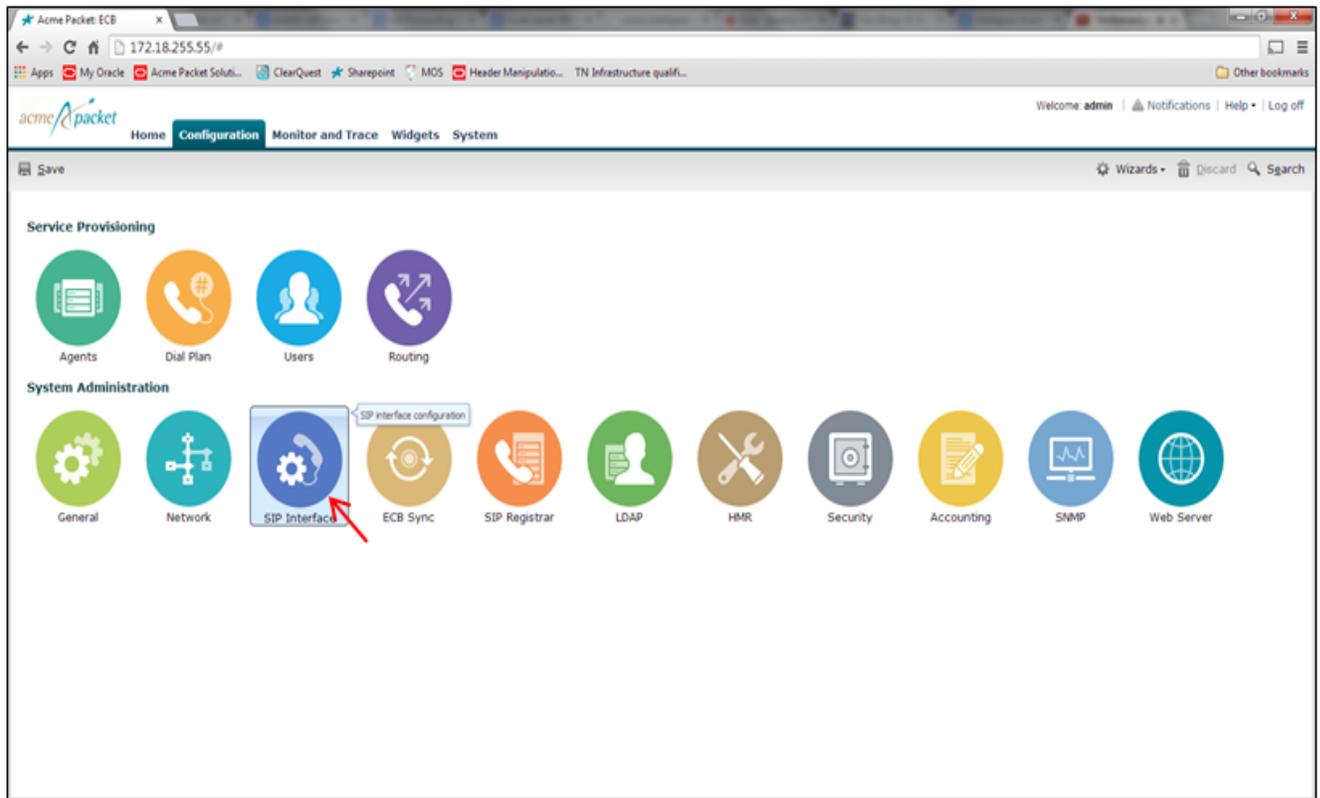
You can verify the network interface settings configured through the `run setup` command by clicking on the **Network** icon under **System Administration**



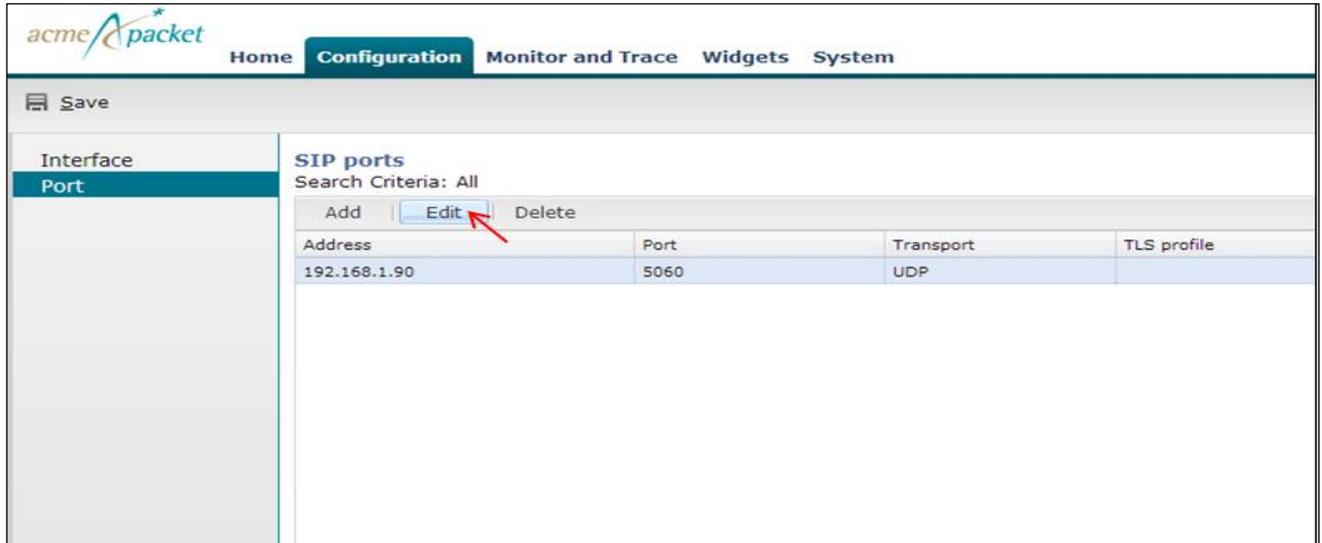
The screenshot shows the 'Modify Network settings' window. It contains several text input fields and checkboxes. The fields are: 'VLAN id' (value: 0, range: 0..4095), 'Network IP address' (value: 192.168.1.90), 'Network IP subnet mask' (value: 255.255.255.0), 'Network IP gateway address' (value: 192.168.1.1), 'DNS server IP address', and 'DNS domain'. There are two checked checkboxes: 'Enable ICMP' and 'Enable gateway heartbeat'. At the bottom, there is a collapsed section for 'High availability settings'.

Configure SIP Interfaces

Click **Configuration** button to go to the **Configuration** tab. Select the **SIP Interface icon** under **System Administration** to make changes to the SIP interface settings configured during initial setup.



Click on the **Port** tab on the left. You will see the sip port 192.168.1.90 with protocol UDP. Click **Edit** to change its protocol to TCP.

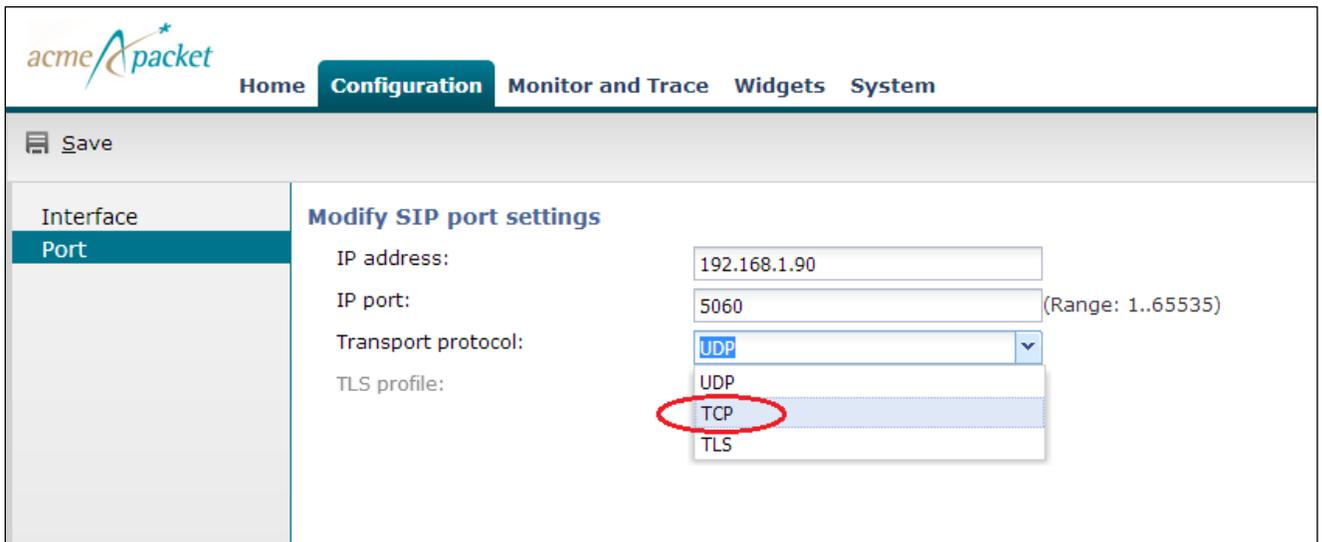


The screenshot shows the acmePacket Configuration page. The 'Configuration' tab is active. On the left, the 'Port' tab is selected under the 'Interface' section. The main content area is titled 'SIP ports' and shows a table with the following data:

Address	Port	Transport	TLS profile
192.168.1.90	5060	UDP	

Buttons for 'Add', 'Edit', and 'Delete' are visible above the table. A red arrow points to the 'Edit' button.

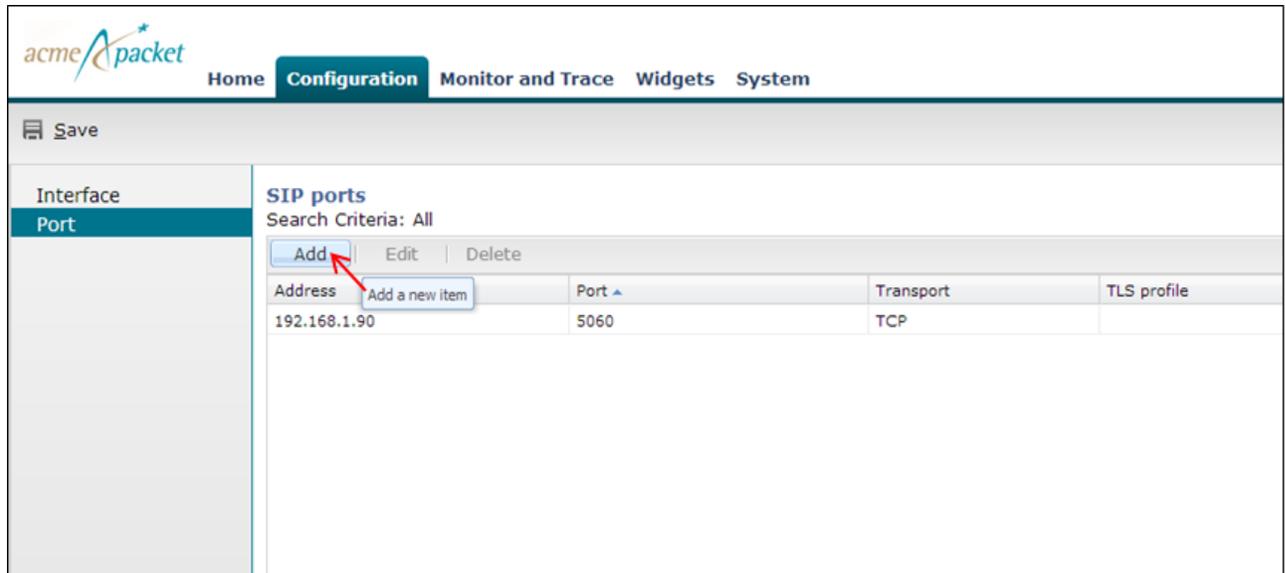
On the **Modify SIP port settings** page, select TCP as the transport protocol from the drop-down menu and click **OK**.



The screenshot shows the 'Modify SIP port settings' page. The form contains the following fields:

- IP address: 192.168.1.90
- IP port: 5060 (Range: 1..65535)
- Transport protocol: A dropdown menu is open, showing options UDP, TCP, and TLS. The 'TCP' option is highlighted and circled in red.
- TLS profile: (empty)

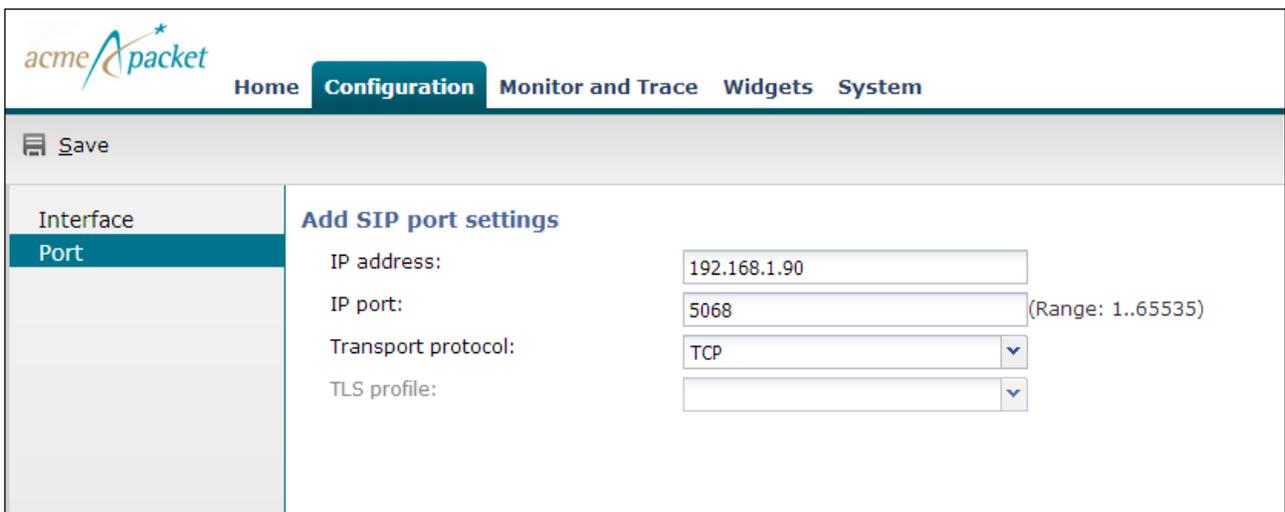
On the SIP ports page, click **Add** to add another sip port.



The screenshot shows the acmePacket Configuration page. The 'SIP ports' section is active, displaying a table with one entry. A red arrow points to the 'Add' button above the table.

Address	Port	Transport	TLS profile
192.168.1.90	5060	TCP	

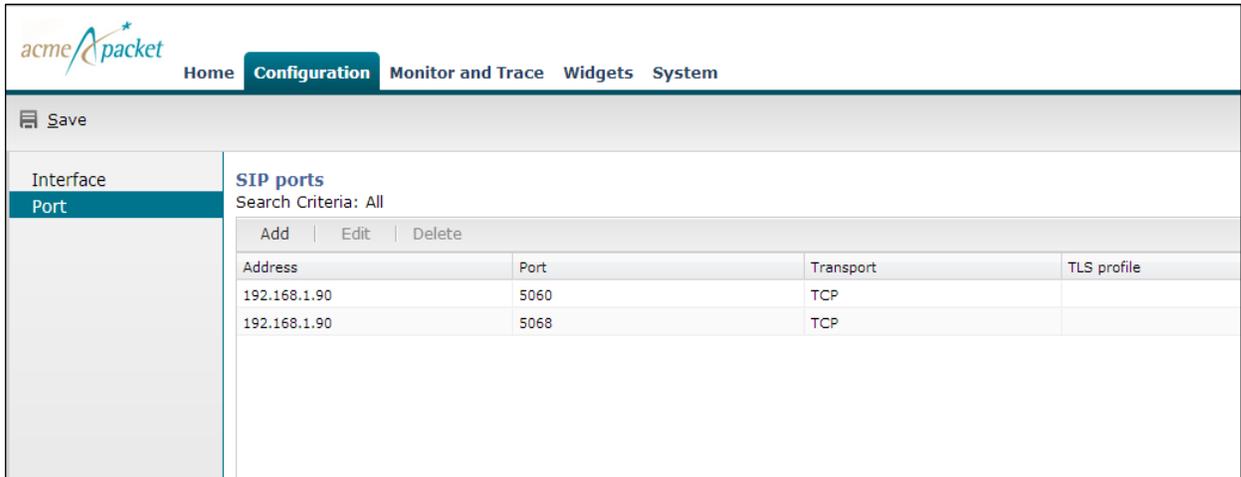
Add a sip port with address 192.168.1.90, port 5068 and transport protocol TCP as shown below and click **OK**.



The screenshot shows the 'Add SIP port settings' form. The fields are filled with the values specified in the instructions.

IP address:	<input type="text" value="192.168.1.90"/>
IP port:	<input type="text" value="5068"/> (Range: 1..65535)
Transport protocol:	<input type="text" value="TCP"/>
TLS profile:	<input type="text"/>

The SIP ports page will be displayed showing the two sip ports we configured.



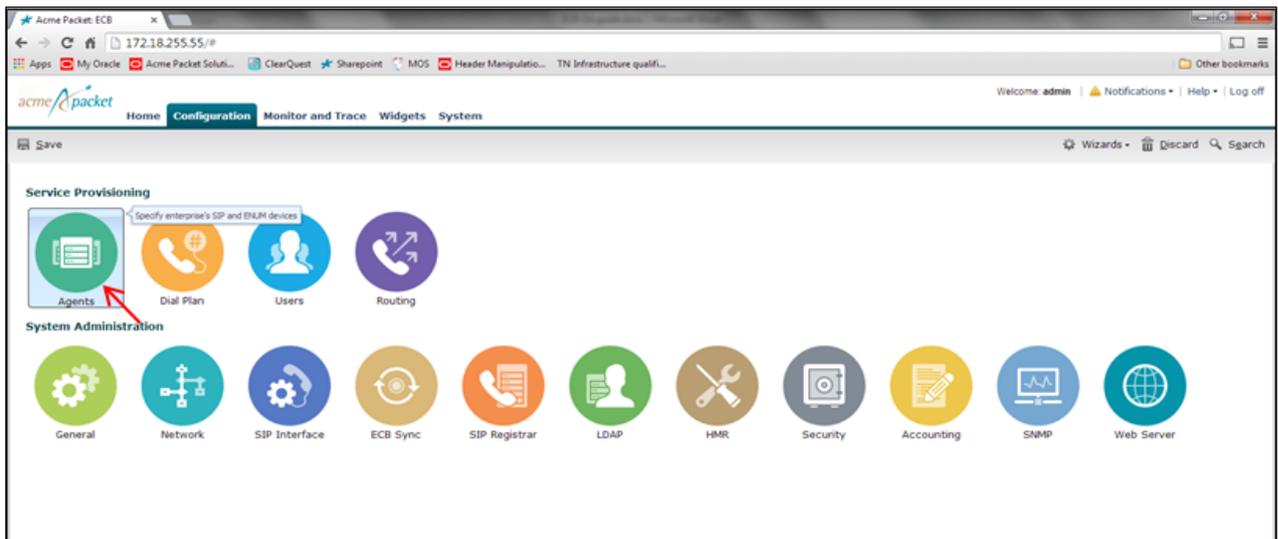
The screenshot shows the Acme Packet web interface. The top navigation bar includes 'Home', 'Configuration', 'Monitor and Trace', 'Widgets', and 'System'. The 'Configuration' tab is active. Below the navigation bar, there is a 'Save' button and a sidebar with 'Interface' and 'Port' options. The main content area is titled 'SIP ports' and shows a table with two rows of data. The table has columns for 'Address', 'Port', 'Transport', and 'TLS profile'. The first row shows address 192.168.1.90, port 5060, and transport TCP. The second row shows address 192.168.1.90, port 5068, and transport TCP.

Address	Port	Transport	TLS profile
192.168.1.90	5060	TCP	
192.168.1.90	5068	TCP	

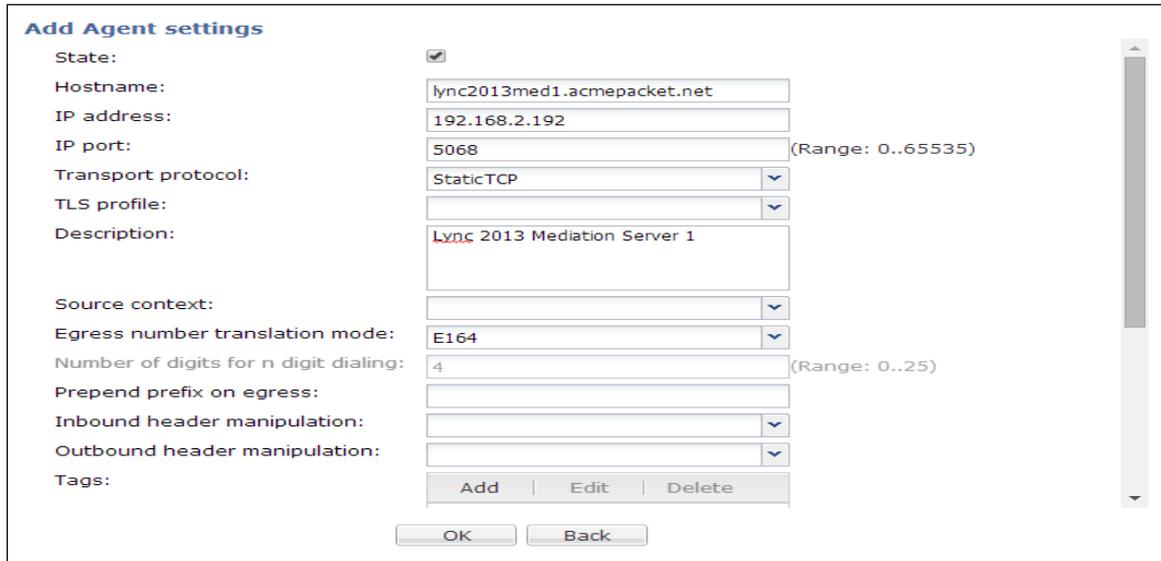
Click on **Configuration** button to back to the **Configuration** tab.

Configure Agents

We will now configure the next hops in our routing paths – the Agents – which in our setup are the Lync Mediation Server, Avaya SM and the E-SBC which connects the ECB to the SIP trunk. Click on **Agents** icon under **Service Provisioning**.



The Agents page will be displayed. Click on the **Add** button. The **Add Agent settings** page is displayed. Add the Lync mediation server by configuring the hostname, ip address and port as shown below.



Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

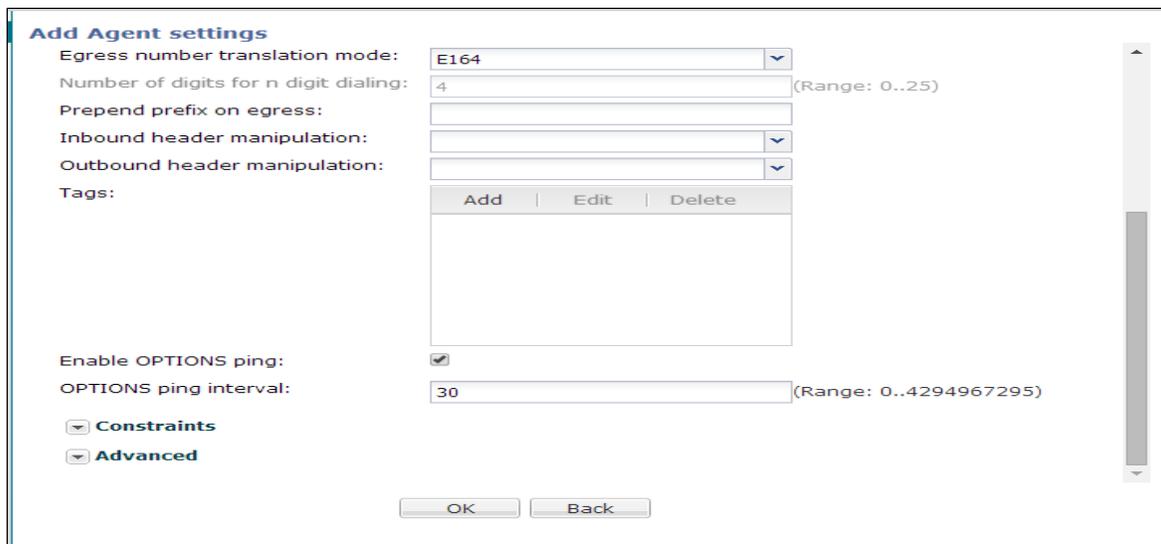
Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags:

Scroll down to enable SIP OPTIONS to monitor agent health locally. Check the **Enable OPTIONS ping** check box and configure the **OPTIONS ping interval** to 30. Click **OK**.



Add Agent settings

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags:

Enable OPTIONS ping:

OPTIONS ping interval: (Range: 0..4294967295)

▼ Constraints

▼ Advanced

You will now see the Lync server listed under **Agents**. Click **Add** to add the second Mediation server from the pool and also enable **OPTIONS** as shown in the previous step.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags: | |

Add another agent with hostname acmepacket.net to route the INVITEs for transfers from Lync and click **OK**.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags: | |

Next add the two Mediation servers from the Lync 2010 setup using the following settings. In order to monitor agent health locally, please enable OPTIONS as shown in the previous steps.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags: | |

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags: | |

Now add the Avaya Session manager as an agent using the following settings. In our setup, the egress number translation mode for Avaya server was set to no-country code.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol:

TLS profile:

Description:

Source context:

Egress number translation mode:

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation:

Outbound header manipulation:

Tags: | |

Now add the CUCM as an agent using the following settings. In our setup, the egress number translation mode for CUCM is E164-no-plus.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol:

TLS profile:

Description:

Source context:

Egress number translation mode:

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation:

Outbound header manipulation:

Tags: | |

Next add the E-SBC which provides the enterprise network access to the SIP trunk and click **OK**.

Add Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags:

Add | Edit | Delete

OK Back

The **Agents page** will be displayed listing the configured agents.

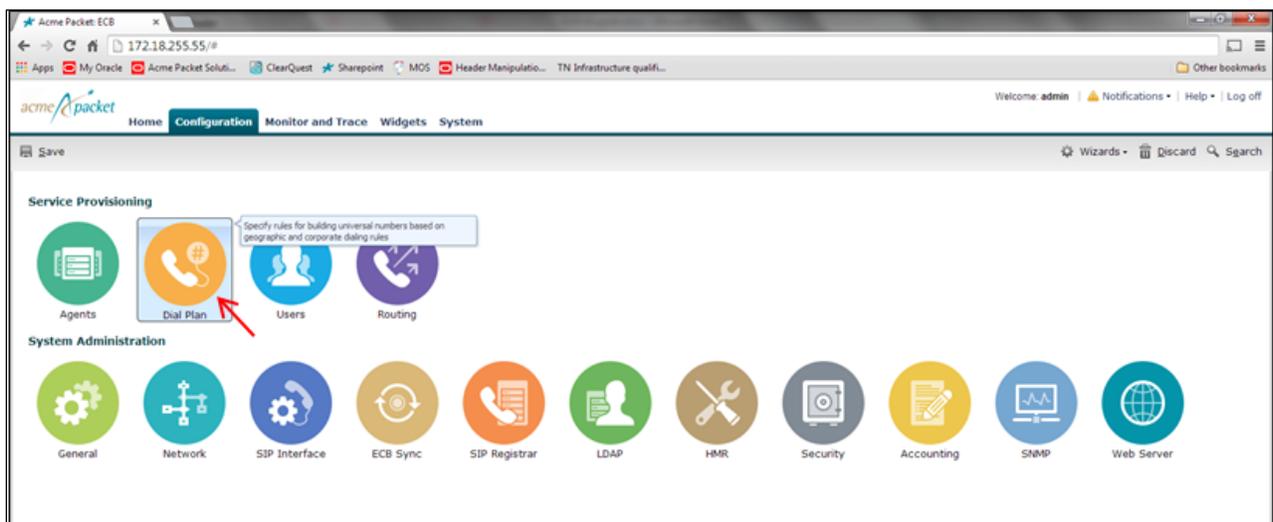
Agents
Search Criteria: All

Add Edit Delete				
Hostname	Address	Port	Transport	Agent state
172.16.101.39	172.16.101.39	5060	StaticTCP	enabled
192.168.1.130	192.168.1.130	5060	StaticTCP	enabled
acmepacket.net		5068	StaticTCP	enabled
aura.com	10.232.50.102	5060	StaticTCP	enabled
lync2013med1.acmepacket.net	192.168.2.192	5068	StaticTCP	enabled
lync2013med2.acmepacket.net	192.168.2.193	5068	StaticTCP	enabled
lyncmedsrv1.selab.com	192.168.1.119	5060	StaticTCP	enabled
lyncmedsrv2.selab.com	192.168.1.120	5060	StaticTCP	enabled

Click on **Configuration** button on the top to go back to the **Configuration** tab.

Configure Dial Plan

We will now configure the dialing contexts and dial plans. Dialing-contexts define the system behavior for calls placed to and from either a corporate or geographic focus. Dialing-contexts include multiple dial-patterns, which define the normalization required to most effectively manage diverse signaling structures. Click on the **Dial Plan** icon under **Service Provisioning**.

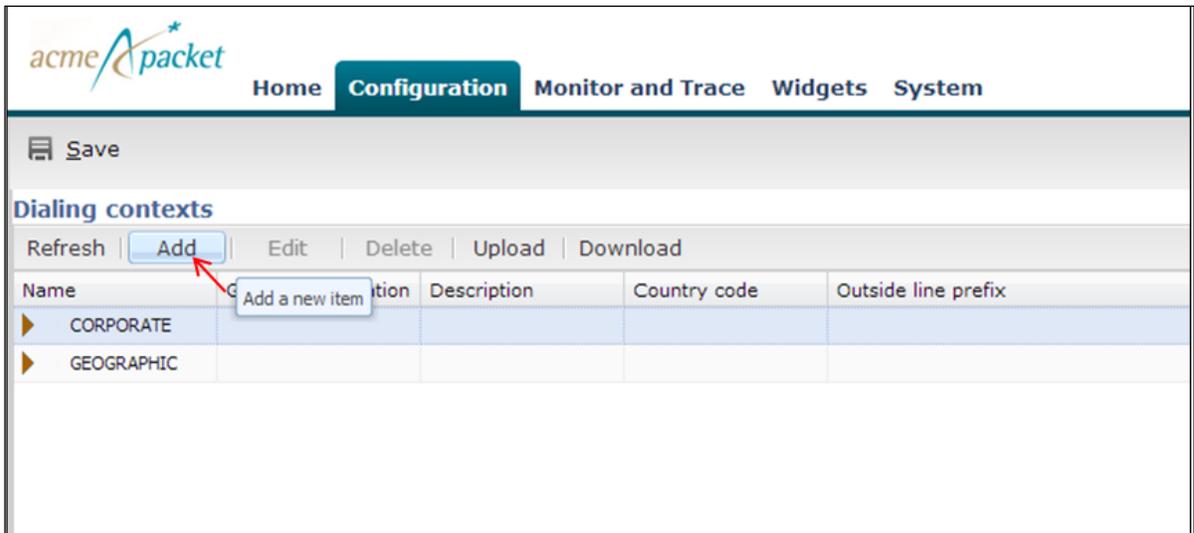


The **Dialing Contexts** page shows the default dialing context parents – Corporate and Geographic.

The screenshot shows the 'Dialing contexts' page in the Acme Packet web interface. The top navigation bar includes 'Home', 'Configuration', 'Monitor and Trace', 'Widgets', and 'System'. The 'Configuration' tab is active. Below the navigation bar, there is a 'Save' button and a search bar. The main content area is titled 'Dialing contexts' and includes a toolbar with 'Refresh', 'Add', 'Edit', 'Delete', 'Upload', and 'Download' buttons. Below the toolbar is a table with the following data:

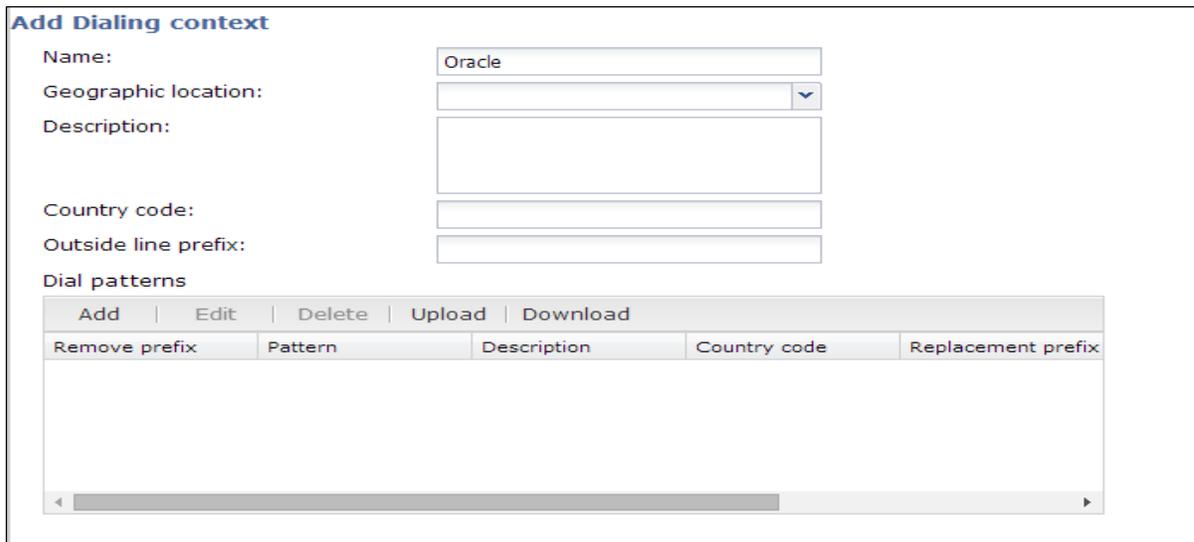
Name	Geographic location	Description	Country code	Outside line prefix
▶ CORPORATE				
▶ GEOGRAPHIC				

To configure a dialing context, select the Corporate context and click **Add**.



The screenshot shows the 'acmeApocket' web interface. The navigation menu includes 'Home', 'Configuration', 'Monitor and Trace', 'Widgets', and 'System'. The 'Configuration' tab is active. Below the navigation is a 'Save' button. The main section is titled 'Dialing contexts' and contains a toolbar with 'Refresh', 'Add', 'Edit', 'Delete', 'Upload', and 'Download'. The 'Add' button is highlighted with a red arrow and a tooltip that says 'Add a new item'. Below the toolbar is a table with the following columns: Name, Geographic location, Description, Country code, and Outside line prefix. The table contains two rows: 'CORPORATE' and 'GEOGRAPHIC', both with expandable arrows on the left.

In the **Add Dialing Context** page, configure a context with the following details and click **OK**.



The screenshot shows the 'Add Dialing context' form. The 'Name' field is filled with 'Oracle'. The 'Geographic location' is a dropdown menu. The 'Description', 'Country code', and 'Outside line prefix' fields are empty. Below the form is a 'Dial patterns' table with the following columns: Remove prefix, Pattern, Description, Country code, and Replacement prefix. The table is currently empty and has a scrollbar at the bottom.

The Dialing Contexts page displays Oracle listed under corporate contexts. We will now configure child contexts under Oracle for our Lync, Avaya and CUCM servers. These can be considered as contexts for the different branches an enterprise has.

Select Oracle under the corporate context and click **Add**.

Dialing contexts

Refresh | **Add** | Edit | Delete | Upload | Download

Name	Geographic location	Description	Country code	Outside line prefix
▲ CORPORATE				
▶ Oracle				
▶ GEOGRAPHIC				

In the **Add Dialing Context** window, configure a context named Bedford-Lync and **Geographic location** as NA. To configure dial patterns, click **Add**.

Add Dialing context

Name:

Geographic location:

Description:

Country code:

Outside line prefix:

Dial patterns

Add | Edit | Delete | Upload | Download

Remove prefix	Add a new item	Description	Country code	Replacement prefix
---------------	----------------	-------------	--------------	--------------------

Add a dial pattern as shown below to enable 4 digit dialing and click **OK**. If the dialed digits match the pattern 72XX, ECB transforms it to a 10 digit number by adding the prefix 781443.

Add Dialing context / dial pattern

Remove prefix:	<input type="text"/>
Pattern:	<input type="text" value="72XX"/>
Description:	<input type="text"/>
Country code:	<input type="text"/>
Replacement prefix:	<input type="text" value="781443"/>
Replacement uri:	<input type="text"/>
Go to context:	<input type="text"/> ▼

Add another dial pattern as shown below to match the pattern 73XX and click **OK**.

Add Dialing context / dial pattern

Remove prefix:	<input type="text"/>
Pattern:	<input type="text" value="73XX"/>
Description:	<input type="text"/>
Country code:	<input type="text"/>
Replacement prefix:	<input type="text" value="781443"/>
Replacement uri:	<input type="text"/>
Go to context:	<input type="text"/> ▼

The Bedford-Lync dialing context displays the configured dial patterns.

Modify Dialing context

Name:

Geographic location:

Description:

Country code:

Outside line prefix:

Dial patterns

Add	Edit	Delete	Upload	Download
Remove prefix	Pattern	Description	Country code	Replacement prefix
	72XX			781443
	73XX			781443

Add another dialing context under Oracle named Burlington-Avaya with the following settings and click **OK**.

Modify Dialing context

Name:

Geographic location:

Description:

Country code:

Outside line prefix:

Dial patterns

Add	Edit	Delete	Upload	Download
Remove prefix	Pattern	Description	Country code	Replacement prefix
	72XX			781443
	73XX			781443

Add a dialing context named Braintree-CUCM with the following settings and click **OK**.

Modify Dialing context

Name:

Geographic location:

Description:

Country code:

Outside line prefix:

Dial patterns

Add Edit Delete Upload Download				
Remove prefix	Pattern	Description	Country code	Replacement prefix
	73XX			781443
	72XX			781443

The **Dialing Contexts** page shows the parent context – Oracle and the child contexts.

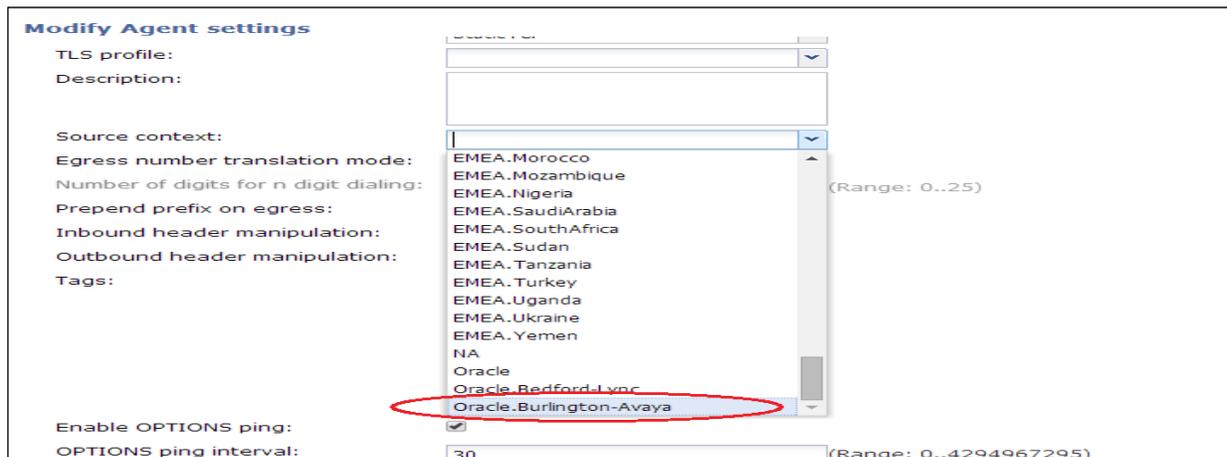
Dialing contexts

Refresh | Add | Edit | Delete | Upload | Download

Name	Geographic location	Description	Country code	Outside line prefix
▲ CORPORATE				
▲ Oracle				
▶ Bedford-Lync	NA			
▶ Braintree-CUCM	NA			
▶ Burlington-Avaya	NA			
▶ GEOGRAPHIC				

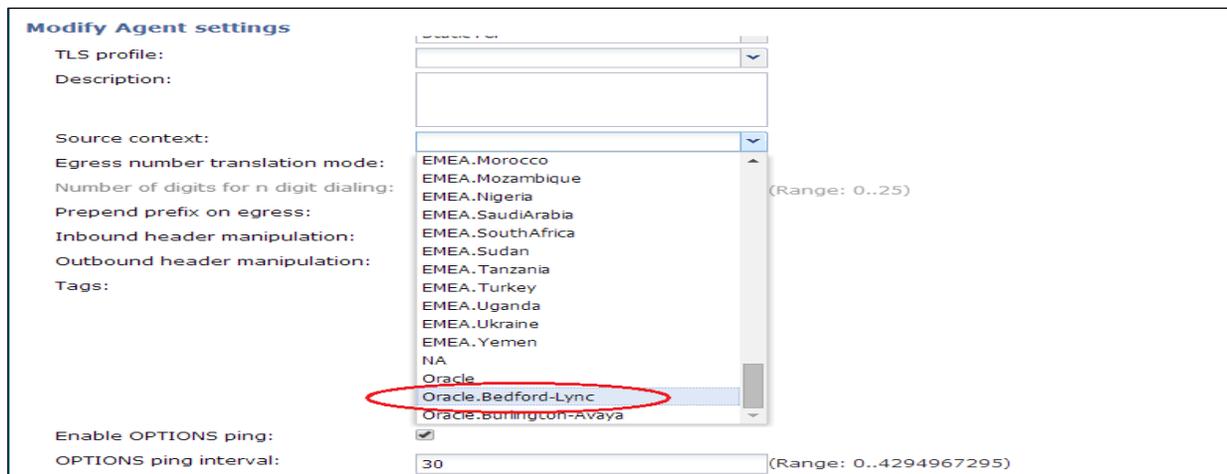
We need to associate the agents with the contexts within which they reside to assign the dialing rules. Click on the **Configuration** button at the top to go to the **Configuration** tab and click on **Agents**.

On the **Agents** page, select the agent configured for Avaya server and click **Edit**. On the **Modify Agent Settings** page, select Oracle.Burlington-Avaya from **Source context** drop-down menu and click **OK**.



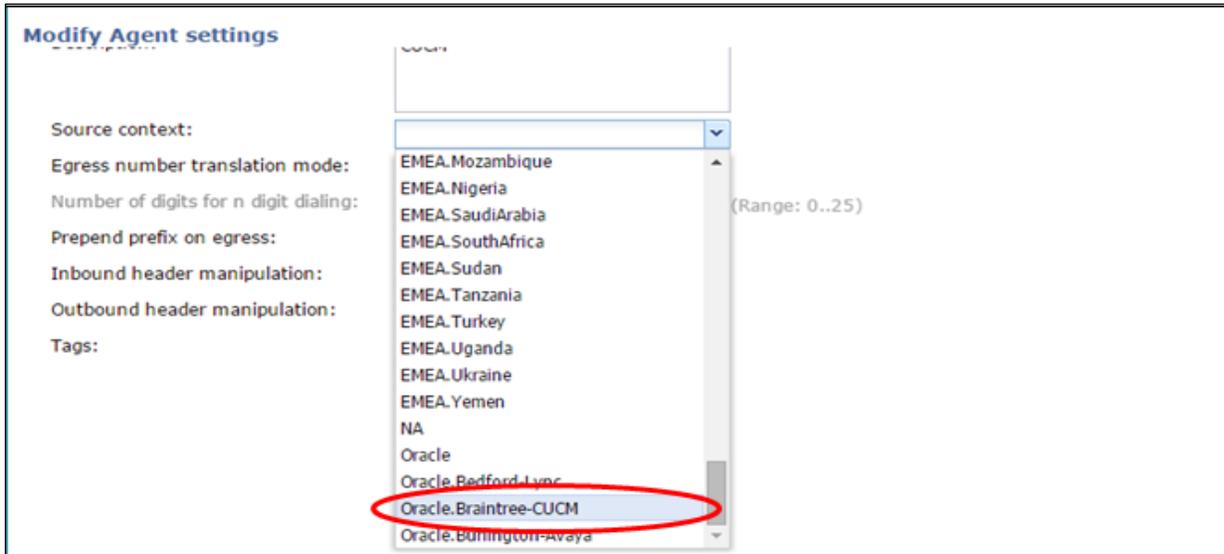
The screenshot shows the 'Modify Agent settings' page. The 'Source context' dropdown menu is open, displaying a list of contexts. 'Oracle.Burlington-Avaya' is highlighted with a red oval. Other contexts in the list include EMEA.Morocco, EMEA.Mozambique, EMEA.Nigeria, EMEA.SaudiArabia, EMEA.SouthAfrica, EMEA.Sudan, EMEA.Tanzania, EMEA.Turkey, EMEA.Uganda, EMEA.Ukraine, EMEA.Yemen, NA, Oracle, Oracle.Bedford-Lync, and Oracle.Burlington-Avaya. The 'Enable OPTIONS ping' checkbox is checked, and the 'OPTIONS ping interval' is set to 30.

Next, configure the Lync 2013/2010 mediation servers with the source context – Oracle.Bedford-Lync since the Lync 2010 and Lync 2013 servers have the same dialing rules.



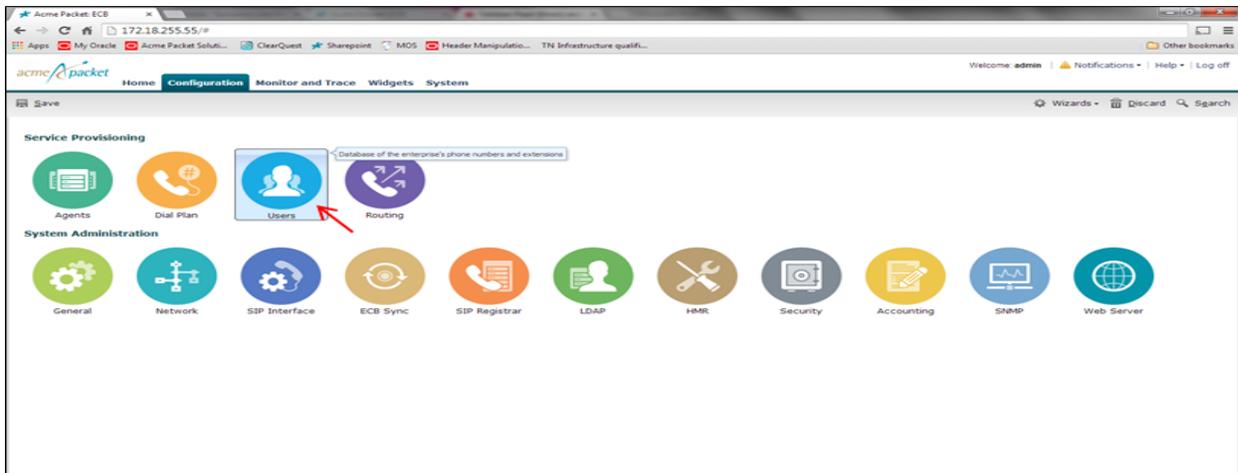
The screenshot shows the 'Modify Agent settings' page. The 'Source context' dropdown menu is open, displaying a list of contexts. 'Oracle.Bedford-Lync' is highlighted with a red oval. Other contexts in the list include EMEA.Morocco, EMEA.Mozambique, EMEA.Nigeria, EMEA.SaudiArabia, EMEA.SouthAfrica, EMEA.Sudan, EMEA.Tanzania, EMEA.Turkey, EMEA.Uganda, EMEA.Ukraine, EMEA.Yemen, NA, Oracle, Oracle.Bedford-Lync, and Oracle.Burlington-Avaya. The 'Enable OPTIONS ping' checkbox is checked, and the 'OPTIONS ping interval' is set to 30.

Finally, configure the CUCM server with the source context – Oracle.Braintree-CUCM.

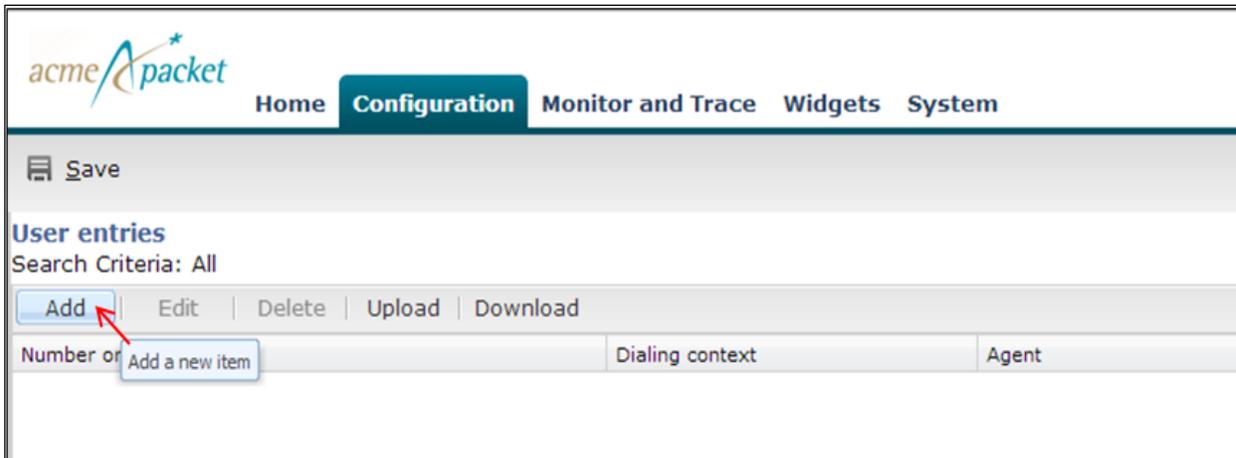


Configure Users

Next we will populate users in the User database. User entries can be added manually or uploaded in a format pre-configured to translate into a user database. Click on the **Users** icon under **Service Provisioning**.



The **User entries** page will be displayed. Click on **Add** to start adding users.



acme packet

Home Configuration Monitor and Trace Widgets System

Save

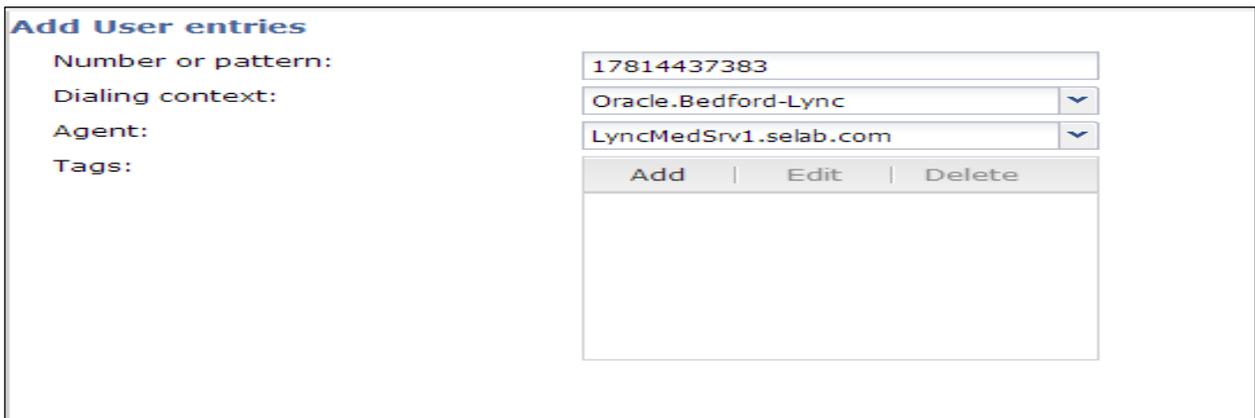
User entries

Search Criteria: All

Add Edit Delete Upload Download

Number of Add a new item Dialing context Agent

The **Add User entries** page will be displayed. You can enter the user numbers in E164 format without the + (17814437383) or a number range (17814437[400-599]) in the **Number** field. Assign the appropriate **Agent** and **Dialing context** and click **OK**.



Add User entries

Number or pattern: 17814437383

Dialing context: Oracle.Bedford-Lync

Agent: LyncMedSrv1.selab.com

Tags:

Add Edit Delete

Continue adding users as shown above using the corresponding agents and dialing contexts.

The User entries page will list all the users configured. Click **Configuration** button on the top to go to the **Configuration** tab

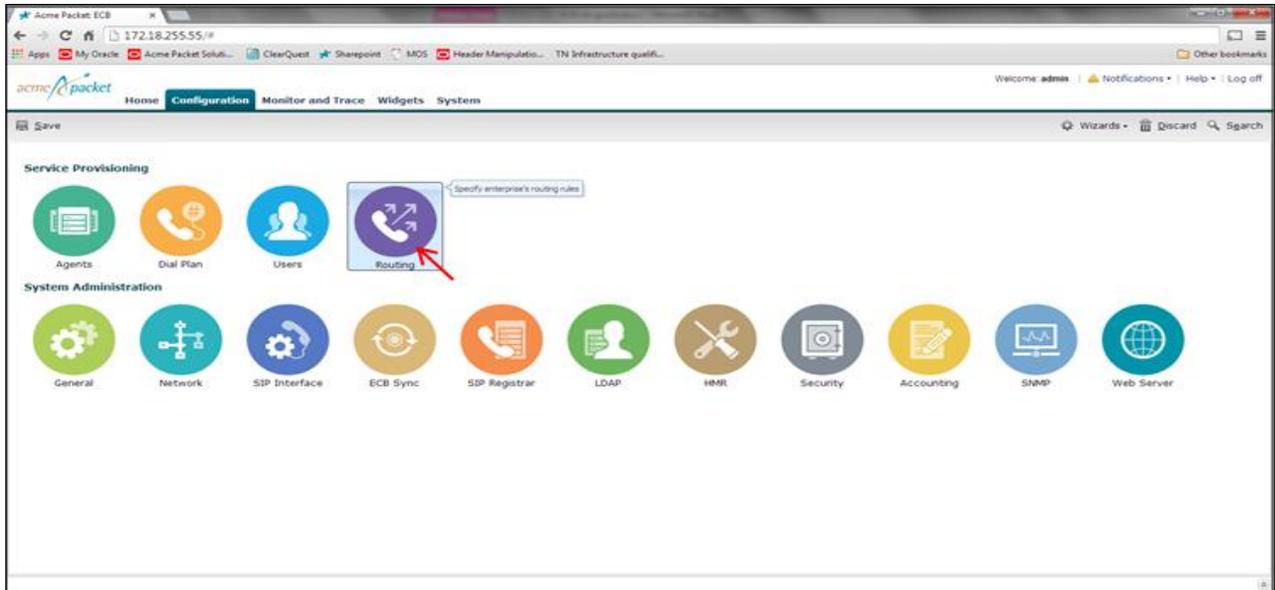
User entries				
Search Criteria: All				
Add	Edit	Delete	Upload	Download
Number or pattern	Dialing context	Agent		
17814437246	Oracle.Burlington-Avaya	aura.com		
17814437247	Oracle.Burlington-Avaya	aura.com		
17814437293	Oracle.Braintree-CUCM	172.16.101.39		
17814437295	Oracle.Braintree-CUCM	172.16.101.39		
17814437383	Oracle.Bedford-Lync	lynccmetsrv1.selab.com		
17814437387	Oracle.Bedford-Lync	lync2013med1.acmepacket.net		
17814437388	Oracle.Bedford-Lync	lynccmetsrv1.selab.com		

Configure Routing

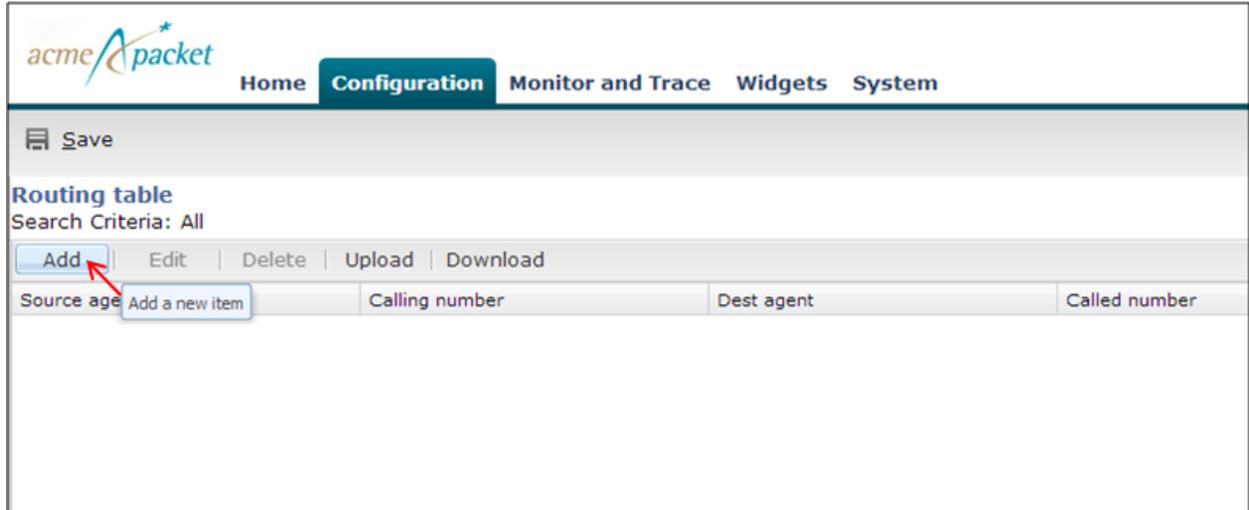
The ECB performs its session routing via the route configuration. Route configuration establishes hop-by-hop paths to signaling endpoints. ECB routing configuration allows the user to specify a route's cost to specify route preference. Cost may or may not be based on monetary considerations. But the reach of an enterprise's network often does allow the user to configure routes that keep session traffic within the enterprise infrastructure rather than incurring cost associated with a service provider.

The ECB allows for a range of route preference criteria to differentiate between routing paths. Criteria include source routing based on the agent or calling number. Target-oriented criteria are also available, allowing the enterprise to designate preferred paths for specific called numbers.

We need not configure a route for the users defined in the user database as the ECB will use their configured agents as next hop to route the calls. Since ECB does not support DNS load balancing as of now, the Lync users are assigned with one mediation server as their agent. To ensure the calls complete if the first mediation server in the pool goes down, we will configure a route to the second agent of the pool with a higher cost. On the **Configuration** tab click on the **Routing** icon under **Service Provisioning**.



On the **Routing table** page, click **Add** to add a route.



Add a routing entry for the Lync 2010 user – 17814437383 with the **Route** set to the second mediation server – LyncMedSrv2.selab.com with a cost of 20 and click **OK**.

Add Routing table

Source agent:	*	▼
Calling number:	*	
Dest agent:	*	▼
Called number:	17814437383	
Route:	LyncMedSrv2.selab.com ▼	
Cost:	20	(Range: 0..100)
Description:	Fail over route to Lync 2010 Mediation Server 2	
Tags:	Add Edit Delete	

When the ECB receives a call for 1781437383, it looks up the user DB and finds that this user is associated to LyncMedSrv1.selab.com and routes the call to it. If this agent is down, ECB will find the above entry and route the call to the second agent of the pool – LyncMedSrv2.selab.com.

Similarly add a routing entry for user 17814437387 pointing to lyncmed2.acmepacket.net as its failover route.

Add Routing table

Source agent:	*	▼
Calling number:	*	
Dest agent:	*	▼
Called number:	17814437387	
Route:	lync2013med2.acmepacket.net ▼	
Cost:	20	(Range: 0..100)
Description:	Fail over route to Lync 2013 Mediation Server	
Tags:	Add Edit Delete	

For calls being made from the Enterprise to the outside world, we will define routes with the SBC (which connects to the SIP trunk) as the next hop.

In the **Add Routing table** page, add a route as shown below, routing all calls from the source agent - Lync mediation server – LyncMed1Srv.selab.com to the trunk SBC with **cost 10** and click **OK**.

Add Routing table

Source agent:

Calling number:

Dest agent:

Called number:

Route:

Cost: (Range: 0..100)

Description:

Tags: | |

The cost for these route needs to be higher than 0 so that the ECB does not route the calls for the configured users (like Avaya/Lync users) to the Trunk SBC.

Add similar entries with remaining servers as source agents as shown below.

Routing table						
Search Criteria: All						
<input type="button" value="Add"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/> <input type="button" value="Upload"/> <input type="button" value="Download"/>						
Source agent	Calling number	Dest agent	Called number	Route	Cost	Description
*	*	*	17814437383	lynccmedsrv2.selab.com	20	Fail over route to Lync 2010 Mediation Server 2
*	*	*	17814437387	lyncc2013med2.acmepacket.net	20	Fail over route to Lync 2013 Mediation Server
172.16.101.39	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 1 to Trunk SBC
acmepacket.net	*	*	*	192.168.1.130	10	Route for Transfer INVITEs from Lync 2013 server
aura.com	*	*	*	192.168.1.130	10	Route from Avaya Server to Trunk SBC
lyncc2013med1.acmepacket.net	*	*	*	192.168.1.130	10	Route from Lync 2013 Med Server 1 to Trunk SBC
lyncc2013med2.acmepacket.net	*	*	*	192.168.1.130	10	Route from Lync 2013 Med Server 2 to Trunk SBC
lynccmedsrv1.selab.com	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 1 to Trunk SBC
lynccmedsrv2.selab.com	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 2 to Trunk SBC

To route the INVITEs in case of transfers from Lync 2013, add a route as shown below routing from source agent – acmepacket.net to the trunk SBC and click **OK**.

Add Routing table				
Source agent:	acmepacket.net			
Calling number:	*			
Dest agent:	*			
Called number:	*			
Route:	192.168.1.130			
Cost:	10 (Range: 0..100)			
Description:	Route for Transfer INVITEs from Lync 2013 server			
Tags:	<table border="1"><tr><td>Add</td><td>Edit</td><td>Delete</td></tr></table>	Add	Edit	Delete
Add	Edit	Delete		

The **Routing Table** page will be displayed listing all the routes added. When you select a specific route, its **Route tree** is displayed at the bottom.

Routing table
Search Criteria: All

Add | Edit | Delete | Upload | Download

Search Search Clear

Source agent	Calling number	Dest agent	Called number	Route	Cost	Description
*	*	*	17814437383	lynxedsrv2.selab.com	20	Fail over route to Lync 2010 Mediation Server 2
*	*	*	17814437387	lync2013med2.acmepacket.net	20	Fail over route to Lync 2013 Mediation Server
172.16.101.39	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 1 to Trunk SBC
acmepacket.net	*	*	*	192.168.1.130	10	Route for Transfer INVITEs from Lync 2013 server
aura.com	*	*	*	192.168.1.130	10	Route from Avaya Server to Trunk SBC
lync2013med1.acmepacket.net	*	*	*	192.168.1.130	10	Route from Lync 2013 Med Server 1 to Trunk SBC
lync2013med2.acmepacket.net	*	*	*	192.168.1.130	10	Route from Lync 2013 Med Server 2 to Trunk SBC
lynxedsrv1.selab.com	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 1 to Trunk SBC
lynxedsrv2.selab.com	*	*	*	192.168.1.130	10	Route from Lync 2010 Med Server 2 to Trunk SBC

Displaying 1 - 9 of 9

[Back](#)

Route tree

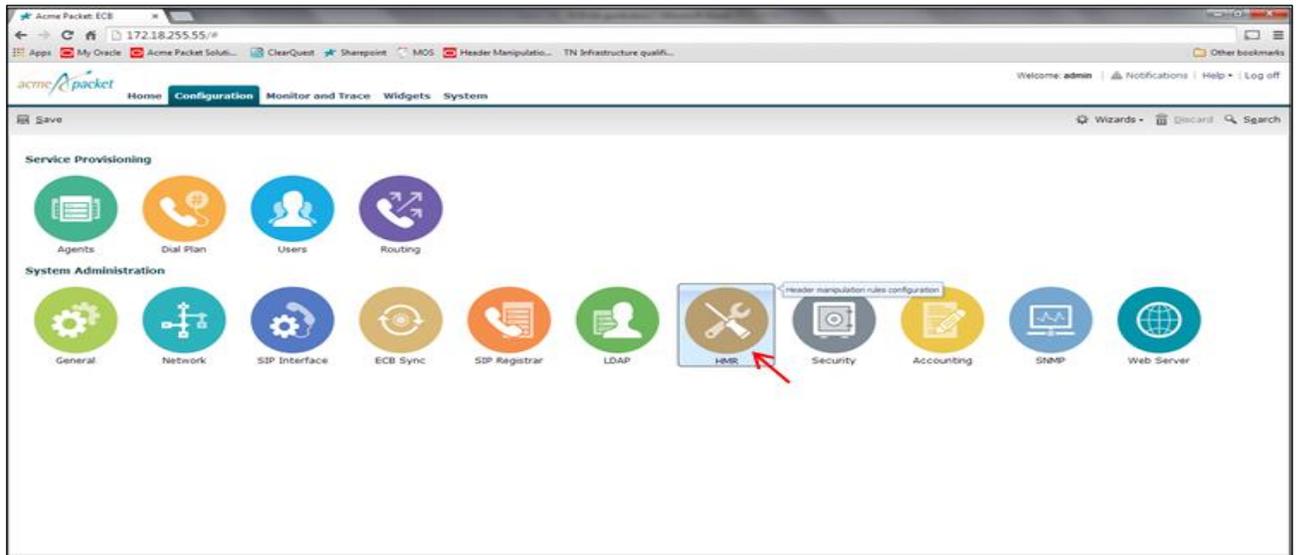
Cost	Hops
10	cost: 10 calling agent: aura.com → 192.168.1.130
20	cost: 20 called number: 17814437387 → lync2013med2.acmepacket.net
20	cost: 20 called number: 17814437383 → lynxedsrv2.selab.com

Click on **Configuration** button on the top to go to the **Configuration** tab.

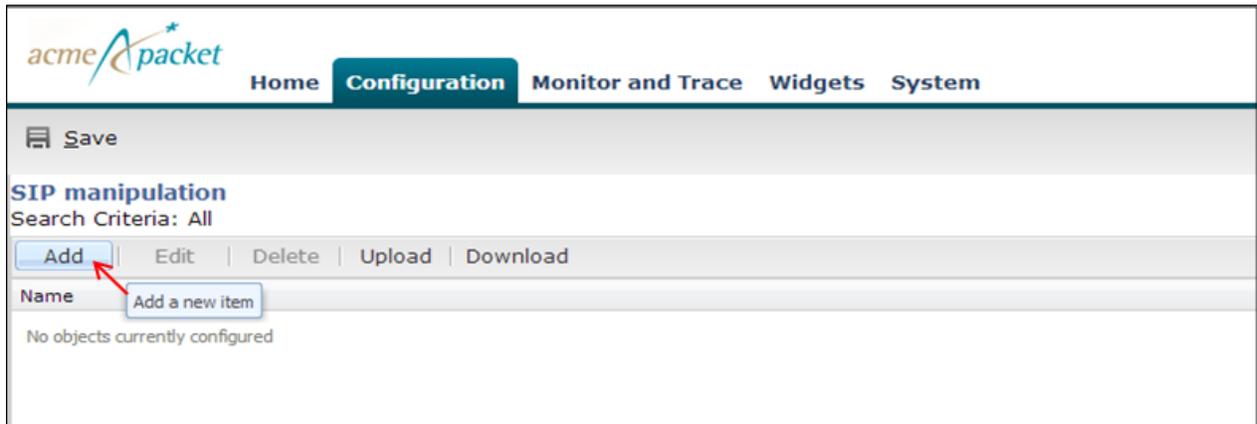
Configure Header manipulation rules

We will now configure header manipulation rules to hide network topology and ensure that the SIP messages sent to all agents cater to their specific signaling standards.

Click on the **HMR** icon under System **Administration** on the **Configuration** tab.



The **SIP manipulation** page is displayed. We need to configure a sip-manipulation to replace the ip-addresses in the From and To headers to hide the network topology. Click Add to add a sip manipulation for this purpose.



In the **Add SIP manipulation** page, enter a name and description for the manipulation. In our case, it is called NATting. To add a header-rule, select **header-rule** from the **Add** drop down menu under the **CfgRules** section.

Add SIP manipulation

Name:

Description:

Split headers:

| |

Join headers:

| |

CfgRules

| | | |

Element type
header-rule
mime-rule
mime-isup-rule

In the **Add SIP manipulation / header rule** page, add a header-rule From as shown below to manipulate the From header. To configure an element-rule for this From Header-rule, select **element-rule** from the **Add** drop down menu under the **CfgRules** section.

Add SIP manipulation / header rule

Name:

Header name:

Action:

Comparison type:

Msg type:

Methods:

| |

Match value:

New value:

CfgRules

| | | |

Element type
element-rule

Add an element-rule named From_header to replace the uri-host with ECB's local ip-address (192.168.1.90) as shown below and click **OK**.

Add SIP manipulation / header rule / element rule	
Name:	<input type="text" value="From_header"/>
Parameter name:	<input type="text"/>
Type:	<input type="text" value="uri-host"/> ▼
Action:	<input type="text" value="replace"/> ▼
Match val type:	<input type="text" value="any"/> ▼
Comparison type:	<input type="text" value="case-sensitive"/> ▼
Match value:	<input type="text"/>
New value:	<input type="text" value="\$LOCAL_IP"/>

The From header-rule page is displayed. Click **OK** and the NATting sip-manipulation page is displayed. Following the steps explained above configure a header-rule to manipulate the To header as shown below.

```

header-rule
  name To
  header-name To
  action manipulate
  comparison-type case-sensitive
  msg-type request
  methods
  match-value
  new-value
  element-rule
    name To
    parameter-name
    type uri-host
    action replace
    match-val-type any
    comparison-type case-sensitive
    match-value
    new-value $REMOTE_IP

```

This completes the configuration of the sip-manipulation for topology hiding.

Apply this sip manipulation as an outbound manipulation to the Trunk SBC agent and click **OK**.

Modify Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol:

TLS profile:

Description:

Source context:

Egress number translation mode:

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation:

Outbound header manipulation:

Tags: | |

Next we will configure the manipulations required for the Avaya server. The Avaya setup in our lab uses a FQDN of aura.com. The uri-host portions of the Request-Uri, From and To headers in the SIP messages sent to the Avaya server need to be changed to aura.com. If the INVITE contains a PAI header, we will need to change the uri-host to aura.com. Configure the following manipulation to change the uri-host portion to aura.com. This will be applied as an outbound manipulation to the Avaya server.

```

sip-manipulation
  name NATtingavaya
  description
  split-headers
  join-headers
  header-rule
    name
    header-name From
    action manipulate
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value
    element-rule

```

```

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

```

```

action                manipulate
comparison-type      case-sensitive
msg-type              any
methods
match-value
new-value
element-rule
  name                Pai_header
  parameter-name
  type                uri-host
  action              replace
  match-val-type      any
  comparison-type     case-sensitive
  match-value
  new-value           aura.com

```

The Request-Uri of the INVITEs sent from the Avaya session manager contains the uri-host as aura.com. This causes issues with the routing in ECB when the dialed numbers are not configured as users in userDB. To resolve this issue, we configure a manipulation to replace the aura.com in the RURI with the ip address of the ECB, in our case – 192.168.1.90 and apply in the inbound direction on the Avaya server agent. This HMR is configured as an out-of-dialog manipulation so that it does not affect the INVITEs for hold and transfers.

```

sip-manipulation
  name                ChangeRURIhost
  description
  split-headers
  join-headers
  header-rule
    name              fixRURI
    header-name       Request-URI
    action            manipulate
    comparison-type   case-sensitive
    msg-type          out-of-dialog
    methods           INVITE
    match-value
    new-value
    element-rule
      name            updateRURI
      parameter-name
      type            uri-host
      action          replace
      match-val-type  any
      comparison-type pattern-rule
      match-value     (.*)$
      new-value       192.168.1.90

```

Apply NATtingavaya as an outbound manipulation and ChangeRURIhost as an inbound manipulation to the Avaya server agent and click OK.

Modify Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol:

TLS profile:

Description:

Source context:

Egress number translation mode:

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation:

Outbound header manipulation:

Tags:

We will now configure manipulations to modify the SIP messages being sent to and received from the Lync server. Lync typically sends mediation server FQDN in the Contact header with no username in the SIP URI which is not acceptable by SIP trunk providers. We configure a manipulation to update the Contact header to include the username appropriately and apply it as an inbound manipulation on the Lync server.

The manipulation consists of two header rules – StoreFromnumber and ChangeContact. The StoreFromnumber header rule stores the uri-user-only element in the From header which is then added as the uri-user in the Contact header in the ChangeContact header rule.

```

sip-manipulation
  name                               ChangeContact
  description
  split-headers
  join-headers
  header-rule
    name                               StoreFromnumber
    header-name                         From
    action                               manipulate
    comparison-type                     case-sensitive
    msg-type                             any
    methods
    match-value
    new-value
    element-rule
      name                               StoreFromnumber_er

```

```

parameter-name
type uri-user-only
action store
match-val-type any
comparison-type case-sensitive
match-value
new-value

header-rule
name ChangeContact
header-name Contact
action manipulate
comparison-type case-sensitive
msg-type any
methods
match-value
new-value
element-rule
name ChangeContact_er
parameter-name
type uri-user
action add
match-val-type any
comparison-type case-sensitive
match-value
new-value
$StoreFromnumber.$StoreFromnumber_er.$0

```

Avaya server sends certain b-lines in the SDP which are not supported by Lync server. We configure the following manipulation to delete these lines from SDP before the messages are being sent out to Lync.

```

sip-manipulation
name Delblines
description Deleting b-lines from Avaya
split-headers
join-headers
header-rule
name manipContentType
header-name Content-Type
action manipulate
comparison-type pattern-rule
msg-type any
methods
match-value
new-value
element-rule
name deleteB
parameter-name application/sdp

```

```

        type mime
        action find-replace-all
        match-val-type any
        comparison-type pattern-rule
        match-value b=CT:.*(\n|\r\n)
        new-value
    element-rule
        name deleteLABEL
        parameter-name application/sdp
        type mime
        action find-replace-all
        match-val-type any
        comparison-type pattern-rule
        match-value b=AS:.*(\n|\r\n)
        new-value
    element-rule
        name deleteLABEL1
        parameter-name application/sdp
        type mime
        action find-replace-all
        match-val-type any
        comparison-type pattern-rule
        match-value b=TIAS:.*(\n|\r\n)
        new-value

```

A nested manipulation named HMRtowardsLync is configured to include the manipulations –NATting and Delblines and applied in the outbound direction to the Lync server.

```

sip-manipulation
  name HMRtowardsLync
  description HMR NAT+deleting the blines
  split-headers
  join-headers
  header-rule
    name doNAT
    header-name From
    action sip-manip
    comparison-type case-sensitive
    msg-type any
    methods
    match-value
    new-value NATting
  header-rule
    name deleteblines
    header-name From
    action sip-manip
    comparison-type case-sensitive

```

msg-type	any
methods	
match-value	
new-value	Delblines

Apply ChangeContact as in inbound manipulation and HMRtowardsLync as an outbound manipulation to all the Lync servers configured as agents.

Modify Agent settings

State:

Hostname:

IP address:

IP port: (Range: 0..65535)

Transport protocol: ▼

TLS profile: ▼

Description:

Source context: ▼

Egress number translation mode: ▼

Number of digits for n digit dialing: (Range: 0..25)

Prepend prefix on egress:

Inbound header manipulation: ▼

Outbound header manipulation: ▼

Tags: | |

Next apply NATting as an outbound manipulation to the CUCM server.

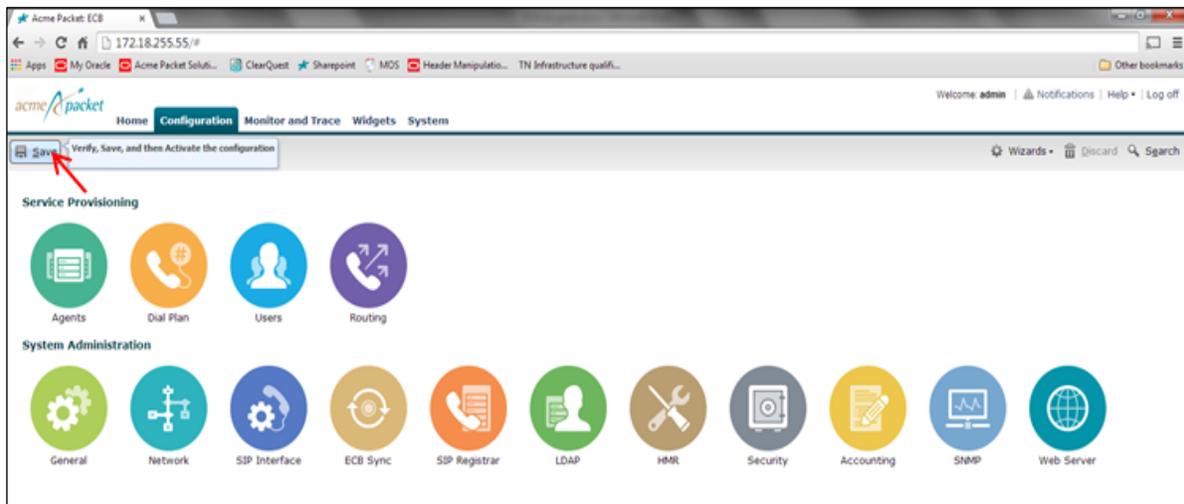
Modify Agent settings

State:	<input checked="" type="checkbox"/>
Hostname:	172.16.101.39
IP address:	172.16.101.39
IP port:	5060 (Range: 0..65535)
Transport protocol:	StaticTCP
TLS profile:	
Description:	CUCM
Source context:	Oracle.Braintree-CUCM
Egress number translation mode:	E164-no-plus
Number of digits for n digit dialing:	4 (Range: 0..25)
Prepend prefix on egress:	
Inbound header manipulation:	
Outbound header manipulation:	NATting
Tags:	<input type="button" value="Add"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/>

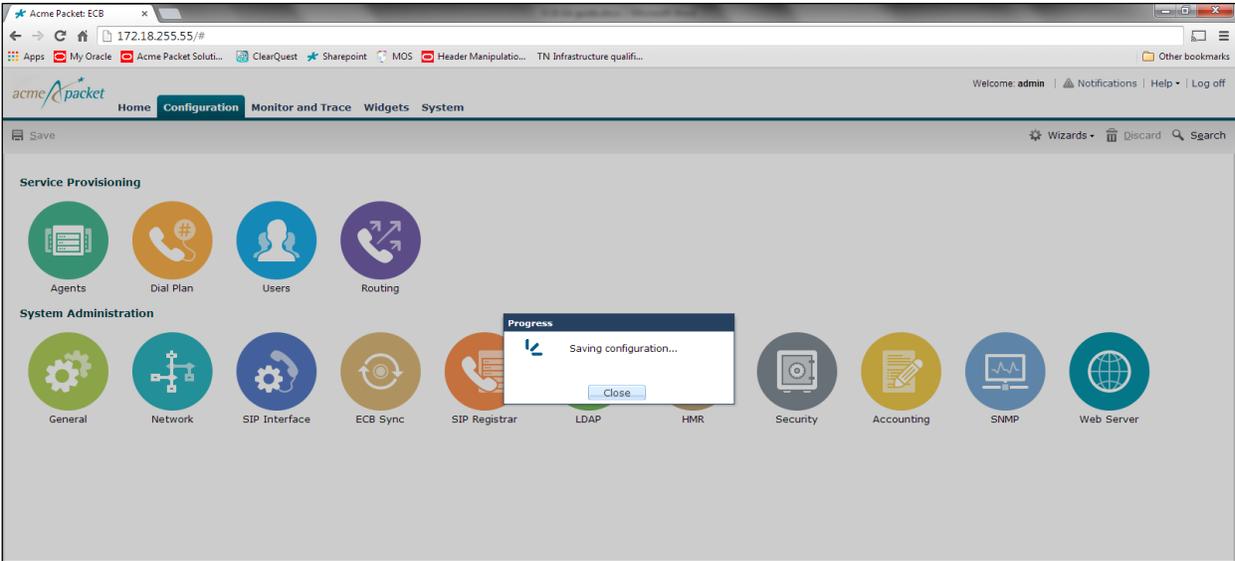
Click **Configuration** on the top to go back to the **Configuration** tab.

Save and activate the configuration

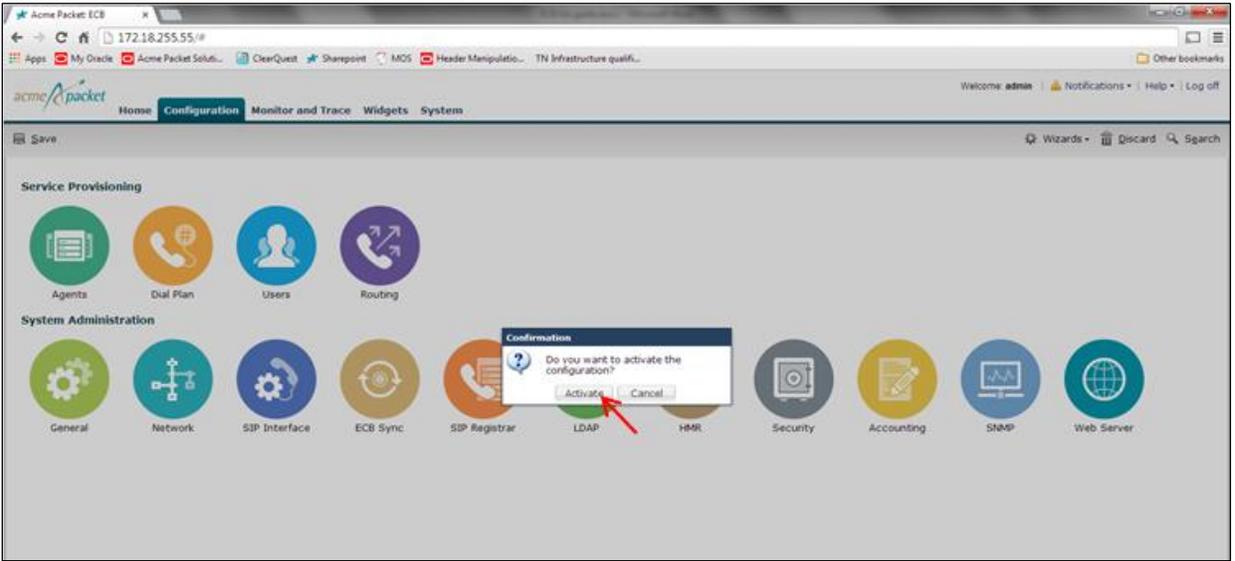
We will now save and activate our ECB configuration. Click **Save** on the top left hand side of the **Configuration** tab.



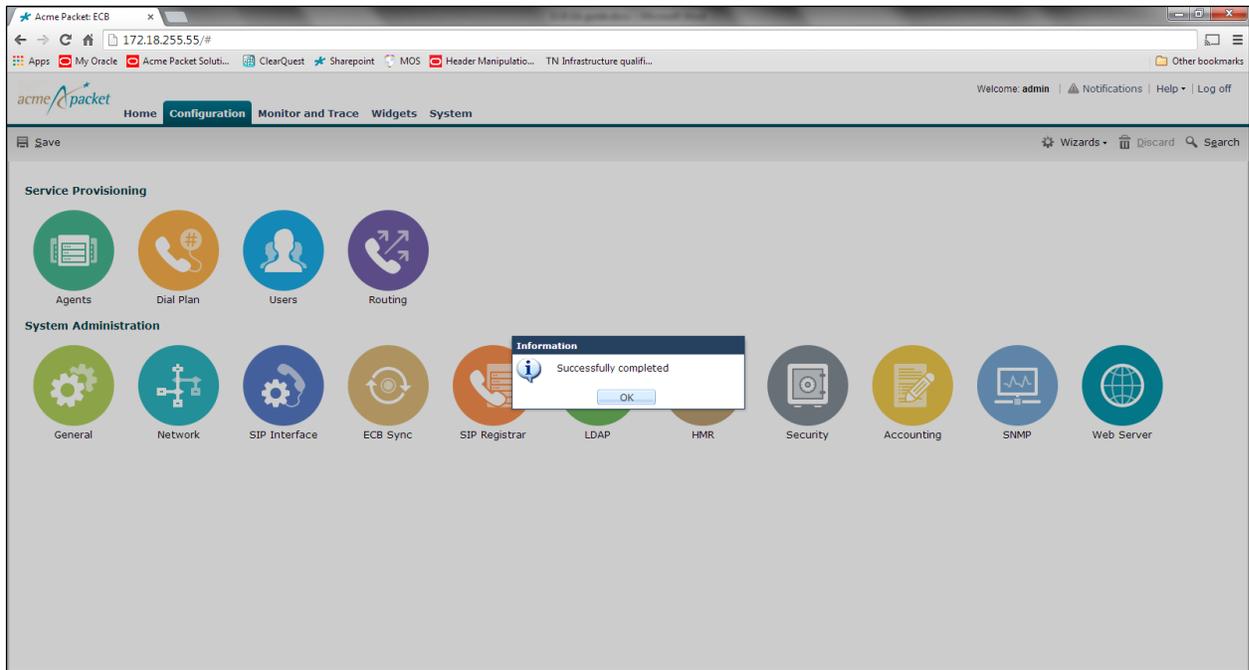
A progress dialog box will appear showing that the configuration is being saved.



You will be asked to confirm if you would like to activate the configuration. Click **Activate**.



After the activation is completed, you will see the screen below



Click OK and the ECB configuration is now complete.

Phase 2 – Configuring the Lync 2013 server

The enterprise will have a fully functioning Lync Server infrastructure with Enterprise Voice deployed and a Mediation Server dedicated to this installation. If there is no Mediation Server present for this purpose, one will have to be deployed.

There are two parts for configuring Lync Server to operate with the Oracle ECB:

- Adding the ECB as a PSTN gateway to the Lync Server infrastructure
- Creating a route within the Lync Server infrastructure to utilize the SIP trunk connected through the ECB.

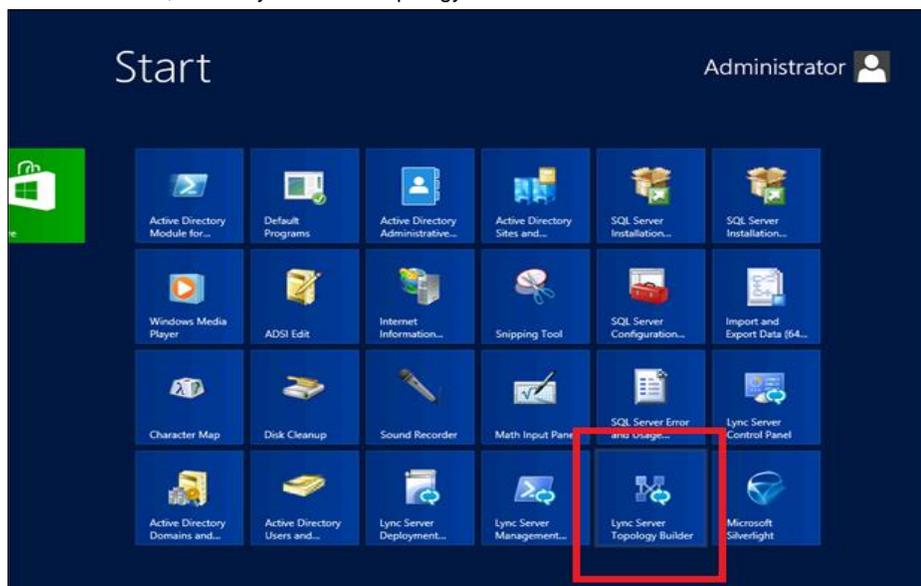
To add the PSTN gateway, we will need:

- IP addresses of the external facing NICs of the Mediation Servers
- IP address of the sip interface of the ECB
- Rights to administer Lync Server Topology Builder
- Access to the Lync Server Topology Builder

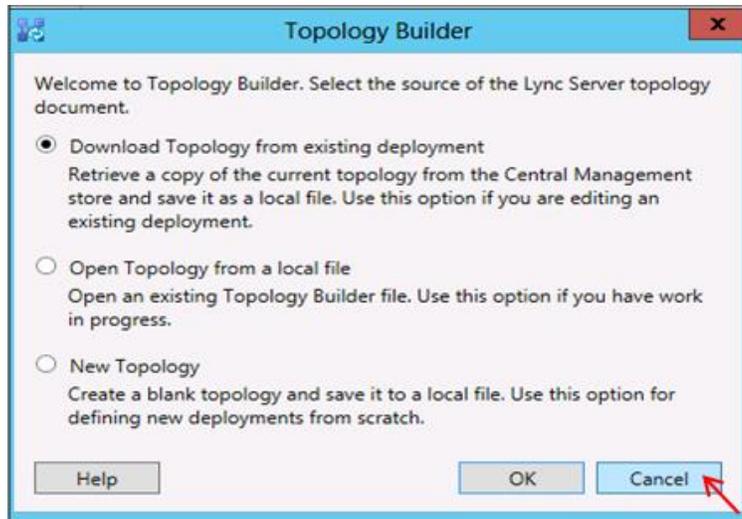
Adding the ECB as a PSTN gateway

The following process details the steps to add the ECB as the PSTN gateway

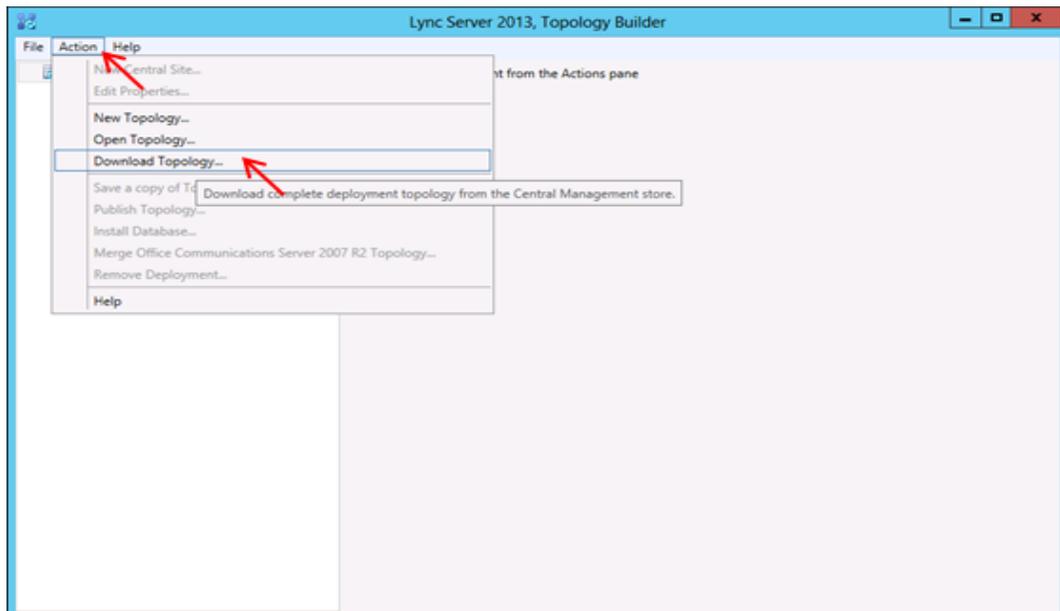
1. On the server where the Topology Builder is located start the console.
2. From the Start bar, select Lync Server Topology Builder.



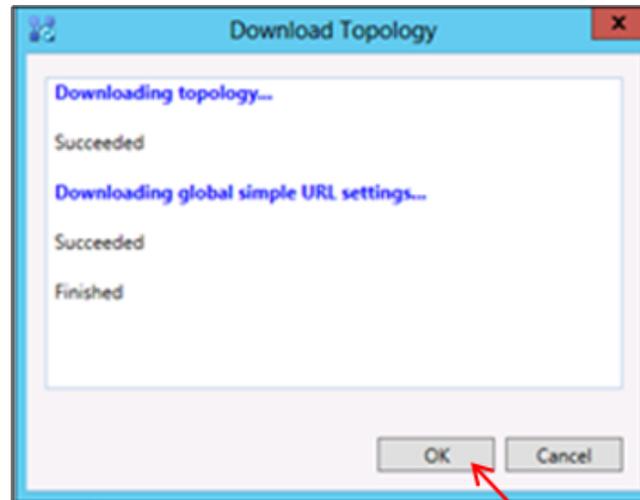
3. The opening screen of the Topology builder will be displayed. Click on the **Cancel** button.



4. The Topology Builder window will now be displayed. Click on **Action** and select **Download Topology**.

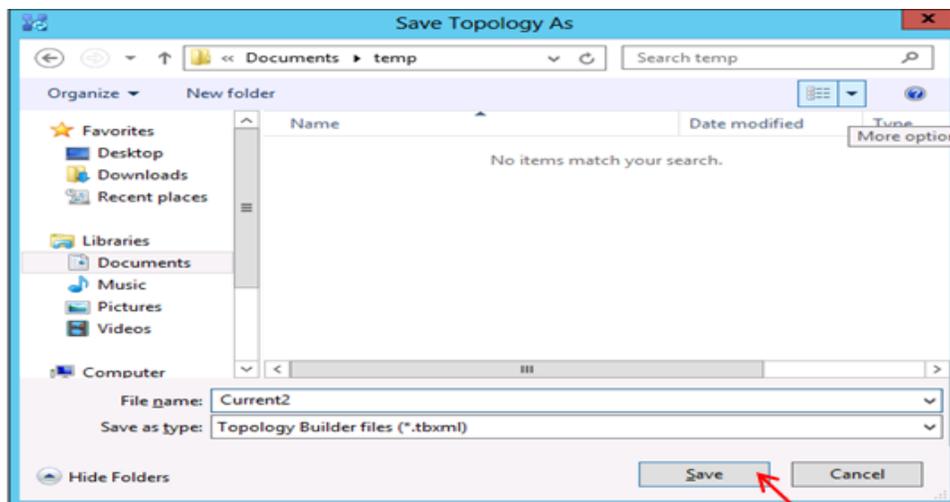


5. You will then see a screen showing that you have successfully imported the topology. Click the **OK** button.

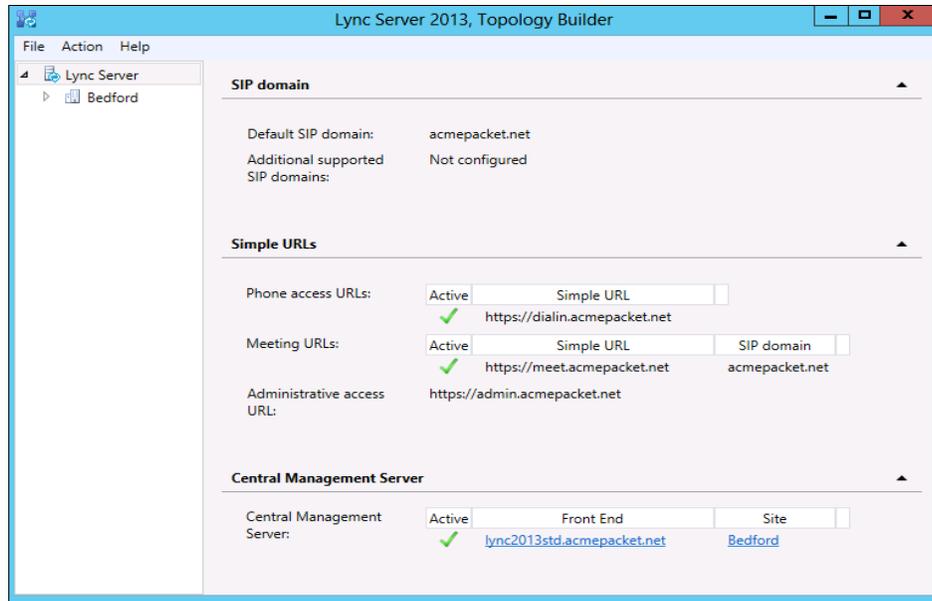


6. Next you will be prompted to save the topology which you have imported. You should revision the name or number of the topology according to the standards used within the enterprise. Click the **Save** button

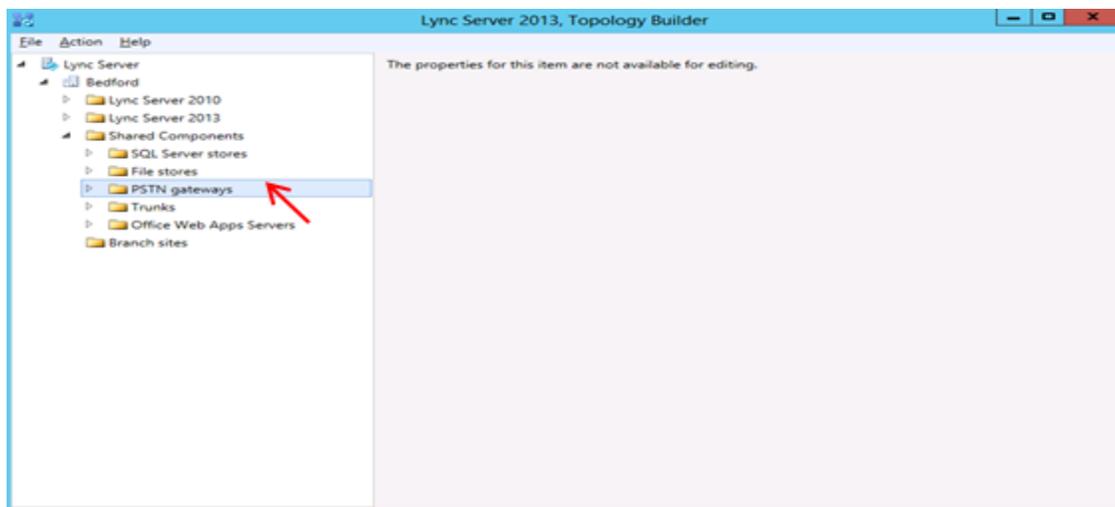
Note: This keeps track of topology changes and, if desired, will allow you to fall back from any changes you make during this installation



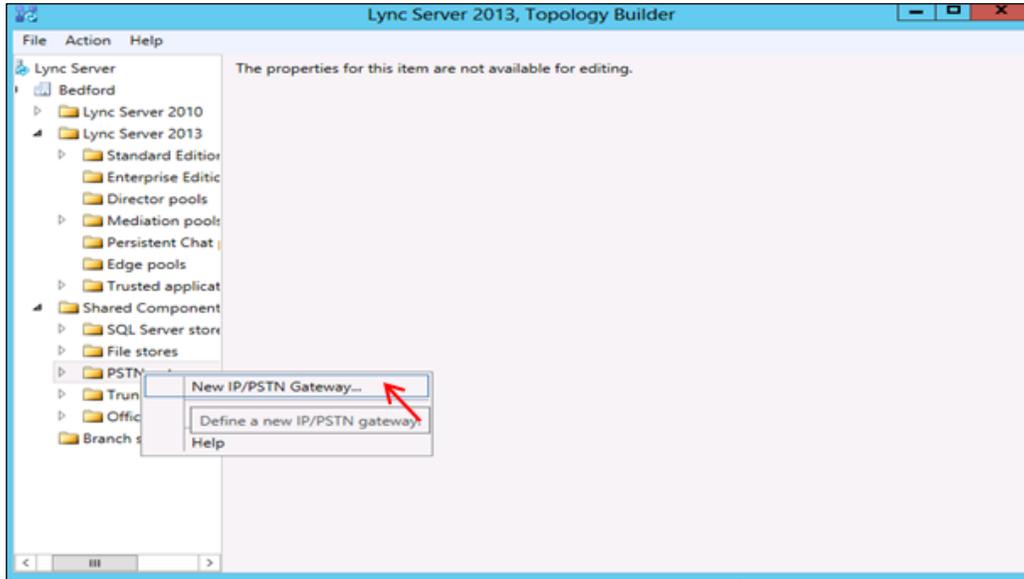
7. You will now see the topology builder screen with the enterprise's topology imported.



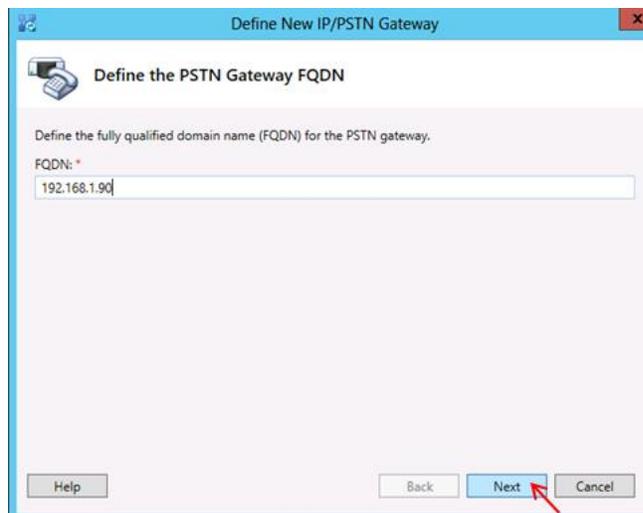
8. In the upper left hand corner, expand the site in which the PSTN gateway will be added. In our case, the site is **Bedford**. Then click on the **PSTN Gateways**



9. Right click on **PSTN gateways** and select **New IP/PSTN Gateway**.



10. In the **Define New IP/PSTN Gateway** window, enter the ip address of the SIP interface of the ECB in the **FQDN** text box and click **Next**.



11. Select **Enable IPv4** in the **Define the IP address** section and click **Next**.

The screenshot shows the 'Define New IP/PSTN Gateway' wizard window. The title bar reads 'Define New IP/PSTN Gateway'. The main heading is 'Define the IP address'. There are two main sections: 'Enable IPv4' and 'Enable IPv6'. Under 'Enable IPv4', the radio button is selected, and the sub-option 'Use all configured IP addresses.' is also selected. Below this, there is a text field for 'PSTN IP address:'. The 'Enable IPv6' section is unselected. At the bottom of the window, there are four buttons: 'Help', 'Back', 'Next', and 'Cancel'. A red arrow points to the 'Next' button.

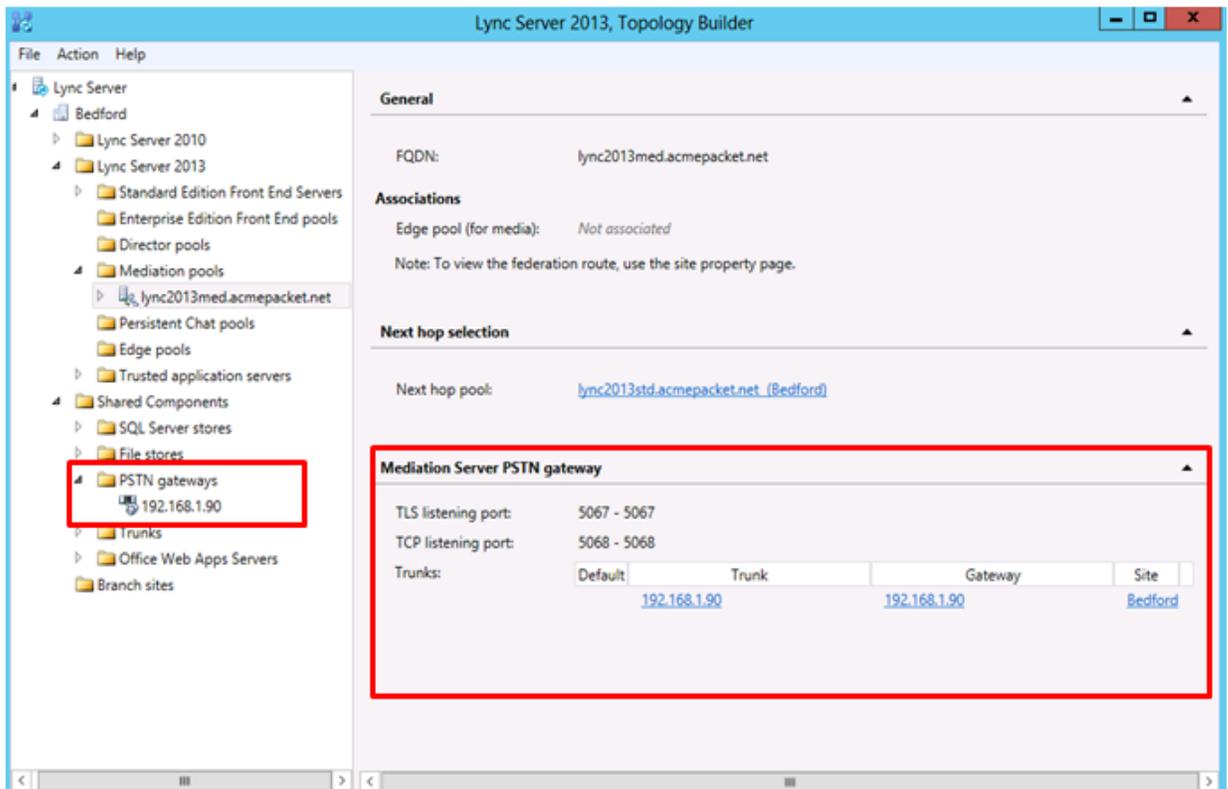
12. In the next section, enter the ip address of the ECB's sip interface under **Trunk name**. Configure the **Listening port for IP/PSTN gateway** as 5068, TCP as the **SIP Transport Protocol** and click **Finish**.

The screenshot shows the 'Define New IP/PSTN Gateway' wizard window at the 'Define the root trunk' step. The title bar reads 'Define New IP/PSTN Gateway'. The main heading is 'Define the root trunk'. There are five fields: 'Trunk name:' with the value '192.168.1.90', 'Listening port for IP/PSTN gateway:' with the value '5068', 'SIP Transport Protocol:' with a dropdown menu set to 'TCP', 'Associated Mediation Server:' with a dropdown menu set to 'lync2013med.acmepacket.net Bedford', and 'Associated Mediation Server port:' with the value '5068'. At the bottom of the window, there are four buttons: 'Help', 'Back', 'Finish', and 'Cancel'. A red arrow points to the 'Finish' button.

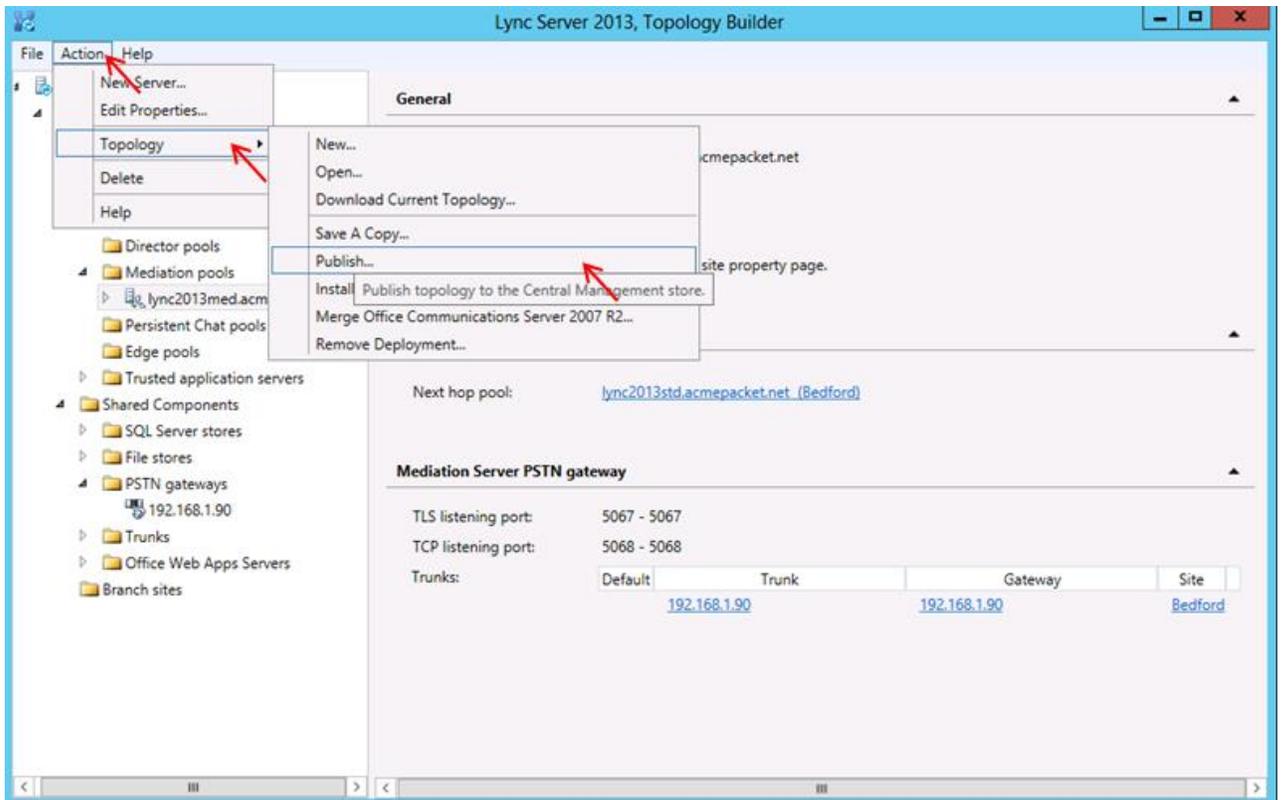
13. The PSTN gateway for the Lync server has been added. It will be listed under **PSTN gateways**.

Expand the **Mediation Pool** list and click on the Mediation Server to be utilized. In our example the Mediation Server is **lync2013med.acmepacket.net**.

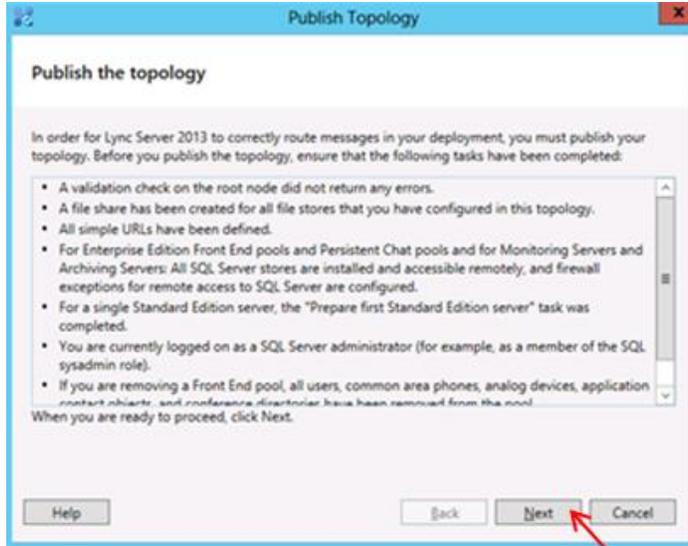
You will see that the PSTN gateway is associated with the Mediation server.



14. In the upper right hand corner of your screen under **Actions** select **Topology** then select **Publish**.

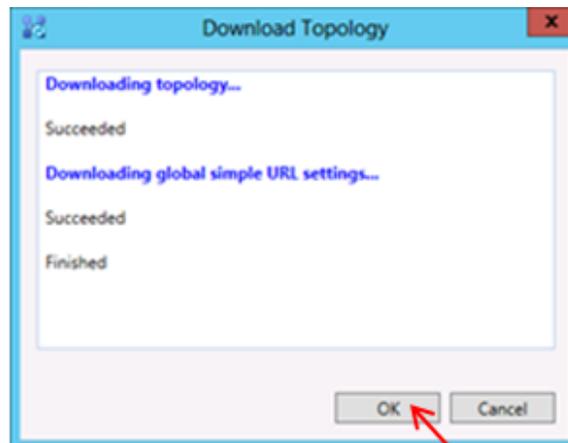


15. You will now see the **Publish Topology** window. Click on the **Next** button



You will now be at a window showing the databases associated with site. Click **Next**.

16. When complete you should see a window from Topology Builder stating that your topology was successfully published. Click the **OK** button.



17. You will be at the Topology Builder main window, expand your site and double check that your PSTN entries are correct and that the appropriate Mediation Server has the PSTN gateway associated.

Creating a route within the Lync Server infrastructure

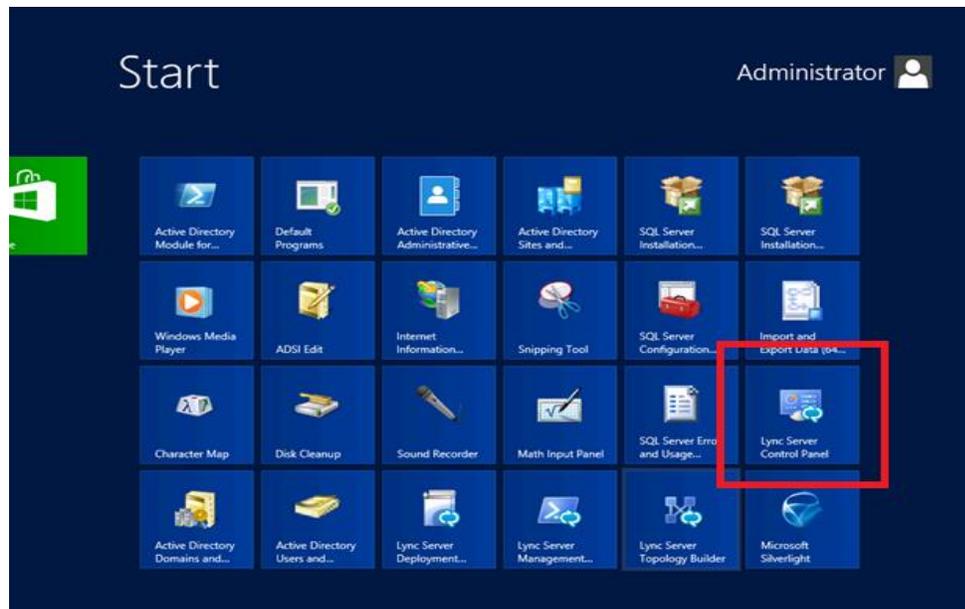
In order for the Lync Server Enterprise Voice clients to utilize the SIP trunking infrastructure that has been put in place, a route will need to be created to allow direction to this egress. Routes specify how Lync Server handles calls placed by enterprise voice users. When a user places a call, the server, if necessary, normalizes the phone number to the E.164 format and then attempts to match that phone number to a SIP Uniform Resource Identifier (URI). If the server is unable to make a match, it applies outgoing call routing logic based on the number. That logic is defined in the form of a separate voice route for each set of target phone numbers listed in the location profile for a locale. For this document we are only describing how to set up a route. Other aspects which apply to Lync Server Enterprise Voice deployments such as dial plans, voice policies, and PSTN usages are not covered.

To add the route we will need:

- Rights to administer Lync Server Control Panel
 - Membership in the CS Administrator Active Directory Group
- Access to the Lync Server Control Panel

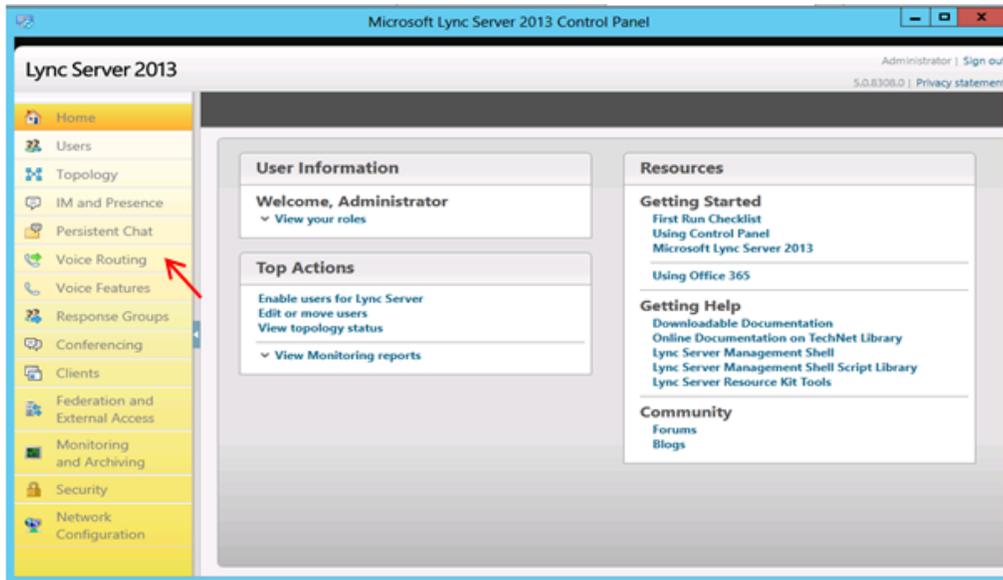
The following process details the steps to create the route:

1. From the Start bar, select Lync Server Control Panel.

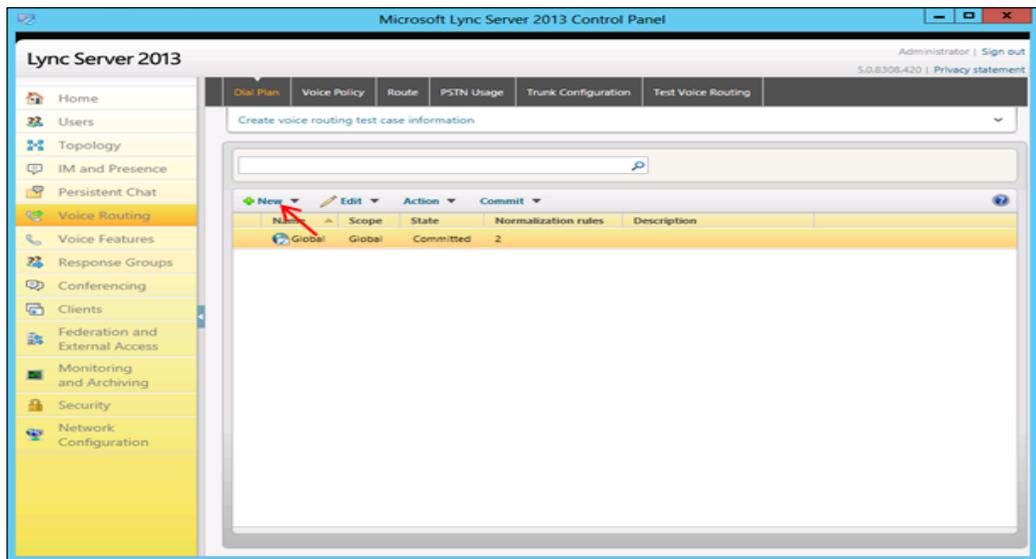


You will be prompted for credential, enter your domain username and password.

- Once logged in, you will now be at the "Welcome Screen". On the left hand side of the window, click on **Voice Routing**.



- The Dial Plan tab in the Voice Routing section will be displayed. On the content area toolbar, click +New.



- Next you build a Dial Plan and a translation rule for the phone numbers you want this route to handle. You have to create two separate dial plans for US and EMEA.

US Dial-plan

Match this pattern: ^\d*

Translation rule: \$1

International call Dial-plan

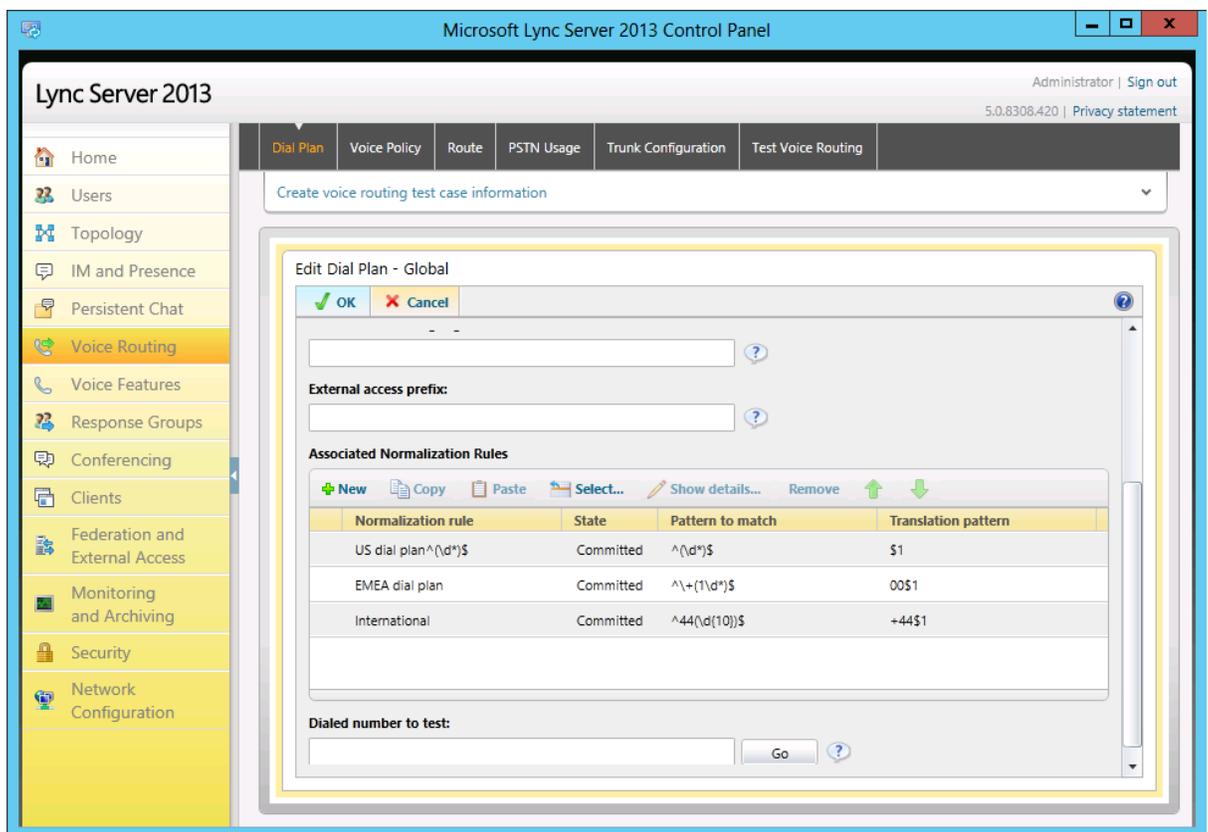
Match this pattern: $\text{^44\d{10}}$

Translation rule: +44\$1

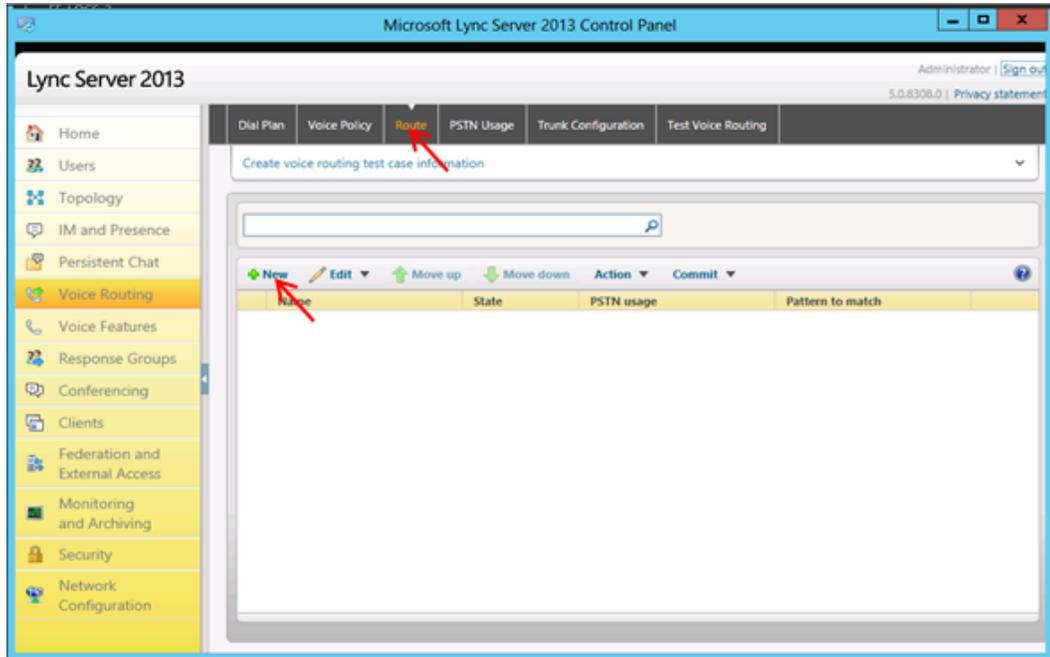
EMEA dial plan

Match this pattern: ^\+(1\d*)

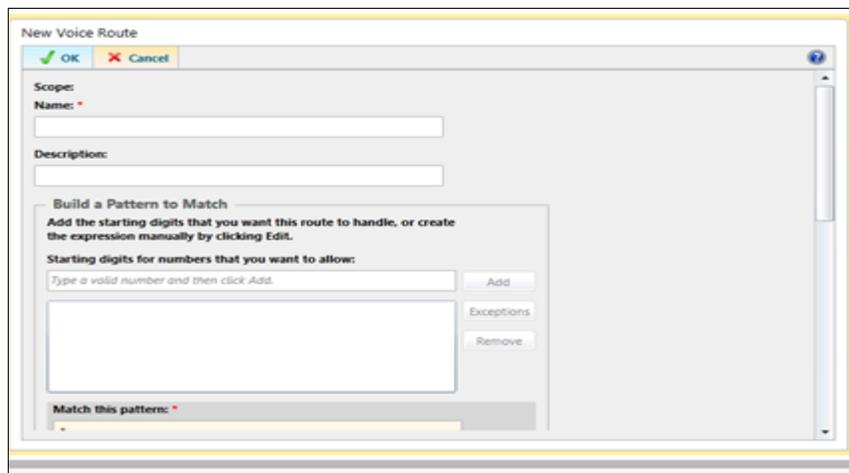
Translation rule: 00\$1



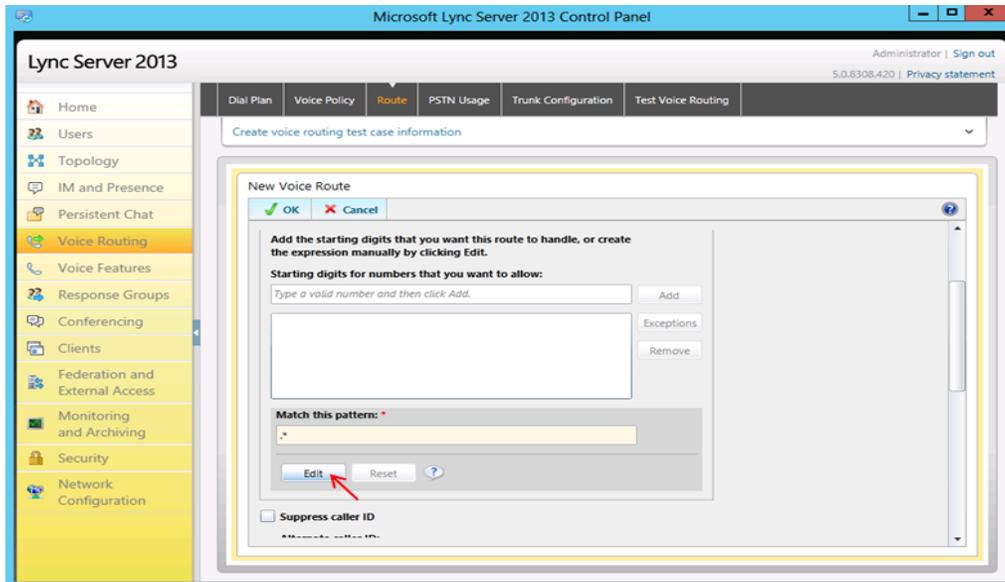
5. On the top row of the tabs, select **Route**. On the content area toolbar, click **+New**.



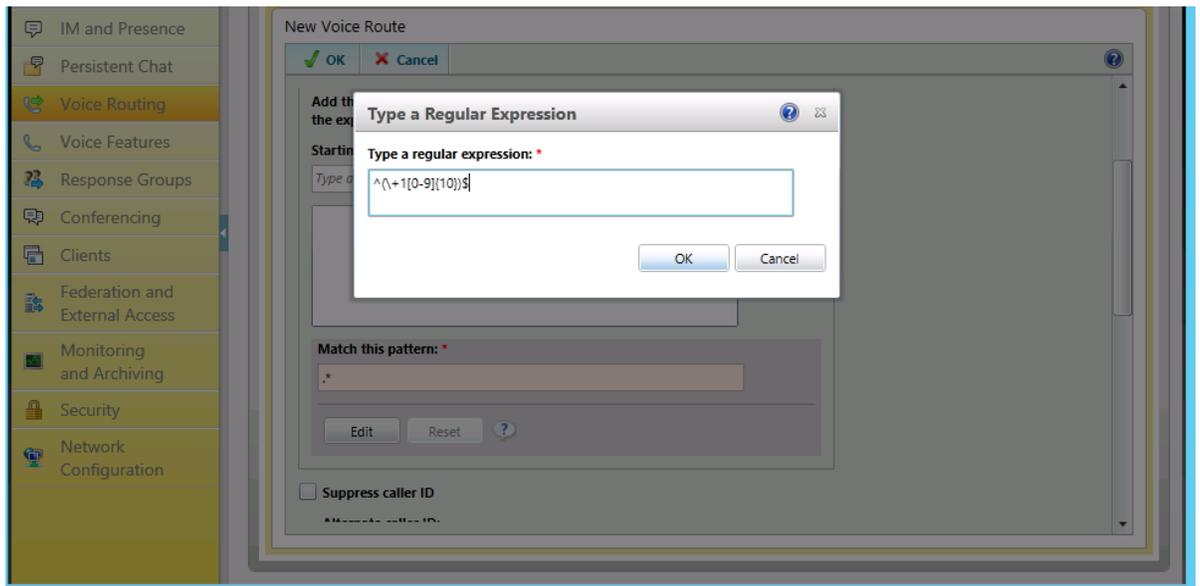
6. On the **New Voice Route** page, in the **Name** field, enter the name you have selected for the Route. In our example, it is US route.



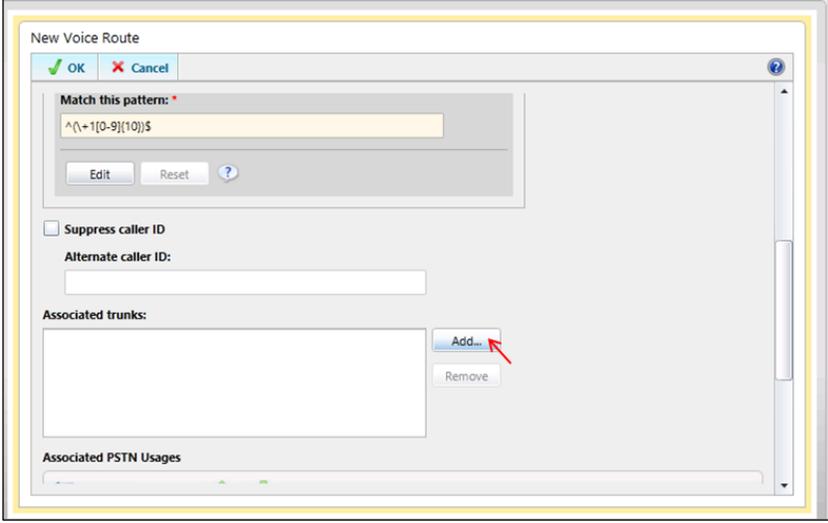
- Next you build a Pattern Match for the phone numbers you want this route to handle. Click **Edit**.



- Enter the pattern for US - `^\(+1[0-9]{10})$` and click **OK**. To enable 4 digit dial, add the pattern `^\(d{4})$`.

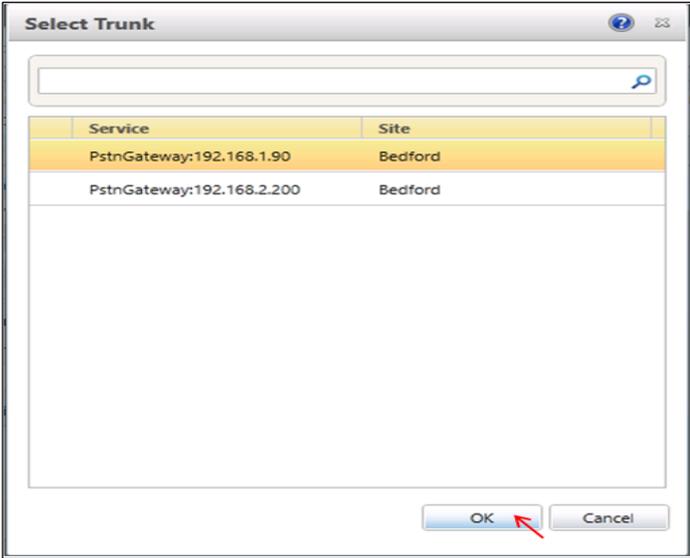


9. Next you want to associate the Voice Route with the **Trunk** you have just created. Scroll down to **Associated Trunks**, click on the **Add** button.



The screenshot shows the 'New Voice Route' dialog box. At the top, there are 'OK' and 'Cancel' buttons. Below that, the 'Match this pattern:' field contains the text '^(\+1[0-9]{10})\$'. There are 'Edit', 'Reset', and a help icon below the pattern field. A checkbox for 'Suppress caller ID' is unchecked. Below it is an 'Alternate caller ID:' field. The 'Associated trunks:' section is empty, and the 'Add...' button is highlighted with a red arrow. There is also a 'Remove' button. At the bottom, there is an 'Associated PSTN Usages' section.

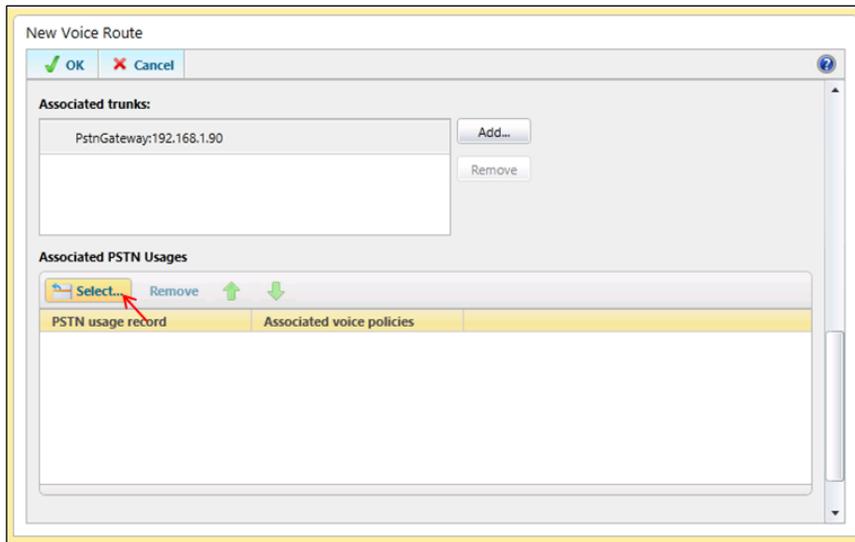
10. You will now be at a window showing available Trunks to associate your Voice Route. Click on the PSTN gateway that you just created and then click the **OK** button.



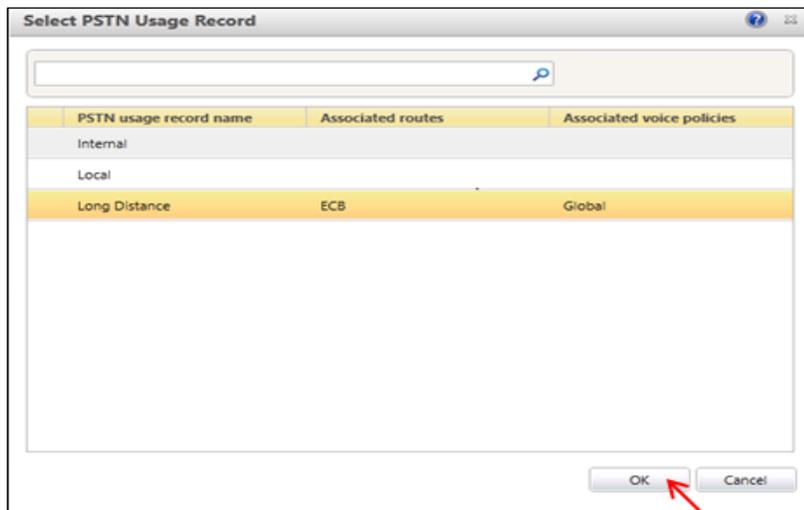
The screenshot shows the 'Select Trunk' dialog box. It has a search bar at the top. Below it is a table with two columns: 'Service' and 'Site'. The first row is highlighted in yellow and shows 'PstnGateway:192.168.1.90' under 'Service' and 'Bedford' under 'Site'. The second row shows 'PstnGateway:192.168.2.200' under 'Service' and 'Bedford' under 'Site'. At the bottom, there are 'OK' and 'Cancel' buttons, with the 'OK' button highlighted by a red arrow.

Service	Site
PstnGateway:192.168.1.90	Bedford
PstnGateway:192.168.2.200	Bedford

11. You can now see that you have associated your trunk with the route you created. An appropriate PSTN usage record will need to be assigned as well. In our example, we use one that was already created in the enterprise. Click on the **Select** button under **Associated PSTN Usages**.



12. In the **Select PSTN Usage Record** window displayed, select the appropriate PSTN Usage Record and click **OK**.



13. You will now see the Associated PSTN Usages which you have added. Click the **OK** button at the top of the **New Voice Route** screen.

New Voice Route

OK Cancel

Associated trunks:

PstnGateway:192.168.1.90

Add... Remove

Associated PSTN Usages

Select... Remove

PSTN usage record	Associated voice policies
Long Distance	Global

Translated number to test:

Go

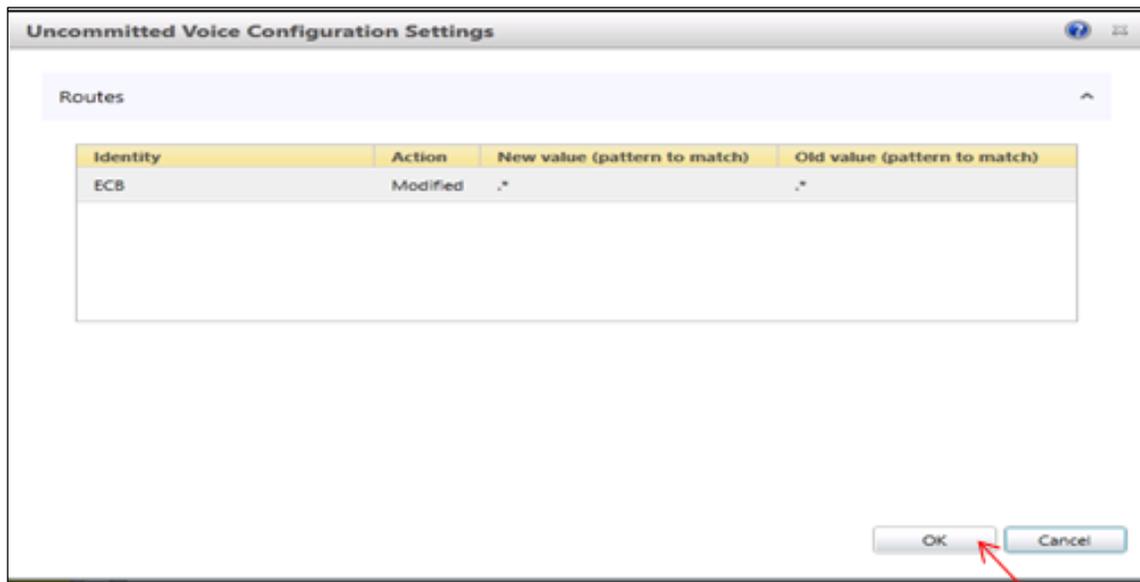
14. You will now be at the Routes page showing the US route. Click the **Commit** drop-down menu, and then **Commit All**.

New Edit Move up Move down Action Commit

Review uncommitted changes
Commit all
Cancel selected changes
Cancel all uncommitted changes

Name	State	PSTN usage	Match
ECB	Uncommitted	Long Distance	

15. On the Uncommitted Voice Configuration Settings window, click OK.



If there are no errors, the new Voice Route has now been successfully created and the state will show as **Committed**.

Additional Steps

There are other aspects to a Lync Server Enterprise Voice deployment such as

- Site, local, and global dial plans
- Voice Policies
- Assigning Voice Policies to users
- PSTN usage policies

Refer to [MSDN technet](#) for relevant information.

Following the same procedure, configure ECB as a PSTN gateway in the Lync 2010 environment.

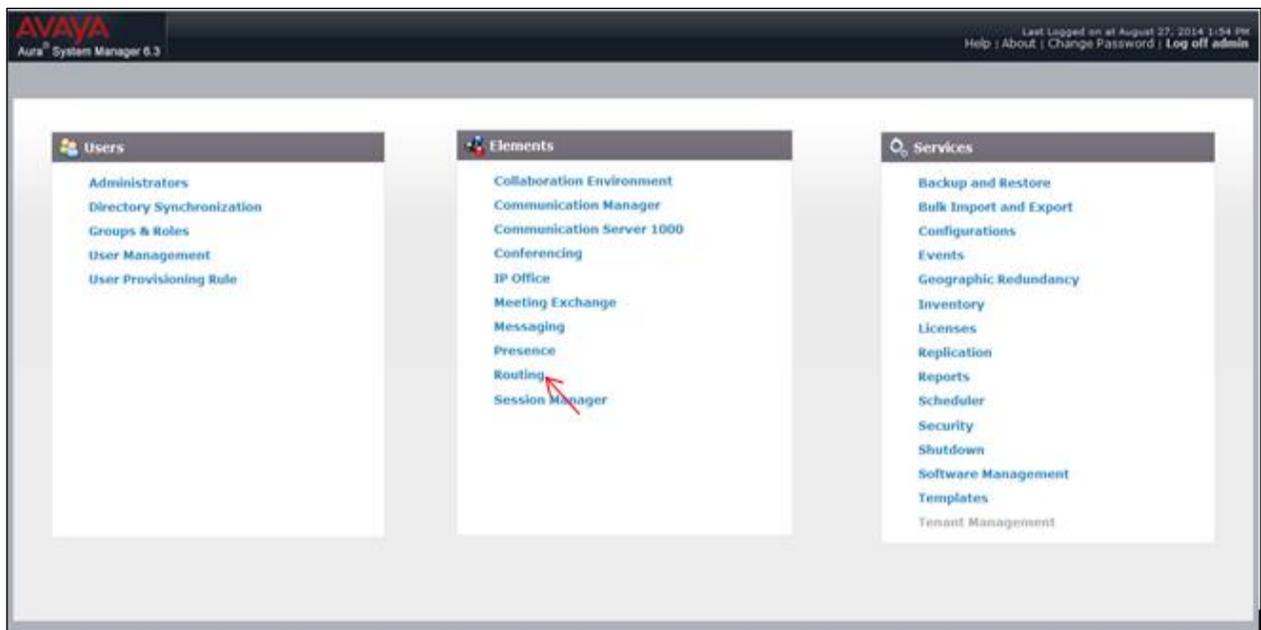
Phase 3 – Configuring the Avaya Session Manager

The enterprise has a fully functional Avaya Aura System Manager. Configuring the System Manager to operate with ECB consists of three steps –

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

Adding the ECB as a SIP Entity

Log in to the Aura System Manager. Click on **Routing** under the **Elements** section.



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

Home Routing * Home / Elements / Routing / SIP Entities Help ?

SIP Entity Details Commit Cancel

General

* Name:

* FQDN or IP Address:

Type: SIP Trunk

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

SIP Link Monitoring

SIP Link Monitoring:

Supports Call Admission Control:

Shared Bandwidth Manager:

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

Override Port & Transport with DNS SRV:

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

Select : All, None

Configuring an Entity link between ECB and Session Manager

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

The screenshot shows the 'Entity Links' configuration page. At the top, there are 'Commit' and 'Cancel' buttons. Below is a table with one item:

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

Below the table, there is a 'Select : All, None' option.

Creating a Routing policy to assign the appropriate routing destination

Select **Routing policies** from the menu and click on **New** to add a routing policy between ECB and SM with the following settings and click **Commit**.

The screenshot shows the 'Routing Policy Details' configuration page. At the top, there are 'Commit', 'Cancel', and 'Help ?' buttons. The page is divided into several sections:

- General**:
 - * Name: Calls to ECB from SM1
 - Disabled:
 - * Retries: 0
 - Notes: Calls to ECB from SM1
- SIP Entity as Destination**:
 - Select

Name	FQDN or IP Address	Type	Notes
acme-ecb	192.168.1.90	SIP Trunk	acme-ecb
- Time of Day**:
 - Add Remove View Gaps/Overlaps

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

At the bottom, there is a 'Select : All, None' option.

Time of Day

1 Item Filter: **Enable**

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

Select : All, None

Dial Patterns

3 Items Filter: **Enable**

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

Select : All, None

Regular Expressions

0 Items Filter: **Enable**

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

The Avaya System Manager is now configured to operate with ECB.

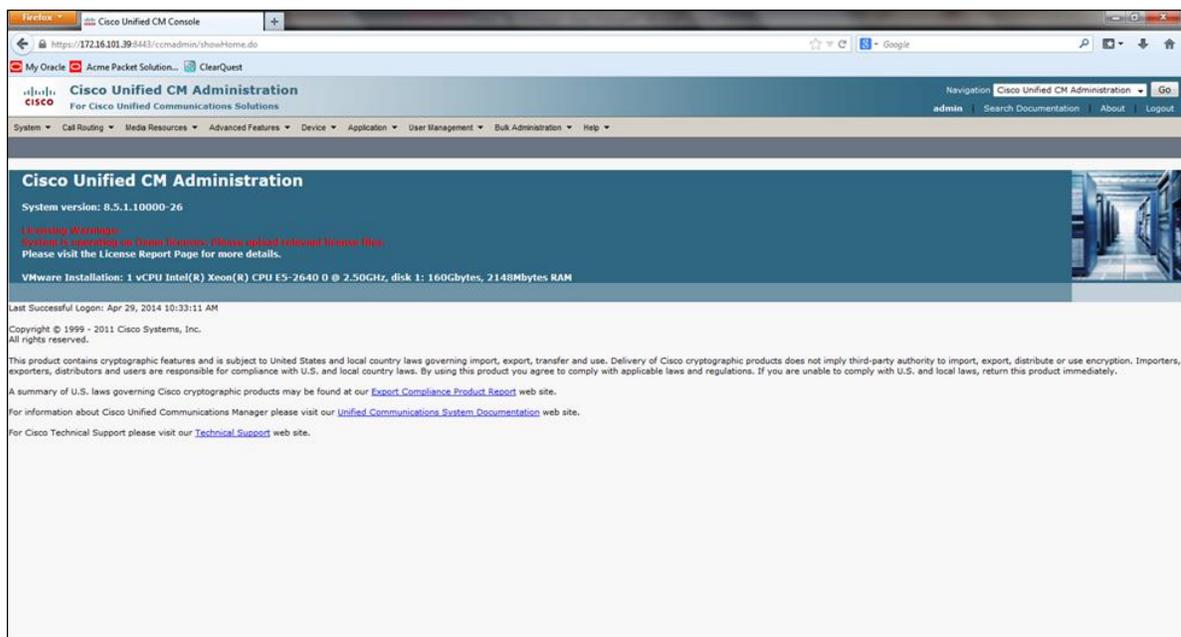
Phase 4 – Configuring the Cisco Unified Communications Manager

The enterprise will have a fully functioning Cisco Unified Communications Manager deployed. We will now configure it to operate with ECB. This consists of the following steps

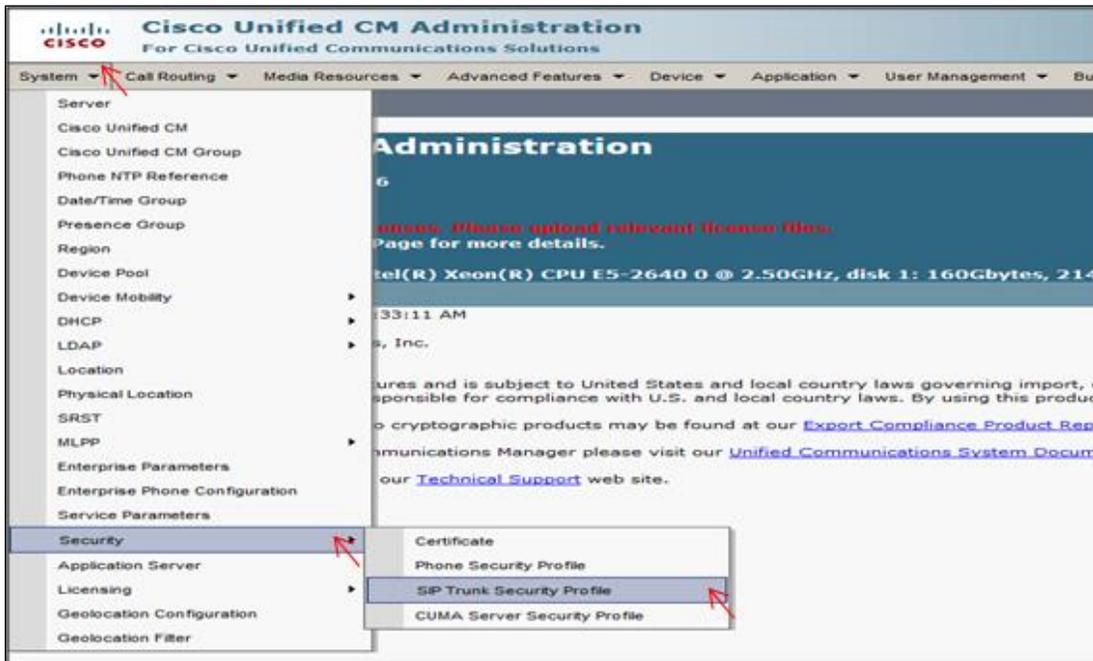
- Configuring the SIP Trunk Security profile
- Configuring the SIP profile
- Configure the Trunk
- Configuring the Route Pattern

Configuring the SIP Trunk Security Profile

1. Log into the Cisco Unified CM administration page using the link <https://server-ip/>.



- To go to the **SIP trunk security profile** page, expand the **System** drop down menu, select **SIP Trunk Security Profile** under **Security**.



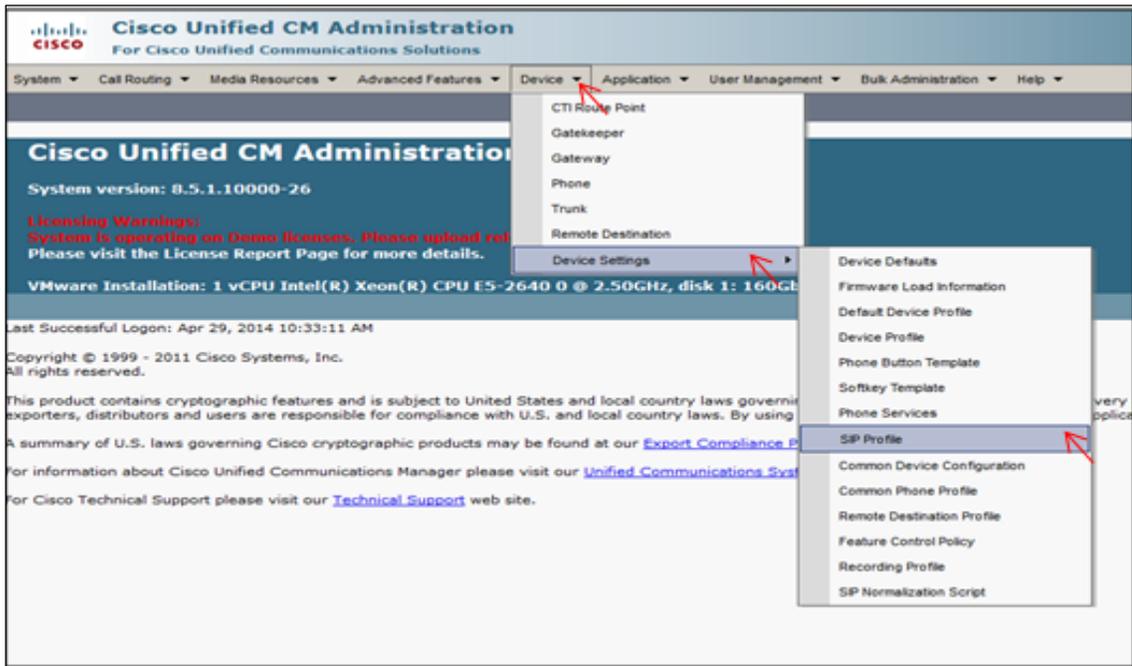
- A Non Secure SIP Trunk security profile should be present, if not create one as shown below

The screenshot displays the 'SIP Trunk Security Profile Configuration' page. At the top, there are action buttons: Save, Delete, Copy, Reset, Apply Config, and Add New. Below this, the 'Status' section shows 'Status: Ready'. The main section is 'SIP Trunk Security Profile Information', which contains the following fields and options:

- Name*: Non Secure SIP Trunk Profile
- Description: Non Secure SIP Trunk Profile authenticated by null String
- Device Security Mode: Non Secure
- Incoming Transport Type*: TCP+UDP
- Outgoing Transport Type: TCP
- Enable Digest Authentication
- Nonce Validity Time (mins)*: 600
- X.509 Subject Name:
- Incoming Port*: 5060
- Enable Application Level Authorization
- Accept Presence Subscription
- Accept Out-of-Dialog REFER**
- Accept Unsolicited Notification
- Accept Replaces Header
- Transmit Security Status

Configuring the SIP Profile

1. To go to the SIP Profile page, expand the **Device** drop down menu and select **SIP Profile** from **Device Settings**.



2. The **Find and List SIP Profiles** page will display the default SIP profile. Click on the **Copy** button to create a new SIP profile.



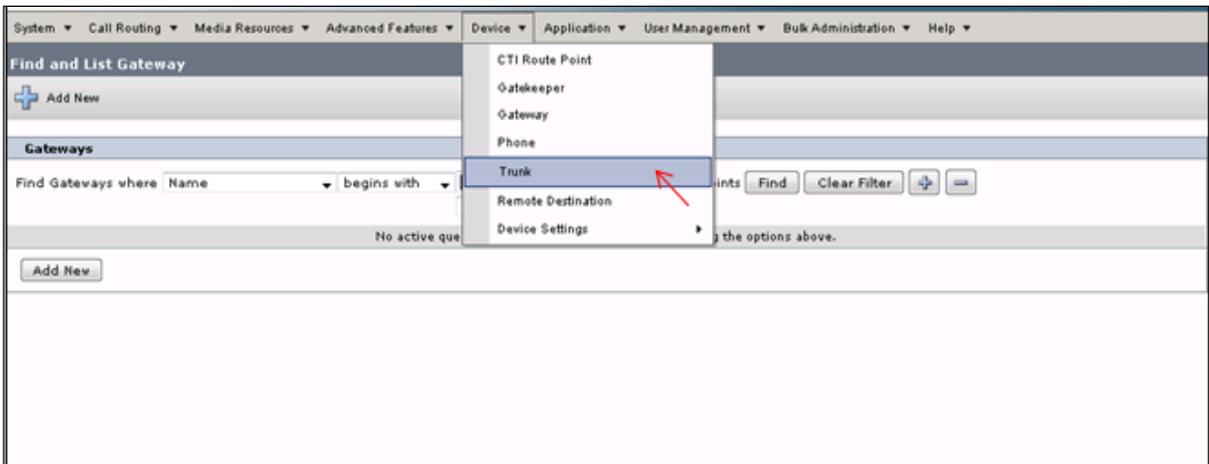
- Add a new SIP profile with the following settings. It is same as the default profile but includes PRACK support.

SIP Profile Information	
Name*	SIP Profile for PRACK
Description	SIP Profile for PRACK
Default MTP Telephony Event Payload Type*	101
Resource Priority Namespace List	< None >
Early Offer for G.Clear Calls*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
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
Start Media Port*	16384
Stop Media Port*	32766
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*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
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
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	

<input type="checkbox"/> RFC 2543 Hold
<input checked="" type="checkbox"/> Semi Attended Transfer
<input type="checkbox"/> Enable VAD
<input type="checkbox"/> Stutter Message Waiting
Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on* <input type="text" value="Never"/>
RSVP Over SIP* <input type="text" value="Local RSVP"/>
<input checked="" type="checkbox"/> Fall back to local RSVP
SIP Rel1XX Options* <input type="text" value="Send PRACK for all 1xx Messages"/>
<input type="checkbox"/> Deliver Conference Bridge Identifier
<input type="checkbox"/> Early Offer support for voice and video calls (insert MTP if needed)
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE
SIP OPTIONS Ping
<input checked="" type="checkbox"/> 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)* <input type="text" value="60"/>
Ping Interval for Out-of-service Trunks (seconds)* <input type="text" value="120"/>
Ping Retry Timer (milliseconds)* <input type="text" value="500"/>
Ping Retry Count* <input type="text" value="6"/>
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Reset"/> <input type="button" value="Apply Config"/> <input type="button" value="Add New"/>

Configuring the Trunk

1. To go to the Trunks page, select **Trunk** from the **Device** drop down menu.



2. Add a trunk with the following settings and click **Save**.

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

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	ECB
Description	ECB
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*	Batch Processing Mode
Packet Capture Duration	0

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

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

Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Transmit UTF-8 Names in QSIG APDU
 Unattended Port
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
 Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*
 Use Trusted Relay Point*
 PSTN Access
 Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain

Call Routing Information

Remote-Party-Id
 Asserted-Identity
 Asserted-Type*

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

Call Routing Information

Remote-Party-Id
 Asserted-Identity
 Asserted-Type*
 SIP Privacy*

Inbound Calls

Significant Digits*
 Connected Line ID Presentation*
 Connected Name Presentation*
 Calling Search Space
 AAR Calling Search Space
 Prefix DN
 Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="< None >"/>	<input checked="" type="checkbox"/>

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

Connected Party Settings

Connected Party Transformation CSS
 Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS
 Use Device Pool Called Party Transformation CSS
 Calling Party Transformation CSS
 Use Device Pool Calling Party Transformation CSS
 Calling Party Selection*
 Calling Line ID Presentation*
 Calling Name Presentation*
 Caller ID DN
 Caller Name
 Redirecting Diversion Header Delivery - Outbound

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

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	192.168.1.90		5068

MTP Preferred Originating Codec*
Presence Group*
SIP Trunk Security Profile*
Rerouting Calling Search Space
Out-Of-Dialog Refer Calling Search Space
SUBSCRIBE Calling Search Space
SIP Profile*
DTMF Signaling Method*

Normalization Script

Normalization Script:

Enable Trace

	Parameter Name	Parameter Value
1		

Geolocation Configuration

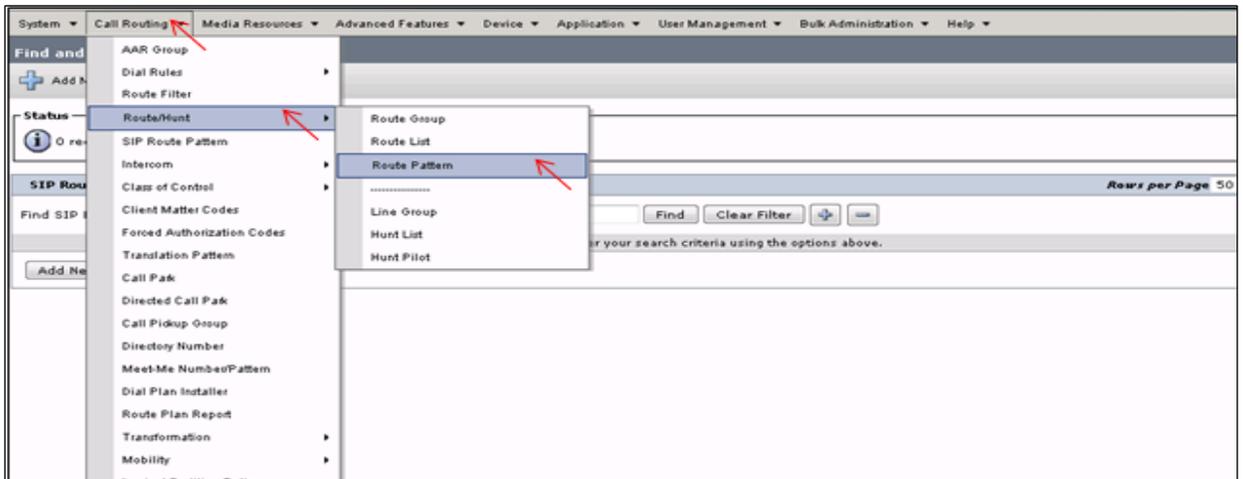
Geolocation

Geolocation Filter

Send Geolocation Information

Configuring the Route Pattern

1. To go to the Route pattern page, click on Call Routing and select Route Pattern from the Route/Hunt drop down menu.



2. In our setup, users dial 8 to dial out. Add a route pattern with the following settings and associate it with the trunk configured in the previous step.

A screenshot of the 'Route Pattern Configuration' page. The page title is 'Route Pattern Configuration' and it includes 'Related Links: Back To Find/List' and a 'Go' button. The page has a toolbar with 'Save', 'Delete', 'Copy', and 'Add New' buttons. The main content area is divided into two sections: 'Pattern Definition' and 'Calling Party Transformations'.
Pattern Definition:
Route Pattern*: 8.@
Route Partition: < None >
Description: Route to SP
Numbering Plan*: NANP
Route Filter: < None >
MLPP Precedence*: Default
Resource Priority Namespace Network Domain: < None >
Route Class*: Default
Gateway/Route List*: ECB (Edit)
Route Option: Route this pattern, Block this pattern No Error
Call Classification*: OffNet
 Allow Device Override, Provide Outside Dial Tone, Allow Overlap Sending, Urgent Priority
 Require Forced Authorization Code
Authorization Level*: 0
 Require Client Matter Code
Calling Party Transformations:
 Use Calling Party's External Phone Number Mask
Calling Party Transform Mask:
Prefix Digits (Outgoing Calls):
Calling Line ID Presentation*: Default

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

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

 Calling Name Presentation*

 Calling Party Number Type*

 Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

 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"/>

The CUCM configuration is now complete.



Phase 5 – Configuring the Oracle Enterprise Session Border Controller

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

In Scope

The following step-by-step guide configuring the E-SBC assumes that this is a newly deployed device dedicated to a single customer. If the enterprise currently has the E-SBC deployed and is adding ECB, then please see the appendix for a better understanding of the Acme Packet Command Line Interface (ACLI).

Note that Oracle offers several models of E-SBC. This document covers the setup for the Acme Packet 3820 and Acme Packet 4500 platform series running Net-Net OS ECX 6.2.0 or later. If instructions are needed for other E-SBC models, please contact your Oracle representative.

Out of Scope

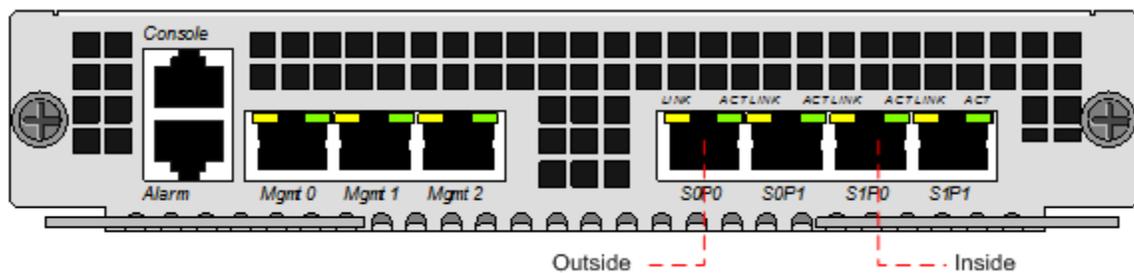
- Configuration of Network management including SNMP and RADIUS; and
- Redundancy configuration

What will you need

- Serial Console cross over cable with RJ-45 connector
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the 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 the ECB
- IP address to be used for the E-SBC internal and external facing ports (Service Interfaces)
- IP address of the next hop gateway in the SIP trunk provider network

Configuring the Oracle Enterprise Session Border Controller (E-SBC)

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

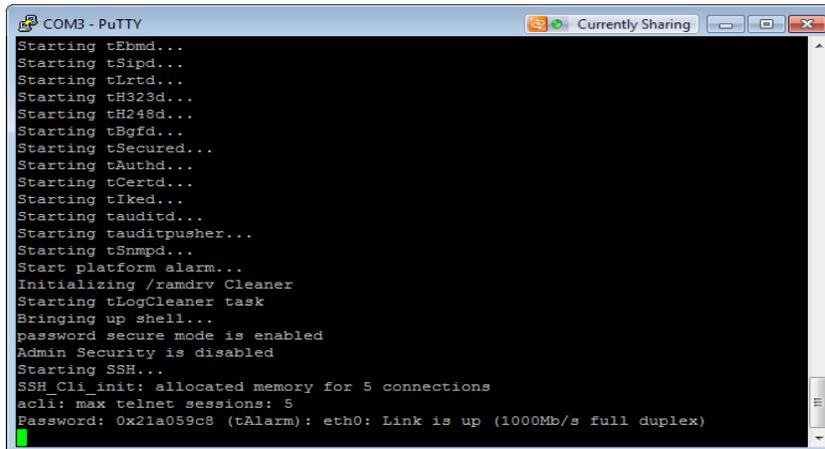
All commands are in bold, such as **configure terminal**; parameters in bold red such as **ORACLE-SBC** are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

Establish the serial connection and logging in the SBC

Confirm the E-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 E-SBC and the other end to console adapter that ships with the E-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 E-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 tCerd...
Starting tIked...
Starting tauditd...
Starting tauditpusher...
Starting tSnmpd...
Start platform alarm...
Initializing /ramdrv Cleaner
Starting tLogCleaner task
Bringing up shell...
password secure mode is enabled
Admin Security is disabled
Starting SSH...
SSH_Cli_init: allocated memory for 5 connections
acl: max telnet sessions: 5
Password: 0x21a059c8 (tAlarm): eth0: Link is up (1000Mb/s full duplex)
```

Enter the following commands to login to the 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
ORACLE-SBC> enable
Password: packet
ORACLE-SBC# configure terminal
ORACLE-SBC (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

Oracle-SBC# 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

```
Oracle-SBC#(configure)bootparam
'.' = clear field; '-' = go to previous field; q = quit
boot device          : eth0
processor number     : 0
host name            : acmesystem
file name            : /code/images/nnECX640m2.tar--- >location where
the software is loaded on the SBC
inet on ethernet (e) : 172.18.255.52:ffffff80 --- > This is the ip
address of the management interface of the SBC, type the IP address and
mask in hex
```

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

Configure System element values

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

```
ORACLE-SBC(configure)# system
ORACLE-SBC(system)# system-config
ORACLE-SBC(system-config)# hostname ORACLE-SBC
ORACLE-SBC(system-config)# description "SBC for SIP Trunking"
ORACLE-SBC(system-config)# location "Bedford, MA"
ORACLE-SBC(system-config)# default-gateway 172.18.0.1
ORACLE-SBC(system-config)# done
```

Once the **system-config** settings have completed and you enter **done**, the E-SBC will output a complete listing of all current settings. This will apply throughout the rest of the configuration and is a function of the **done** command. Confirm the output reflects the values you just entered as well as any configuration defaults.

```
system-config
  hostname
  description                SBC for SIP Trunking
  location                    Bedford, MA
  mib-system-contact
  mib-system-name
  mib-system-location
  snmp-enabled                enabled
  enable-snmp-auth-traps     disabled
  enable-snmp-syslog-notify  disabled
  enable-snmp-monitor-traps  disabled
  enable-env-monitor-traps   disabled
  snmp-syslog-his-table-length 1
  snmp-syslog-level          WARNING
  system-log-level           WARNING
```

```

process-log-level          DEBUG
process-log-ip-address    0.0.0.0
process-log-port          0
collect
    sample-interval       5
    push-interval         15
    boot-state             disabled
    start-time             now
    end-time               never
    red-collect-state      disabled
    red-max-trans          1000
    red-sync-start-time    5000
    red-sync-comp-time     1000
    push-success-trap-state disabled
call-trace                 disabled
internal-trace             disabled
log-filter                 all
default-gateway            172.18.0.1
restart                    enabled
exceptions
telnet-timeout             0
console-timeout           0
remote-control             enabled
cli-audit-trail           enabled
link-redundancy-state     disabled
source-routing             disabled
cli-more                  disabled
terminal-height           24
debug-timeout             0
trap-event-lifetime       0
default-v6-gateway        ::
ipv6-signaling-mtu        1500
ipv4-signaling-mtu        1500
cleanup-time-of-day       00:00
snmp-engine-id-suffix
snmp-agent-mode           v1v2
comm-monitor
    state                 disabled
    qos-enable             enabled
    sbc-grp-id            0
    tls-profile

```

Configure Physical Interface values

To configure physical Interface values, use the `phy-interface` command under the system branch. To enter the system branch from system-config, you issue the `exit` command then the `phy-interface` command.

You will first configure the slot 0, port 0 interface designated with the name `s0p0`. This will be the port plugged into your outside (connection to the trunk) interface.

```
ORACLE-SBC(system-config)# exit
ORACLE-SBC(system)# phy-interface
ORACLE-SBC(phy-interface)# name M00
ORACLE-SBC(phy-interface)# operation-type media
ORACLE-SBC(phy-interface)# slot 0
ORACLE-SBC(phy-interface)# port 0
ORACLE-SBC(phy-interface)# done
```

Once the **phy-interface** settings have completed for slot 0 port 0 and you enter **done**, the E-SBC will output a complete listing of all current settings. Confirm the output reflects the values you just entered.

```
phy-interface
  name                M00
  operation-type      Media
  port                0
  slot                0
  virtual-mac
  admin-state         enabled
  auto-negotiation    enabled
  duplex-mode         FULL
  speed               100
  overload-protection disabled
```

You will now configure the slot 1 port 0 phy-interface, specifying the appropriate values. This will be the port plugged into your inside (connection to the ECB) interface.

```
ORACLE-SBC(phy-interface)# name M10
ORACLE-SBC(phy-interface)# operation-type media
ORACLE-SBC(phy-interface)# slot 1
ORACLE-SBC(phy-interface)# port 0
ORACLE-SBC(phy-interface)# done
phy-interface
  name                M10
  operation-type      Media
  port                0
```

slot	1
virtual-mac	
admin-state	enabled
auto-negotiation	enabled
duplex-mode	FULL
speed	100
overload-protection	disabled

Configure Network Interface values

To configure Network Interface values, use the `network-interface` command under the system branch. To enter the system branch from phy-interface, you issue the `exit` command, then the `network-interface` command.

You will first configure the IP characteristics for the M10 interface defined above

```

ORACLE-SBC (phy-interface) # exit
ORACLE-SBC (system) # network-interface
ORACLE-SBC (network-interface) # name slp0
ORACLE-SBC (network-interface) # description "ECB-facing inside interface"
ORACLE-SBC (network-interface) # ip-address 192.168.1.130
ORACLE-SBC (network-interface) # netmask 255.255.255.0
ORACLE-SBC (network-interface) # gateway 192.168.1.1
ORACLE-SBC (network-interface) # pri-utility-addr 192.168.1.131
ORACLE-SBC (network-interface) # sec-utility-addr 192.168.1.132
ORACLE-SBC (network-interface) # add-hip-ip 192.168.1.130
ORACLE-SBC (network-interface) # add-icmp-ip 192.168.1.130
ORACLE-SBC (network-interface) # done

network-interface
  name                slp0
  sub-port-id         0
  description         ECB-facing inside interface
  hostname
  ip-address          192.168.1.130
  pri-utility-addr    192.168.1.131
  sec-utility-addr    192.168.1.132
  netmask             255.255.255.0
  gateway             192.168.1.1
  sec-gateway
  gw-heartbeat
    state             disabled
    heartbeat         0

```

```

        retry-count          0
        retry-timeout        1
        health-score         0
    dns-ip-primary
    dns-ip-backup1
    dns-ip-backup2
    dns-domain
    dns-timeout              11
    hip-ip-list              192.168.1.130
    ftp-address
    icmp-address             192.168.1.130
    snmp-address
    telnet-address
    ssh-address

```

You will now configure the slot 0 port 0 sub port 0 network-interface, specifying the appropriate values.

```

ORACLE-SBC(network-interface)# name s0p0
ORACLE-SBC(network-interface)# description "VoIP gateway-facing outside
interface"
ORACLE-SBC(network-interface)# ip-address 192.20.0.108
ORACLE-SBC(network-interface)# netmask 255.255.255.0
ORACLE-SBC(network-interface)# gateway 192.20.0.1
ORACLE-SBC(network-interface)# pri-utility-addr 192.20.0.109
ORACLE-SBC(network-interface)# sec-utility-addr 192.20.0.110
ORACLE-SBC(network-interface)# dns-ip-primary 8.8.8.8
ORACLE-SBC(network-interface)# dns-ip-backup1 8.8.4.4
ORACLE-SBC(network-interface)# dns-domain tsengr.com
ORACLE-SBC(network-interface)# add-hip-ip 192.20.0.108
ORACLE-SBC(network-interface)# add-icmp-ip 192.20.0.108
ORACLE-SBC(network-interface)# done

network-interface
    name                s0p0
    sub-port-id         0
    description         VoIP gateway-facing outside
interface
    hostname
    ip-address          192.20.0.108
    pri-utility-addr    192.20.0.109
    sec-utility-addr    192.20.0.110
    netmask             255.255.255.0
    gateway             192.20.0.1

```

```

sec-gateway
gw-heartbeat
    state                disabled
    heartbeat            0
    retry-count          0
    retry-timeout        1
    health-score         0
dns-ip-primary          8.8.8.8
dns-ip-backup1         8.8.4.4
dns-ip-backup2
dns-domain              tsengr.com
dns-timeout            11
hip-ip-list             192.20.0.108
ftp-address
icmp-address            192.20.0.108
snmp-address
telnet-address
ssh-address

```

You will now configure the wancom1 and wancom2 for redundancy, specifying the appropriate values.

```

ORACLE-SBC (network-interface) # name wancom1
ORACLE-SBC (network-interface) # netmask 255.255.255.252
ORACLE-SBC (network-interface) # pri-utility-addr 169.254.1.1
ORACLE-SBC (network-interface) # sec-utility-addr 169.254.1.2
ORACLE-SBC (network-interface) # done

network-interface
    name                wancom1
    sub-port-id         0
    description
    hostname
    ip-address
    pri-utility-addr    169.254.1.1
    sec-utility-addr    169.254.1.2
    netmask             255.255.255.252
    gateway
    sec-gateway
    gw-heartbeat
        state          disabled
        heartbeat      0
        retry-count    0

```

```
        retry-timeout          1
        health-score           0
    dns-ip-primary
    dns-ip-backup1
    dns-ip-backup2
    dns-domain
    dns-timeout                 11
    hip-ip-list
    ftp-address
    icmp-address
    snmp-address
    telnet-address
    ssh-address

ORACLE-SBC(network-interface)# name wancom2
ORACLE-SBC(network-interface)# netmask 255.255.255.252
ORACLE-SBC(network-interface)# pri-utility-addr 169.254.2.1
ORACLE-SBC(network-interface)# sec-utility-addr 169.254.2.2
ORACLE-SBC(network-interface)# done

network-interface
    name                       wancom2
    sub-port-id                0
    description
    hostname
    ip-address
    pri-utility-addr           169.254.2.1
    sec-utility-addr           169.254.2.2
    netmask                    255.255.255.252
    gateway
    sec-gateway
    gw-heartbeat
        state                  disabled
        heartbeat              0
        retry-count            0
        retry-timeout          1
        health-score           0
    dns-ip-primary
    dns-ip-backup1
    dns-ip-backup2
    dns-domain
```

```
dns-timeout          11
hip-ip-list
ftp-address
icmp-address
snmp-address
telnet-address
ssh-address
```

Configure Global SIP configuration

To configure the Global SIP values, use the **sip-config** command under the session-router branch. To enter the session-router branch from network-interface, you issue the **exit** command twice, followed by the **sip-config** command.

```
ORACLE-SBC(network-interface)# exit
ORACLE-SBC(system)# exit
ORACLE-SBC(configure)# session-router
ORACLE-SBC(session-router)# sip-config
ORACLE-SBC(sip-config)# operation-mode dialog
ORACLE-SBC(sip-config)# done

sip-config
state                enabled
operation-mode       dialog
dialog-transparency  enabled
home-realm-id
egress-realm-id
nat-mode              None
registrar-domain
registrar-host
registrar-port       0
register-service-route always
init-timer            500
max-timer             4000
trans-expire          32
invite-expire         180
inactive-dynamic-conn 32
enforcement-profile
pac-method
pac-interval          10
pac-strategy          PropDist
pac-load-weight       1
```

pac-session-weight	1
pac-route-weight	1
pac-callid-lifetime	600
pac-user-lifetime	3600
red-sip-port	1988
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
add-reason-header	disabled
sip-message-len	4096
enum-sag-match	disabled
extra-method-stats	disabled
rph-feature	disabled
nsep-user-sessions-rate	0
nsep-sa-sessions-rate	0
registration-cache-limit	0
register-use-to-for-lp	disabled
refer-src-routing	disabled
add-ucid-header	disabled
proxy-sub-events	
pass-gruu-contact	disabled
sag-lookup-on-redirect	disabled
set-disconnect-time-on-bye	disabled

Configure Global Media configuration

To configure the Media values, use the **media-manager** command under the media-manager branch. To enter the media-manager branch from sip-config, you issue the **exit** command twice, followed by the **media-manager** command twice.

By issuing the **select** then **done** commands at this level, you will be creating the media-manager element, enabling the media management functions in the E-SBC with the default values.

```

ORACLE-SBC (sip-config)# exit
ORACLE-SBC (session-router)# exit
ORACLE-SBC (configure)# media-manager
ORACLE-SBC (media-manager)# media-manager
ORACLE-SBC (media-manager)# select
ORACLE-SBC (media-manager-config)# done

media-manager
  state          enabled
  latching      enabled

```

flow-time-limit	86400
initial-guard-timer	300
subsq-guard-timer	300
tcp-flow-time-limit	86400
tcp-initial-guard-timer	300
tcp-subsq-guard-timer	300
tcp-number-of-ports-per-flow	2
hnt-rtcp	disabled
algd-log-level	NOTICE
mbcd-log-level	NOTICE
red-flow-port	1985
red-mgcp-port	1986
red-max-trans	10000
red-sync-start-time	5000
red-sync-comp-time	1000
media-policing	enabled
max-signaling-bandwidth	10000000
max-untrusted-signaling	100
min-untrusted-signaling	30
app-signaling-bandwidth	0
tolerance-window	30
rtcp-rate-limit	0
trap-on-demote-to-deny	disabled
min-media-allocation	2000
min-trusted-allocation	4000
deny-allocation	64000
anonymous-sdp	disabled
arp-msg-bandwidth	32000
fragment-msg-bandwidth	0
rfc2833-timestamp	disabled
default-2833-duration	100
rfc2833-end-pkts-only-for-non-sig	enabled
translate-non-rfc2833-event	disabled
media-supervision-traps	disabled
dnssalg-server-failover	disabled

Configure Realms

To configure the realm values, use the **realm-config** command under the media-manager branch. To enter the media-manager branch from media-manager-config, you issue the **exit** command, followed by the **realm-config** command.

You will create two realms:

- The ECB- Peer, which represents the ECB-facing (inside) network; and
- The SIP-trunk, which represents the gateway-facing (outside) network.

```
ORACLE-SBC (media-manager-config)# exit
ORACLE-SBC (media-manager)# realm-config
ORACLE-SBC (realm-config)# identifier ECB-Peer
ORACLE-SBC (realm-config)# description "ECB-facing (Inside) "
ORACLE-SBC (realm-config)# network-interfaces slp0:0
ORACLE-SBC (realm-config)# done

realm-config
    identifier                ECB-Peer
    description                ECB-facing (Inside)
    addr-prefix                0.0.0.0
    network-interfaces        slp0:0
    mm-in-realm                enabled
    mm-in-network              enabled
    mm-same-ip                 enabled
    mm-in-system               enabled
    bw-cac-non-mm              disabled
    msm-release                 disabled
    qos-enable                  disabled
    generate-UDP-checksum      disabled
    max-bandwidth               0
    fallback-bandwidth         0
    max-priority-bandwidth     0
    max-latency                 0
    max-jitter                  0
    max-packet-loss            0
    observ-window-size         0
    parent-realm
    dns-realm
    media-policy
    media-sec-policy
    in-translationid
    out-translationid
    in-manipulationid
    out-manipulationid
    manipulation-string
```



```

stun-server-ip          0.0.0.0
stun-server-port       3478
stun-changed-ip        0.0.0.0
stun-changed-port     3479
match-media-profiles
qos-constraint
sip-profile
sip-isup-profile
block-rtcp             disabled
hide-egress-media-update disabled

```

You will now configure the realm for SIP Trunk side of the E-SBC, specifying the appropriate values.

```

ORACLE-SBC (realm-config)# identifier SIP-trunk
ORACLE-SBC (realm-config)# description "Gateway(Outside)"
ORACLE-SBC (realm-config)# network-interfaces s0p0:0
ORACLE-SBC (realm-config)# done

realm-config
  identifier          SIP-trunk
  description         Gateway(Outside)
  addr-prefix         0.0.0.0
  network-interfaces

                        s0p0:0
  mm-in-realm        enabled
  mm-in-network       enabled
  mm-same-ip          enabled
  mm-in-system        enabled
  bw-cac-non-mm      disabled
  msm-release         disabled
  qos-enable          disabled
  generate-UDP-checksum disabled
  max-bandwidth       0
  fallback-bandwidth  0
  max-priority-bandwidth 0
  max-latency         0
  max-jitter          0
  max-packet-loss     0
  observ-window-size  0
  parent-realm
  dns-realm
  media-policy

```

```
media-sec-policy
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit          0
access-control-trust-level  none
invalid-signal-threshold    0
maximum-signal-threshold    0
untrusted-signal-threshold  0
nat-trust-threshold         0
deny-period                 30
cac-failure-threshold       0
untrust-cac-failure-threshold 0
ext-policy-svr
diam-e2-address-realm
symmetric-latching         disabled
pai-strip                   disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching        none
restriction-mask            32
accounting-enable           enabled
user-cac-mode               none
user-cac-bandwidth         0
user-cac-sessions           0
icmp-detect-multiplier      0
icmp-advertisement-interval 0
icmp-target-ip
monthly-minutes             0
net-management-control      disabled
delay-media-update          disabled
refer-call-transfer         disabled
dyn-refer-term              disabled
codec-policy
codec-manip-in-realm        disabled
```

```

codec-manip-in-network      disabled
constraint-name
call-recording-server-id
xnq-state                   xnq-unknown
hairpin-id                 0
stun-enable                disabled
stun-server-ip            0.0.0.0
stun-server-port          3478
stun-changed-ip           0.0.0.0
stun-changed-port         3479
match-media-profiles
qos-constraint
sip-profile
sip-isup-profile
block-rtcp                 disabled
hide-egress-media-update   disabled

```

Configure E-SBC redundancy configuration

To configure the E-SBC redundancy configuration, use the **redundancy-config** command under **system** element.

```

ORACLE-SBC (realm-config) # exit
ORACLE-SBC (media-manager) # exit
ORACLE-SBC (configure) # system
ORACLE-SBC (system) # redundancy
ORACLE-SBC (redundancy) # state enabled
ORACLE-SBC (redundancy) # peer
ORACLE-SBC (rdncy-peer) # name Oracle-SBC
ORACLE-SBC (rdncy-peer) # state enabled
ORACLE-SBC (rdncy-peer) # type Primary
ORACLE-SBC (rdncy-peer) # destination
ORACLE-SBC (rdncy-peer-dest) # address 169.254.1.1:9090
ORACLE-SBC (rdncy-peer-dest) # network-interface wancom1:0
ORACLE-SBC (rdncy-peer-dest) # done
destination
  address                169.254.1.1:9090
  network-interface      wancom1:0
ORACLE-SBC (rdncy-peer-dest) # address 169.254.2.1:9090
ORACLE-SBC (rdncy-peer-dest) # network-interface wancom2:0
ORACLE-SBC (rdncy-peer-dest) # done
destination
  address                169.254.2.1:9090
  network-interface      wancom2:0

```

```

ORACLE-SBC (rdncy-peer-dest) # exit
ORACLE-SBC (rdncy-peer) # done
peer
  name                Oracle-SBC
  state               enabled
  type                Primary
  destination
    address            169.254.1.1:9090
    network-interface  wancom1:0
  destination
    address            169.254.2.1:9090
    network-interface  wancom2:0
ORACLE-SBC (rdncy-peer) # name SN1Secondary
ORACLE-SBC (rdncy-peer) # state enabled
ORACLE-SBC (rdncy-peer) # type Secondary
ORACLE-SBC (rdncy-peer) # destination
ORACLE-SBC (rdncy-peer-dest) # address 169.254.1.2:9090
ORACLE-SBC (rdncy-peer-dest) # network-interface wancom1:0
ORACLE-SBC (rdncy-peer-dest) # done
destination
  address            169.254.1.2:9090
  network-interface  wancom1:0
ORACLE-SBC (rdncy-peer-dest) # address 169.254.2.2:9090
ORACLE-SBC (rdncy-peer-dest) # network-interface wancom2:0
ORACLE-SBC (rdncy-peer-dest) # done
destination
  address            169.254.2.2:9090
  network-interface  wancom2:0
ORACLE-SBC (rdncy-peer-dest) # exit
ORACLE-SBC (rdncy-peer) # done
peer
  name                SN1Secondary
  state               enabled
  type                Secondary
  destination
    address            169.254.1.2:9090
    network-interface  wancom1:0
  destination
    address            169.254.2.2:9090
    network-interface  wancom2:0
ORACLE-SBC (rdncy-peer) # exit

```

```

ORACLE-SBC (redundancy) # done
redundancy-config
    state                enabled
    log-level            INFO
    health-threshold    75
    emergency-threshold 50
    port                 9090
    advertisement-time   500
    percent-drift        210
    initial-time         1250
    becoming-standby-time 180000
    becoming-active-time 100
    cfg-port             1987
    cfg-max-trans        10000
    cfg-sync-start-time  5000
    cfg-sync-comp-time   1000
    gateway-heartbeat-interval 10
    gateway-heartbeat-retry 3
    gateway-heartbeat-timeout 1
    gateway-heartbeat-health 1
    media-if-peercheck-time 0
peer
    name                 SN1Secondary
    state                enabled
    type                 Secondary
    destination
        address          169.254.1.2:9090
        network-interface wancom1:0
    destination
        address          169.254.2.2:9090
        network-interface wancom2:0
peer
    name                 Oracle-SBC
    state                enabled
    type                 Primary
    destination
        address          169.254.1.1:9090
        network-interface wancom1:0
    destination
        address          169.254.2.1:9090
        network-interface wancom2:0

```

```
ORACLE-SBC (redundancy) # exit
```

Configure SIP signaling configuration

To configure the SIP signaling values, use the **sip-interface** command under the session-router branch. To enter the session-router branch from realm-config, you issue the **exit** command twice, followed by the **sip-interface** command.

Here you will be configuring the IP addresses and TCP ports on which the E-SBC will listen for and transmit SIP messages. These will be the same IP addresses as configured on the associated network-interface elements.

```
ORACLE-SBC (realm-config) # exit
ORACLE-SBC (media-manager) # exit
ORACLE-SBC (configure) # session-router
ORACLE-SBC (session-router) # sip-interface
ORACLE-SBC (sip-interface) # realm SIP-trunk
ORACLE-SBC (sip-interface) # description "SIP Trunk-facing (Outside)"
ORACLE-SBC (sip-interface) # sip-ports
ORACLE-SBC (sip-port) # address 192.20.0.108
ORACLE-SBC (sip-port) # done

sip-port
address                192.20.0.108
port                   5060
transport-protocol    UDP
tls-profile
allow-anonymous       all
ims-aka-profile

ORACLE-SBC (sip-port) # exit
ORACLE-SBC (sip-interface) # done

sip-interface
state                  enabled
realm-id              SIP-trunk
description           SIP Trunk-facing (Outside)
sip-port
  address              192.20.0.108
  port                 5060
  transport-protocol  UDP
  tls-profile
  allow-anonymous     all
  ims-aka-profile
carriers
```

trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass

```

ccf-address
ecf-address
term-tgrp-mode           none
implicit-service-route   disabled
rfc2833-payload          101
rfc2833-mode             transparent
constraint-name
response-map
local-response-map
ims-aka-feature          disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive           none
add-sdp-invite           disabled
add-sdp-profiles
sip-profile
sip-isup-profile

```

You will now configure the ECB-facing SIP interface.

```

ORACLE-SBC (sip-interface)# realm-id ECB-Peer
ORACLE-SBC (sip-interface)# description "ECB-Facing(Inside)"
ORACLE-SBC (sip-interface)# sip-ports
ORACLE-SBC (sip-port)# address 192.168.1.130
ORACLE-SBC (sip-port)# transport-protocol TCP
ORACLE-SBC (sip-port)# done
sip-port
address                192.168.1.130
port                   5060
transport-protocol    TCP
tls-profile
allow-anonymous        all
ims-aka-profile

ORACLE-SBC (sip-port)# exit
ORACLE-SBCORACLE-SBC (sip-interface)# done

sip-interface
state                  enabled
realm-id               ECB-Peer
description             ECB-Facing(Inside)
sip-port

```

address		192.168.1.130
port		5060
transport-protocol		TCP
tls-profile		
allow-anonymous		all
ims-aka-profile		
carriers		
trans-expire	0	
invite-expire	0	
max-redirect-contacts	0	
proxy-mode		
redirect-action		
contact-mode	none	
nat-traversal	none	
nat-interval	30	
tcp-nat-interval	90	
registration-caching	disabled	
min-reg-expire	300	
registration-interval	3600	
route-to-registrar	disabled	
secured-network	disabled	
teluri-scheme	disabled	
uri-fqdn-domain		
trust-mode	all	
max-nat-interval	3600	
nat-int-increment	10	
nat-test-increment	30	
sip-dynamic-hnt	disabled	
stop-recurse	401,407	
port-map-start	0	
port-map-end	0	
in-manipulationid		
out-manipulationid		
manipulation-string		
manipulation-pattern		
sip-ims-feature	disabled	
operator-identifier		
anonymous-priority	none	
max-incoming-conns	0	
per-src-ip-max-incoming-conns	0	
inactive-conn-timeout	0	

```

untrusted-conn-timeout      0
network-id
ext-policy-server
default-location-string
charging-vector-mode        pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode              none
implicit-service-route      disabled
rfc2833-payload             101
rfc2833-mode                 transparent
constraint-name
response-map
local-response-map
ims-aka-feature              disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive               none
add-sdp-invite               disabled
add-sdp-profiles
sip-profile
sip-isup-profile

```

Configure Next-hop signaling configuration

To configure the next-hop signaling elements (i.e., the ECB and PSTN gateway) you define session-agents. Use the **session-agent** command under the session-router branch. To enter the session-router branch from sip-interface, you issue the **exit** command, followed by the **session-agent** command.

Here you will be configuring the IP addresses and TCP ports to which the E-SBC will send and from which it will expect to receive SIP messages for your next-hop signaling elements.

We will first configure the PSTN gateway.

```

ORACLE-SBCORACLE-SBC(sip-interface)# exit
ORACLE-SBC(session-router)# hostname 10.10.1.8
ORACLE-SBC(session-router)# ip-address 10.10.1.8
ORACLE-SBC(session-router)# port 5060
ORACLE-SBC(session-router)# realm-id SIP-trunk
ORACLE-SBC(session-router)# done

```

```
session-agent
  hostname                10.10.1.8
  ip-address              10.10.1.8
  port                   5060
  state                  enabled
  app-protocol           SIP
  app-type
  transport-method      UDP
  realm-id               SIP-trunk
  egress-realm-id
  description
  carriers
  allow-next-hop-lp     enabled
  constraints            disabled
  max-sessions          0
  max-inbound-sessions  0
  max-outbound-sessions 0
  max-burst-rate        0
  max-inbound-burst-rate 0
  max-outbound-burst-rate 0
  max-sustain-rate      0
  max-inbound-sustain-rate 0
  max-outbound-sustain-rate 0
  min-seizures          5
  min-asr                0
  time-to-resume        0
  ttr-no-response       0
  in-service-period     0
  burst-rate-window     0
  sustain-rate-window   0
  req-uri-carrier-mode  None
  proxy-mode
  redirect-action
  loose-routing         enabled
  send-media-session    enabled
  response-map
  ping-method
  ping-interval         0
  ping-send-mode        keep-alive
  ping-all-addresses   disabled
  ping-in-service-response-codes
```

```

out-service-response-codes
media-profiles
in-translationid
out-translationid
trust-me disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me disabled
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate 0
early-media-allow
invalidate-registrations disabled
rfc2833-mode none
rfc2833-payload 0
codec-policy
enforcement-profile
refer-call-transfer disabled
reuse-connections NONE
tcp-keepalive none
tcp-reconn-interval 0
max-register-burst-rate 0
register-burst-window 0
sip-profile
sip-isup-profile

```

We will now configure the ECB as the second session agent.

```

ORACLE-SBC (session-router) # ip-address 192.168.1.90
ORACLE-SBC (session-router) # port 5060
ORACLE-SBC (session-router) # realm-id ECB-Peer
ORACLE-SBC (session-router) # transport-method StaticTCP
ORACLE-SBC (session-router) # ping-method OPTIONS+hops=0
Oracle-SBC (session-agent) # refer-call-transfer enabled
ORACLE-SBC (session-router) # done

```

```
session-agent
  hostname
  ip-address          192.168.1.90
  port                5060
  state               enabled
  app-protocol        SIP
  app-type
  transport-method    StaticTCP
  realm-id             ECB-Peer
  egress-realm-id
  description
  carriers
  allow-next-hop-lp   enabled
  constraints          disabled
  max-sessions         0
  max-inbound-sessions 0
  max-outbound-sessions 0
  max-burst-rate      0
  max-inbound-burst-rate 0
  max-outbound-burst-rate 0
  max-sustain-rate    0
  max-inbound-sustain-rate 0
  max-outbound-sustain-rate 0
  min-seizures        5
  min-asr              0
  time-to-resume      0
  ttr-no-response     0
  in-service-period   0
  burst-rate-window   0
  sustain-rate-window 0
  req-uri-carrier-mode None
  proxy-mode
  redirect-action
  loose-routing        enabled
  send-media-session   enabled
  response-map
  ping-method          OPTIONS;hops=0
  ping-interval        30
  ping-send-mode       keep-alive
  ping-all-addresses  disabled
```

```

ping-in-service-response-codes
out-service-response-codes
media-profiles
in-translationid
out-translationid
trust-me                disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me             disabled
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate    0
early-media-allow
invalidate-registrations     disabled
rfc2833-mode                 none
rfc2833-payload              0
codec-policy
enforcement-profile
refer-call-transfer          enabled
reuse-connections            NONE
tcp-keepalive                none
tcp-reconn-interval         0
max-register-burst-rate     0
register-burst-window        0
sip-profile
sip-isup-profile

```

Configure SIP routing

To configure the SIP routing, use the `local-policy` command under the session-router branch. To enter the session-router branch from session-agent, you issue the `exit` command, followed by the `local-policy` command.

We will first configure the route from the gateway to the ECB.

```

ORACLE-SBC(session-agent)# exit
ORACLE-SBC(session-router)# local-policy
ORACLE-SBC(local-policy)# from-address *
ORACLE-SBC(local-policy)# to-address *
ORACLE-SBC(local-policy)# source-realm SIP-trunk
ORACLE-SBC(local-policy)# policy-attributes
ORACLE-SBC(local-policy-attributes)#next-hop 192.168.1.90
ORACLE-SBC(local-policy-attributes)# realm ECB-Peer
ORACLE-SBC(local-policy-attributes)# app-protocol sip
ORACLE-SBC(local-policy-attributes)# done

```

```

policy-attribute
      next-hop          192.168.1.90
      realm             ECB-Peer
      action            none
      terminate-recursion disabled
      carrier
      start-time        0000
      end-time          2400
      days-of-week      U-S
      cost              0
      app-protocol      SIP
      state             enabled
      methods
      media-profiles
      lookup            single
      next-key
      eloc-str-lkup     disabled
      eloc-str-match

```

```

ORACLE-SBC(local-policy-attributes)# exit
ORACLE-SBC(local-policy)# done
local-policy
      from-address
      to-address
      source-realm     SIP-trunk
      description
      activate-time    N/A

```

```

deactivate-time      N/A
state                enabled
policy-priority      none

policy-attribute
  next-hop           192.168.1.90
  realm              ECB-Peer
  action             none
  terminate-recursion disabled
  carrier
  start-time         0000
  end-time           2400
  days-of-week       U-S
  cost               0
  app-protocol       SIP
  state              enabled
  methods
  media-profiles
  lookup             single
  next-key
  eloc-str-lkup      disabled
  eloc-str-match

```

We will first configure the route from the ECB to the gateway.

```

ORACLE-SBC(local-policy)# from-address *
ORACLE-SBC(local-policy)# to-address *
ORACLE-SBC(local-policy)# source-realm ECB-Peer
ORACLE-SBC(local-policy)# policy-attributes
ORACLE-SBC(local-policy-attributes)# next-hop 10.10.1.8
ORACLE-SBC(local-policy-attributes)# realm SIP-trunk
ORACLE-SBC(local-policy-attributes)# app-protocol sip
ORACLE-SBC(local-policy-attributes)# done
policy-attribute
  next-hop           10.10.1.8
  realm              SIP-trunk
  action             none
  terminate-recursion disabled
  carrier
  start-time         0000
  end-time           2400
  days-of-week       U-S

```

```
cost 0
app-protocol SIP
state enabled
methods
media-profiles
lookup single
next-key
eloc-str-lkup disabled
eloc-str-match
```

```
ORACLE-SBC(local-policy-attributes)# exit
ORACLE-SBC(local-policy)# done
```

```
local-policy
  from-address *
  to-address *
  source-realm ECB-Peer
  description
  activate-time N/A
  deactivate-time N/A
  state enabled
  policy-priority none
  last-modified-by admin@172.41.0.11
  last-modified-date 2012-03-06 11:43:03
  policy-attribute
    next-hop 10.10.1.8
    realm SIP-trunk
    action none
    terminate-recursion disabled
    carrier
    start-time 0000
    end-time 2400
    days-of-week U-S
    cost 0
    app-protocol
    state enabled
    methods
    media-profiles
```

lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	

We will need a route to handle call transfer and refer scenarios (local refer handling by the E-SBC) when Lync client 1 refers/transfers the call to Lync Client 2.

```

local-policy
  from-address
  to-address *
  source-realm 192.168.1.90
  description SIP-trunk
  activate-time For referred party header
  deactivate-time N/A
  state N/A
  policy-priority enabled
  last-modified-by none
  last-modified-date admin@console
  policy-attribute 2012-02-28 13:05:51
    next-hop 192.168.1.90
    realm ECB-Peer
    action replace-uri
    terminate-recursion disabled
    carriers
    start-time 0000
    end-time 2400
    days-of-week U-S
    cost 0
    app-protocol SIP
    state enabled
    methods
    media-profiles
    lookup single
    next-key
    eloc-str-lkup disabled
    eloc-str-match

```

Configure Media handling

To configure the media handling, use the **steering-pool** command under the media-manager branch. To enter the steering-pool branch from local-policy, you issue the **exit** command twice, followed by the **media-manager** then the **steering-pool** command.

You will use the same IP address for the steering pool as the one used for the SIP interface. Note that the port ranges provide a means of limiting the number of concurrent media sessions within a given realm. For example, assigning 100 ports to a realm would limit it to 50 concurrent bidirectional calls, where two ports are assigned (one per unidirectional media stream).

```
ORACLE-SBC(local-policy)# exit
ORACLE-SBC(session-router)# exit
ORACLE-SBC(configure)# media-manager
ORACLE-SBC(media-manager)# steering-pool
ORACLE-SBC(steering-pool)# ip-address 192.168.1.130
ORACLE-SBC(steering-pool)# start-port 30000
ORACLE-SBC(steering-pool)# end-port 40000
ORACLE-SBC(steering-pool)# realm-id ECB-Peer
ORACLE-SBC(steering-pool)# network-interface slp0:0
ORACLE-SBC(steering-pool)# done
steering-pool
  ip-address          192.168.1.130
  start-port          30000
  end-port             40000
  realm-id             ECB-Peer
  network-interface    slp0:0
```

You will now configure the media handling for the pstn realm

```
ORACLE-SBC(steering-pool)# ip-address 192.20.0.108
ORACLE-SBC(steering-pool)# start-port 40000
ORACLE-SBC(steering-pool)# end-port 50000
ORACLE-SBC(steering-pool)# realm-id SIP-trunk
ORACLE-SBC(steering-pool)# network-interface s0p0:0
ORACLE-SBC(steering-pool)# done
steering-pool
  ip-address          192.20.0.108
  start-port          40000
  end-port             50000
  realm-id             SIP-trunk
  network-interface    s0p0:0
```

Configure Sip-manipulations and translation rules

To ensure that the E-SBC is doing topology hiding and replacing host-portions in SIP URIs of From and To headers, a sip manipulation will need to be created and configured as the out-manipulation id on the sip-interface.

The sip-manipulation element can be found under the session-router element.

```

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

```

The sip-manipulation then needs to be applied on the realm or sip-interface or session-agent towards the trunk and ECB side. We apply it on the sip-interface here:

```
ORACLE-SBC(session-router)# sip-interface
Oracle-SBC(sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 2
Oracle-SBC(sip-interface)# out-manipulationid NATting
Oracle-SBC(sip-interface)# done

sip-interface
state enabled
realm-id SIP-trunk
description SIP Trunk-facing (Outside)
sip-port
address 192.20.0.108
port 5060
transport-protocol UDP
tls-profile
allow-anonymous all
ims-aka-profile

carriers
trans-expire 0
invite-expire 0
max-redirect-contacts 0
proxy-mode
redirect-action
contact-mode none
nat-traversal none
nat-interval 30
tcp-nat-interval 90
registration-caching disabled
min-reg-expire 300
registration-interval 3600
route-to-registrar disabled
secured-network disabled
teluri-scheme disabled
uri-fqdn-domain
options
```

trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	NATting
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	
route-unauthorized-calls	
tcp-keepalive	none
add-sdp-invite	disabled
add-sdp-profiles	
sip-profile	
sip-isup-profile	

```

ORACLE-SBC(session-router)# sip-interface
Oracle-SBC(sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 1
Oracle-SBC(sip-interface)# out-manipulationid NATting
Oracle-SBC(sip-interface)# done

sip-interface
state enabled
realm-id ECB-Peer
description ECB-Facing(Inside)
sip-port
address 192.168.1.130
port 5060
transport-protocol TCP
tls-profile
allow-anonymous all
ims-aka-profile
carriers
trans-expire 0
invite-expire 0
max-redirect-contacts 0
proxy-mode
redirect-action
contact-mode none
nat-traversal none
nat-interval 30
tcp-nat-interval 90
registration-caching disabled
min-reg-expire 300
registration-interval 3600
route-to-registrar disabled
secured-network disabled
teluri-scheme disabled
uri-fqdn-domain
trust-mode all
max-nat-interval 3600

```

```
nat-int-increment      10
nat-test-increment     30
sip-dynamic-hnt        disabled
stop-recurse           401,407
port-map-start         0
port-map-end           0
in-manipulationid
out-manipulationid     NATting
manipulation-string
manipulation-pattern
sip-ims-feature        disabled
operator-identifier
anonymous-priority     none
max-incoming-conns    0
per-src-ip-max-incoming-conns 0
inactive-conn-timeout  0
untrusted-conn-timeout 0
network-id
ext-policy-server
default-location-string
charging-vector-mode   pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode         none
implicit-service-route disabled
rfc2833-payload        101
rfc2833-mode           transparent
constraint-name
response-map
local-response-map
ims-aka-feature        disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive          none
add-sdp-invite         disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```

Lync does send E164 numbers in the To and From headers whereas the trunk does not accept E164 numbers. Hence we need a translation rule on the E-SBC to translate the E164 phone numbers into a regular one by stripping off the +1 from the phone numbers. The translation rule then needs to be added on the session-translation which then gets called from the trunk session-agent.

```
Oracle-SBC(sip-interface)# exit
Oracle-SBC(session-router)# translation-rules
Oracle-SBC(translation-rules)# id stripplus1
Oracle-SBC(translation-rules)# type delete
Oracle-SBC(translation-rules)# delete-string +1
Oracle-SBC(translation-rules)# done

translation-rules
  id                stripplus1
  type              delete
  add-string
  add-index         0
  delete-string     +1
  delete-index      0

Oracle-SBC(translation-rules)# exit
Oracle-SBC(session-router)# session-translation
Oracle-SBC(session-translation)# id stripplus1
Oracle-SBC(session-translation)# rules-calling stripplus1
Oracle-SBC(session-translation)# rules-called stripplus1
Oracle-SBC(session-translation)# done

session-translation
  id                stripplus1
  rules-calling     stripplus1
  rules-called      stripplus1
  last-modified-by  admin@console
  last-modified-date 2012-01-26 18:28:59

Oracle-SBC(session-translation)# exit
Oracle-SBC(session-router)# session-agent
Oracle-SBC(session-agent)# sel
<hostname>:
1: 10.10.1.8          realm=SIP-trunk
2: 192.168.1.130     realm=ECB-Lync-Peer ip=192.168.1.130
```

```
selection: 1
Oracle-SBC(session-agent)# out-translationid stripplus1
Oracle-SBC(session-agent)# done
```

```
session-agent
  hostname                10.10.1.8
  ip-address              10.10.1.8
  port                    5060
  state                   enabled
  app-protocol            SIP
  app-type
  transport-method       UDP
  realm-id                SIP-trunk
  egress-realm-id
  description
  carriers
  allow-next-hop-lp      enabled
  constraints             disabled
  max-sessions            0
  max-inbound-sessions   0
  max-outbound-sessions  0
  max-burst-rate         0
  max-inbound-burst-rate 0
  max-outbound-burst-rate 0
  max-sustain-rate       0
  max-inbound-sustain-rate 0
  max-outbound-sustain-rate 0
  min-seizures           5
  min-asr                 0
  time-to-resume         0
  ttr-no-response        0
  in-service-period      0
  burst-rate-window      0
  sustain-rate-window    0
  req-uri-carrier-mode   None
  proxy-mode
  redirect-action
  loose-routing           enabled
  send-media-session     enabled
  response-map
```

```
ping-method
ping-interval          0
ping-send-mode         keep-alive
ping-all-addresses    disabled
ping-in-service-response-codes
out-service-response-codes
media-profiles
in-translationid
out-translationid      stripplus1
trust-me               disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me           disabled
in-manipulationid
out-manipulationid     NATting
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate  0
early-media-allow
invalidate-registrations  disabled
rfc2833-mode          none
rfc2833-payload       0
codec-policy
enforcement-profile
refer-call-transfer    disabled
reuse-connections      NONE
tcp-keepalive         none
tcp-reconn-interval    0
max-register-burst-rate  0
register-burst-window   0
sip-profile
sip-isup-profile
```

Configure SIP PRACK Interworking

In order to establish an early media session for outbound calls, Lync Server gateway specification mandates the PSTN gateways to offer a reliable provisional response and for inbound calls offer INVITEs with a supported header. The E-SBC can interwork and provide RFC 3262 PRACK interworking towards Lync and it is a mandatory configuration in all Oracle – Microsoft Lync deployments. For this, the following need to be configured

- Configure option 100rel-interworking on the sip-interface facing ECB
- Configure a sip-feature to pass the 100-rel in Supported and Required headers
- Configure a sip-manipulation to add a Require:100rel header in incoming SIP INVITE from mediation server and delete the Supported:100rel header.

```
ORACLE-SBC(session-router)# sip-interface
Oracle-SBC(sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 1
Oracle-SBC(sip-interface)# options 100rel-interworking
Oracle-SBC(sip-interface)# done

sip-interface
  state                enabled
  realm-id             ECB-Peer
  description          ECB-Facing(Inside)
  sip-port
    address            192.168.1.130
    port               5060
    transport-protocol TCP
    tls-profile
    allow-anonymous    all
    ims-aka-profile
  carriers
  trans-expire         0
  invite-expire        0
  max-redirect-contacts 0
  proxy-mode
  redirect-action
```

contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	100rel-interworking
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	NATting
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101

```

rfc2833-mode          transparent
constraint-name
response-map
local-response-map
ims-aka-feature       disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive         none
add-sdp-invite        disabled
add-sdp-profiles
sip-profile
sip-isup-profile

```

Configure Sip-feature to pass Supported and Require headers in SIP messages. The sip-feature element can be found under session-router.

```

ORACLE-SBC (session-router) #sip-feature
ORACLE-SBC (sip-feature) #name 100rel
ORACLE-SBC (sip-feature) #realm pstn
ORACLE-SBC (sip-feature) # support-mode-inbound Pass
ORACLE-SBC (sip-feature) # require-mode-inbound Pass
ORACLE-SBC (sip-feature) # proxy-require-mode-inbound Pass
ORACLE-SBC (sip-feature) # support-mode-outbound Pass
ORACLE-SBC (sip-feature) # require-mode-outbound Pass
ORACLE-SBC (sip-feature) # proxy-require-mode-outbound Pass
ORACLE-SBC (sip-feature) # done

sip-feature
  name          100rel
  realm         SIP-Trunk
  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

```

Configure the sip-manipulation For early media to delete the Supported header and add the Required header and apply it as an in-manipulation in the interface facing ECB.

```

sip-manipulation
  name                               Forearlymedia
  description
  split-headers
  join-headers
  header-rule
    name                               delsupported
    header-name                         Supported
    action                               delete
    comparison-type                     case-sensitive
    msg-type                             request
    methods                              INVITE
    match-value
    new-value
  header-rule
    name                               addrequireinINVITE
    header-name                         Require
    action                               add
    comparison-type                     case-sensitive
    msg-type                             request
    methods                              INVITE
    new-value                            100rel

ORACLE-SBC (session-router)# sip-interface
Oracle-SBC (sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 1
Oracle-SBC (sip-interface)# in-manipulationid Forearlymedia
Oracle-SBC (sip-interface)# done

sip-interface
  state                               enabled
  realm-id                             ECB-Peer
  description                           ECB-Facing (Inside)
  sip-port
  address                               192.168.1.130
```

port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	Forearlymedia
out-manipulationid	NATting
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0

```

network-id
ext-policy-server
default-location-string
charging-vector-mode          pass
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode                none
implicit-service-route        disabled
rfc2833-payload               101
rfc2833-mode                  transparent
constraint-name
response-map
local-response-map
ims-aka-feature               disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive                 none
add-sdp-invite                disabled
add-sdp-profiles
sip-profile
sip-isup-profile

```

Configuring REFER Handling for Transfers

Lync Server authorizes transfers of all Lync initiated calls whether it is Lync to Lync or Lync to PSTN. The E-SBC provides REFER handling by terminating the REFER from Lync and generating an INVITE for the referred party back towards the Lync Mediation server. Lync then process the INVITE, authorizes the call transfer and sends either a new INVITE (for calls transferred to PSTN) to the E- SBC or transfers call internally to the transferred Lync client

To handle the call transfer and refer scenarios – when Lync client 1 refers/transfers the call to Lync Client 2 or to a party on the PSTN, we will need a route to the ECB.

```

local-policy
  from-address                 *
  to-address                   192.168.1.90
  source-realm                 Trunk
  description                  For referred party
  activate-time
  deactivate-time
  state                        enabled
  policy-priority              none

```

```

policy-attribute
  next-hop          192.168.1.90
  realm             ECB-Peer
  action            replace-uri
  terminate-recursion disabled
  carrier
  start-time        0000
  end-time          2400
  days-of-week      U-S
  cost              0
  state             enabled
  app-protocol      SIP
  methods
  media-profiles
  lookup            single
  next-key
  eloc-str-lkup     disabled
  eloc-str-match

```

Addressing No Ringback tone on Transfers

During call transfer to a PSTN party, the transfer completes but the calling party does not hear a ring back tone during the process of transfer. The INVITE Lync sends to the E-SBC to initiate the transfer contains the SDP attribute, a=inactive which is forwarded to the trunk and as a result of which the E-SBC cannot play the ring back tone to the original PSTN caller (while call is being transferred). A send only attribute is required for MoH and transfer scenarios for the calling party to be able to hear ringback or MoH when it is kept on hold. The E-SBC is able to signal appropriately towards the SIP trunk by changing the a=inactive SDP attribute in the INVITE to sendonly towards PSTN.

Sip manipulations are configured to make the necessary changes. The manipulation Changeinactosendonly is configured to change the SDP attribute from a=inactive to a=sendonly in the INVITES sent to the calling party for transfer.

```

sip-manipulation
  name              Changeinactosendonly
  description       Change inactive to sendonly for transfer
  split-headers
  join-headers
  header-rule
    name            changeSDP
    header-name     Content-Type
    action          manipulate
    comparison-type case-sensitive
    msg-type        request
    methods         INVITE
    match-value
    new-value

```

```

element-rule
  name                inacttosendonly
  parameter-name      application/sdp
  type                mime
  action              find-replace-all
  match-val-type      any
  comparison-type     pattern-rule
  match-value         a=inactive
  new-value           a=sendonly

```

The above manipulation is applied as a nested manip in the manipulation – Forearlymedia that is applied inbound on the sip-interface facing ECB.

```

sip-manipulation
  name                Forearlymedia
  description
  split-headers
  join-headers
  header-rule
    name              delsupported
    header-name       Supported
    action            delete
    comparison-type   case-sensitive
    msg-type          request
    methods           INVITE
    match-value
    new-value
  header-rule
    name              addrequireinINVITE
    header-name       Require
    action            add
    comparison-type   case-sensitive
    msg-type          request
    methods           INVITE
    match-value
    new-value         100rel
  header-rule
    name              inactosendonly
    header-name       From
    action            sip-manip
    comparison-type   case-sensitive
    msg-type          request
    methods
    match-value
    new-value         Changeinactosendonly

```

We utilize the local playback feature of the E-SBC to play ring back tone during transfers. The ringback tone is played based on REFER. You must upload a file containing to /code/media on the E-SBC for the media you want played. This file must be raw media binary containing data for the desired codec. A separate file is required for each different codec type, even if the media itself is the same.

The playback configuration is defined listing the media files that you want to play. The playback-config element is configured under media-manager.

```
playback-config
  name                transferrbt
  entry
    encoding          PCMU
    filename           US_ringbackPCMU.raw
    bytes-per-sec     8000
```

The playback options can be applied to realms, sip-interfaces or session agents using the `spl-options` command.

```
ORACLE-SBC(session-router)# sip-interface
Oracle-SBC(sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 1
Oracle-SBC(sip-interface)# spl-options playback-on-refer="transferrbt"
Oracle-SBC(sip-interface)# done

sip-interface
  state                enabled
  realm-id             ECB-Peer
  description          ECB-Facing(Inside)
  sip-port
    address            192.168.1.130
    port               5060
    transport-protocol TCP
    tls-profile
    allow-anonymous    all
    ims-aka-profile
  carriers
  trans-expire         0
  invite-expire        0
  max-redirect-contacts 0
  proxy-mode
  redirect-action
  contact-mode         none
```

nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	100rel-interworking
spl-options	playback-on-refer="transferrbt"
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	NATting
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass
charging-function-address-mode	pass
ccf-address	
ecf-address	
term-tgrp-mode	none
implicit-service-route	disabled
rfc2833-payload	101
rfc2833-mode	transparent
constraint-name	
response-map	
local-response-map	
ims-aka-feature	disabled
enforcement-profile	

```
route-unauthorized-calls
tcp-keepalive           none
add-sdp-invite          disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```

Verify configuration integrity

You will verify your configuration referential integrity before saving and activating it with the `verify-config` command. This command is available from Superuser Mode. To enter the Superuser Mode from session-agent, you issue the `exit` command three times.

```
ORACLE-SBC(session-agent)# exit
ORACLE-SBC(session-router)# exit
ORACLE-SBC(configuration)# exit
ORACLE-SBC# verify-config
-----
Verification successful! No errors nor warnings in the configuration
```

Save and activate your configuration

You will now save your configuration with the `save-config` command. This will make it persistent through reboots, but it will not take effect until after you issue the `activate-config` command.

```
ORACLE-SBC# save-config
checking configuration
Save-Config received, processing.
waiting 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'.
ORACLE-SBC# activate-config
Activate-Config received, processing.
waiting for request to finish
Setting phy0 on Slot=0, Port=0, MAC=00:08:25:03:FC:43,
VMAC=00:08:25:03:FC:43
Setting phy1 on Slot=1, Port=0, MAC=00:08:25:03:FC:45,
VMAC=00:08:25:03:FC:45
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
```

E-SBC configuration is now complete.

Interoperability testing

Interoperability between Avaya and Lync

The following observations were made during testing –

- In versions prior to the 6.2 release, Avaya SM uses INVITEs for session refreshes. These INVITEs do not include SDP. This results in an interoperability issue with Microsoft Lync when media bypass is enabled as it does not accept INVITEs without SDP.
- In releases 6.2 and later, UPDATE message can be used for session refresh, to enable UPDATEs from the Avaya SM instead of the INVITEs without SDP, the signaling group was configured as follows

change signaling-group 2 Page 1 of 2

SIGNALING GROUP

Group Number: 2 Group Type: sip

IMS Enabled? Transport Method:

Q-SIP?

IP Video? Enforce SIPS URI for SRTP?

Peer Detection Enabled? Peer Server: SM

Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers?

Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers?

Near-end Node Name: Far-end Node Name:

Near-end Listen Port: Far-end Listen Port:

Far-end Network Region:

Far-end Domain:

Incoming Dialog Loopbacks: Bypass If IP Threshold Exceeded?

DTMF over IP: RFC 3389 Comfort Noise?

Session Establishment Timer(min): **Direct IP-IP Audio Connections?**

Enable Layer 3 Test? **IP Audio Hairpinning?**

Alternate Route Timer(sec):

- With UPDATES enabled, the calls flows for hold and resume were as follows:
 - When Avaya server places a call on hold during the inbound calls, the call flow does not include INVITE messages to signal the hold and resume. It utilizes the UPDATE messages.
 - When Avaya server performs a call hold and resume during the outbound calls, it sends INVITE message to signal the hold and resume.
- For lync initiated transfers, Avaya does not support REFER from Lync. The Refer-To in the REFER from Lync does not contain the uri-user. Avaya sends a **403 Forbidden (Refer-to user is null)** error.
 - In Lync environments with REFER enabled, the calls will need to be sent through the E-SBC to the ECB for REFER termination on the E-SBC.
 - Or Lync needs to be reconfigured to use INVITEs for transfer scenarios.

Interoperability between CUCM and Lync

During an inbound call from Lync, when CUCM performs a call hold and resume, CUCM sends an INVITE without SDP. When media bypass is enabled on Lync, the mediation server does not accept INVITEs without SDP and responds with *488 GatewayCall is not in Connected State*. As a result the hold and resume fails. To resolve this issue we either have to insert SDP or disable media bypass. To insert SDP, the call will need to be placed through a E-SBC.

When media bypass is disabled, Lync accepts the INVITE when it is sent without SDP. However, during call resume, the INVITE sent from CUCM contains SDP without the telephone event. Lync responds with the error *488 Invalid SDP: Gateway ParseSdpOffer Error: No DTMF support on Gateway side*.

To resolve this issue, we configure a sip-manipulation to add the SDP line for telephone event if it does not exist before forwarding the INVITE to Lync.

```

sip-manipulation
  name                fixSDP
  description         To bypass the 488 due to missing DTMF from CUCM during hold
  split-headers
  join-headers
  header-rule
    name              Checkfordtmf
    header-name       Content-type
    action            store
    comparison-type   case-sensitive
    msg-type          any
    methods           INVITE
    match-value
    new-value
    element-rule
      name            Checkdtmfexists
      parameter-name  application/sdp
      type            mime
      action          store
      match-val-type  any
  
```

```

comparison-type      case-sensitive
match-value          (a=rtmpmap:101 telephone-event/8000)
new-value

header-rule
  name                AddPtime10
  header-name         Content-Type
  action              manipulate
  comparison-type     boolean
  msg-type            any
  methods             INVITE
  match-value         !$Checkfordtmf.$Checkdtmfexists
  new-value
  element-rule
    name              Adddtmf
    parameter-name    application/sdp
    type              mime
    action            find-replace-all
    match-val-type    any
    comparison-type   pattern-rule
    match-value       (\\,*)
    new-value         $0+"a=rtmpmap:101 telephone-event/8000"+$CRLF+"a=fmtp:101 0-15"+$CRLF

header-rule
  name                Modifymline
  header-name         From
  action              sip-manip
  comparison-type     case-sensitive
  msg-type            request
  methods
  match-value
  new-value           Modmline
last-modified-by     web@
last-modified-date   2014-07-23 15:43:00

```

This manipulation is then nested into the HMR, `HMRtowardsLync` that is applied in the outbound directions towards the Lync servers.

```

sip-manipulation
  name                HMRtowardsLync
  description         HMR NAT+deleting the blines+addDTMF
  split-headers
  join-headers
  header-rule
    name              doNAT
    header-name       From
    action            sip-manip
    comparison-type   case-sensitive
    msg-type          any
    methods
    match-value
    new-value         NATting
  header-rule
    name              deleteblines

```

header-name	From
action	sip-manip
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	Delblines
header-rule	
name	adddtmfines
header-name	From
action	sip-manip
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	fixSDP

Please note that when CUCM calls Lync and the call is placed on hold by Lync, the issue mentioned above does not occur.

In Lync environments with media bypass and REFER enabled, the calls will need to be sent through the E-SBC to the ECB to utilize the add-sdp-profile and REFER termination features of the E-SBC.

Test Plan & Results

Test Plan

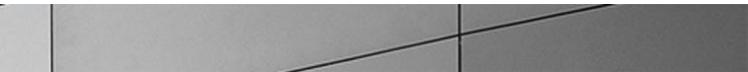
The testing was done with TCP/RTP. Lync 2010 and Lync 2013 servers are two different environments independent of each other. Lync 2010 has media bypass and refer enabled. Lync 2013 setup has media bypass and refer disabled.

The test plan consisted of the following test cases.

Test case	Result	Notes
Basic inbound and Outbound calls		
PSTN calls Lync 2010	Pass	
PSTN calls Lync 2013	Pass	
PSTN calls Avaya	Pass	
PSTN calls CUCM	Pass	
Lync 2013 calls CUCM	Pass	
Lync 2013 calls Avaya	Pass	
Lync 2013 calls PSTN	Pass	
CUCM calls Avaya	Pass	
CUCM calls PSTN	Pass	
CUCM calls Lync 2010	Pass	
CUCM calls Lync 2013	Pass	
Lync 2010 calls CUCM	Pass	
Lync 2010 calls Avaya	Pass	
Lync 2010 calls PSTN	Pass	
Avaya calls PSTN	Pass	
Avaya calls Lync 2010	Pass	
Avaya calls Lync 2013	Pass	
Avaya calls CUCM	Pass	
Back up route (replicating the SAG in ECB)		
PSTN calls Lync 2010 (home agent med1 paused)	Pass	

4 digit dialing		
Lync 2010 calls Avaya	Pass	
Lync 2010 calls CUCM	Pass	
CUCM calls Avaya	Pass	
CUCM calls Lync 2010	Pass	
Avaya calls Lync 2010	Pass	
Avaya calls CUCM	Pass	
Transfers		
PSTN calls Lync 2010 and Lync transfers to PSTN	Pass	
PSTN calls Lync 2013 and Lync transfers to PSTN	Pass	
PSTN calls CUCM and CUCM transfers to PSTN	Pass	The transfer was successful but the calling number displayed to the second PSTN party is that of Cisco phone and not the Original PSTN party
PSTN calls Lync 2010 and Lync transfers to CUCM using 4 digit dial	Pass	
PSTN calls Avaya and Avaya transfers to Lync using 4 digit dial	Pass	
PSTN calls Avaya and Avaya transfers to Lync 2013 using 10 digits	Pass	
PSTN calls Lync 2013 and Lync transfers to CUCM using 10 digit dial	Pass	
Lync 2013 calls Lync 2010 and transfers to CUCM	Pass	
Lync 2013 calls CUCM(10 digit) and CUCM transfers to Lync 2010 (4 digit) - no media bypass on Lync 2013	Pass	The transfer was successful but the calling number displayed to the second PSTN party is that of Cisco phone and not the Original PSTN party
Lync calls Avaya and transfer to CUCM (refer disabled)	Pass	Transfer works with refer disabled
CUCM calls Avaya and Avaya transfers to Lync	Pass	
CUCM calls Lync 2013 and Lync transfers to Lync 2010 (refer disabled)	Pass	
CUCM calls Avaya and CUCM transfers to Lync	Pass	
CUCM calls Avaya and CUCM transfers to PSTN	Pass	
Avaya calls Lync and transfers to CUCM	Pass	

Call Hold/Resume		
PSTN calls Lync and Lync places call on hold	Pass	
PSTN call Avaya and Avaya places call on hold	Pass	
PSTN calls CUCM and CUCM places call on hold	Pass	
Lync 2013 calls Avaya and Lync places call on hold	Pass	
Lync 2013 calls Avaya and Avaya places call on hold	Pass	
Lync 2013 calls CUCM and CUCM places call on hold	Pass	During the call hold process, one of the re-invites with sdp from CUCM does not contain a lines for DTMF support to which Lync responds with a 488 DTMF not supported. To overcome this issue we have an HMR towards Lync to add the a lines for DTMF in SDP if not present
CUCM calls Lync 2010 and CUCM places the call on hold	Pass	
Avaya calls Lync 2013 and Avaya places the call on hold	Pass	
Lync 2010 calls Avaya and Avaya places the call on hold	Pass	
CUCM calls Avaya and CUCM places the call on hold	Pass	
Avaya calls PSTN and Avaya places the call on hold	Pass	
Avaya calls CUCM and Avaya places the call on hold	Pass	



Troubleshooting Tools

If you find that you are not able to complete calls or have problems with the test cases, there are a few tools available for Windows Server, Lync Server, and the E-SBC like logging and tracing which may be of assistance. In this section we will provide a list of tools which you can use to aid in troubleshooting any issues you may encounter.

Since we are concerned with communication between the Lync Server mediation server and the E-SBC we will focus on the troubleshooting tools to use between those devices if calls are not working or tests are not passing.

Microsoft Network Monitor (NetMon)

NetMon is a network protocol analyzer which is freely downloadable from Microsoft. It can be found at www.microsoft.com/downloads. NetMon could be installed on the Lync Server mediation server, the Lync Server Standard Edition server, or Enterprise Edition front end server.

Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from www.wireshark.org. Wireshark could be installed on the Lync Server mediation server, the Lync Server Standard Edition server, or MCS Enterprise Edition front end server.

Eventviewer

There are several locations in the event viewer where you can find valuable information to aid in troubleshooting issues with your deployment.

With the requirement that there is a completely functioning Lync Server with Enterprise Voice deployment in place, there are only a few areas in which one would use the Event Viewer for troubleshooting:

- The Enterprise Voice client;
- The Lync Server Front End server;
- A Lync Server Standard Edition Server; and
- A Lync Server Mediation Server.

On the Oracle Enterprise Communications Broker and Oracle Enterprise Session Border Controller

The Oracle Enterprise Session Border Controller and Oracle Enterprise Communications Broker provide 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:

```
ORACLE-SBC# reset sipd
ORACLE-SBC# notify sipd debug
ORACLE-SBC#
enabled SIP Debugging
ORACLE-SBC# 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 192.168.5.24
Connected to 192.168.85.55.
220 ORACLE-SBCFTP server (VxWorks 6.4) ready.
User (192.168.85.55:(none)): user
331 Password required for user.
Password: acme
230 User user logged in.
ftp> cd /ramdrv/logs
250 CWD command successful.
ftp> get sipmsg.log
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/sipmsg.log' (3353
bytes).
226 Transfer complete.
ftp: 3447 bytes received in 0.00Seconds 3447000.00Kbytes/sec.
ftp> get log.sipd
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/log.sipd' (204681
```

```
bytes).  
226 Transfer complete.  
ftp: 206823 bytes received in 0.11Seconds 1897.46Kbytes/sec.  
ftp> bye  
221 Goodbye.
```

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

Telnet

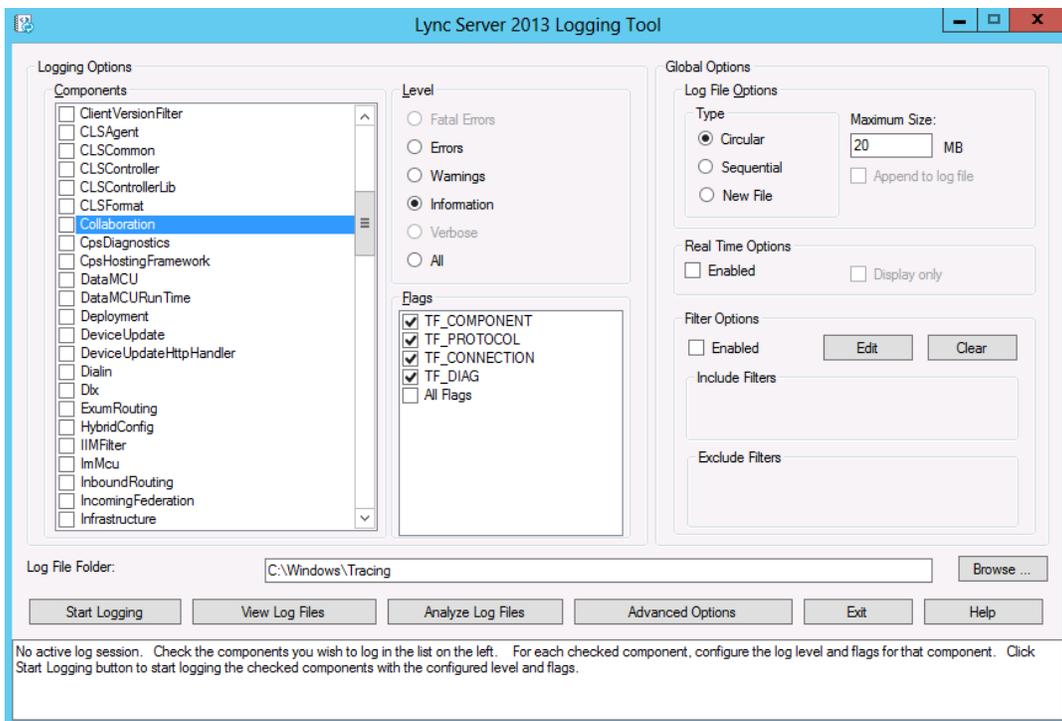
Since we are working within an architecture which uses bound TCP listening ports for functionality, the simplest form of troubleshooting can be seeing if the devices are listening on a particular port, as well as confirming that there is nothing blocking them such as firewalls. Ensure that you have a TELNET client available on a workstation as well as on the Lync Server mediation server.

The Lync Server mediation server will listen on TCP port 5067 by default for SIP signaling. In our example we are listening on 5060 on the PSTN facing NIC. From the Standard Edition pool or Enterprise Edition pool the Mediation Server would be listening on port 5061. Tests may include:

- Client to pool server: `telnet <servername> 5061`
- Pool server to Mediation Server: `telnet <servername> 5061`

Lync Server Logging Tool

The Lync Server 2013 Logging Tool provides internal traces and messaging between different Lync Server 2013 elements like Front-end, Mediation server, Lync Clients, etc. File name is OCSReskit.msi. Once installed, it can be accessed from any one of the Lync Server servers by running Start/Microsoft Lync Server 2013/Lync Server Logging Tool.



Appendix A

No Ring Back Tone heard for inbound calls from PSTN to MS Lync through E-SBC

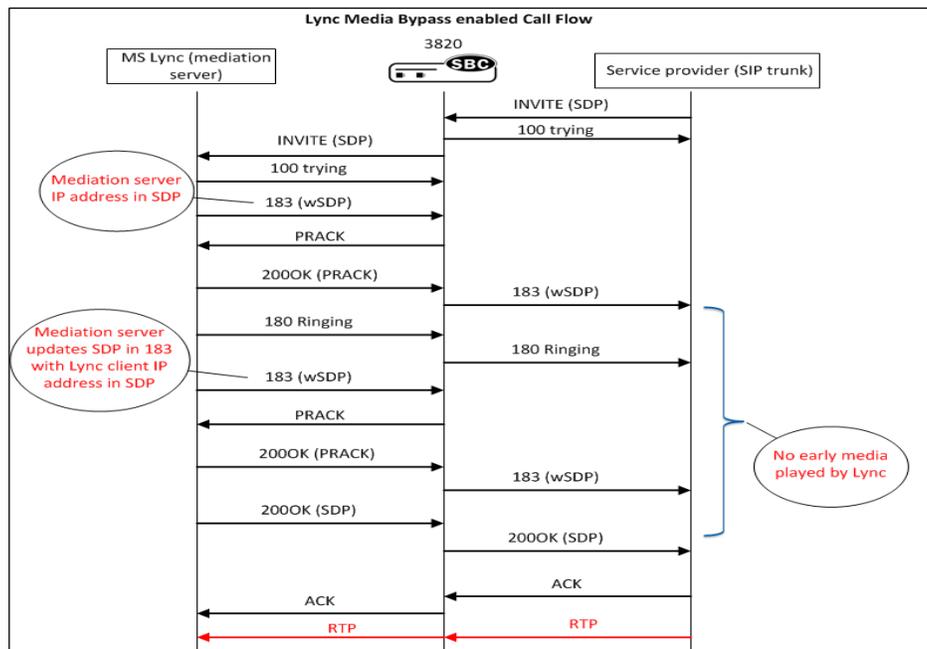
Recently, in some accounts where MS Lync and E-SBCs are deployed for enterprise voice and SIP trunk termination to an enterprise, there have been complaints of the PSTN caller hearing a silence when a call is placed from PSTN to a Lync user on the enterprise especially when Media Bypass is enabled on MS Lync

The configuration note below aims to explain this scenario briefly, steps taken to rectify this issue and proposed workaround by Oracle. The workaround is an interim solution while a permanent solution is being researched and developed by Oracle Engineering

Media Bypass

As explained earlier in the document, in order for Media Bypass to work, both Client and gateway (E-SBC) need to use the same RTP format, either SRTP (by default) or RTP. In default configuration of MS Lync, Lync client is required to use media encryption, so Media Bypass is mainly when media is encrypted (SRTP) and exchanged between Lync client and PSTN gateway (E-SBC).

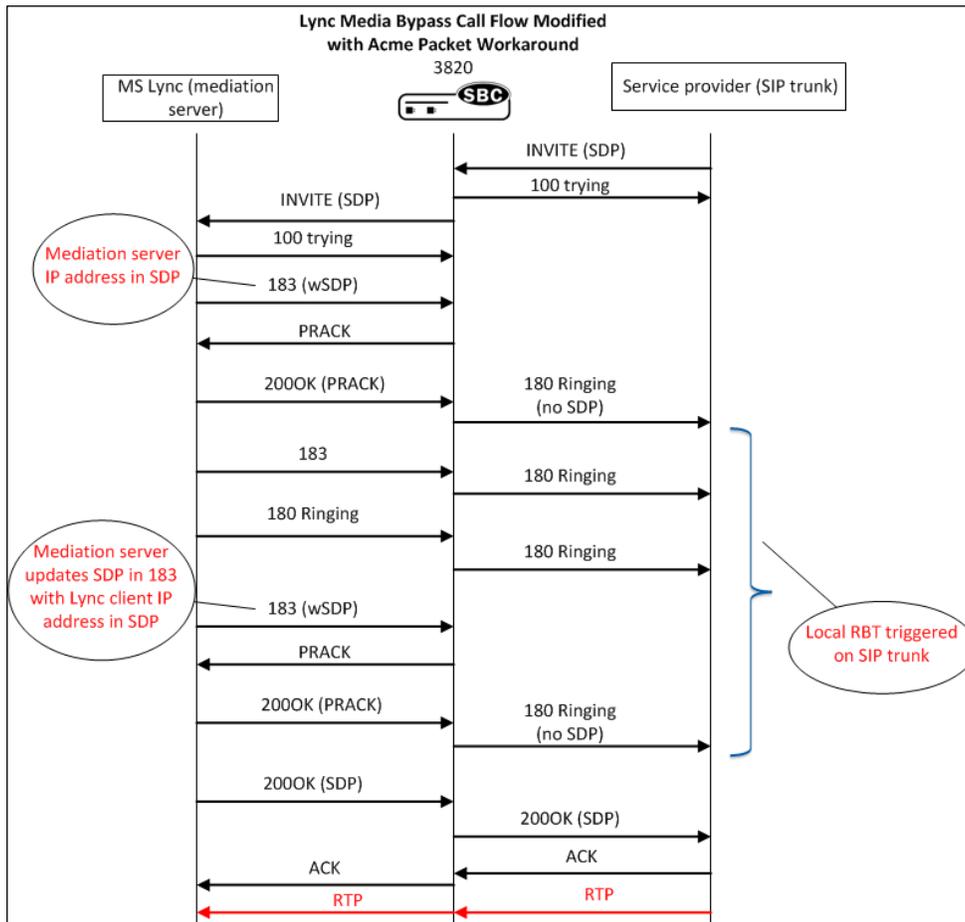
Signaling between mediation server and E-SBC is a little different (Two 183s with SDP coming from mediation server) when media bypass is enabled on Lync. The following is the call flow



Note that after signaling 183 with SDP, Lync never plays any early media and expects gateway (E-SBC) to signal appropriately to the SIP Trunk provider to follow RFC 3960 and play local RBT. The second 183w SDP coming from Mediation server which is forwarded to the SIP trunk and stops the local RBT which was started after 180 Ringing was sent, hence PSTN caller would hear a silence before Lync client answers call.

Acme Packet Work Around

The interim solution is to present 180 ringing (convert all 183s on lync side to 180 ringing towards SIP trunk and strip the SDP) to trigger RBT in ISUP. The call flow is modified with the help of Oracle's robust Sip Manipulation and Sip Response Map features to the following:



A sip manipulation, Stripsdp183, is configured to delete the SDP from the 183 messages.

```

sip-manipulation
  name                               Stripsdp183
  description                         For incoming 183 from Lync,
strip SDP
  split-headers
  join-headers
  header-rule
    name                               check183
    header-name                         @status-line
    action                               store
    comparison-type                     pattern-rule
    msg-type                             any
    methods
    match-value
    new-value
    element-rule
      name                               is183
      parameter-name
      type                               status-code
      action                             store
      match-val-type                     any
      comparison-type                   pattern-rule
      match-value                         183
      new-value
    header-rule
      name                               delSDP
      header-name                       Content-Type
      action                             manipulate
      comparison-type                   case-insensitive
      msg-type                             any
      methods
      match-value                       $check183.$is183
      new-value
    element-rule
      name                               del183SDP
      parameter-name                   application/sdp
      type                               mime
      action                             delete-element
      match-val-type                     any
      comparison-type                   boolean
      match-value
      new-value
    header-rule
      name                               delContentType
      header-name                       Content-Type
      action                             manipulate

```

comparison-type	boolean
msg-type	any
methods	
match-value	\$check183.\$is183
new-value	
element-rule	
name	delCT
parameter-name	*
type	header-param
action	delete-header
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	

The above manipulation is applied as a nested manip in the manipulation – Forearlymedia that is applied inbound on the sip-interface facing ECB.

sip-manipulation	
name	Forearlymedia
description	
split-headers	
join-headers	
header-rule	
name	delsupported
header-name	Supported
action	delete
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	
header-rule	
name	addrequireinINVITE
header-name	Require
action	add
comparison-type	case-sensitive
msg-type	request
methods	INVITE
match-value	
new-value	100rel
header-rule	
name	inactosendonly
header-name	From
action	sip-manip
comparison-type	case-sensitive

msg-type	request
methods	
match-value	
new-value	Changeinactosendonly
header-rule	
name	mod183
header-name	From
action	sip-manip
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	Stripsdp183

A sip response map is configured to change the 183s to 180 and applied on the sip-interface facing the trunk.

```

response-map
  name                change183to180
  entries
    recv-code         183
    xmit-code          180
    reason             Ringing
    method
    register-response-expires

ORACLE-SBC (session-router)# sip-interface
Oracle-SBC (sip-interface)# sel
<realm-id>:
1: ECB-Peer 192.168.1.130:5060
2: SIP-trunk 192.20.0.108:5060

selection: 1
Oracle-SBC (sip-interface)# response-map change183to180
Oracle-SBC (sip-interface)# done

sip-interface
  state                enabled
  realm-id             ECB-Peer
  description          ECB-Facing (Inside)
  sip-port
  address              192.168.1.130

```

port	5060
transport-protocol	TCP
tls-profile	
allow-anonymous	all
ims-aka-profile	
carriers	
trans-expire	0
invite-expire	0
max-redirect-contacts	0
proxy-mode	
redirect-action	
contact-mode	none
nat-traversal	none
nat-interval	30
tcp-nat-interval	90
registration-caching	disabled
min-reg-expire	300
registration-interval	3600
route-to-registrar	disabled
secured-network	disabled
teluri-scheme	disabled
uri-fqdn-domain	
options	100rel-interworking
spl-options	playback-on-refer="transferrbt"
trust-mode	all
max-nat-interval	3600
nat-int-increment	10
nat-test-increment	30
sip-dynamic-hnt	disabled
stop-recurse	401,407
port-map-start	0
port-map-end	0
in-manipulationid	
out-manipulationid	NATting
manipulation-string	
manipulation-pattern	
sip-ims-feature	disabled
operator-identifier	
anonymous-priority	none
max-incoming-conns	0
per-src-ip-max-incoming-conns	0
inactive-conn-timeout	0
untrusted-conn-timeout	0
network-id	
ext-policy-server	
default-location-string	
charging-vector-mode	pass

```
charging-function-address-mode pass
ccf-address
ecf-address
term-tgrp-mode none
implicit-service-route disabled
rfc2833-payload 101
rfc2833-mode transparent
constraint-name
response-map change183to180
local-response-map
ims-aka-feature disabled
enforcement-profile
route-unauthorized-calls
tcp-keepalive none
add-sdp-invite disabled
add-sdp-profiles
sip-profile
sip-isup-profile
```

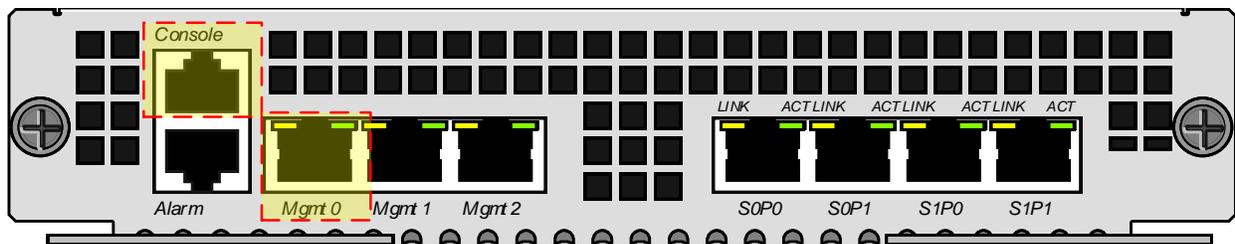
Appendix B

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.

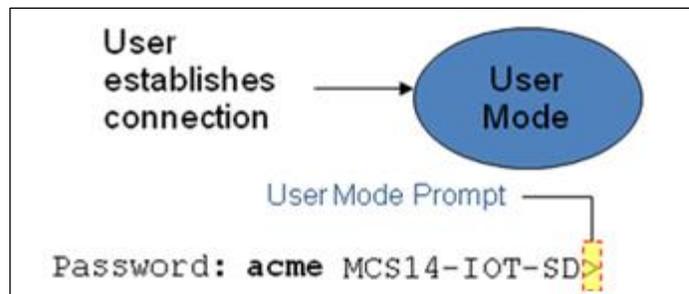


ACLI Basics

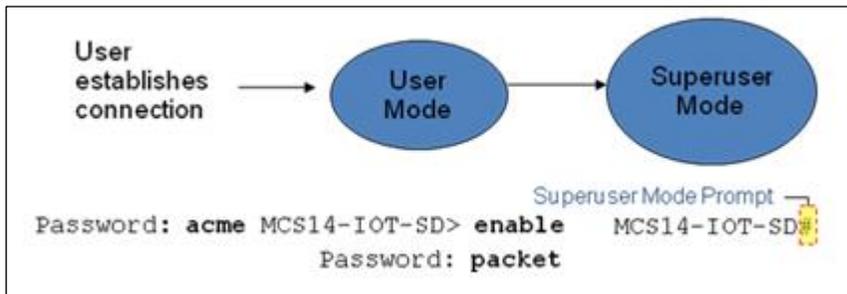
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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.



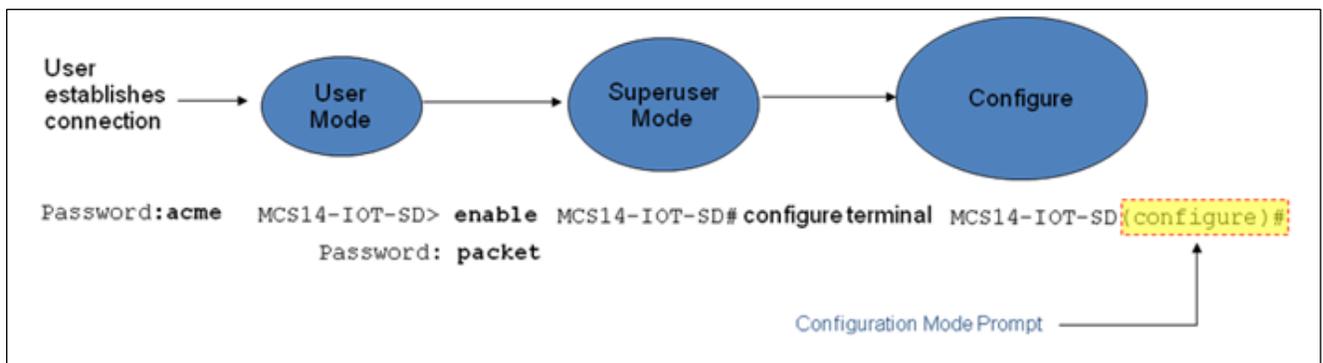
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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.

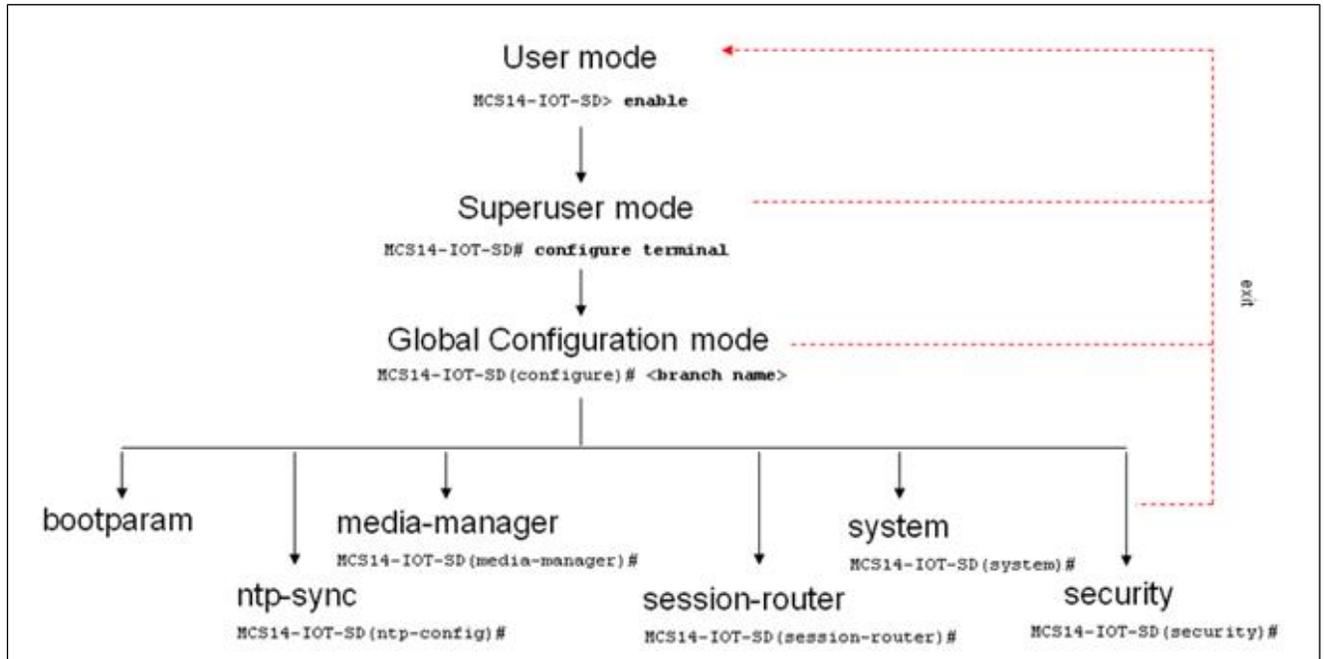
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the **configure terminal** command in the Superuser mode.

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

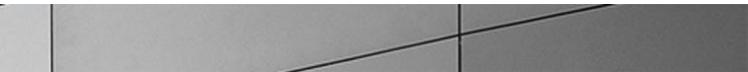
The security branch provides access to security configuration.

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

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 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.

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 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,

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 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 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.

```
MCS14-IOT-SD# 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'.
MCS14-IOT-SD#
```

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.

```
MCS14-IOT-SD# activate-config
Activate-Config received, processing.
waiting 120000 for request to finish
Request to 'ACTIVATE-CONFIG' has Finished,
Activate Complete
MCS14-IOT-SD#
```



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Hardware and Software, Engineered to Work Together

