ORACLE COMMUNICATIONS

Oracle Enterprise Communications Broker & Oracle Enterprise Session Border Controller with Avaya's Aura 6.3 & Aura 7.0, Cisco's UCM 10.5 & UCM 11.0, Microsoft's Lync 2013 & Skype for Business

Technical Application Note

Disclaimer

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

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Intended Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Communications Enterprise-SBC, Enterprise Communications Broker, Microsoft Lync and Skype for Business, Avaya Aura Session 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.

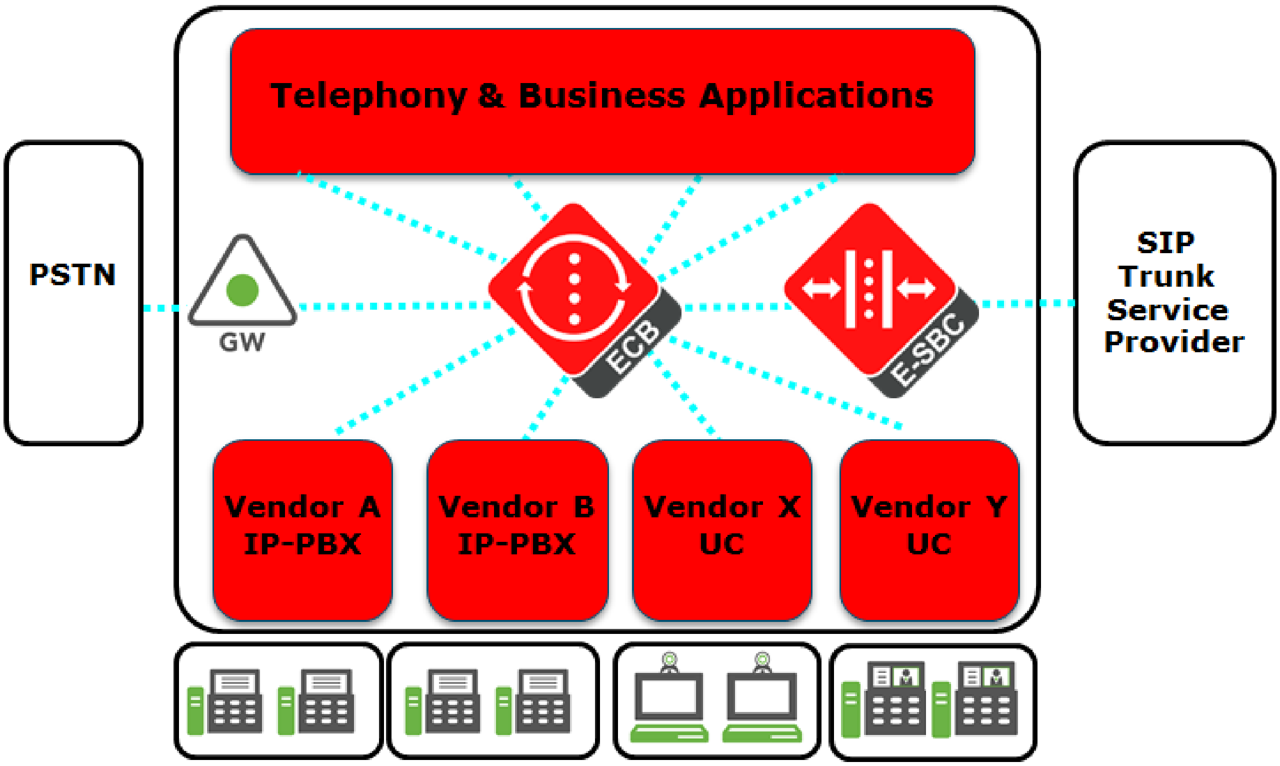
Document Overview

This technical application note documents the implementation of the Oracle Enterprise Communications Broker (Oracle ECB) in an Enterprise network consisting of multi-vendor Unified Communications platforms - Microsoft Lync 2013, Microsoft Skype for Business 2015, Avaya Aura Session Manager and Cisco Unified Communications Manager - connecting to a SIP trunk through an Enterprise Session Border Controller.

Introduction

Enterprise Communications Broker Overview

The Oracle 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.



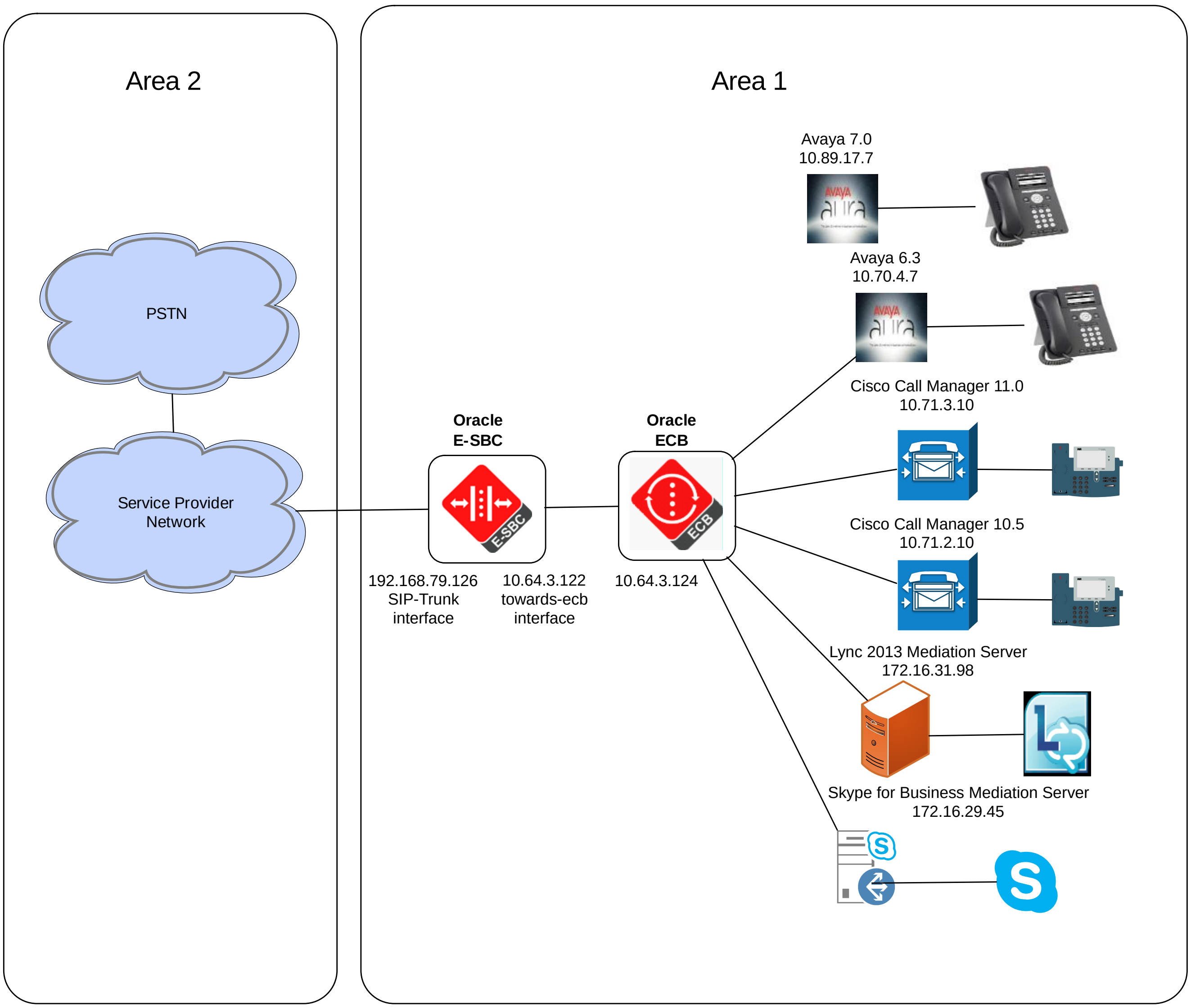
The Oracle ECB is typically deployed in the core of a multi-vendor communications network where multiple UC, PBX and service provider trunk interfaces must be interconnected. It normalizes communications between disparate premise-based systems and connects them to service provider networks and hosted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

Requirements

- Oracle Enterprise Communications Broker PCZ2.0.0 MR-2 Patch 1
- Oracle Enterprise Session Border Controller ECZ7.3.0 MR-1
- Microsoft Lync 2013 and/or Skype for Business 2015
- Avaya Aura 6.3 and/or 7.0
- Cisco Unified Communications Manager 10.5 and/or 11.0

Lab Configuration

The following diagram illustrates the lab environment created by tekVizion to facilitate certification testing. tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Their core services include consulting/solution design, interoperability/verification testing, integration, custom software development and solution support services.



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 the ECB at its core connecting together multiple UC platforms. The ECB connects to the Enterprise SBC which provides the Enterprise network access to the 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 nine phases:

- Phase 1 – Configuring the Oracle ECB
- Phase 2 – Configuring the Oracle E-SBC
- Phase 3 – Configuring Active Directory for LDAP Integration with the ECB
- Phase 4 – Configuring the Lync 2013 server
- Phase 5 – Configuring the Skype for Business server

- Phase 6 – Configuring the Avaya Aura Session Manager 6.3
- Phase 7 – Configuring the Avaya Aura Session Manager 7.0
- Phase 8 – Configuring the Cisco Unified Communications Manager 10.5
- Phase 9 – Configuring the Cisco Unified Communications Manager 11.0

Phase 1 – Configuring the Oracle ECB

The Oracle ECB is available either as an appliance or as an application for operation on virtual machines. When running as an appliance, the Oracle ECB software is packaged with the Netra Server X3-2 and delivered to the end customers. When running as a virtual application, the Oracle 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 Oracle 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 via a VGA monitor and USB keyboard.

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] : 172.18.255.82
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 : 10.64.3.124
Subnet mask on SIP interface [255.255.255.0] : 255.255.0.0
Gateway IP address on SIP interface : 10.64.1.1
```

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.
#?2
```

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 timezones. 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.

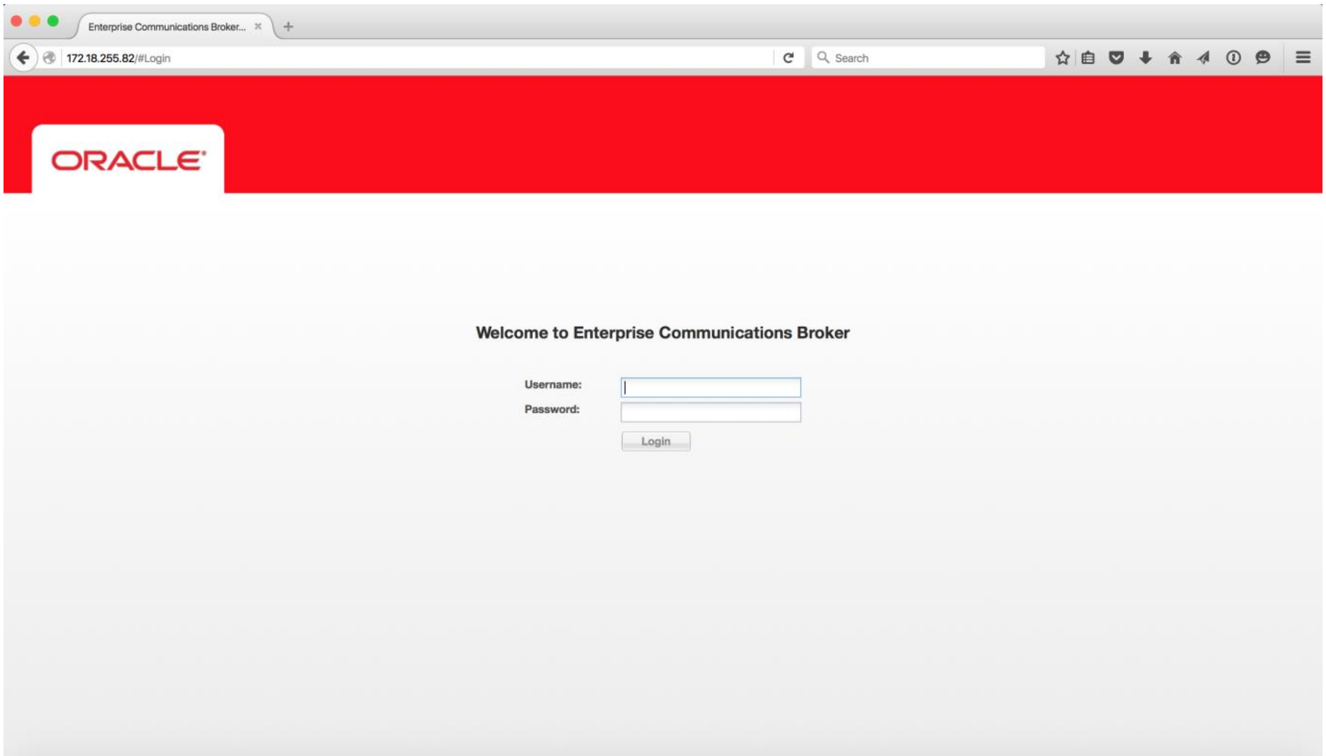
```
Saved configuration. ----------------------------
HIGH AVAILABILITY
 2 : ECB mode                                          : standalone
 3 : ECB role                                          : N/A

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                   : 172.18.255.82
 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                          : 10.64.3.124
 15: SIP interface subnet mask                         : 255.255.0.0
 16: SIP interface gateway IP address                  : 10.64.1.1

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://172.18.255.82:80/ 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.



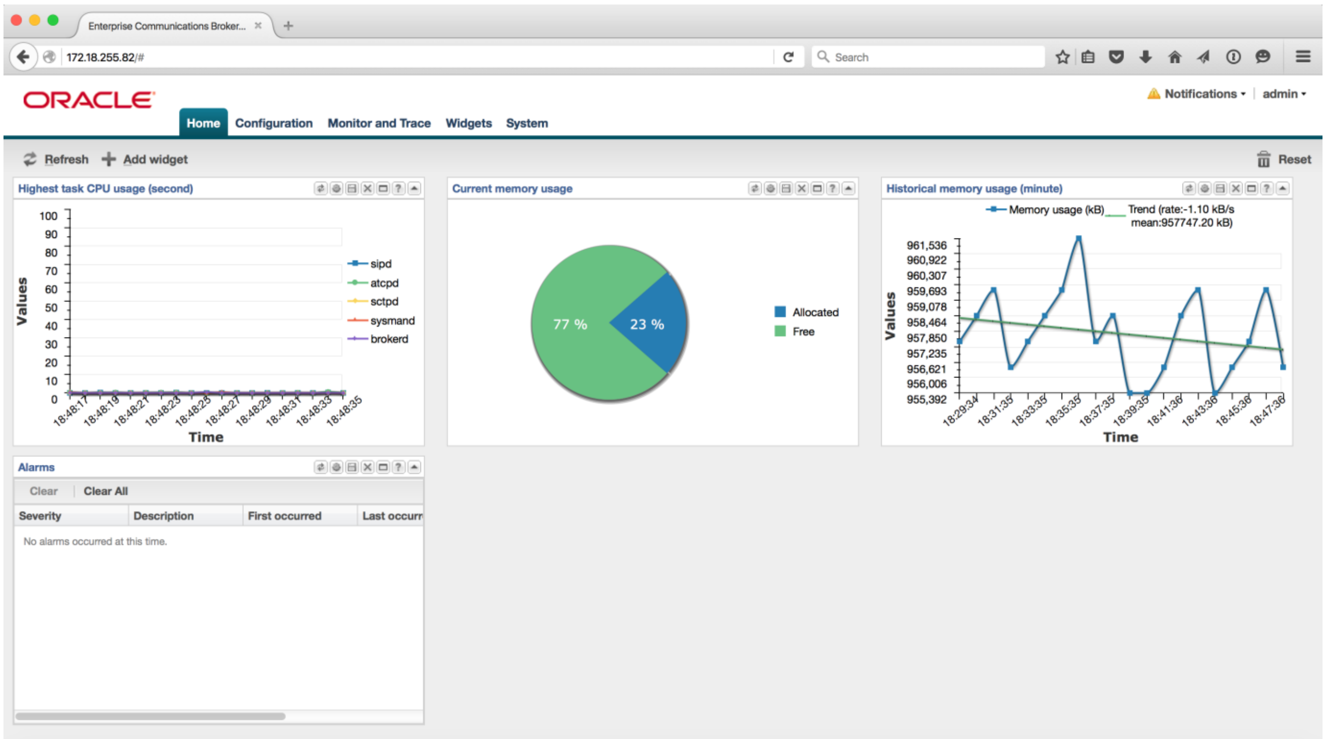
Enter your GUI username and password. The default username for the User level is “user” and the default password is “acme”.

The default username for an Administrator level is “admin”, and the default password is "packet".

Configuring the ECB

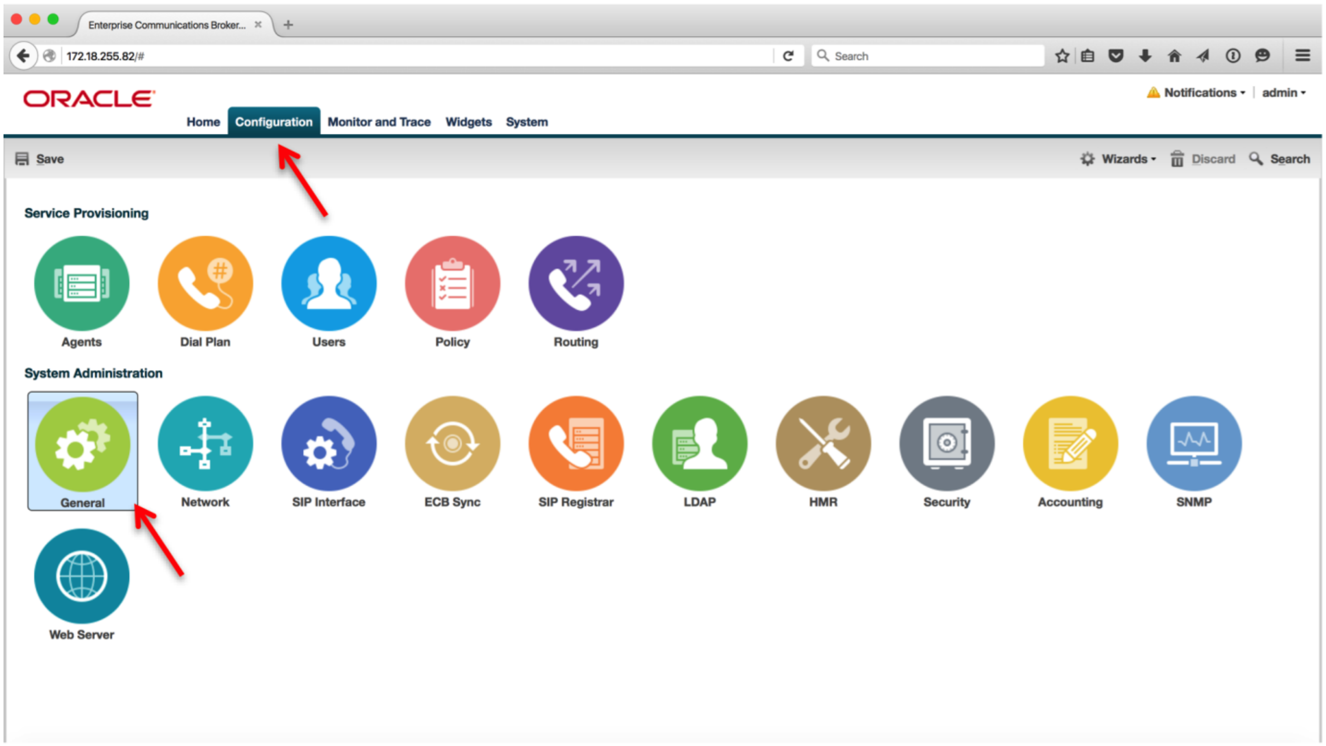
After logging into the ECB, the **Home** screen will be displayed. The Oracle ECB GUI has five tabs across the top – **Home, Configuration, Monitor and Trace, Widgets** and **System**.

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

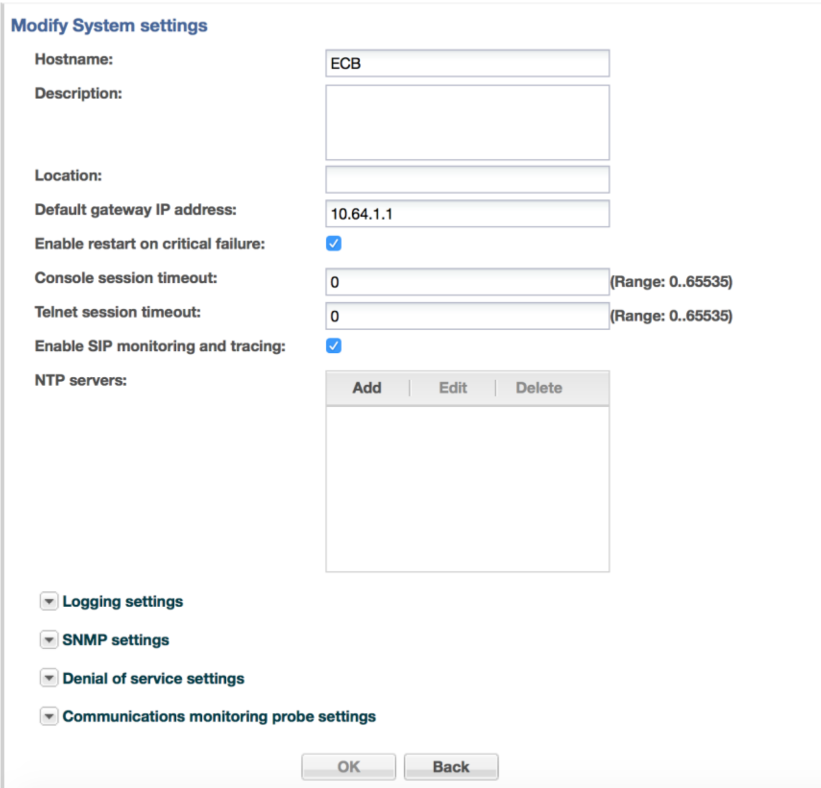


System Settings

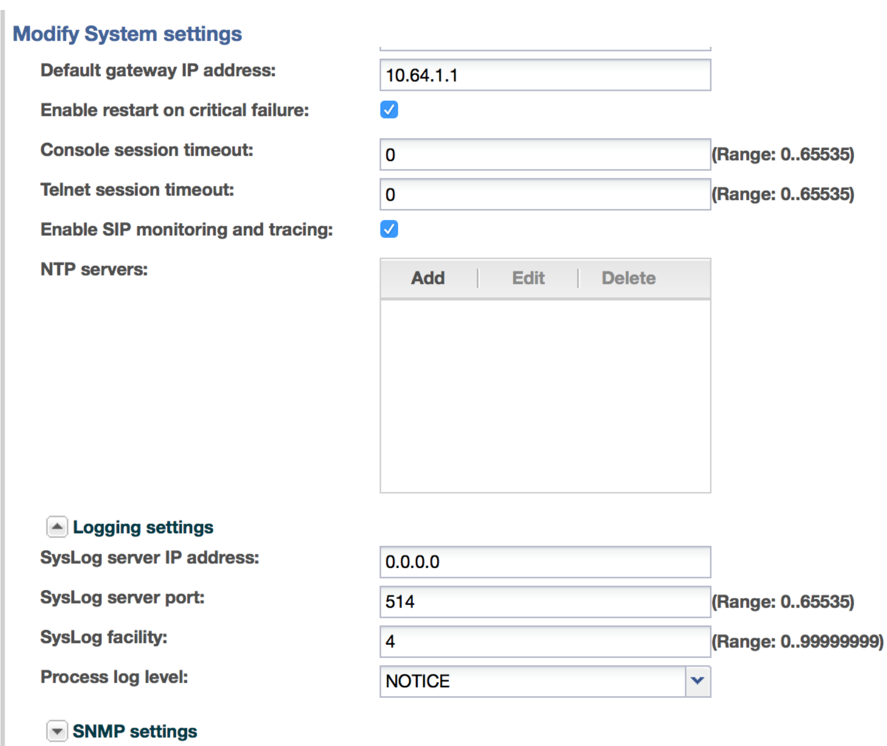
Select the **Configuration** tab. This tab displays the configurable elements in the ECB in two sections – **Service Provisioning** and **System Administration**. Click on the **General** icon under **System Administration**.



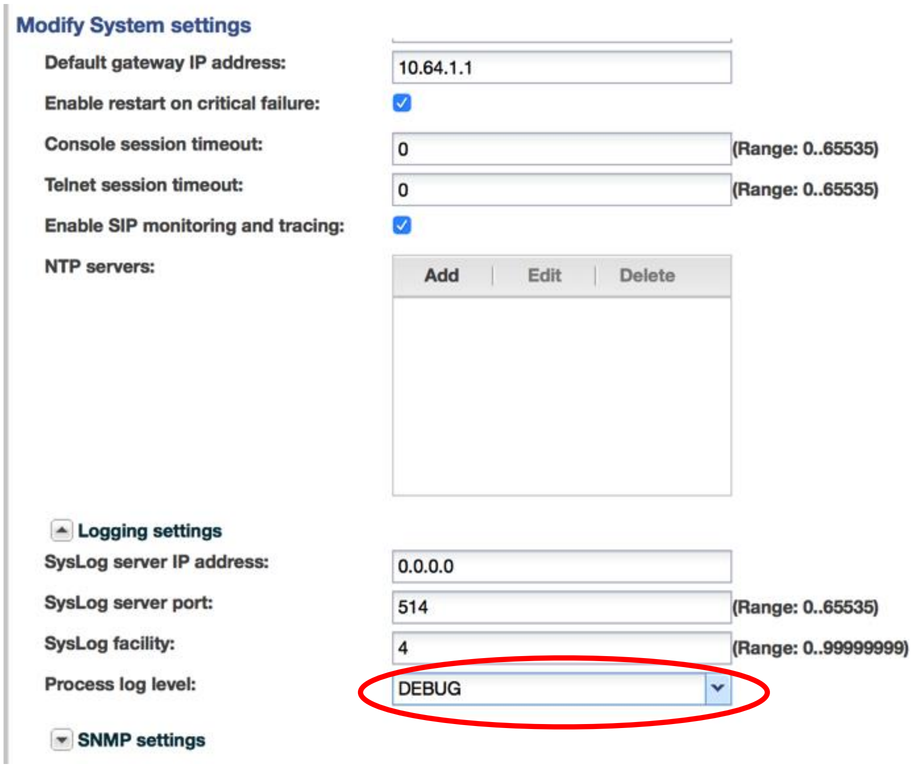
The Modify System Settings page is displayed.



Expand the **Logging settings** section.

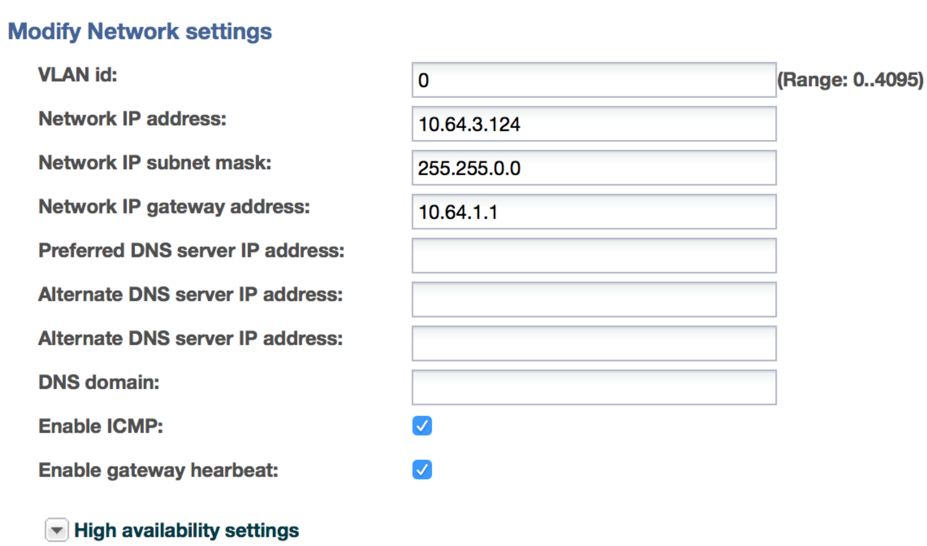


Process log level is set at **NOTICE**. Change the setting to **DEBUG** by selecting the option from the drop down menu and click **OK**. This should be changed back to **NOTICE** after testing is complete.



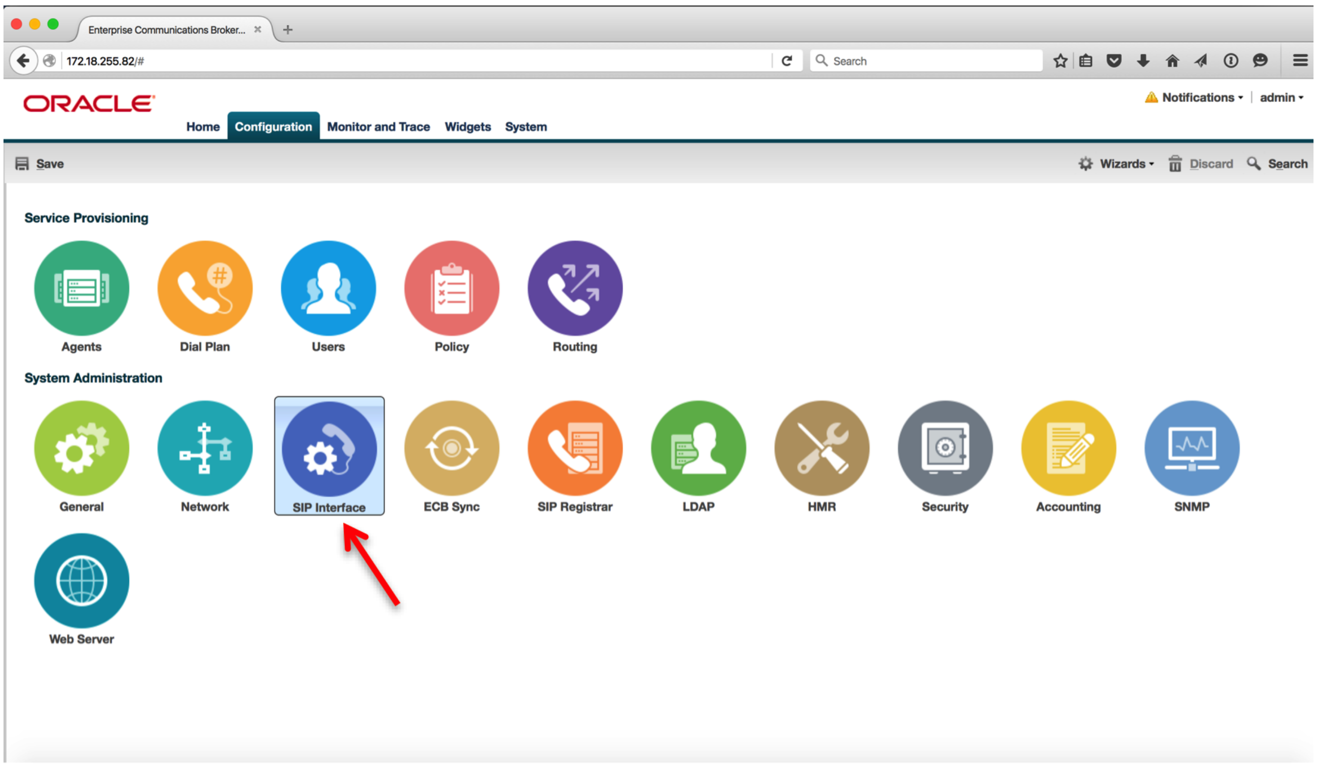
Click the **Configuration** button at the top to go to the **Configuration** tab.

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

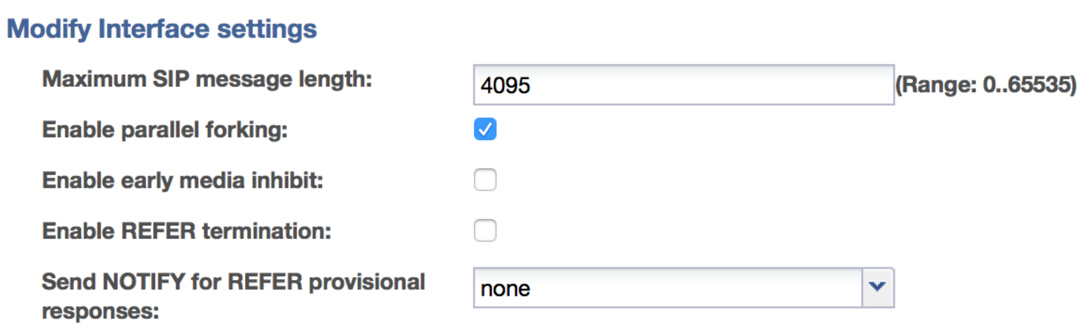


Configure SIP Interfaces

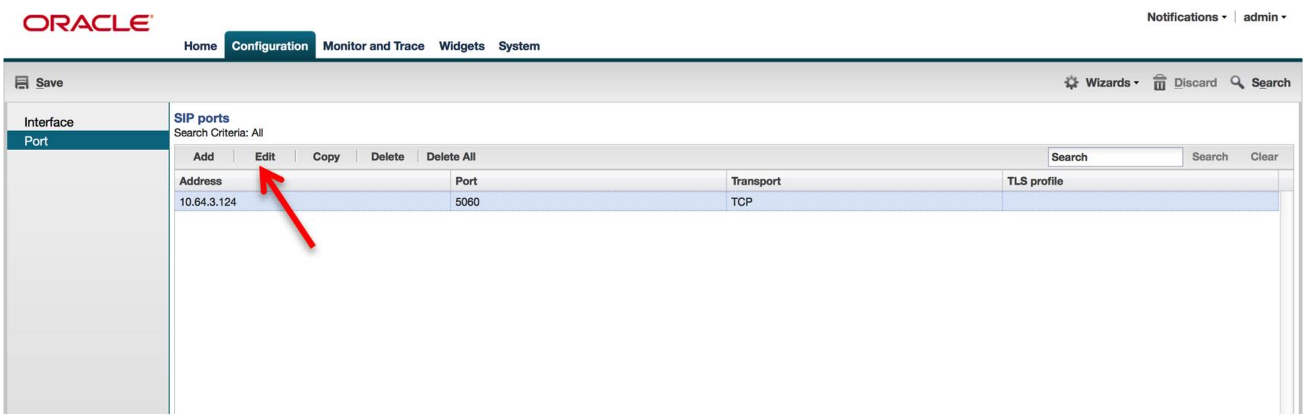
Click **Configuration** button to go to the **Configuration** tab. Select the **SIP Interface icon** under **System Administration** to make changes to the SIP interface settings configured during initial setup.



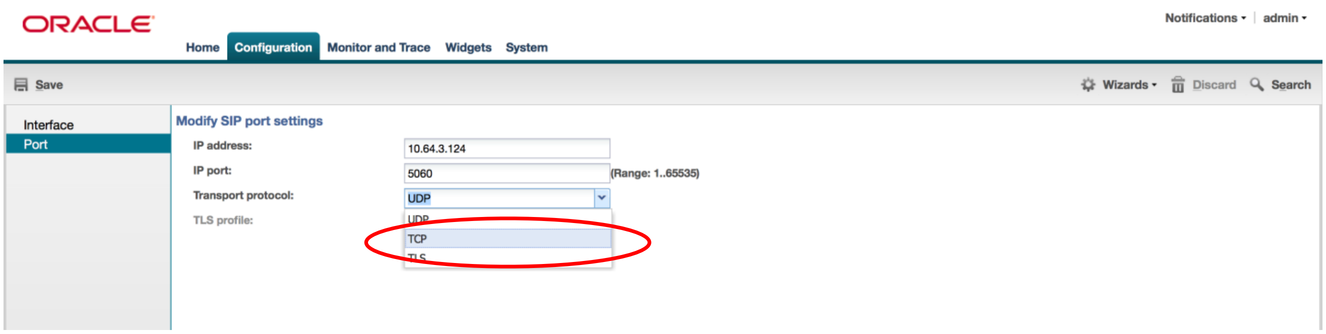
Click the “enable parallel forking” checkbox to enable parallel forking, i.e. calling a user on two devices at once, or leave it unchecked for serial forking. See the Configure LDAP Integration with Active Directory section of this document for more information.



Click on the **Port** tab on the left. You will see the sip port 10.64.3.124 with protocol UDP. Click **Edit** to change its protocol to TCP.



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

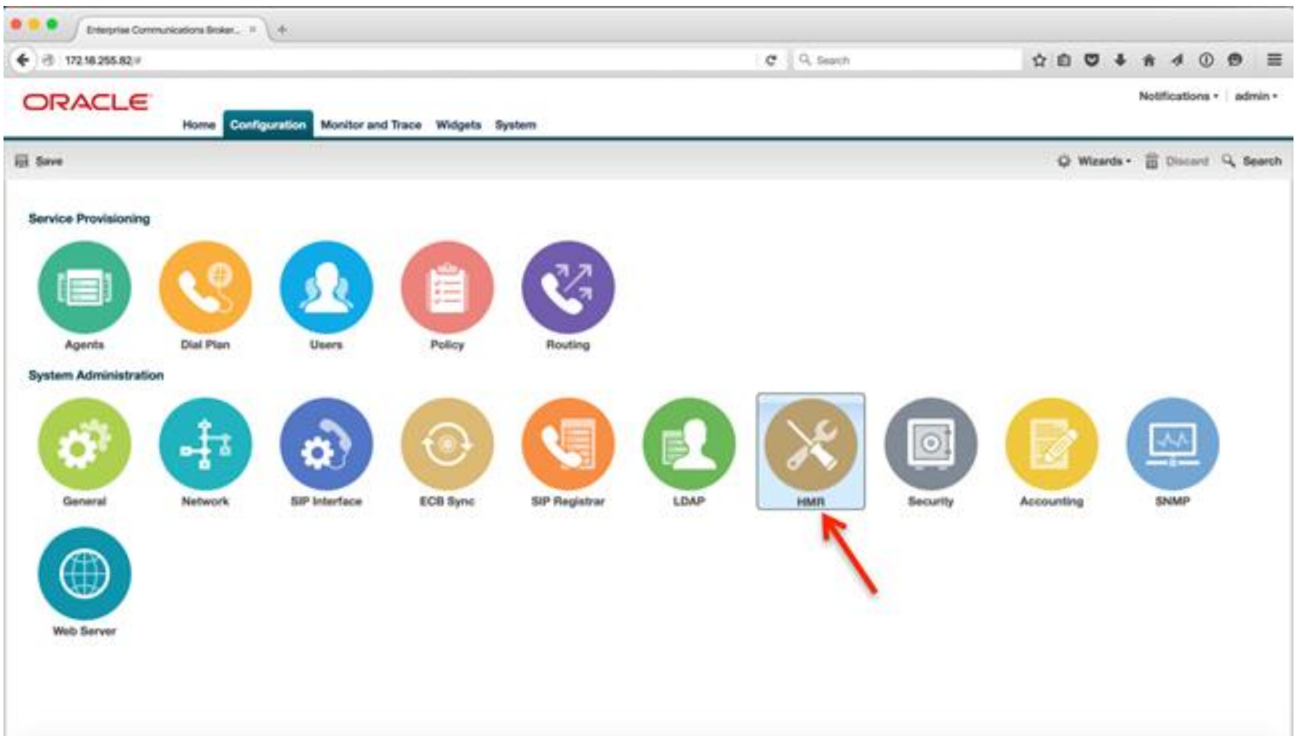


Click on the **Configuration** button to go back to the **Configuration** tab.

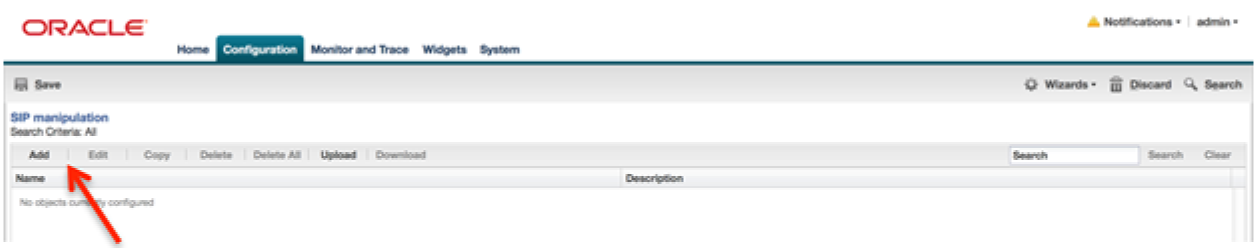
Configure Header Manipulation Rules (HMR)

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 needs.

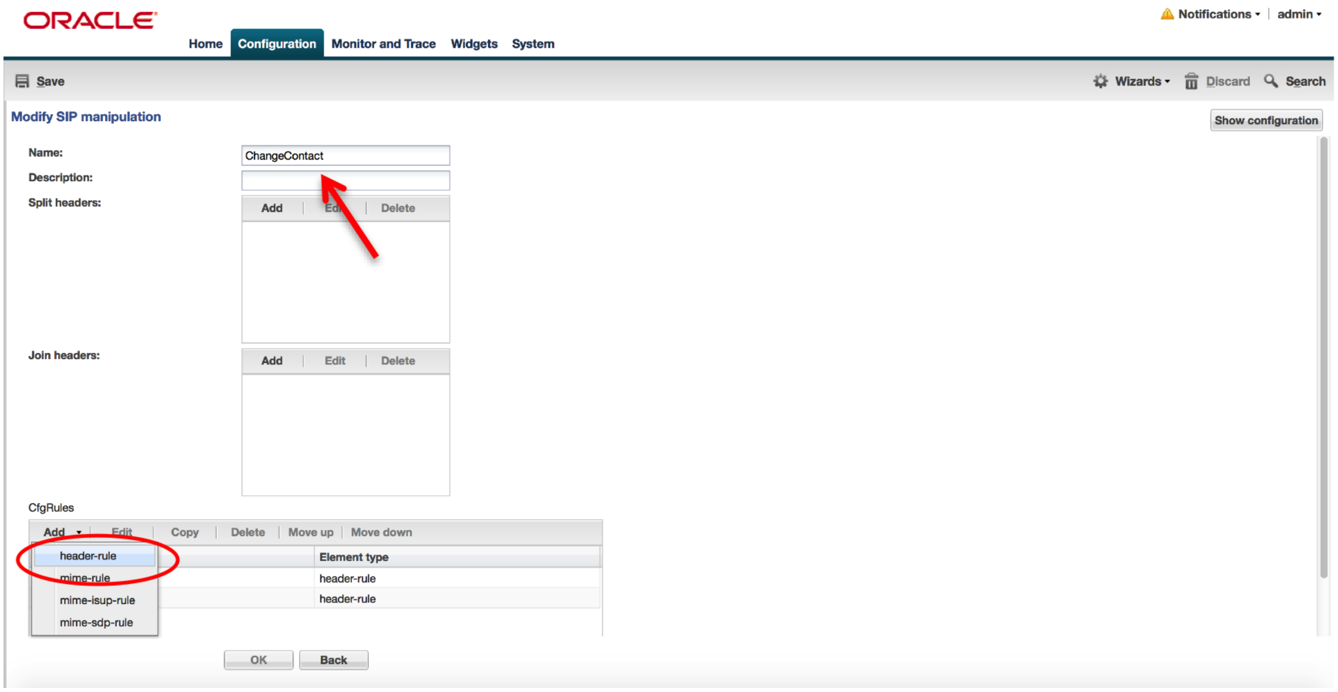
Click on the **HMR** icon under **System Administration** on the **Configuration** tab.



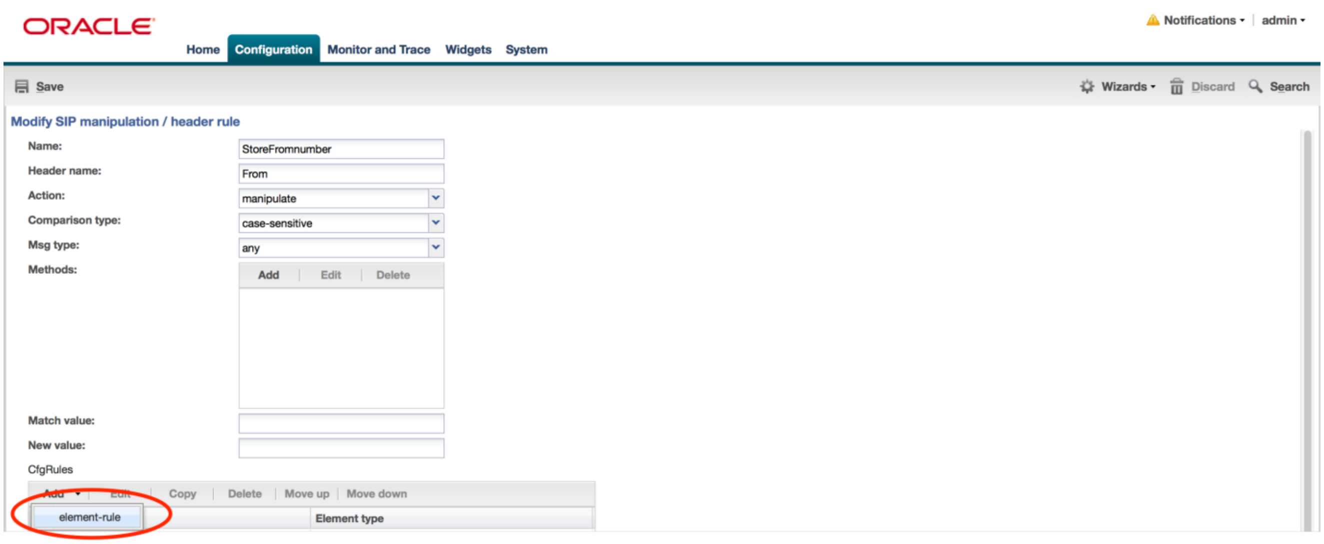
The **SIP manipulation** page is displayed. Click **Add** to add a SIP manipulation.



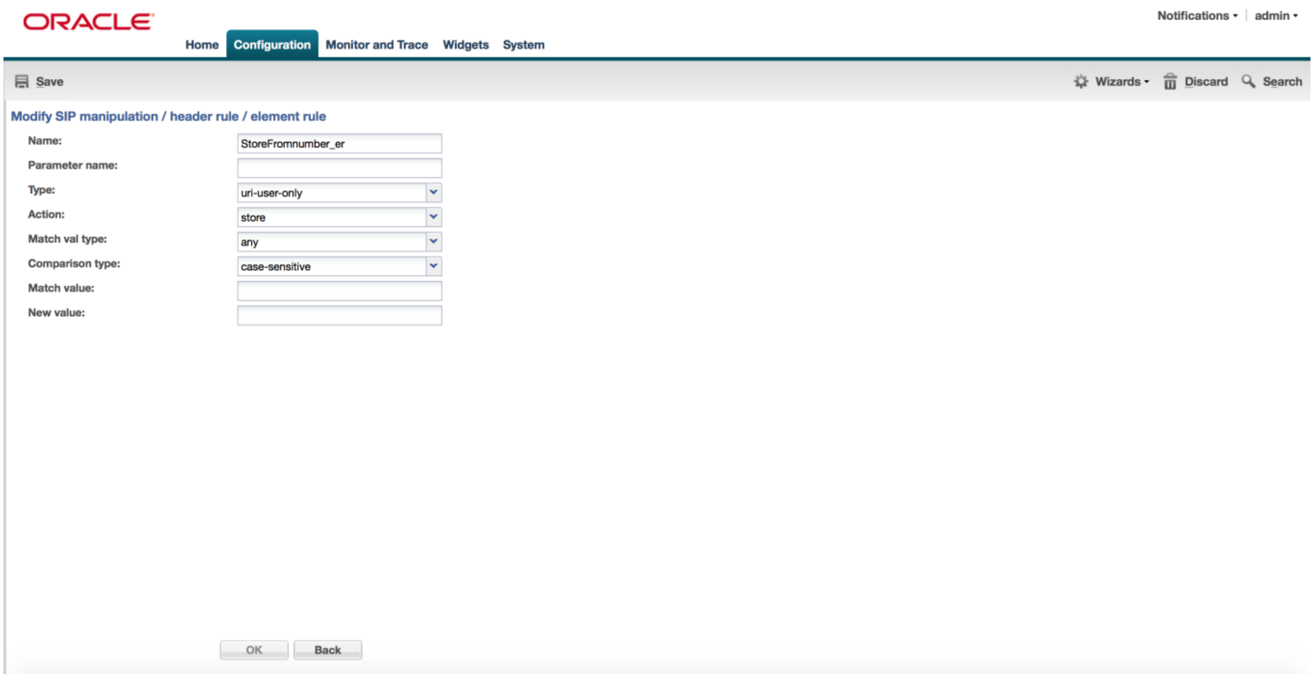
Type the name of the HMR rule, ChangeContact in this instance, and then click Add under CfgRules, then click header-rule. The manipulation consists of two header rules – StoreFromnumber and ChangeContact. The StoreFromnumber header rule stores the uri-user-only element in the From header which is then added as the uri-user in the Contact header in the ChangeContact header rule.



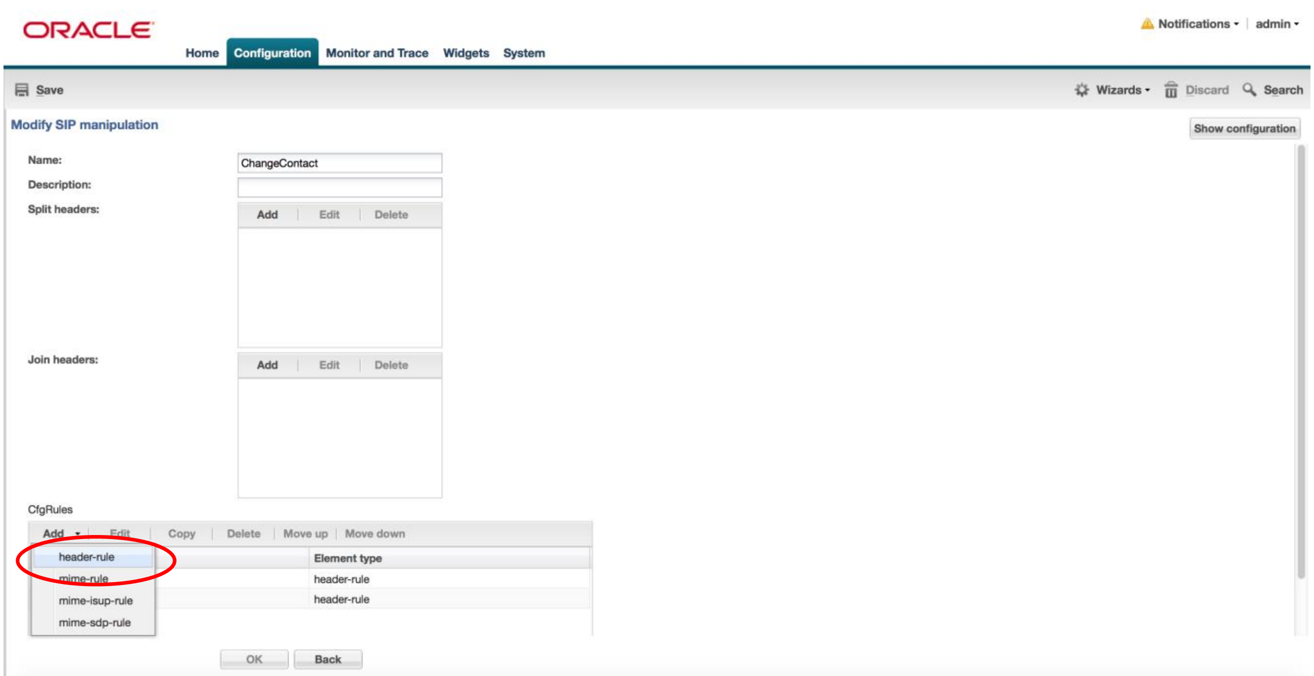
Enter the Name, Header name, and Action to match the following screenshot, then click on Add under CfgRules, then element-rule.



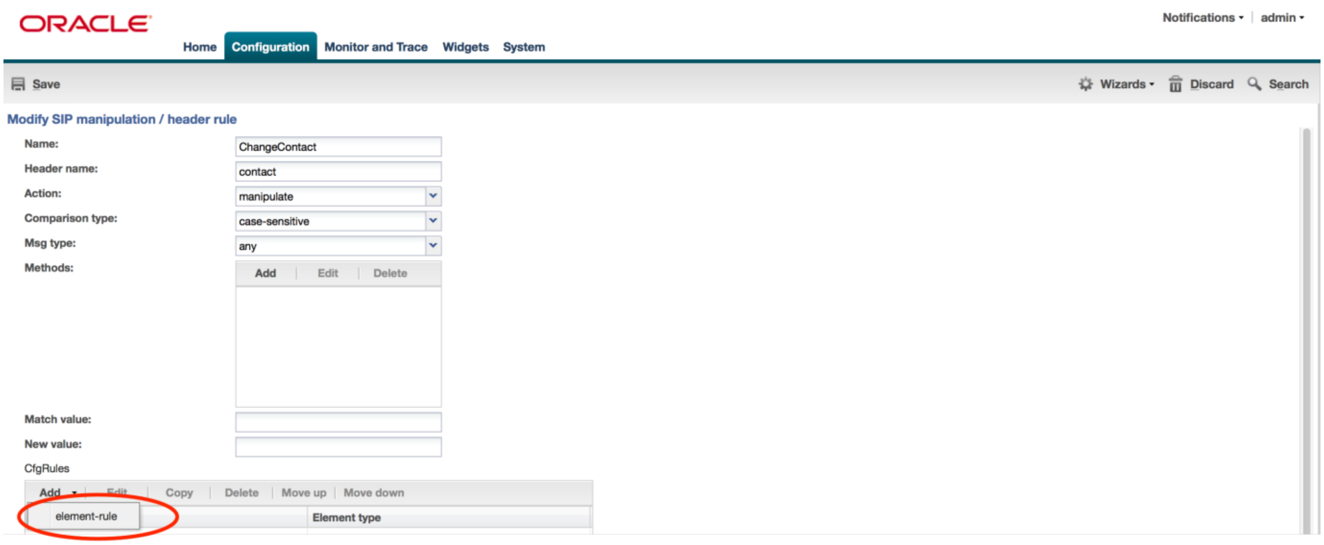
Then enter the following element-rule and click OK.



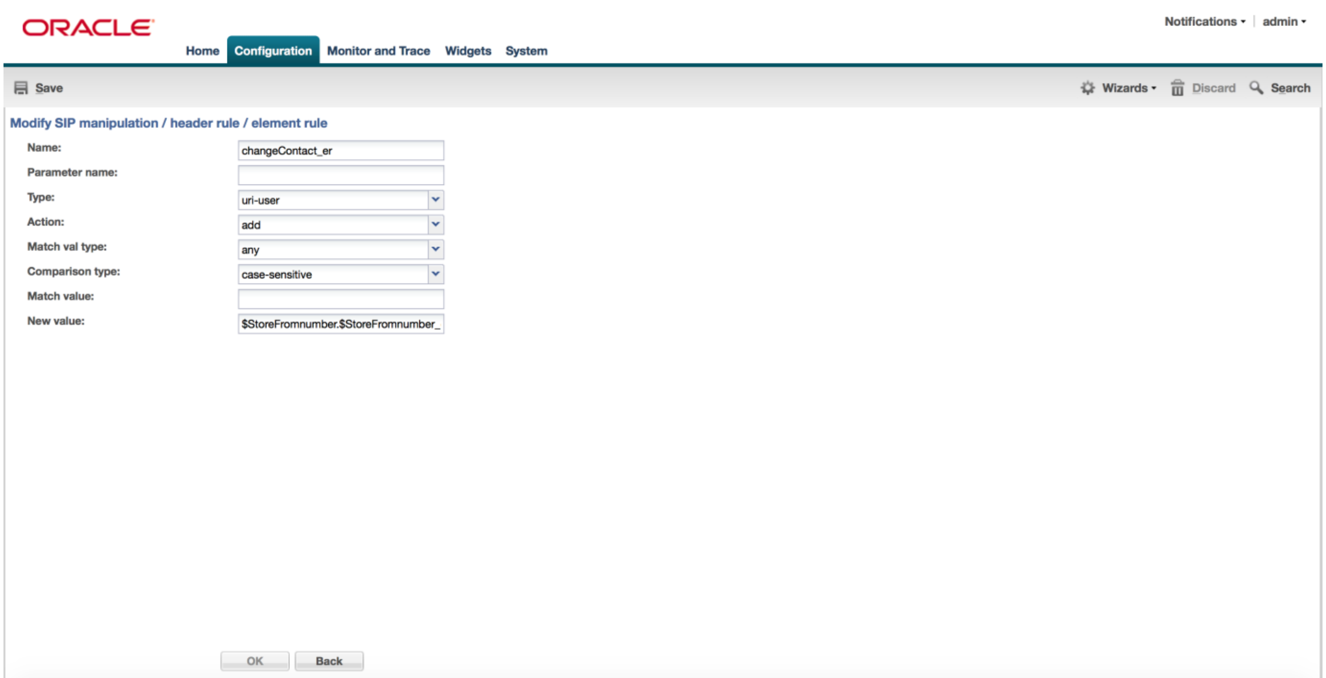
Add another header-rule:



Add the following header-rule, then click on Add > element-rule.



Add the following element-rule, then click OK. The New value displayed below is truncated and should be: $StoreFromnumber.$StoreFromnumber_er.$0



Here is a table of the HMR rules being configured on the ECB.

HMR Rule	Description
ChangeContact	Adds a user to the Contact header
HMRfromLync	Solves ringback issue and references ChangeContact rule
HMRtowardsAvaya	Changes From and To to 10 digits. Adds PAI to UPDATE requests.
HMRtowardsCUCM	Changes From and To to 10 digits. NATs IPs in those headers.
HMRtowardsLync	NAT, delete SDP b= lines, adds DTMF, changes to E.164, adds PAI to UPDATE
HMRtowardsSBC	Removes 9, removes +, removes PAI header
Modmline	Adds DTMF to SDP m= line
NATing	HMR for topology hiding
addPAItoUpdate	Adds P-Asserted-Identity to UPDATE requests
changeFromtoExt	Changes From and PAI to Extension only (optional)
changeTo10Digit	Changes From and To to 10 digits
changeToE164	Changes From and To to E.164
delblines	Deletes SDP b= lines from Avaya
fixSDP	Bypasses the 488 due to missing DTMF from CUCM during hold

For reference, here is the ChangeContact HMR rule in text format from the CLI. The text highlighted in **bold** are the non-default fields.

```
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
```

The following HMR rule will be applied as an inbound manipulation from Lync and SFB. It changes “183 Session Progess” to “180 Ringing” to solve a ringback issue, and it references the ChangeContact rule as a nested HMR.

```
sip-manipulation
        name                                        HMRfromLync
        description
        split-headers
        join-headers
        header-rule
                name                                        change183to180
                header-name                                 @status-line
                action                                      manipulate
                comparison-type                             case-sensitive
                msg-type                                    reply
                methods
                match-value
                new-value
                element-rule
                        name                                        mod183to180
                        parameter-name
                        type                                        status-code
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             case-sensitive
                        match-value                                 183
                        new-value                                   180
                element-rule
                        name                                  sessionProgressToRinging
                        parameter-name
                        type                                        reason-phrase
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             case-sensitive
                        match-value                                 Session Progress
                        new-value                                   Ringing
        header-rule
                name                                        ChangeContact
                header-name                                 To
                action                                      sip-manip
                comparison-type                             case-sensitive
                msg-type                                    any
                methods
                match-value
                new-value                                   ChangeContact
```

The following HMR rule is applied as an outbound manipulation towards Avaya. It changes the From and To headers to 10 digits and adds a P-Asserted-Identity to UPDATE requests. **The last header-rule (changeFromtoExt) is optional and changes the From header to be a four digit extension. This can be used if internal Caller-ID needs to be extension based instead of 10 digits.** See the "changeFromtoExt" HMR rule later in this document for more details.

```
sip-manipulation
        name                                    HMRtowardsAvaya
        description
        split-headers
        join-headers
        header-rule
                name                                    changeTo10Digit
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeTo10Digit
        header-rule
                name                                    addPAItoUpdate
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 UPDATE
                match-value
                new-value                               addPAItoUpdate
        header-rule (optional)
                name                                    changeFromtoExt
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeFromtoExt
```

The following HMR rule is applied as an outbound manipulation towards Cisco CUCM. It changes the From and To headers to 10 digits and NATs IPs in those headers as well. **The last header-rule (changeFromtoExt) is optional and changes the From header to be a four digit extension. This can be used if internal Caller-ID needs to be extension based instead of 10 digits.** See the "changeFromtoExt" HMR rule later in this document for more details.

```
sip-manipulation
        name                                    HMRtowardsCUCM
        description
        split-headers
        join-headers
        header-rule
                name                                    changeTo10Digit
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeTo10Digit
        header-rule
                name                                    NATing
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               NATing
        header-rule (optional)
                name                                    changeFromtoExt
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeFromtoExt
```

The following HMR rule is applied as an outbound manipulation towards Microsoft Lync and SFB. It changes the From and To headers to E.164 format, NATs IPs in those headers, and modifies the SDP. **The last header-rule (changeFromtoExt) is optional and changes the From header to be a four digit extension. This can be used if internal Caller-ID needs to be extension based instead of 10 digits.** See the “changeFromtoExt” HMR rule later in this document for more details.

```
sip-manipulation
        name                                    HMRtowardsLync
        description
        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                               NATing
        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                                    adddtmflines
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods                                 INVITE
                match-value
                new-value                               fixSDP
        header-rule
                name                                    changeToE164
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeToE164
        header-rule
                name                                    addPAItoUpdate
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods                                 UPDATE
                match-value
                new-value                               addPAItoUpdate
        header-rule (optional)
                name                                    changeFromtoExt
                header-name                             To
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                request
                methods
                match-value
                new-value                               changeFromtoExt
```

The following HMR rule is applied as an outbound manipulation towards the Oracle E-SBC. It removes 9 from the beginning of the Request- and To- URIs, NATs the From and To headers headers, removes the plus sign in the From header, and removes P-Asserted-Identity.

```
sip-manipulation
        name                                        HMRtowardsSBC
        description
        split-headers
        join-headers
        header-rule
                name                                        Remove9
                header-name                                 request-uri
                action                                      manipulate
                comparison-type                             pattern-rule
                msg-type                                    request
                methods
                match-value
                new-value
                element-rule
                        name                                        remove9FromRuri
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 91(\d{10})
                        new-value                                   $1
        header-rule
                name                                        NATing
                header-name                                 To
                action                                      sip-manip
                comparison-type                             case-sensitive
                msg-type                                    request
                methods
                match-value
                new-value                                   NATing
        header-rule
                name                                        remove9fromTOURI
                header-name                                 to
                action                                      manipulate
                comparison-type                             pattern-rule
                msg-type                                    request
                methods
                match-value
                new-value
                element-rule
                        name                                        remove9inTOuri
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 91(\d{10})
                        new-value                                   $1
        header-rule
                name                                        reomvePlusInFrom
                header-name                                 From
                action                                      manipulate
                comparison-type                             pattern-rule
                msg-type                                    request
                methods
                match-value
                new-value
```

```
            element-rule
                    name                                        reomvePlus
                    parameter-name
                    type                                        uri-user
                    action                                      replace
                    match-val-type                              any
                    comparison-type                             pattern-rule
                    match-value                                 ^\+([0-9]{10,11})$
                    new-value                                   $1
        header-rule
            name                                        removePAI
            header-name                                 P-Asserted-Identity
            action                                      delete
            comparison-type                             case-sensitive
            msg-type                                    request
            methods
            match-value
            new-value
```

The following HMR rule is referenced in another HMR rule later in this document and adds DTMF to the m= line in SDP.

```
sip-manipulation
        name                                        Modmline
        description                                 Add DTMF to m line
        split-headers
        join-headers
        mime-sdp-rule
            name                                        modmline
            msg-type                                    any
            methods
            action                                      manipulate
            comparison-type                             pattern-rule
            match-value
            new-value
            sdp-media-rule
                    name                                        modmline_m
                    media-type                                  audio
                    action                                      manipulate
                    comparison-type                             pattern-rule
                    match-value
                    new-value
                    sdp-line-rule
                            name                                    change_payload
                            type                                        m
                            action                                  find-replace-all
                            comparison-type                             pattern-rule
                            match-value     ^(audio)( [0-9]{4,5})( RTP/AVP 0)$
                            new-value                  audio+$2+" RTP/AVP 0 101"
```

The following HMR rule is referenced in other HMR rules in this document, and it NATs the IP addresses in the From and To headers.

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

The following HMR rule is referenced in other HMR rules in this document, and it adds a P-Asserted-Identity header to UPDATE requests based on the user in the Contact-URI.

```
sip-manipulation
        name                                    addPAItoUpdate
        description
        split-headers
        join-headers
        header-rule
                name                                    storeContact
                header-name                             Contact
                action                                  store
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 UPDATE
                match-value
                new-value
                element-rule
                        name                                    storeContactUser
                        parameter-name
                        type                                    uri-user
                        action                                  store
                        match-val-type                          any
                        comparison-type                         case-sensitive
                        match-value
                        new-value
        header-rule
                name                                    addPAIheader
                header-name                             P-Asserted-Identity
                action                                  add
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 UPDATE
                match-value
                new-value
                element-rule
                        name                                    addPAI
                        parameter-name
                        type                                    header-value
                        action                                  add
                        match-val-type                          any
                        comparison-type                         case-sensitive
                        match-value
                        new-value "<sip:"+$storeContact.$storeContactUser.$0+"@"+$LOCAL_IP+">"
```

The following HMR rule is referenced in other HMR rules in this document, and it changes the From and P-Asserted-Identity headers to 4-digit extensions. **It is optional and may be used to ensure internal Caller-ID shows extensions only instead of 10-digits. In our lab config, it looks for 571293, 1571293, +571293, and +1571293, and strips these prefixes off if they are present. You will need to change this to be your network's prefix. It is looking for 4 digit extensions, which is why the {4} appears in the match-value. This may be changed to meet your extension length requirements. The match-value is a regular expression (regex) and it looks for an optional plus sign, an optional 1, and then 571293, followed by 4 digits. If you need it to look for a country code of +61 folllowed by 5712 followed by a 5 digit extension, for example, the regex would be: `^\+615712([0-9]{5})$`**

```
sip-manipulation (optional)
        name                                    changeFromtoExt
        description                             change From and PAI to Extension only
        split-headers
        join-headers
        header-rule
                name                                    removePlusandPrefix
                header-name                             From
                action                                  manipulate
                comparison-type                         pattern-rule
                msg-type                                request
                methods
                match-value
                new-value
                element-rule
                        name                                    changeFromUri
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^\+?1?571293([0-9]{4})$
                        new-value                               $1
        header-rule
                name                                    removePAIplusandPrefix
                header-name                             P-Asserted-Identity
                action                                  manipulate
                comparison-type                         pattern-rule
                msg-type                                request
                methods
                match-value
                new-value
                element-rule
                        name                                    modifyPaiUri
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^\+?1?571293([0-9]{4})$
                        new-value                               $1
```

The following HMR rule is referenced in other HMR rules in this document, and it converts the From and To headers to 10 digit dialing. **Change 571293 to match your prefix. {4} represents a 4 digit extension. Change the length of the extension if needed.**

```
sip-manipulation
        name                                        changeTo10Digit
        description
        split-headers
        join-headers
        header-rule (do not use this header-rule if using the changeFromtoExt rule above)
                name                                        changeFrom10Digit
                header-name                                 From
                action                                      manipulate
                comparison-type                             pattern-rule
                msg-type                                    request
                methods
                match-value
                new-value
                element-rule
                        name                                        changeExtTo10Digit
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 ^([0-9]{4})$
                        new-value                                   571293+$1
                element-rule
                        name                                        change11DigitTo10Digit
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 ^1([0-9){10})$
                        new-value                                   $1
                element-rule
                        name                                        changeE164To10Digit
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 ^\+1([0-9]{10})$
                        new-value                                   $1
        header-rule
                name                                        changeTo10Digit
                header-name                                 To
                action                                      manipulate
                comparison-type                             pattern-rule
                msg-type                                    request
                methods
                match-value
                new-value
                element-rule
                        name                                        changeExtTo10Digit
                        parameter-name
                        type                                        uri-user
                        action                                      replace
                        match-val-type                              any
                        comparison-type                             pattern-rule
                        match-value                                 ^([0-9]{4})$
                        new-value                                   571293+$1
                element-rule
```

```
                name                                        change11DigitTo10Digit
                parameter-name
                type                                        uri-user
                action                                      replace
                match-val-type                              any
                comparison-type                             pattern-rule
                match-value                                 ^1([0-9]{10})$
                new-value                                   $1
        element-rule
                name                                        changeE164to10Digit
                parameter-name
                type                                        uri-user
                action                                      replace
                match-val-type                              any
                comparison-type                             pattern-rule
                match-value                                 ^\+1([0-9]{10})$
                new-value                                   $1
```

The following HMR rule is referenced in other HMR rules in this document, and it converts the From and To headers to E.164 dialing. **Change 571293 to match your prefix. {4} represents a 4 digit extension. Change the length of the extension if needed.**

```
sip-manipulation
        name                                    changeToE164
        description
        split-headers
        join-headers
        header-rule
                name                                    changeToE164
                header-name                             To
                action                                  manipulate
                comparison-type                         pattern-rule
                msg-type                                request
                methods
                match-value
                new-value
                element-rule
                        name                                    changeExtToE164
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^([0-9]{4})$
                        new-value                               \+1571293+$1
                element-rule
                        name                                    change11DigitToE164
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^([0-9]{11})$
                        new-value                               \++$1
                element-rule
                        name                                    change10DigitToE164
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^([0-9]{10})$
                        new-value                               \+1+$1
        header-rule (do not use this header-rule if using the changeFromtoExt rule above)
                name                                    changeFromE164
                header-name                             From
                action                                  manipulate
                comparison-type                         pattern-rule
                msg-type                                request
                methods
                match-value
                new-value
                element-rule
                        name                                    changeExtToE164
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             ^([0-9]{4})$
                        new-value                               \+1571293+$1
                element-rule
```

```
            name                                    change11DigitToE164
            parameter-name
            type                                    uri-user
            action                                  replace
            match-val-type                          any
            comparison-type                         pattern-rule
            match-value                             ^([0-9]{11})$
            new-value                               \++$1
        element-rule
            name                                    change10DigitToE164
            parameter-name
            type                                    uri-user
            action                                  replace
            match-val-type                          any
            comparison-type                         pattern-rule
            match-value                             ^([0-9]{10})$
            new-value                               \+1+$1
```

The following HMR rule is referenced in other HMR rules in this document, and it removes b= lines in the SDP coming from Avaya.

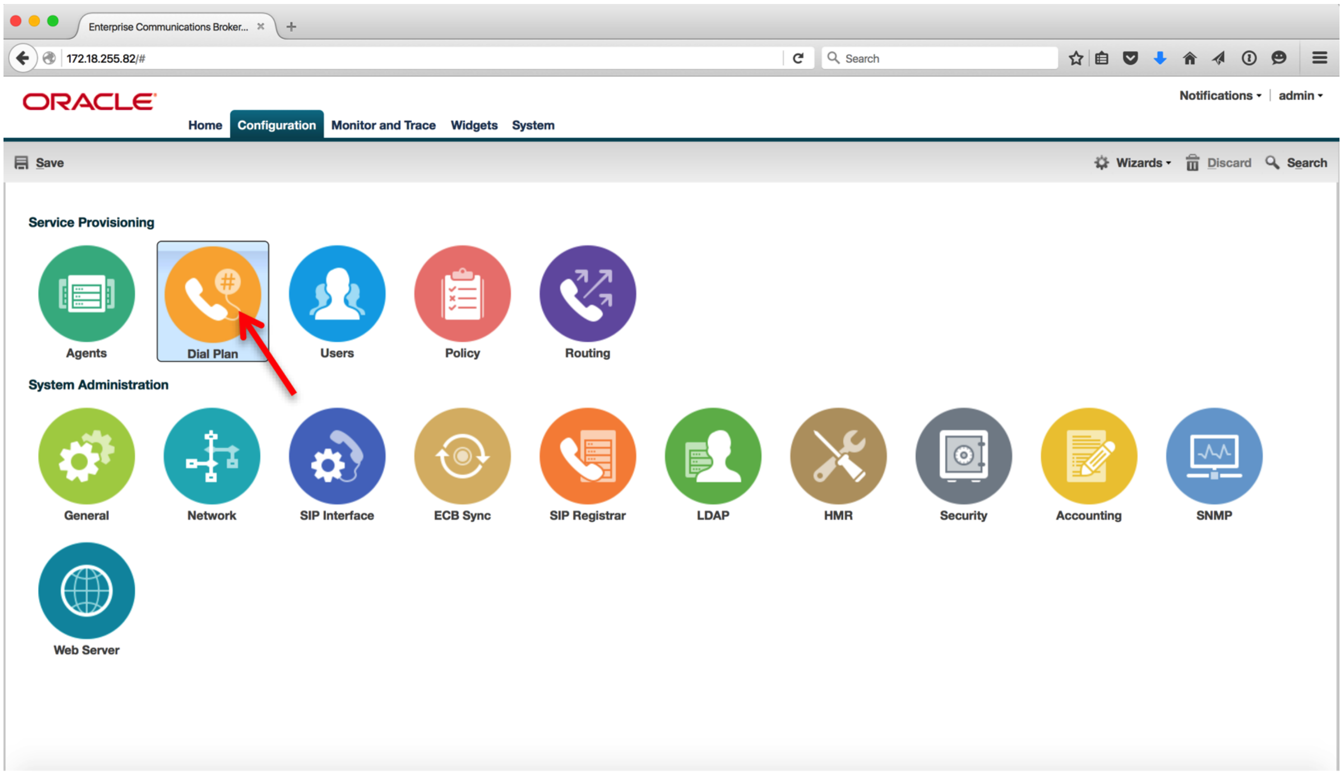
```
sip-manipulation
        name                                    delblines
        description                             Deleting b-lines from Avaya
        split-headers
        join-headers
        header-rule
                name                                    manipContentType
                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=TIAS:.*(\n|\r\n)
                        new-value
```

The following HMR rule is referenced in other HMR rules in this document, and it adds DTMF to the SDP.

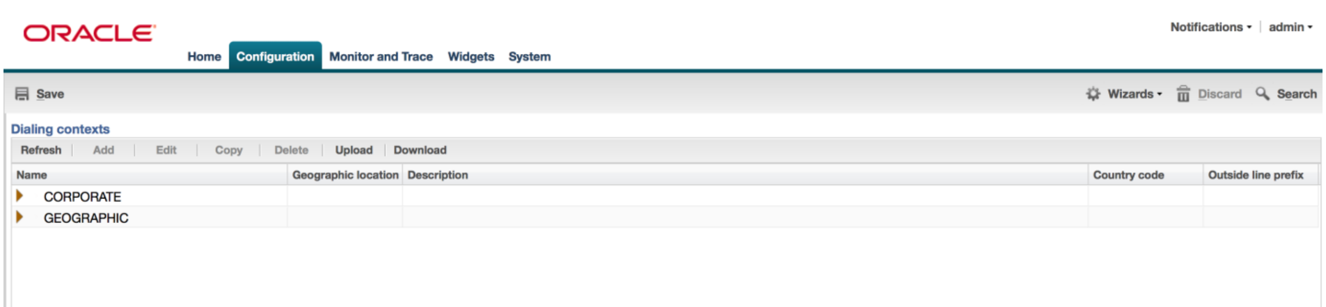
```
sip-manipulation
        name                                    fixSDP
        description                             To bypass the 488 due to missing DTMF from
CUCM during hold
        split-headers
        join-headers
        header-rule
                name                                    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=rtpmap: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 $0+"a=rtpmap:101 telephone-event/8000"+$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
```

Configure Dial Plan

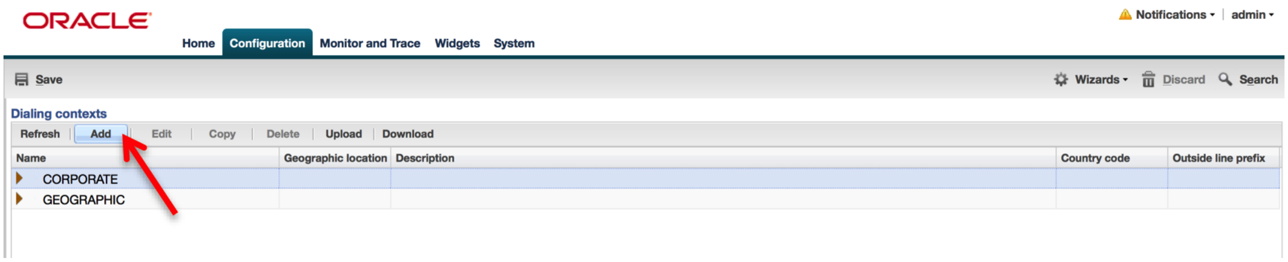
We will now configure the dialing contexts and dial plans. Dialing-contexts define the system behavior for calls placed to and from either a corporate or geographic focus. Dialing-contexts include multiple dial-patterns, which define the normalization required to most effectively manage diverse signaling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



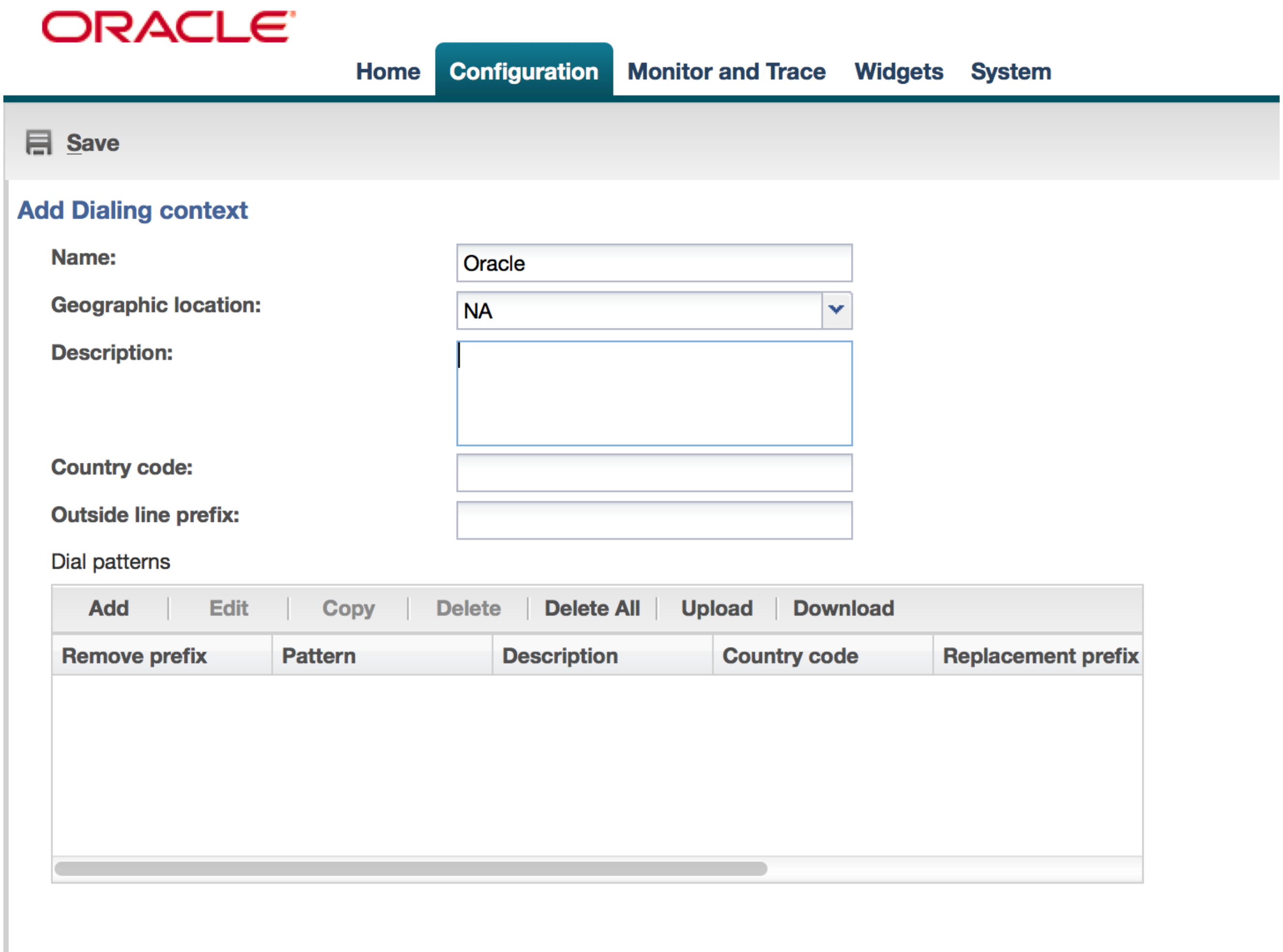
The **Dialing Contexts** page shows the default dialing context parents – Corporate and Geographic.



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

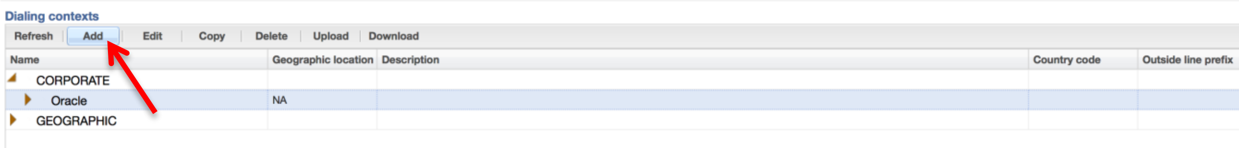


In the **Add Dialing Context** page, configure a context with the following details and click **OK**.

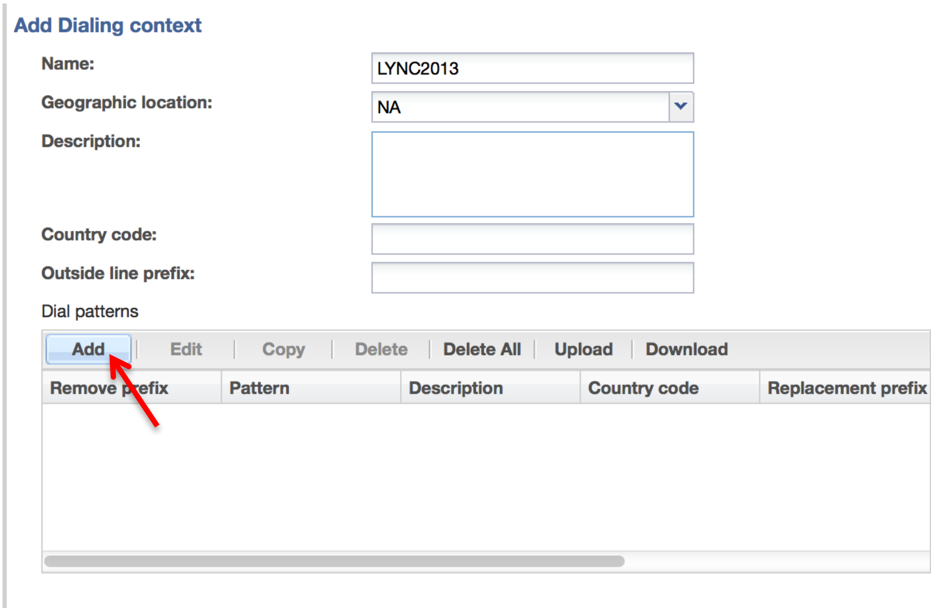


The Dialing Contexts page displays Oracle listed under Corporate contexts. We will now configure child contexts under Oracle for our Lync, SFB, Avaya and CUCM servers. These can be considered as contexts for the different branches an enterprise has.

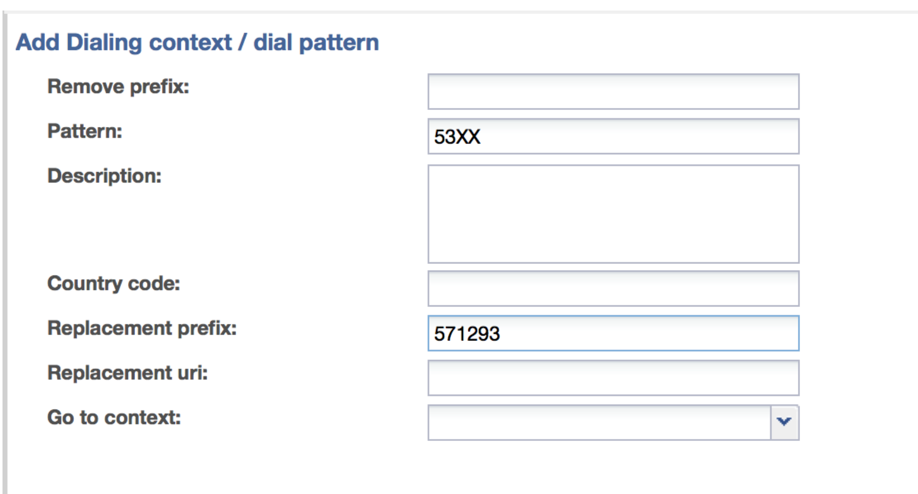
Select Oracle under the Corporate context and click **Add**.



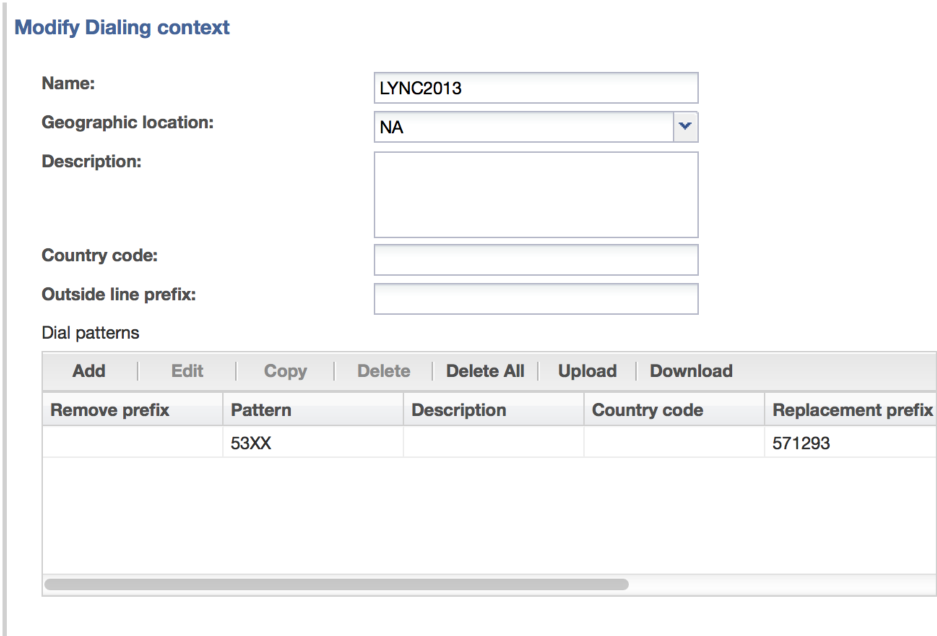
In the **Add Dialing Context** window, configure a context named LYNC2013 and **Geographic location** as NA. To configure dial patterns, click **Add**.



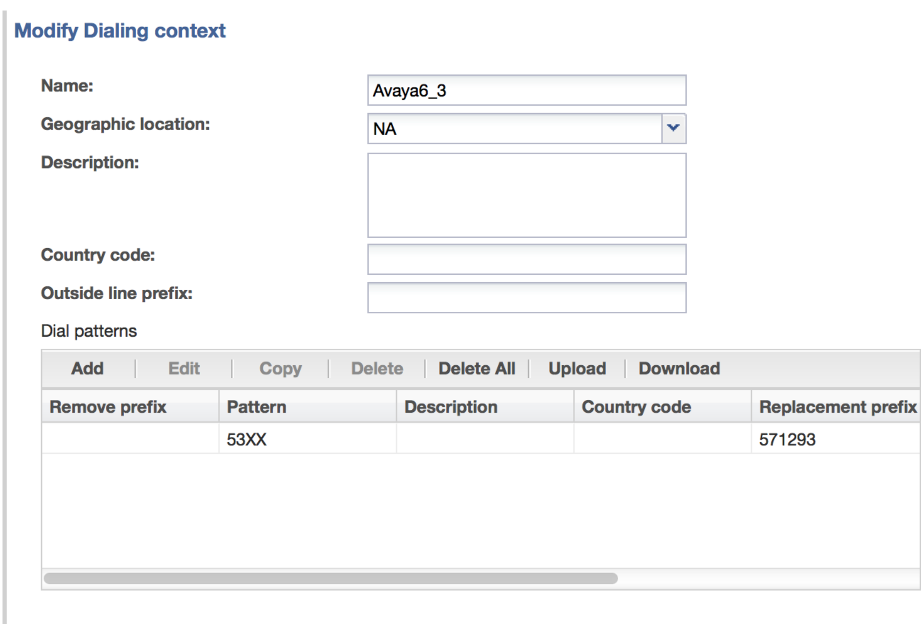
Add a dial pattern as shown below to enable 4 digit dialing and click **OK**. If the dialed digits match the pattern 53XX, the ECB transforms it to a 10 digit number by adding the prefix 571293.



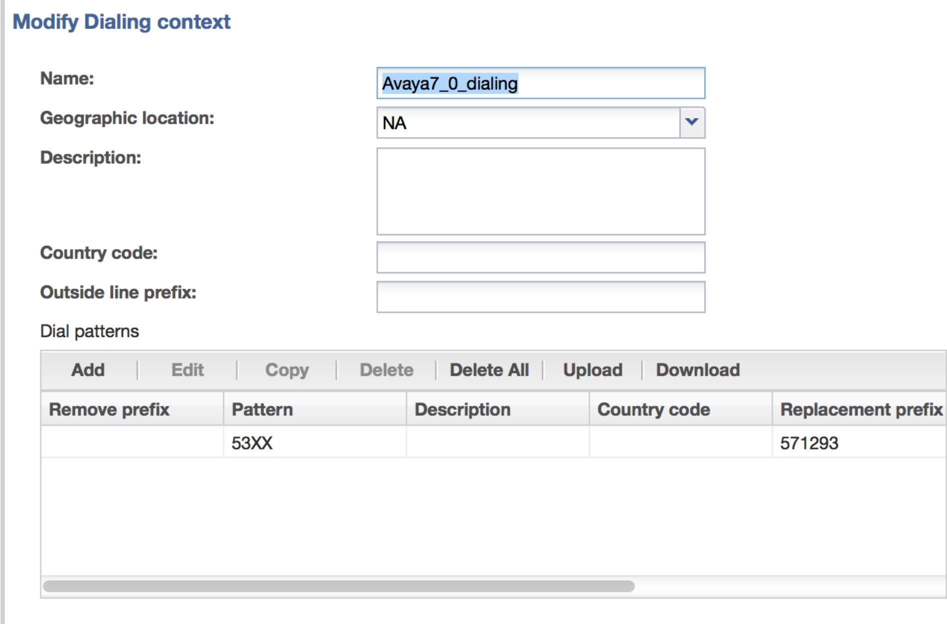
The LYNC2013 dialing context displays the configured dial pattern.



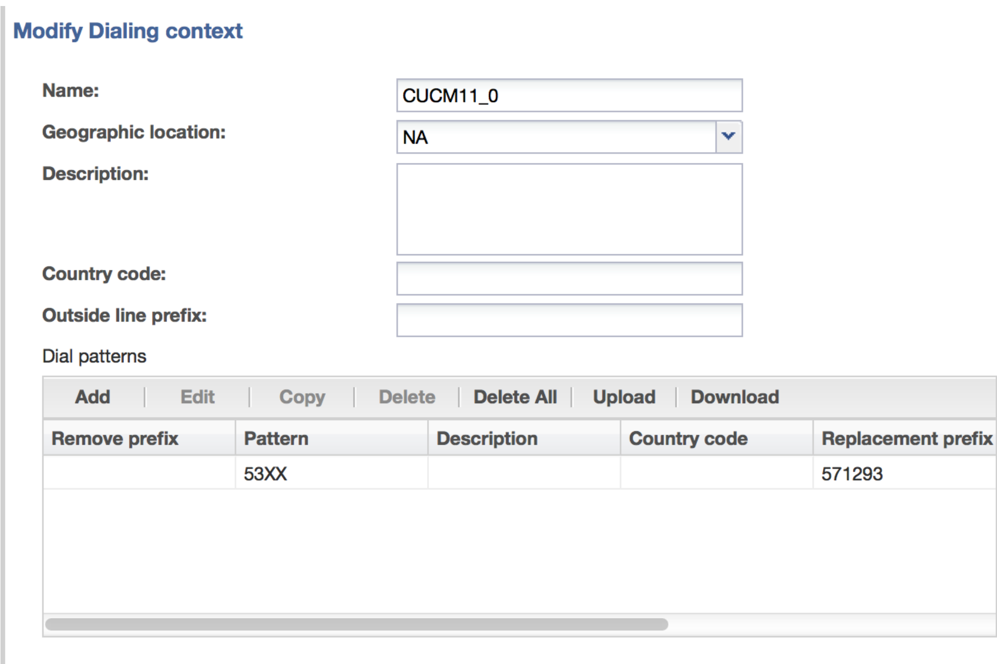
Add another dialing context under Oracle named Avaya6_3 with the following settings and click **OK**.



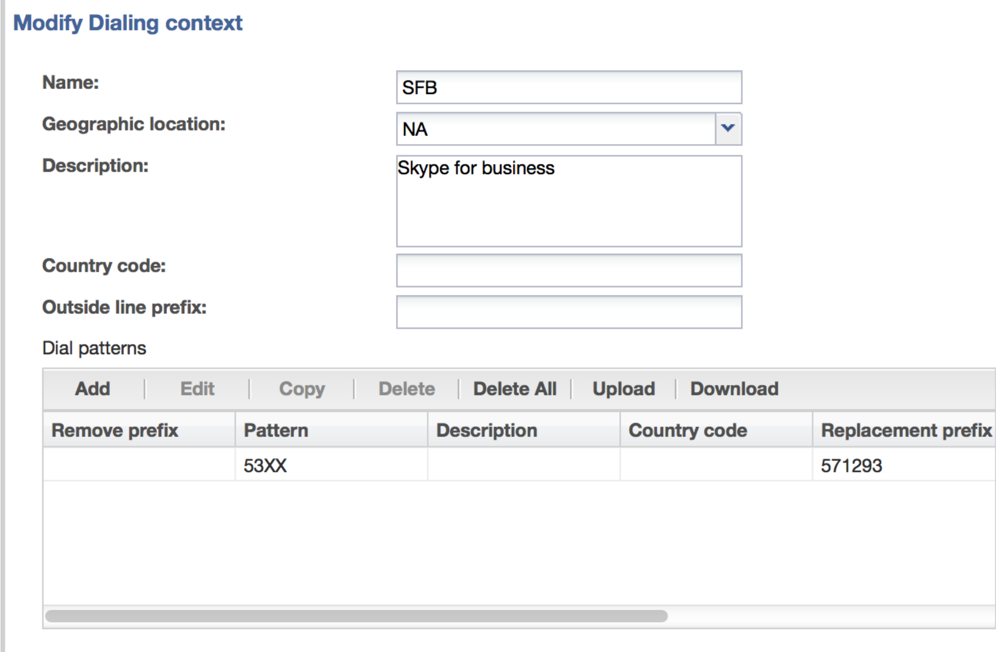
Add another dialing context under Oracle named Avaya7_0_dialing with the following settings and click **OK**.



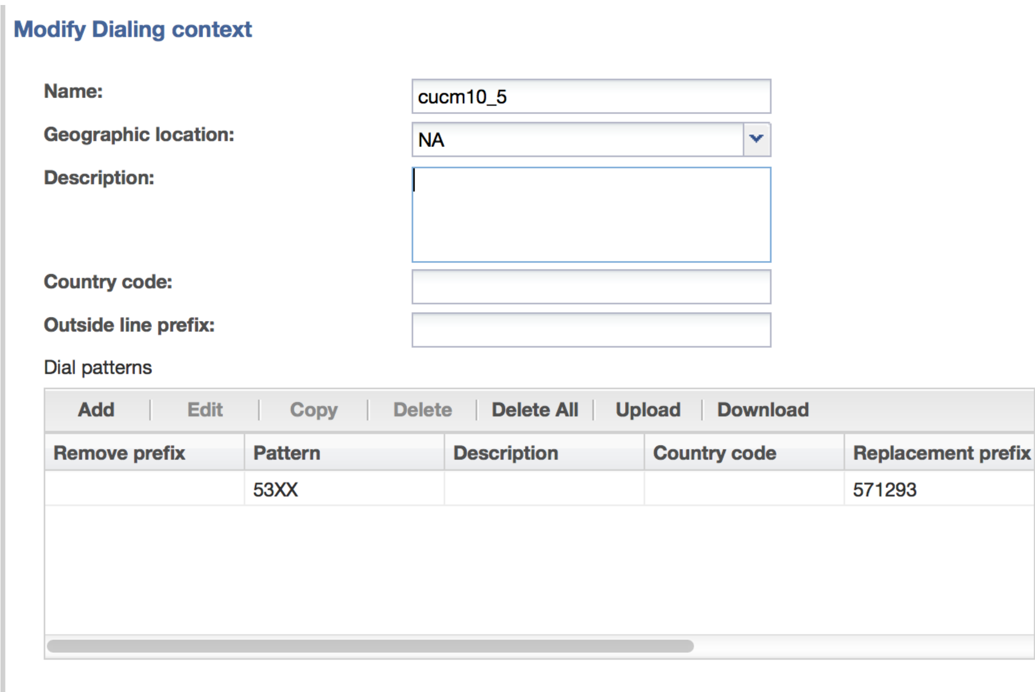
Add another dialing context under Oracle named CUCM11_0 with the following settings and click **OK**.



Add another dialing context under Oracle named Skype for Business with the following settings and click **OK**.



Add another dialing context under Oracle named cucm10_5 with the following settings and click **OK**.



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

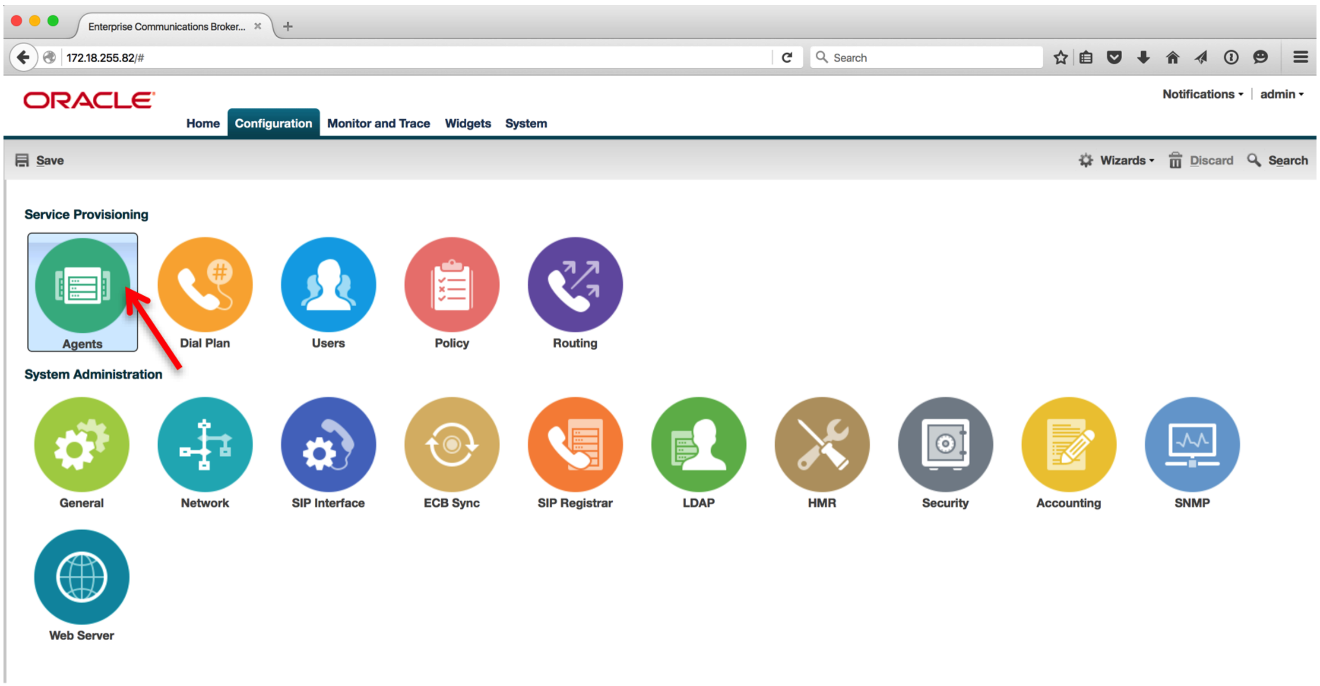
Dialing contexts

Refresh | Add | Edit | Copy | Delete | Upload | Download

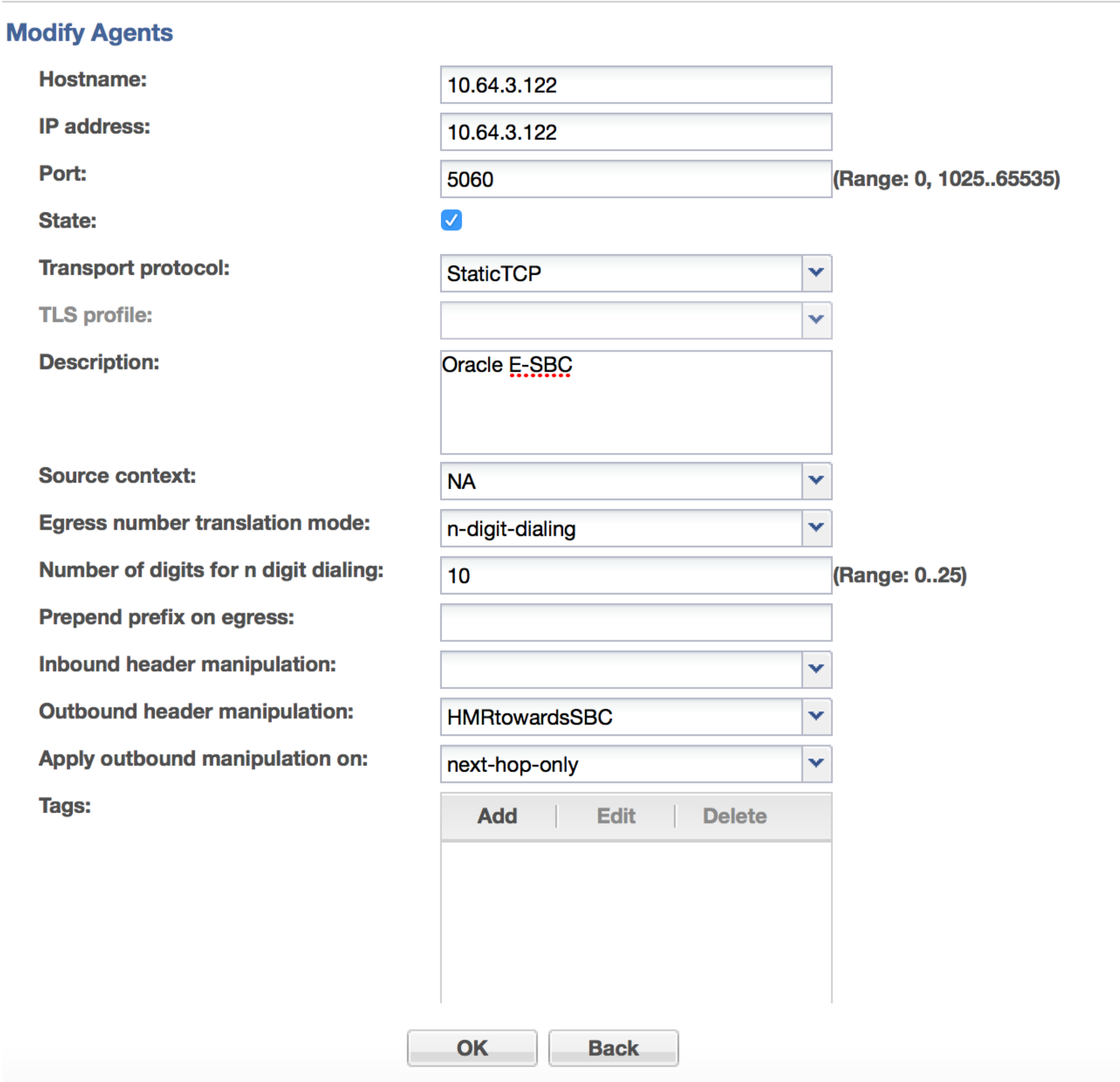
Name	Geographic location	Description	Country code	Outside line prefix
CORPORATE				
Oracle	NA			
Avaya6_3	NA			
Avaya7_0_dialing	NA			
CUCM11_0	NA			
LYNC2013	NA			
SFB	NA	Skype for business		
cucm10_5	NA			
GEOGRAPHIC				

Configure Agents

We will now configure the next hops in our routing paths – the Agents – which in our setup are the Cisco CUCM, Lync and SFB Mediation Servers, Avaya SM and the 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 Oracle E-SBC by configuring the hostname, IP address, port, transport protocol, egress number translation mode, number of digits for n digit dialing, source context, and Header Manipulation Rule as shown below.

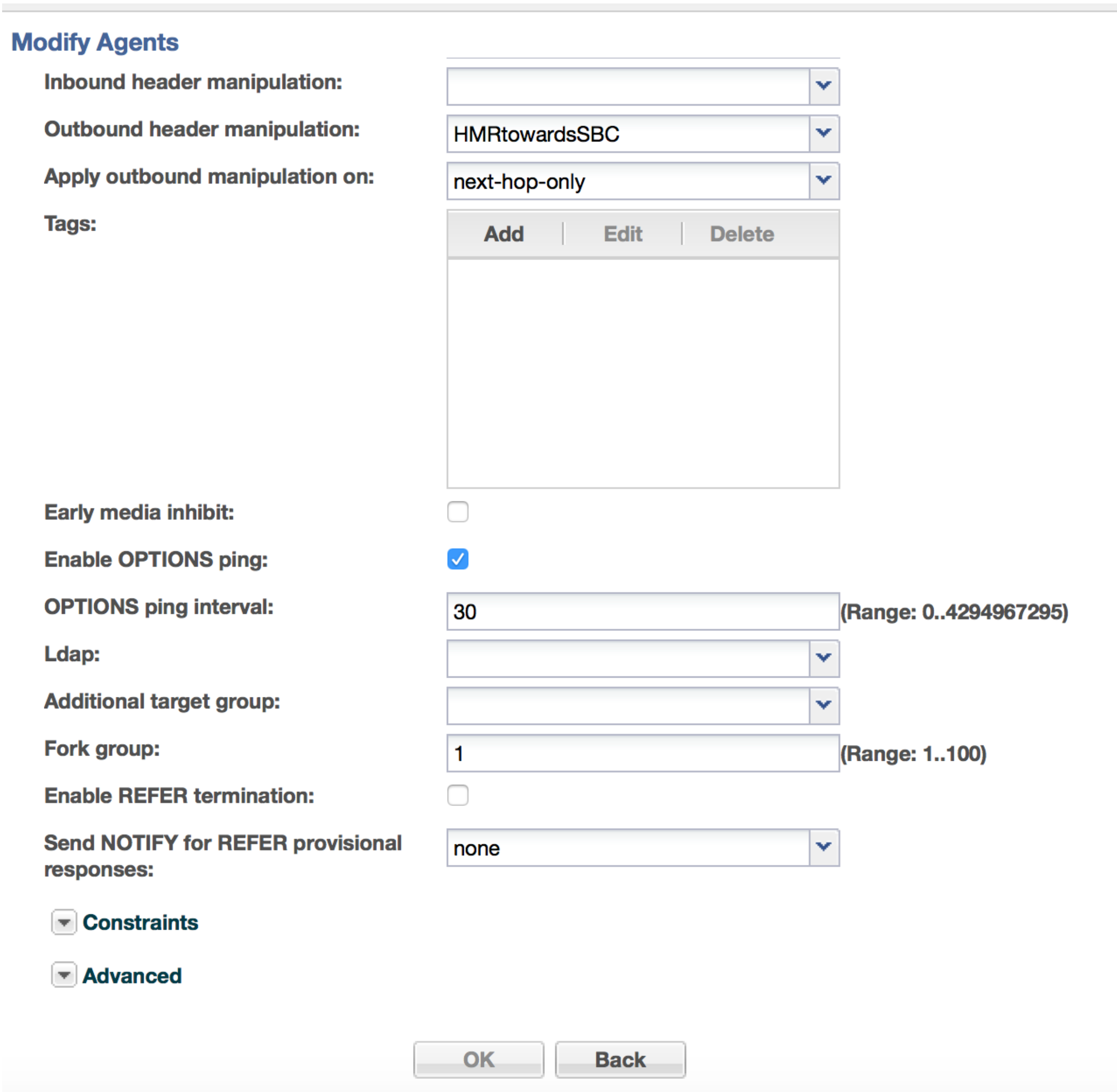


Modify Agents

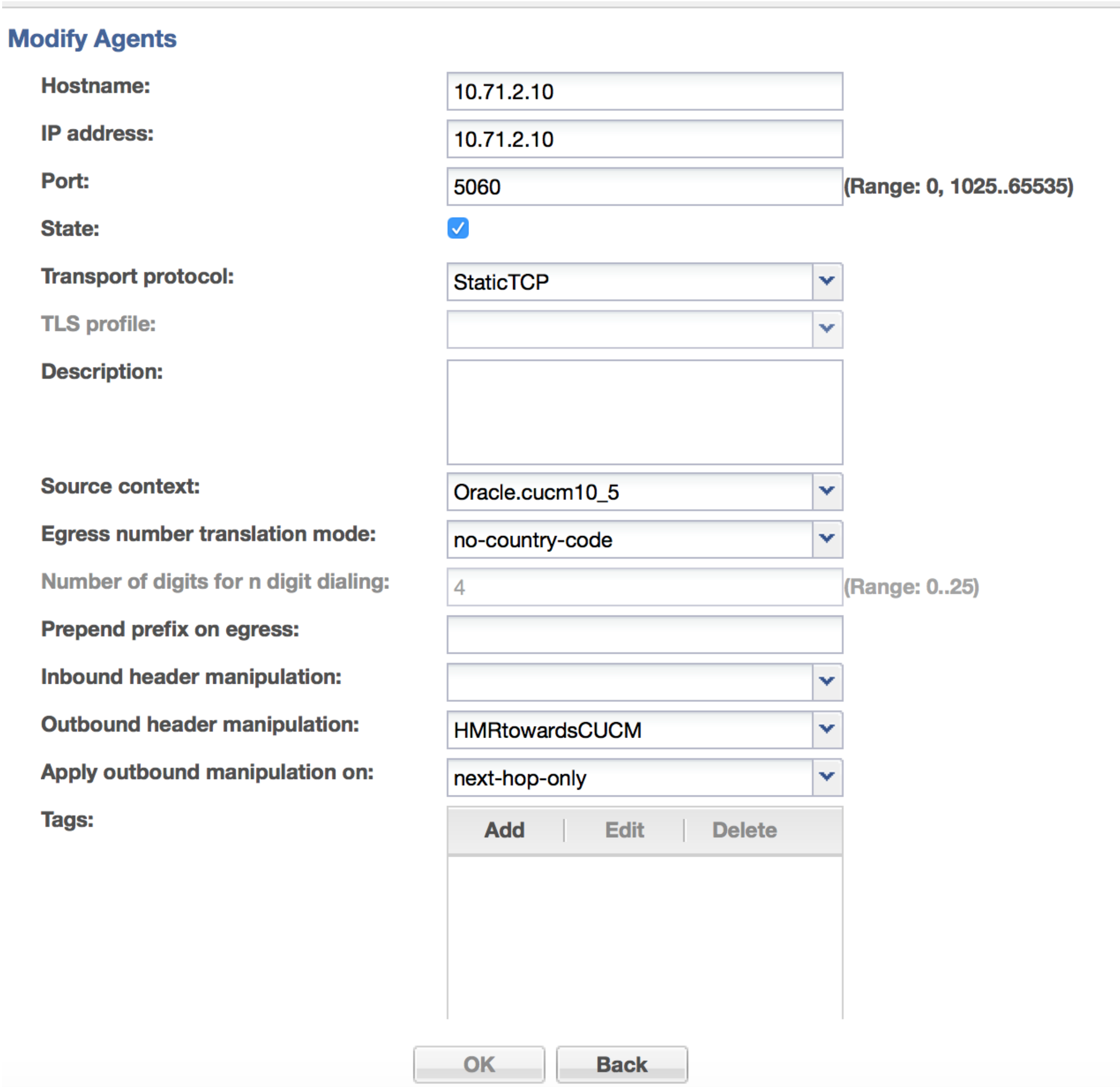
Hostname:	10.64.3.122
IP address:	10.64.3.122
Port:	5060 (Range: 0, 1025..65535)
State:	☑
Transport protocol:	StaticTCP
TLS profile:	
Description:	Oracle E-SBC
Source context:	NA
Egress number translation mode:	n-digit-dialing
Number of digits for n digit dialing:	10 (Range: 0..25)
Prepend prefix on egress:	
Inbound header manipulation:	
Outbound header manipulation:	HMRtowardsSBC
Apply outbound manipulation on:	next-hop-only
Tags:	Add | Edit | Delete

OK Back

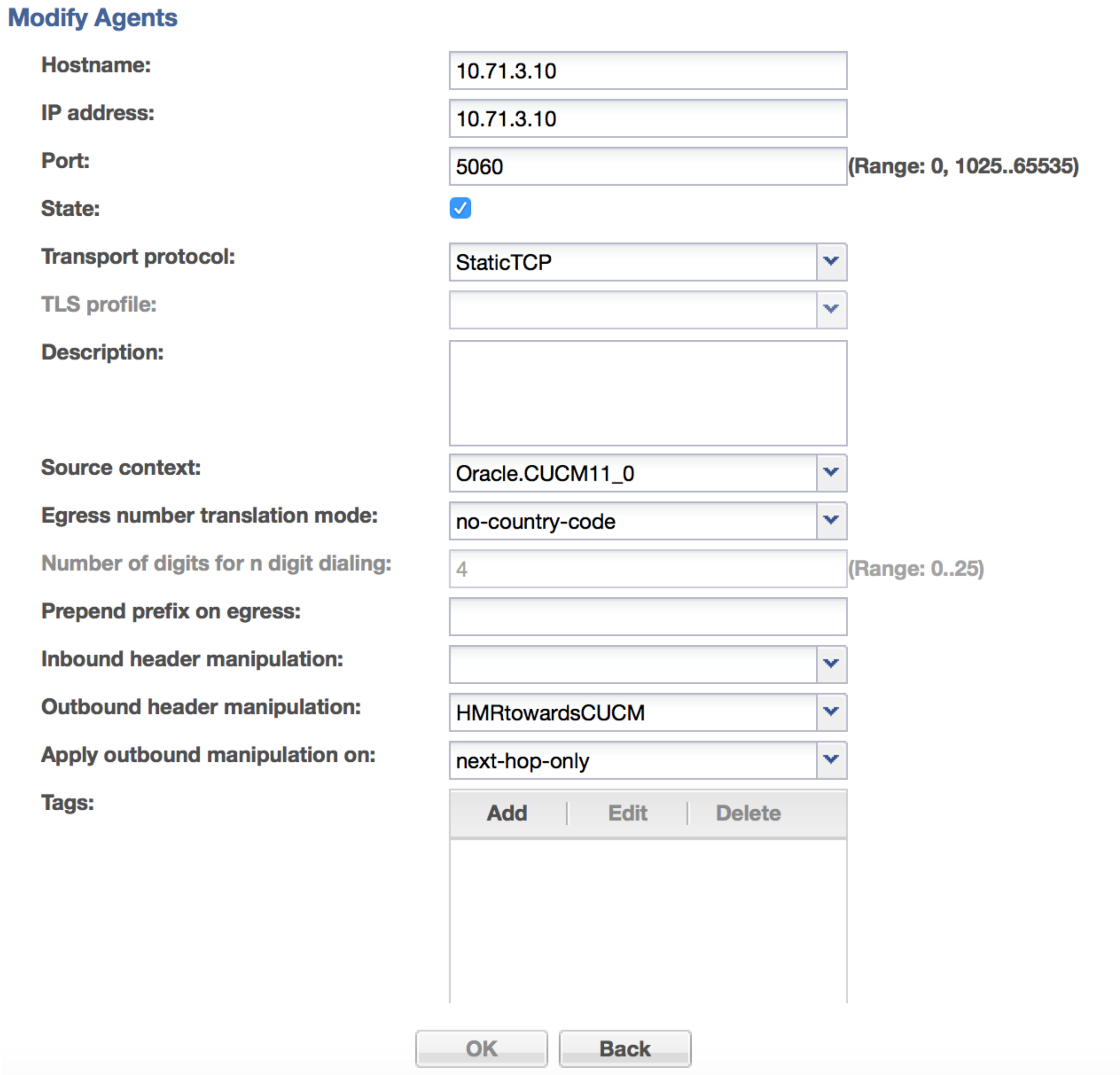
Scroll down to enable SIP OPTIONS to monitor agent health locally. Check the Enable OPTIONS ping check box and configure the OPTIONS ping interval to 30. Click OK.



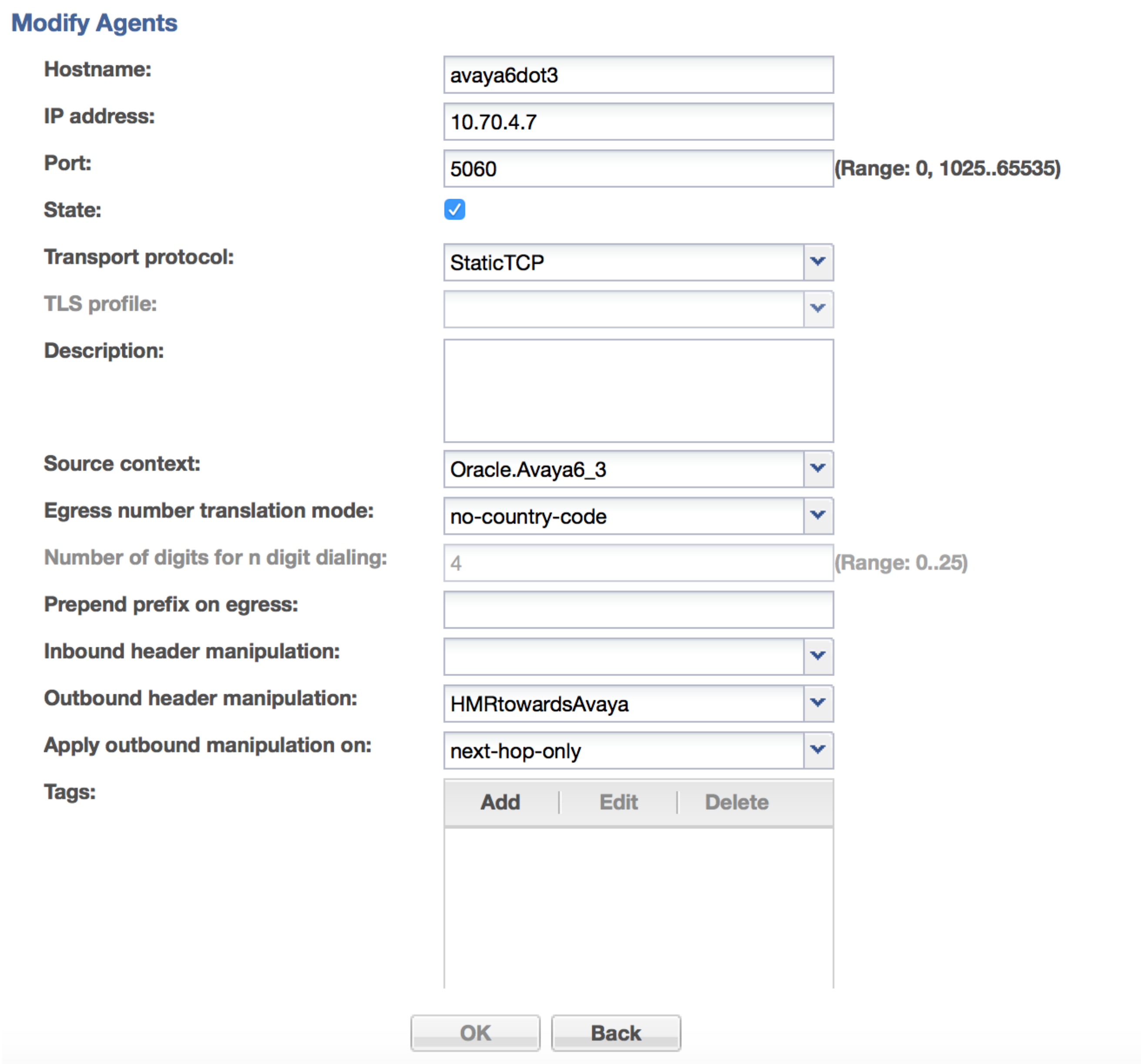
You will now see the Oracle E-SBC listed under **Agents**. Click **Add** to add the Cisco CUCM 10.5 server and also enable OPTIONS as shown in the previous step.



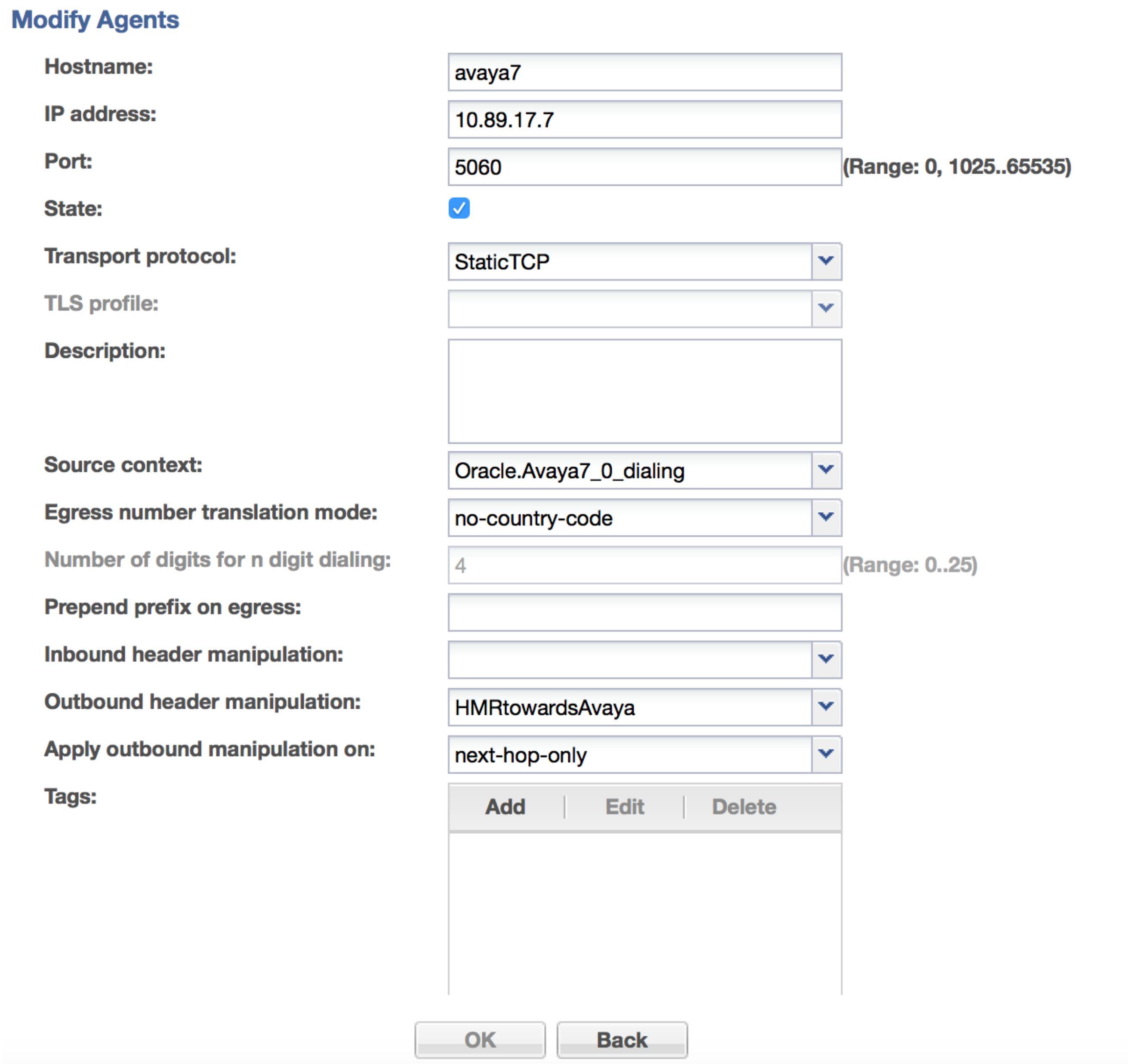
Click **Add** to add the Cisco CUCM 11.0 server and also enable OPTIONS as shown in the previous step.



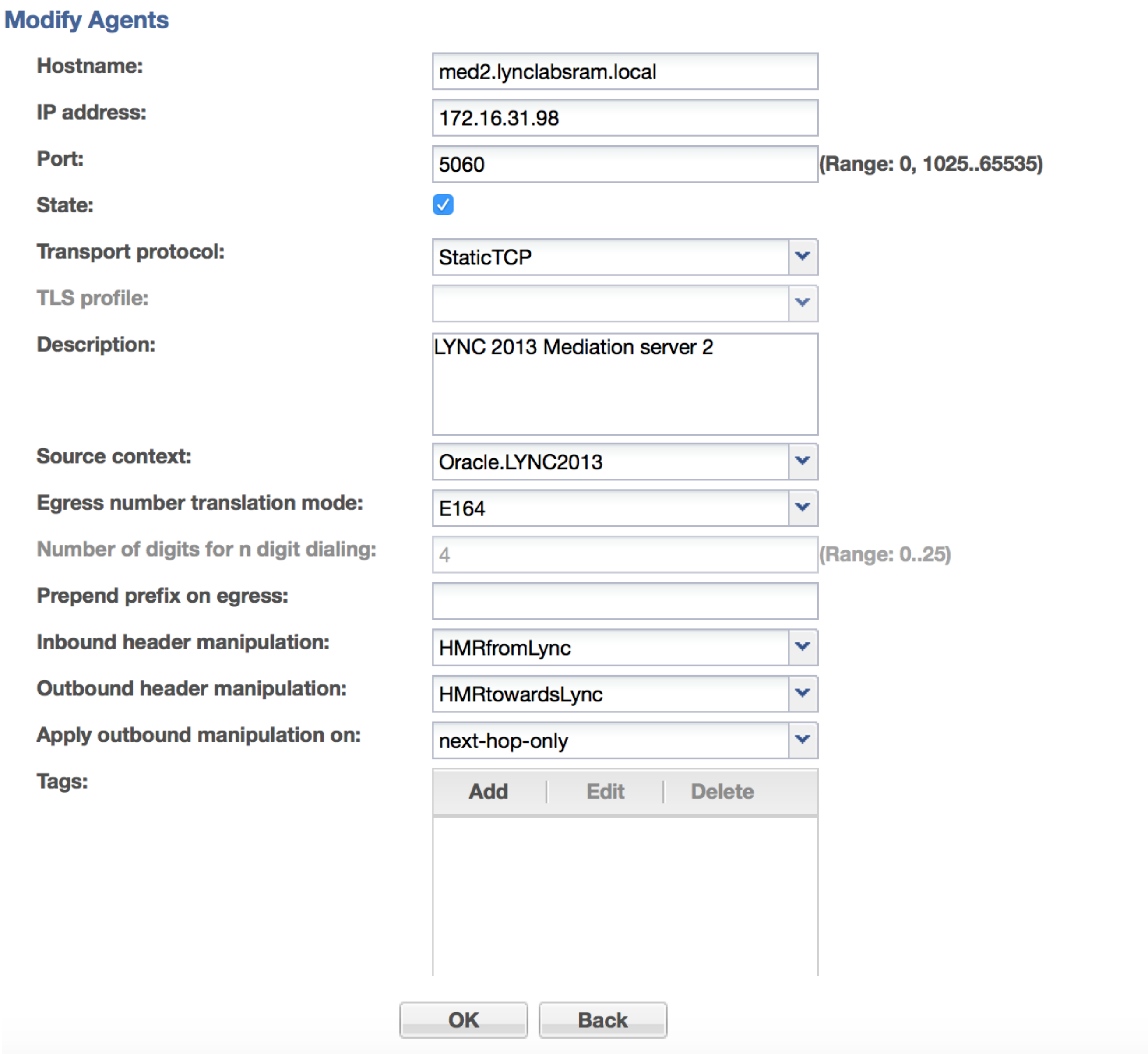
Click **Add** to add Avaya 6.3 server and also enable OPTIONS as shown in the previous step.



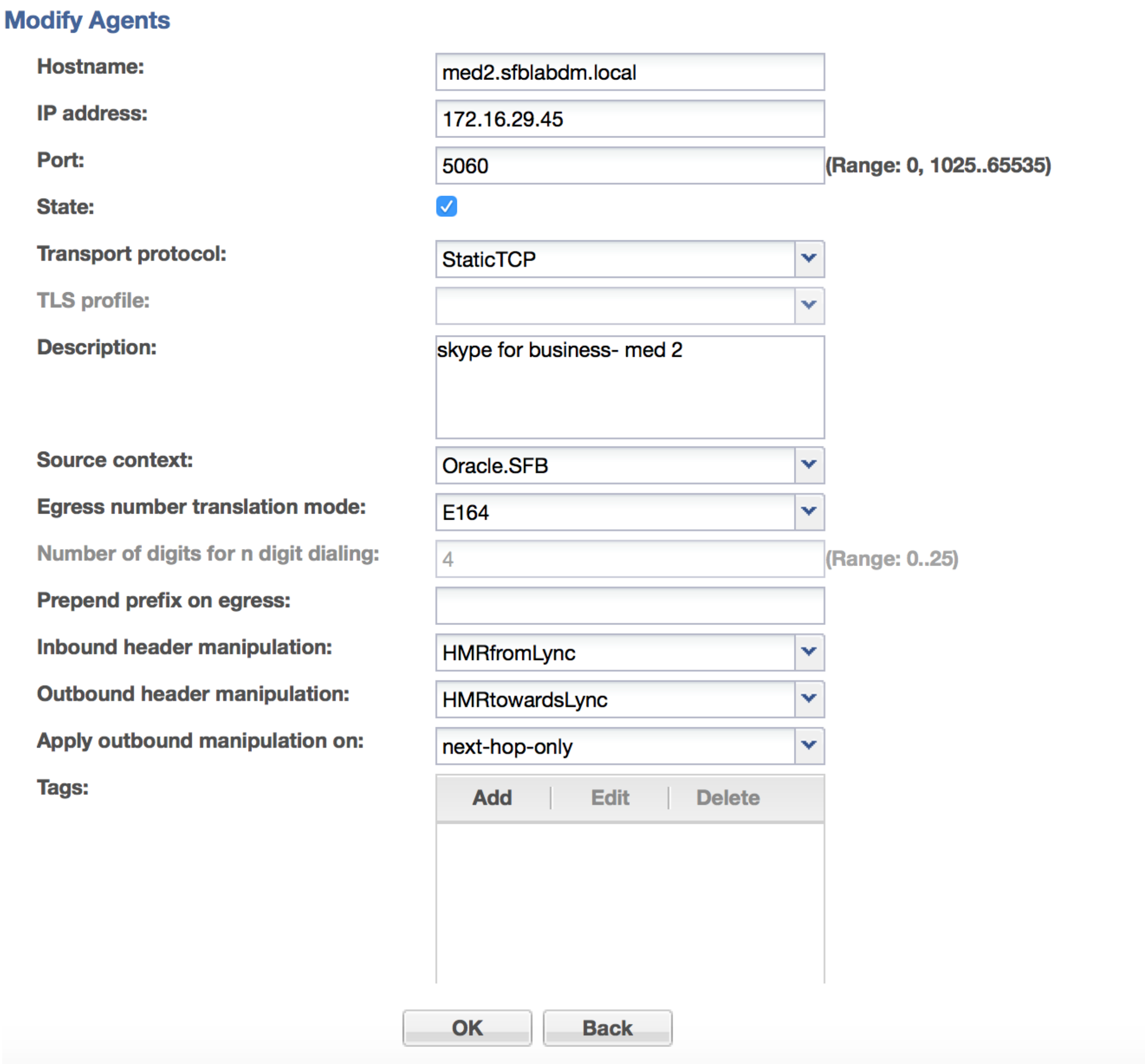
Click **Add** to add the Avaya 7.0 server and also enable OPTIONS as shown in the previous step.



Click **Add** to add the Lync 2013 mediation server and also enable OPTIONS as shown in the previous step.



Click **Add** to add the Skype for Business mediation server and also enable OPTIONS as shown in the previous step.

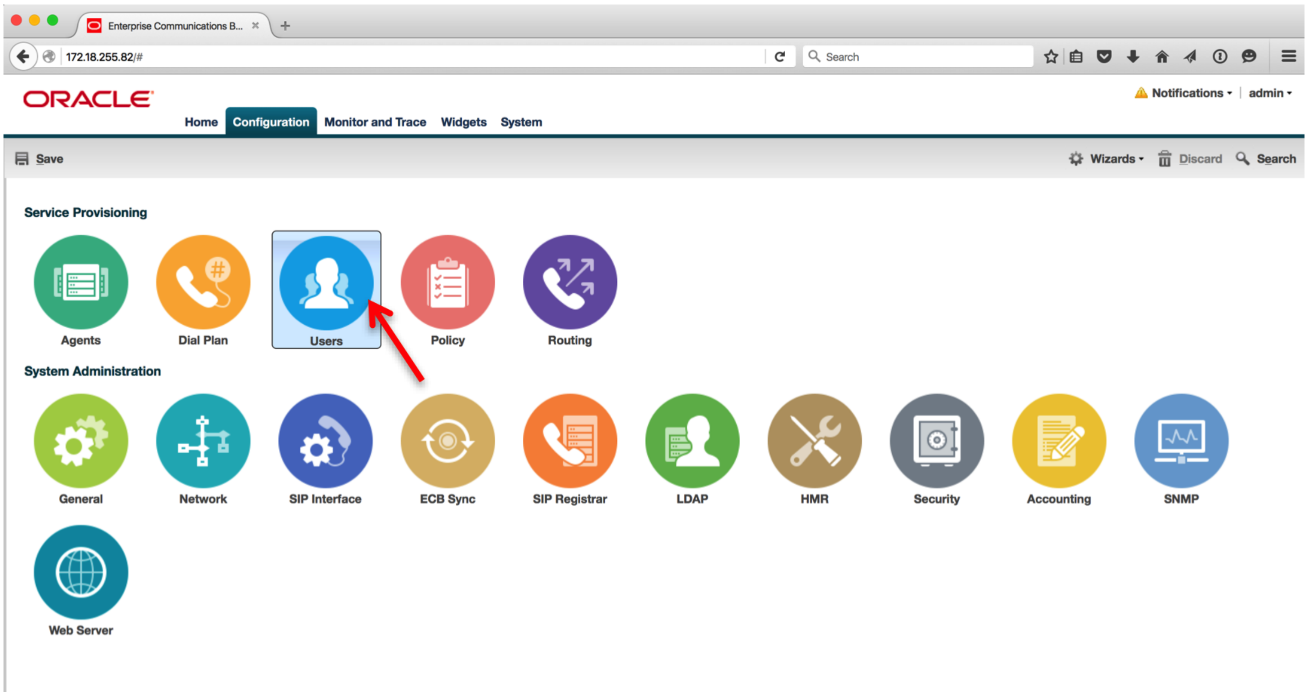


Configure Users

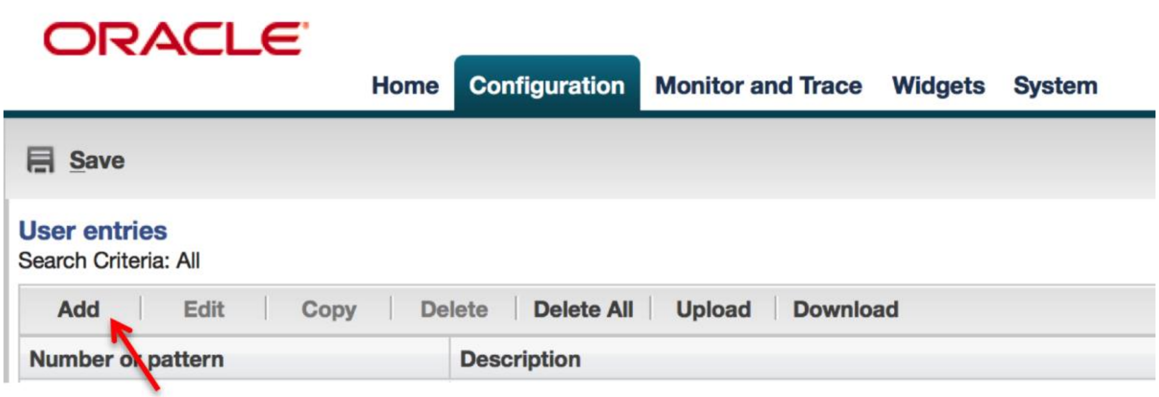
Next we will populate users in the User database. User entries can be added manually or uploaded in a format pre-configured to translate into a user database.

If the ECB and Active Directory are configured for LDAP integration, then it is NOT necessary to define users in the User database on the ECB.

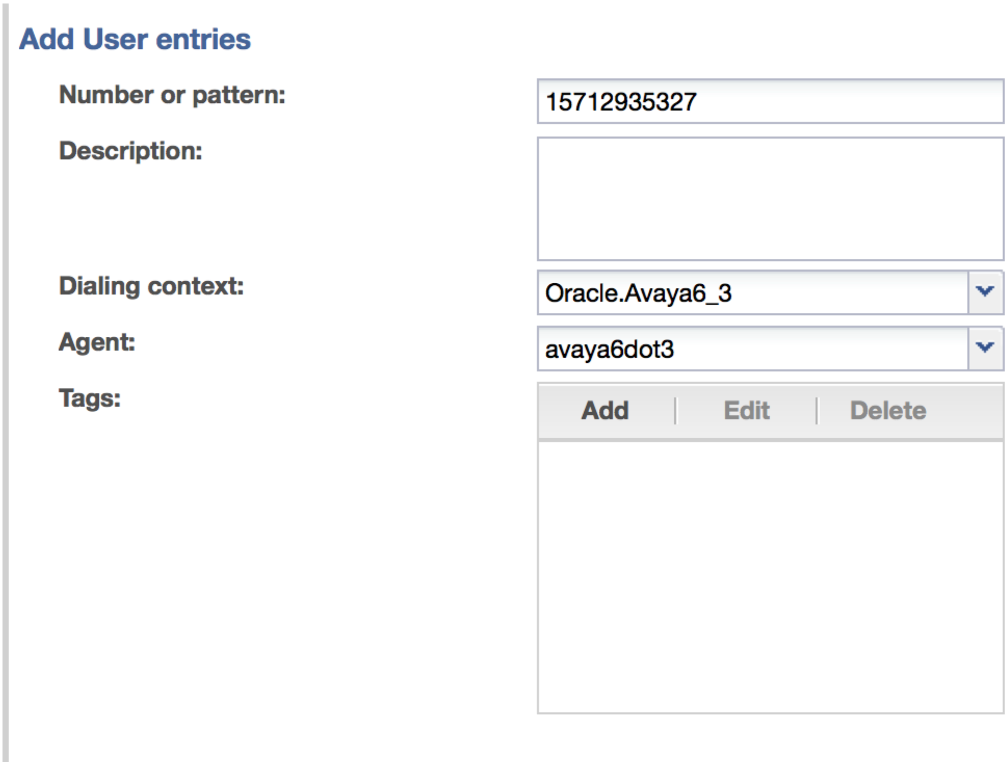
Click on the **Users** icon under **Service Provisioning**.



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



The **Add User entries** page will be displayed. You can enter the user numbers in E.164 format without the + (15712935327) or a number range (1571293[400-599]) in the **Number** field. Assign the appropriate **Agent** and **Dialing context** and click **OK**.



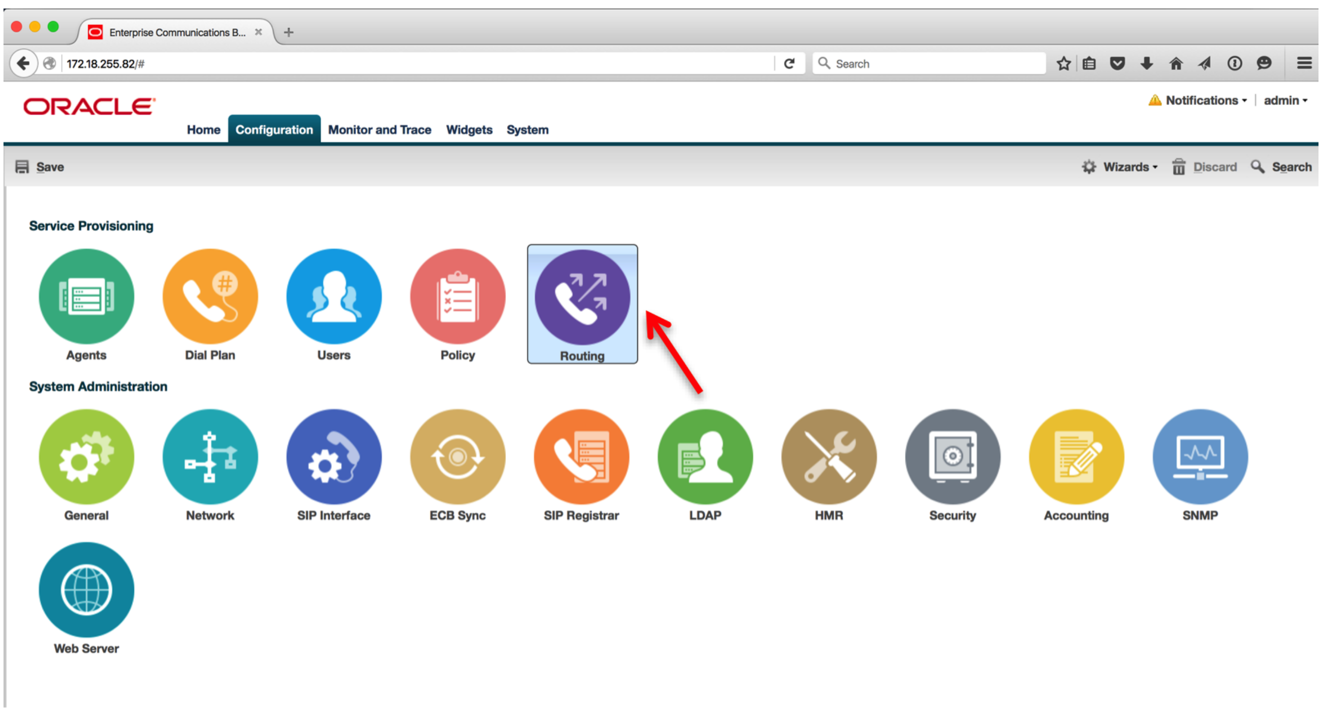
Continue adding users as shown above using the corresponding agents and dialing contexts.

Configure Routing

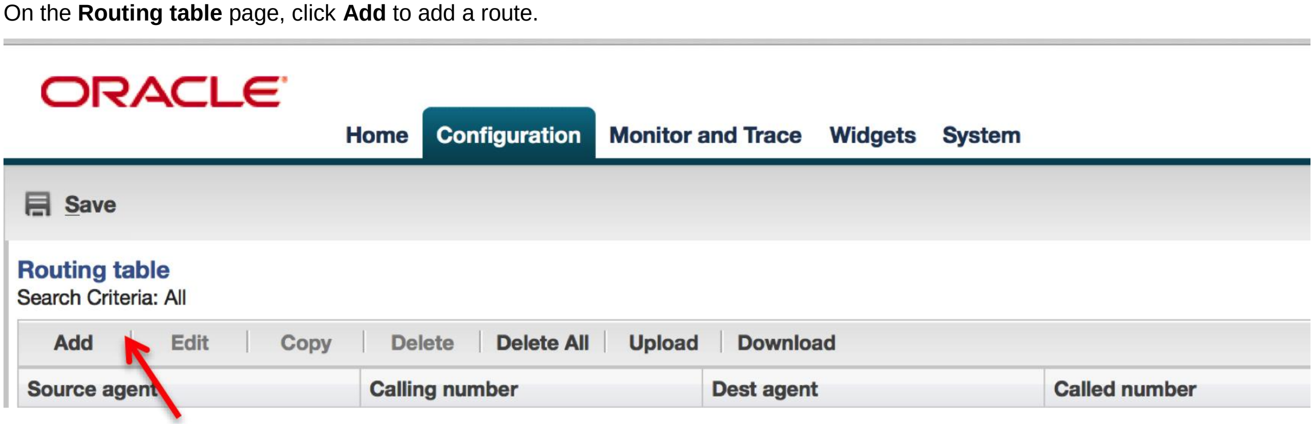
The ECB performs its session routing via the route configuration. The route configuration establishes hop-by-hop paths to signaling endpoints. Oracle ECB routing configuration allows the user to specify a route's cost to specify route preference. Cost may or may not be based on monetary considerations. 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 Oracle 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.

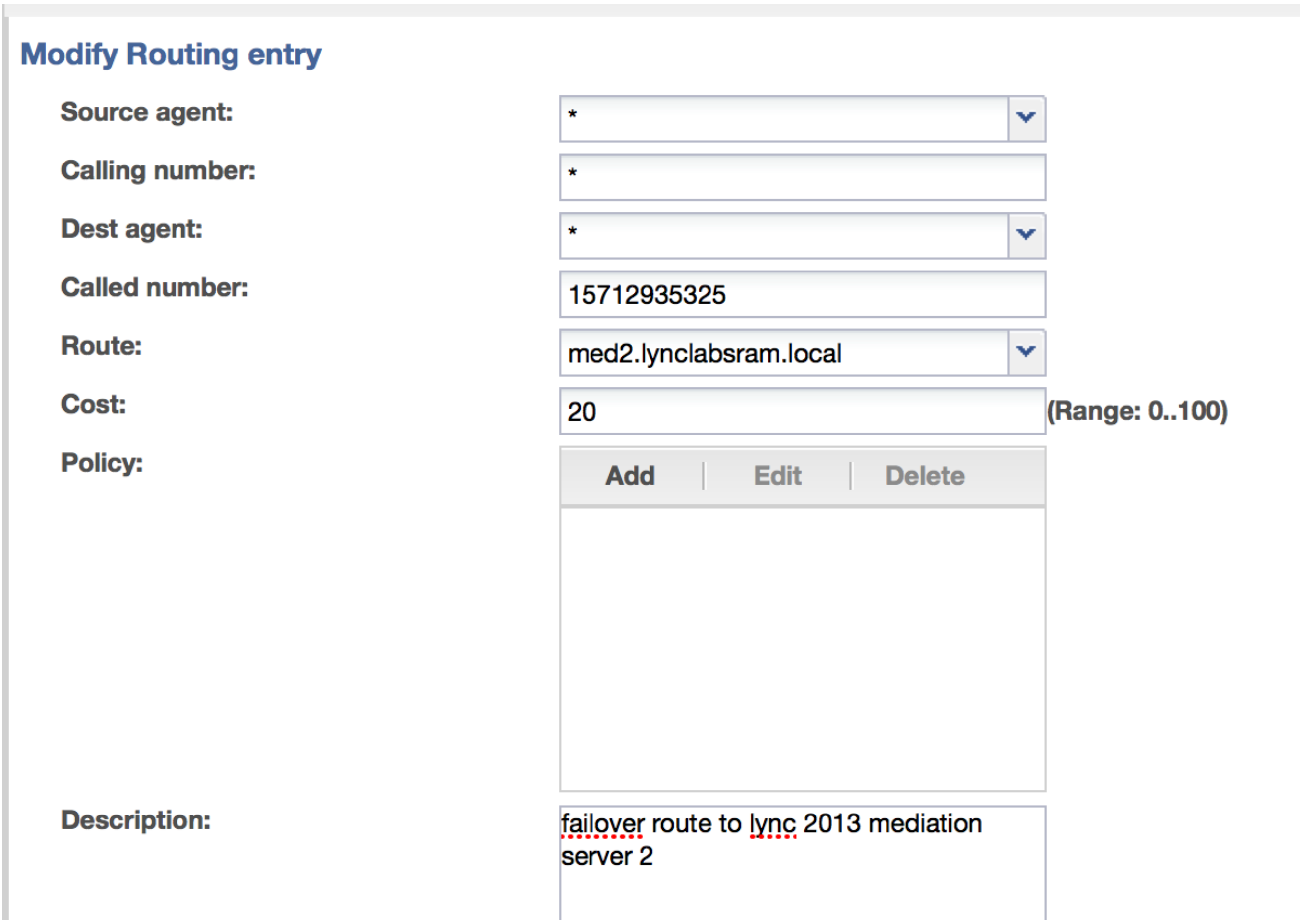
We need not configure a route for the users defined in the user database as the ECB will use their configured agents as next hop to route the calls. Since ECB does not support DNS load balancing as of now, the Lync users are assigned with one mediation server as their agent. To ensure the calls complete 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.

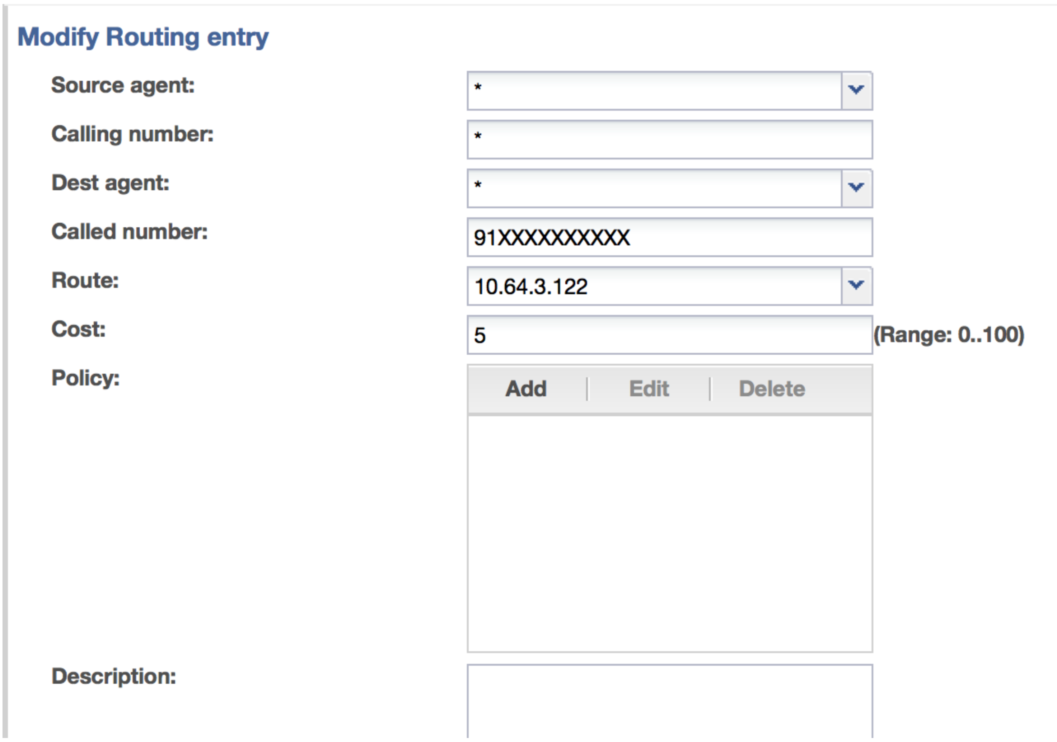


Add a routing entry for the Lync 2013 user – 15712935325 with the **Route** set to the second mediation server – med2.lynclabsram.local with a cost of 20 and click **OK**.

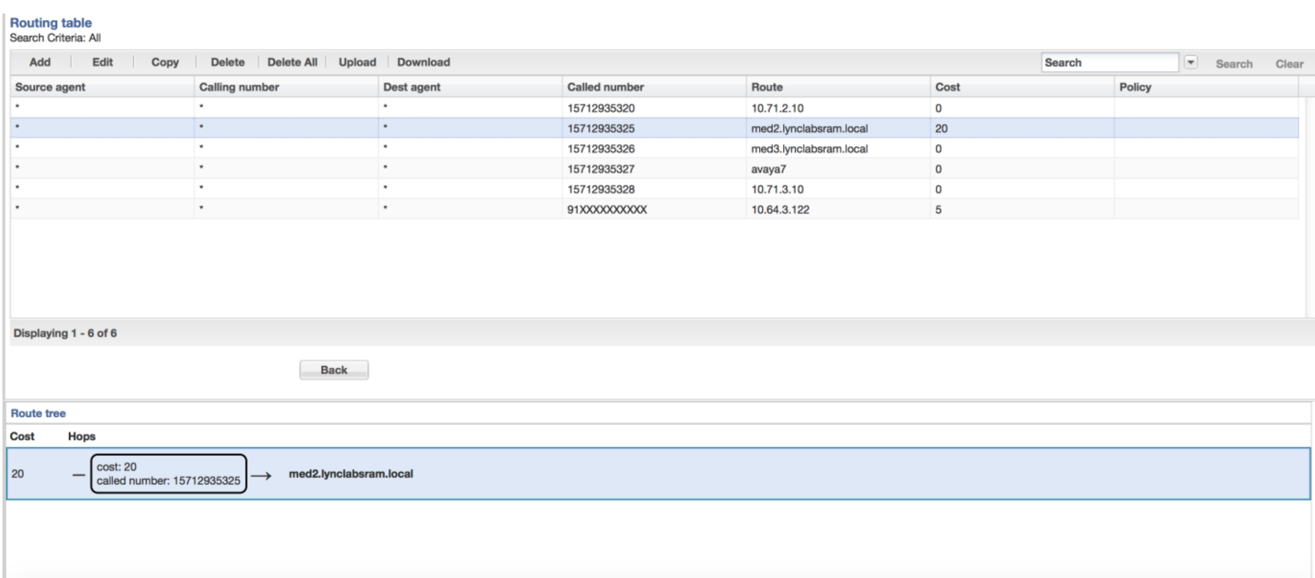


When the ECB receives a call for 15712935325, it looks up the user DB and finds that this user is associated to med1.lynclabsram.local and routes the call to it. If this agent is down, ECB will find the above entry and route the call to the second agent of the pool – med2.lynclabsram.local.

Configure a route for called number 91XXXXXXXXXX to point to the Oracle E-SBC. If a user dials an external number by dialing 9 and then 1 and the number, the ECB will route the call to the E-SBC to get to the service provider network.



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.



Source agent	Calling number	Dest agent	Called number	Route	Cost	Policy
*	*	*	15712935320	10.71.2.10	0	
*	*	*	15712935325	med2.lynclabsram.local	20	
*	*	*	15712935326	med3.lynclabsram.local	0	
*	*	*	15712935327	avaya7	0	
*	*	*	15712935328	10.71.3.10	0	
*	*	*	91XXXXXXXXXX	10.64.3.122	5	

Configure LDAP Integration with Active Directory

This is an optional step. If LDAP is used, then users do not need to be defined in the ECB's User database or in the Routing database. The Oracle ECB supports LDAP as a communications mechanism for interaction with an LDAP server. For many enterprises, this means utilizing Active Directory, a common LDAP-based service, to request information used in SIP session routing and authentication. The Oracle ECB's LDAP client requires configuration on the Oracle ECB and the LDAP server.

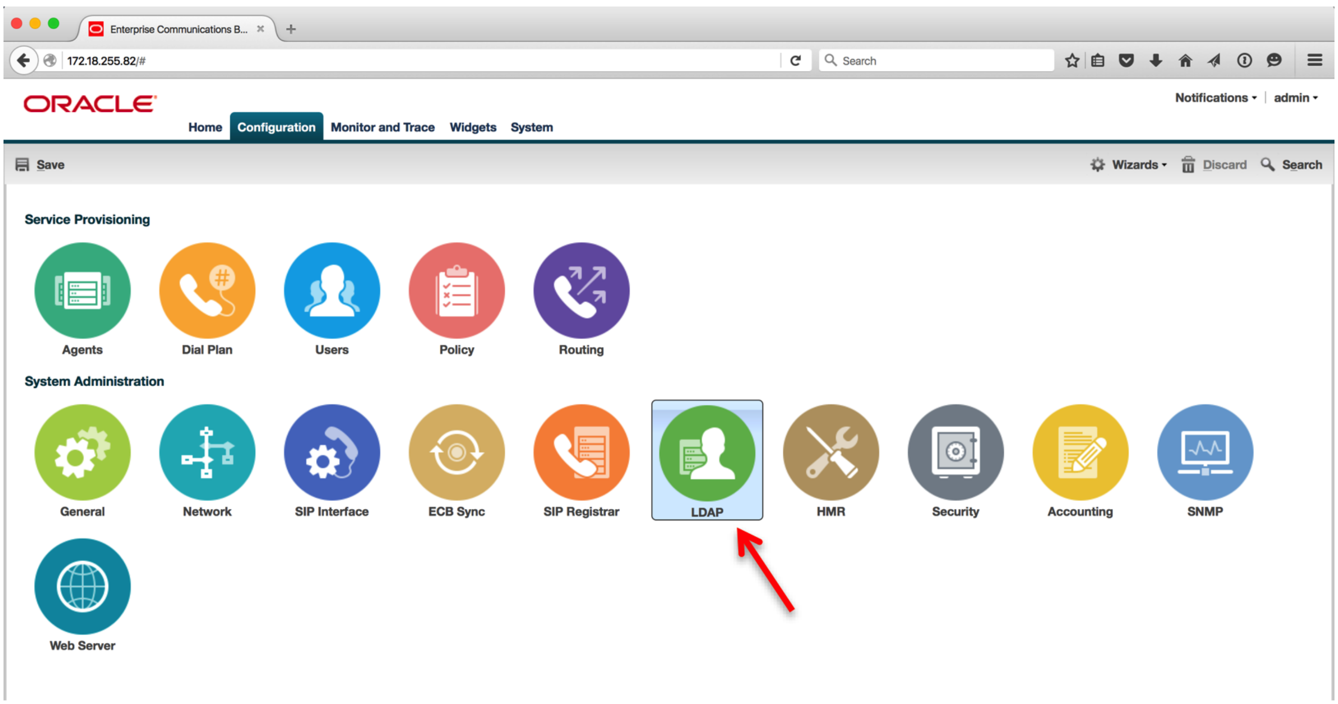
Configuration aspects of LDAP client configuration include:

• LDAP server access—The user specifies LDAP server location and access preferences.

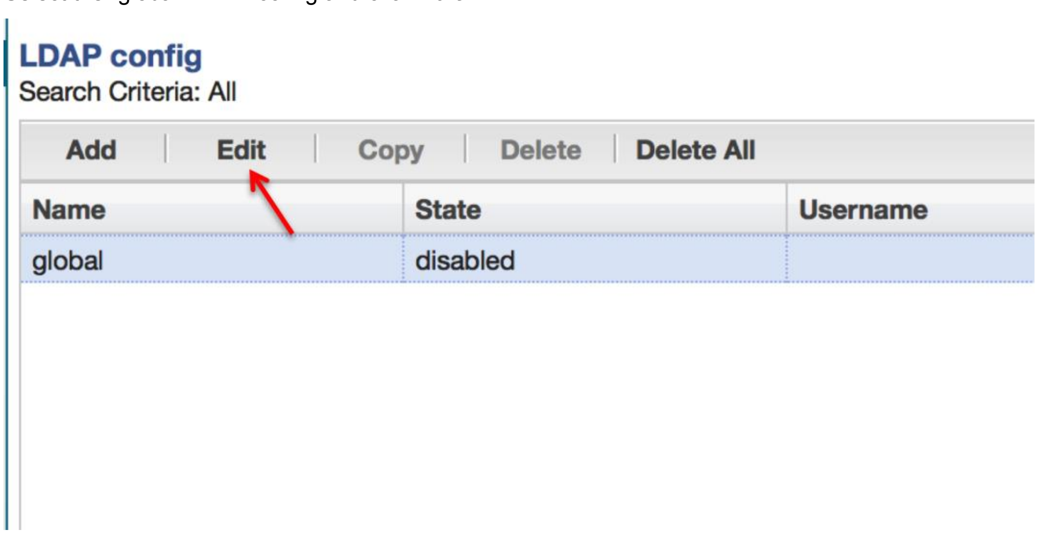
• Routing queries—The user specifies the conditions wherein the Oracle ECB performs an LDAP dip to obtain location information (home agent) for FROM and REQUEST-URIs.

• AoR queries—Optionally searches for additional AoR matches in Active Directory so that it can create additional routes to target users that have contacts stored in separate records.

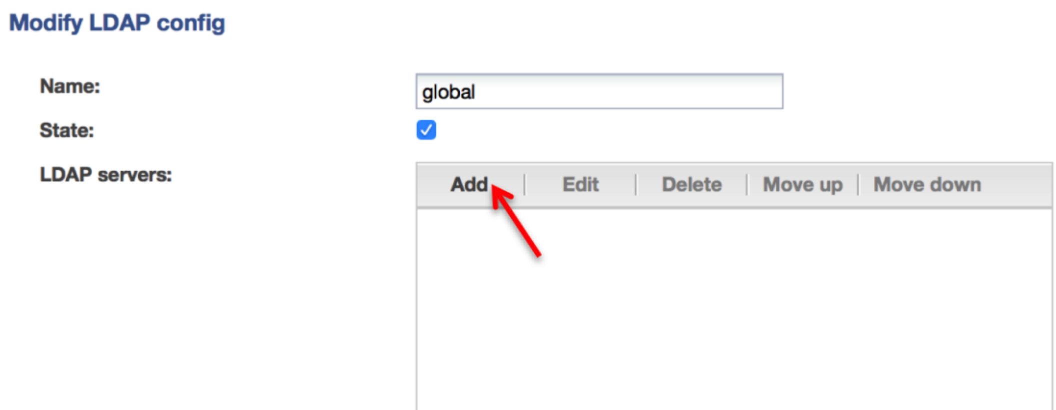
Click on the **LDAP** icon under **System Administration**.



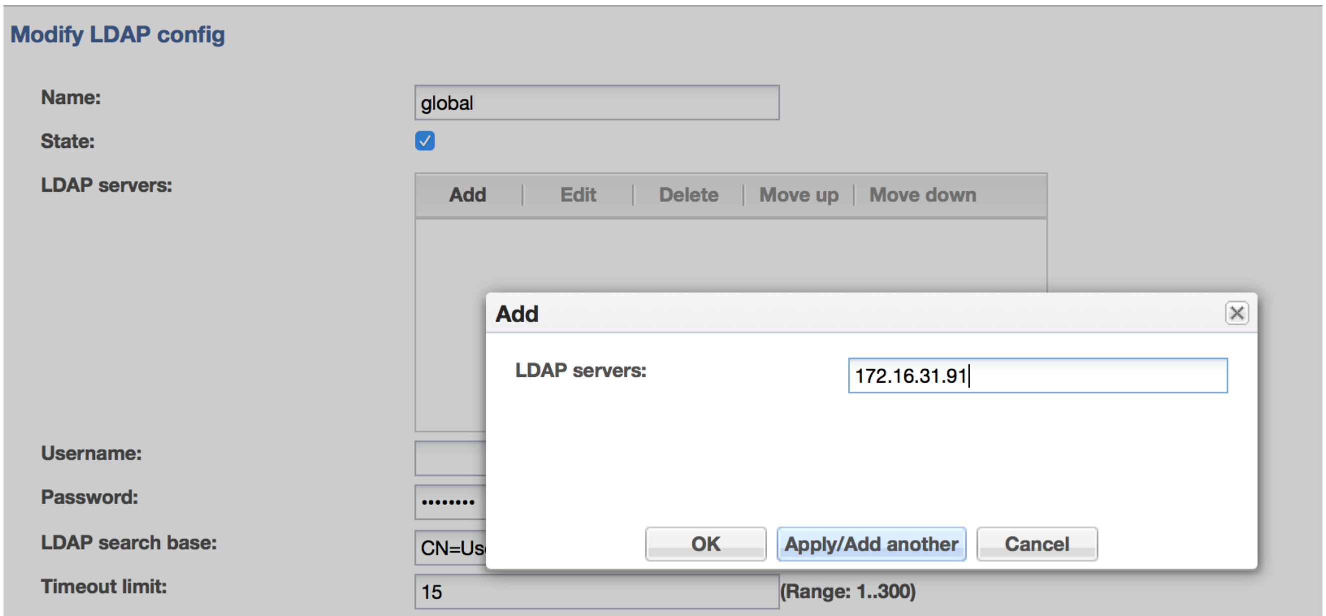
Select the "global" LDAP config and click **Edit**.



Check the **State** checkbox to enable LDAP, then under **LDAP servers** click **Add**:

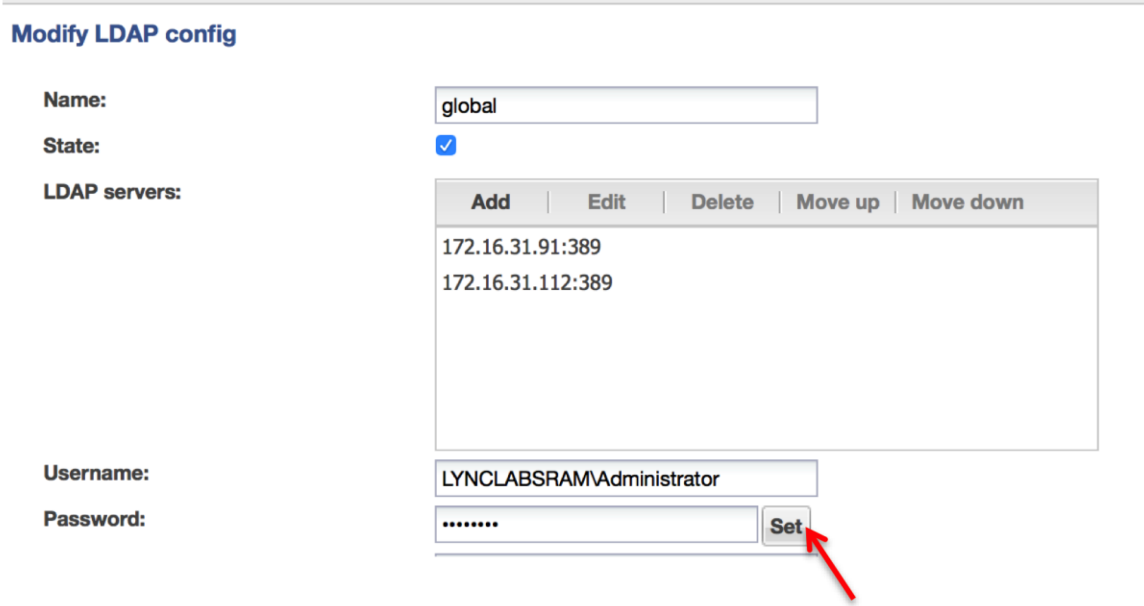


Enter the LDAP server's IP address. If no port is specified, the ECB will use the default of 389.

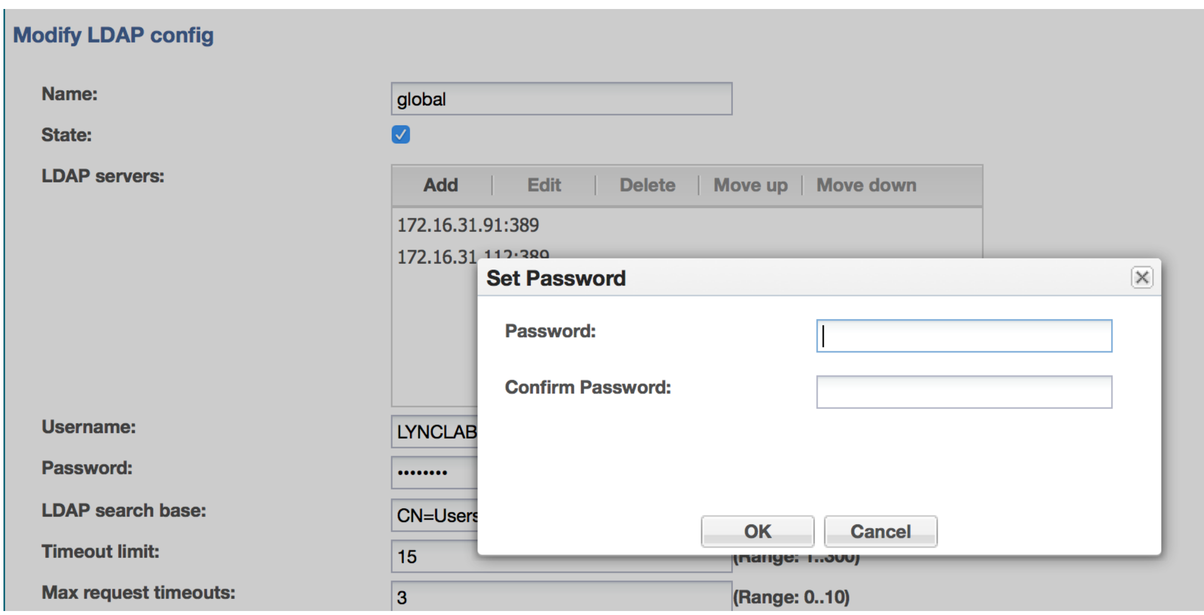


Click **Apply/Add another** to enter a secondary LDAP server IP, or click **OK** to use only one. If two are entered, the ECB will attempt to communicate with the first one, and if there is a failure, it will try the next server in the list on the next call.

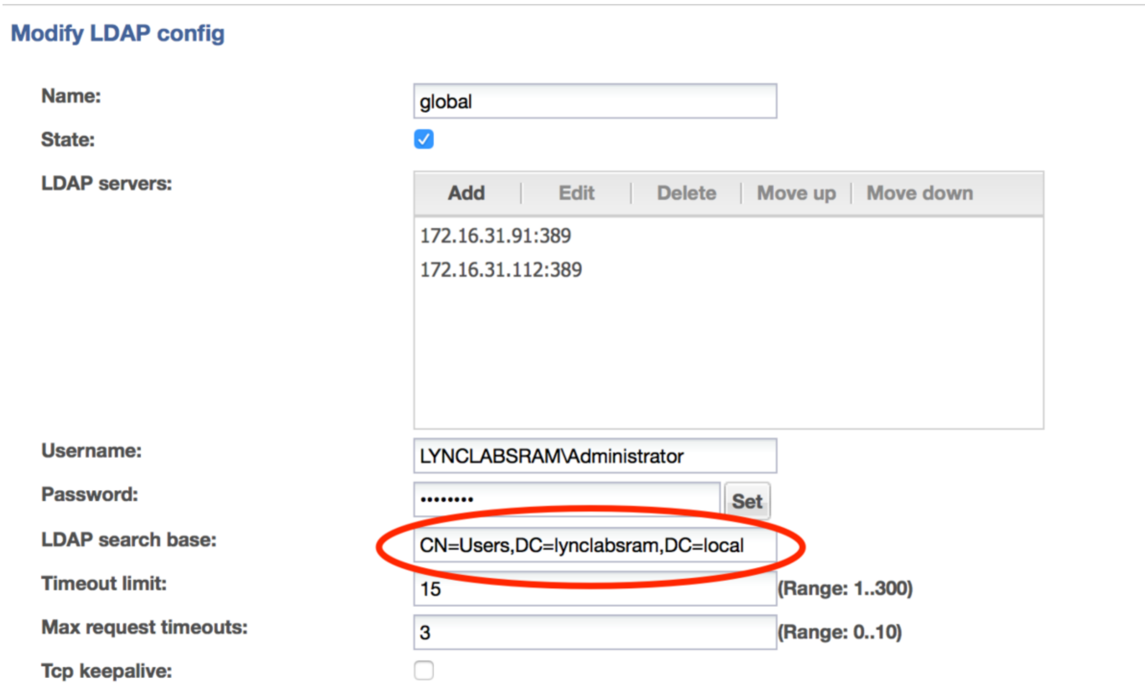
Enter the LDAP sever username in the Username field, then click on Set next to the Password field to enter the LDAP server password.



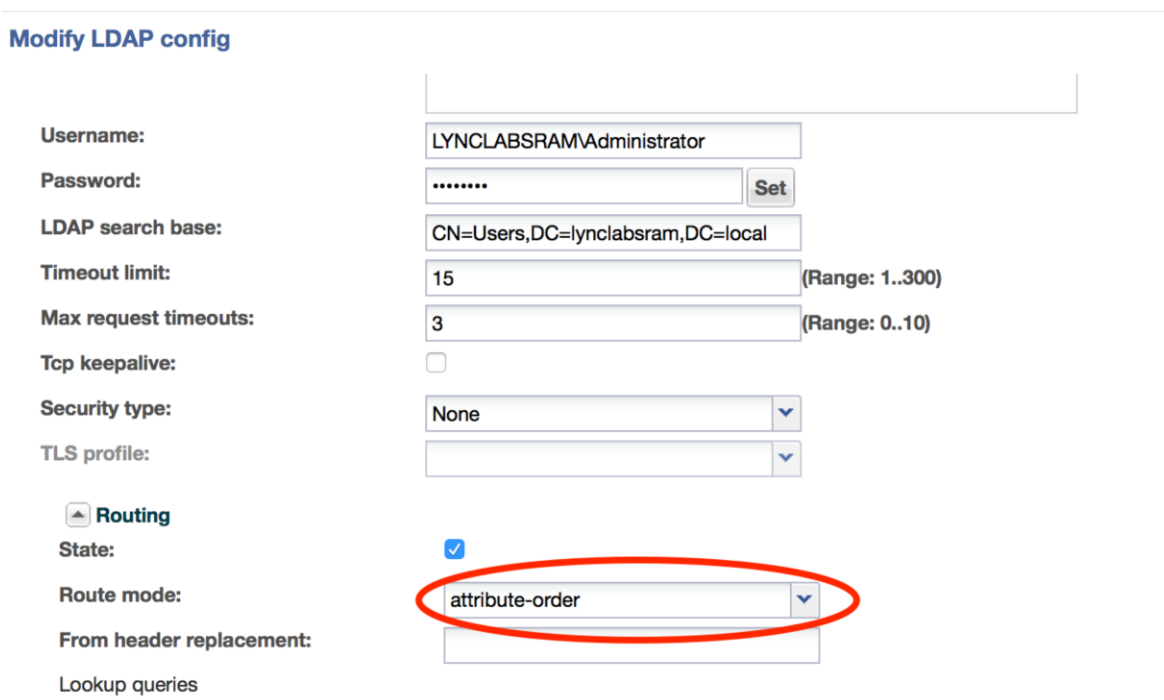
Enter and re-enter the password, then click OK.



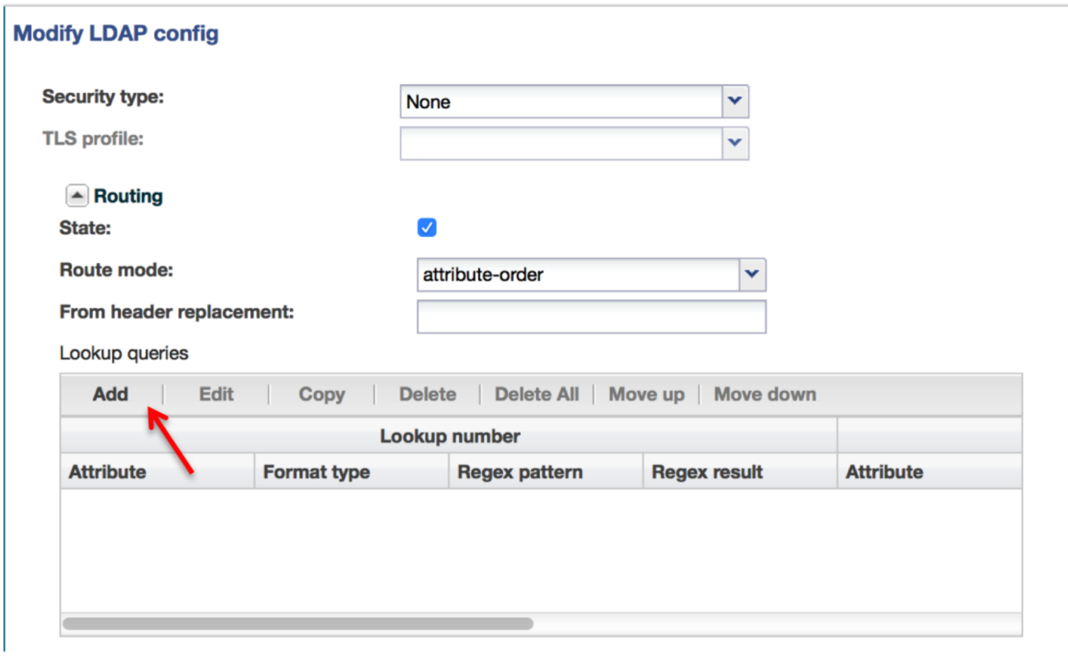
Enter the **LDAP search base**. In the test lab, we used “CN=Users,DC=lynclabsram,DC=local” as shown in the following screenshot, where CN stands for Common Name and DC stands for Domain Component.



Scroll down and select “attribute-order” under the **Route mode**.

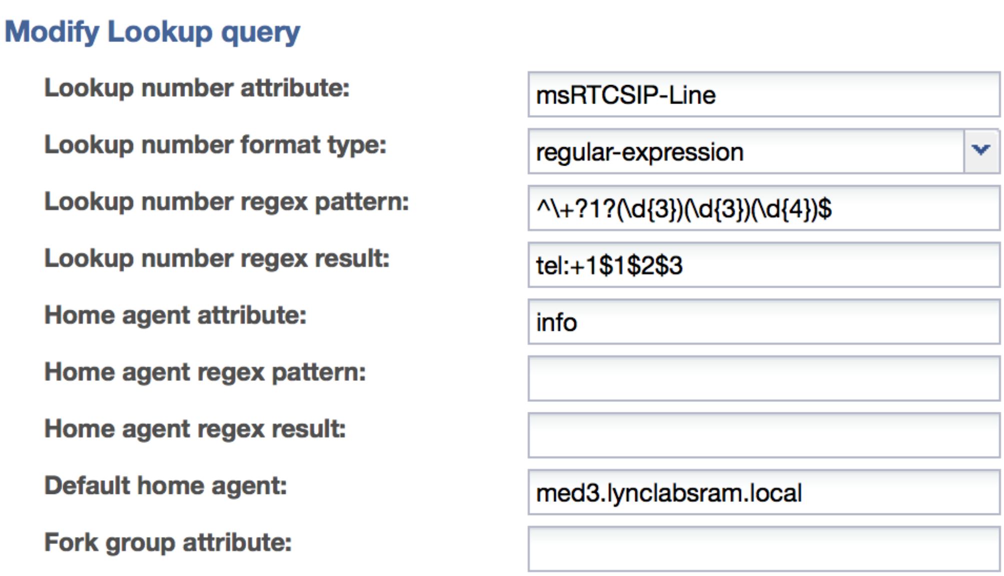


Under the Lookup queries, click **Add**:

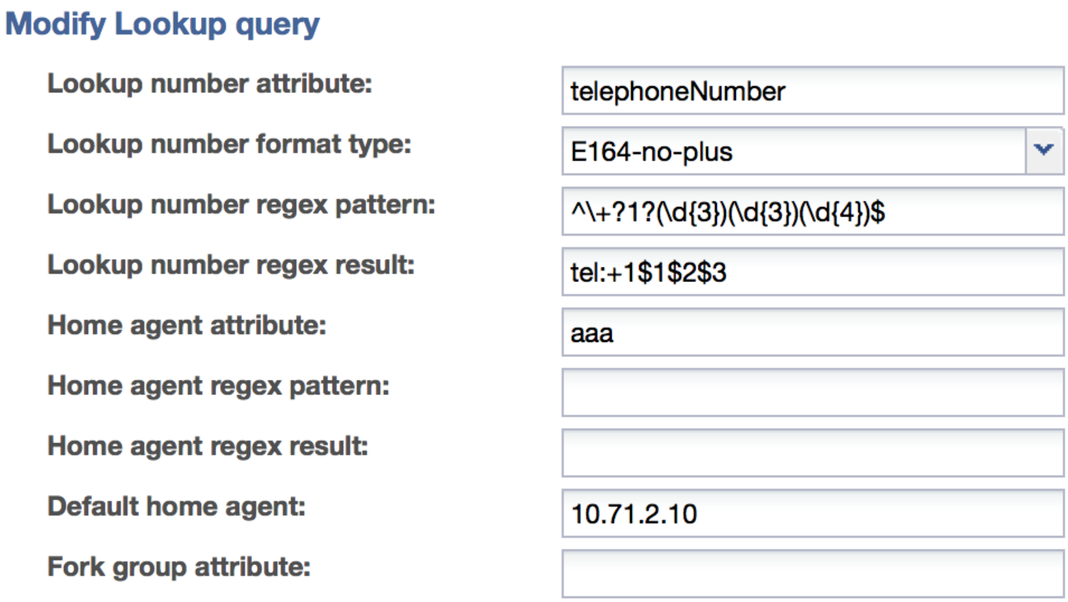


This is where attributes are referenced that determine the agent a user is assigned to. In the following example, msRTCSIP-Line is assigned to Lync 2013 and telephoneNumber is assigned to Cisco CUCM. When the ECB does an LDAP query, it will send these attributes. In the response, the server will return the attributes assigned to a particular user. Let's say the LDAP response returns both msRTCSIP-Line and telephoneNumber with value 15712935329, then the ECB knows to route the call to the same number on both Lync 2013 and CUCM. Whether it does this serially or in parallel depends on the SIP Interface "enable parallel forking" setting.

Add the following query and assign it to Lync 2013 (med3.lynclabsram.local in our test lab). The Lookup number format type should be regular-expression, and the Home agent attribute can be anything. The Lookup number regex pattern and result are default values.

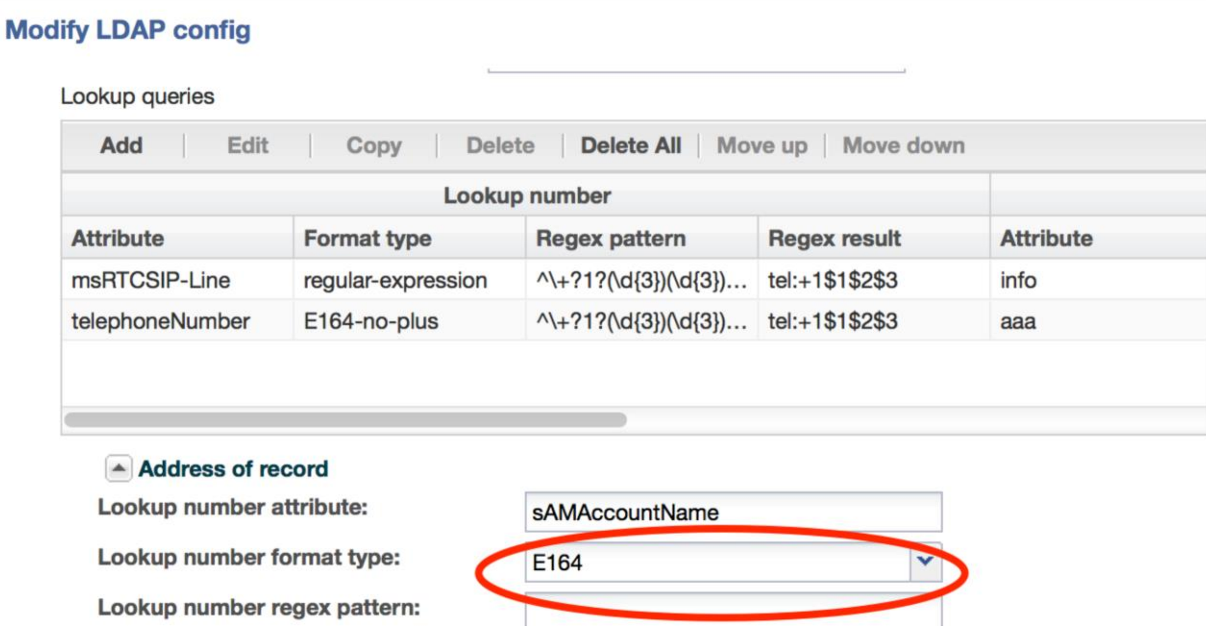


Add the following lookup query and assign it to Cisco CUCM:



Add other lookup queries as needed and determined by your Active Directory configuration.

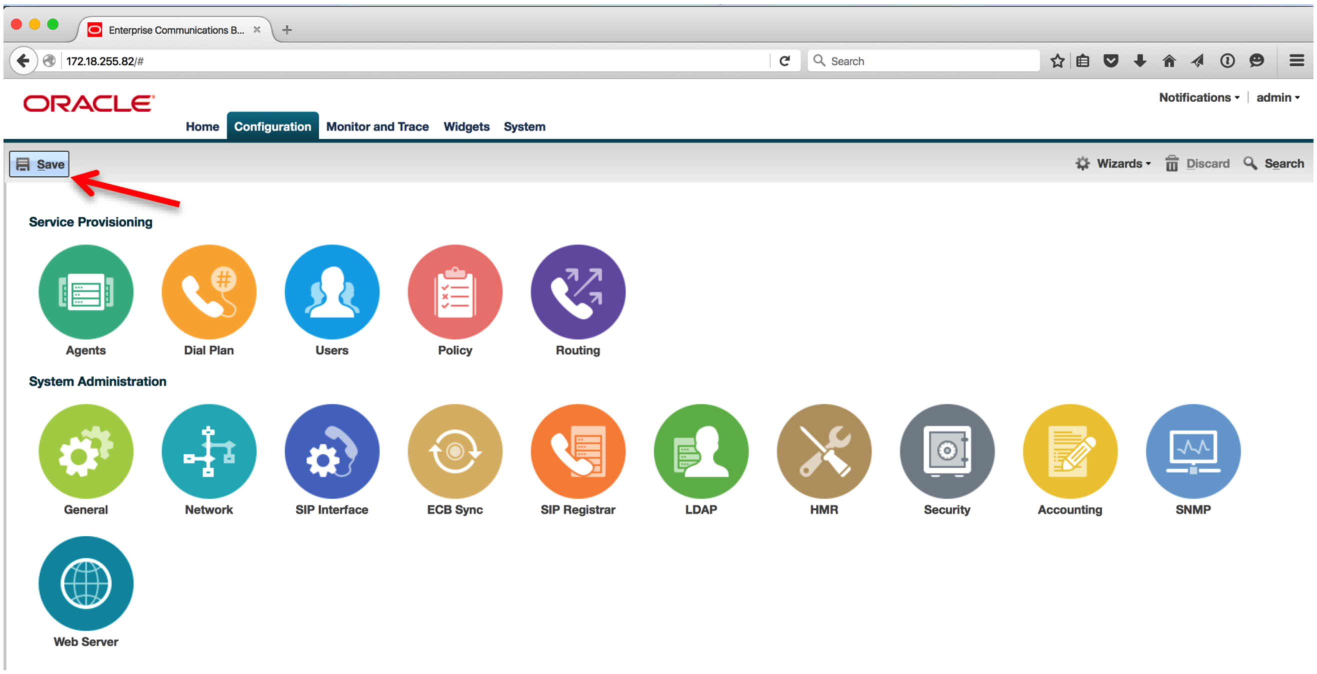
When finished adding lookup queries, set the Lookup number format type to E164. Leave the Lookup number attribute at its default value of sAMAccountName.



Click **OK** to finish the LDAP configuration.

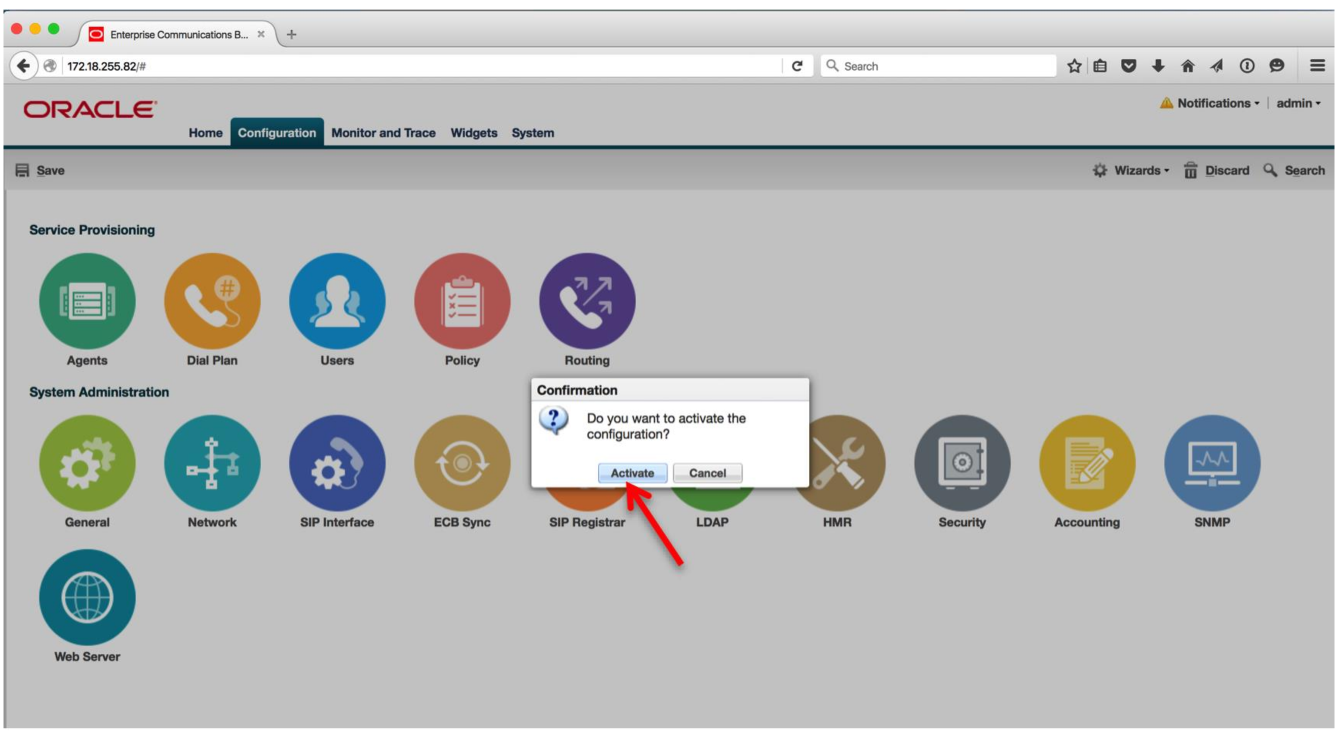
Save and activate the configuration

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

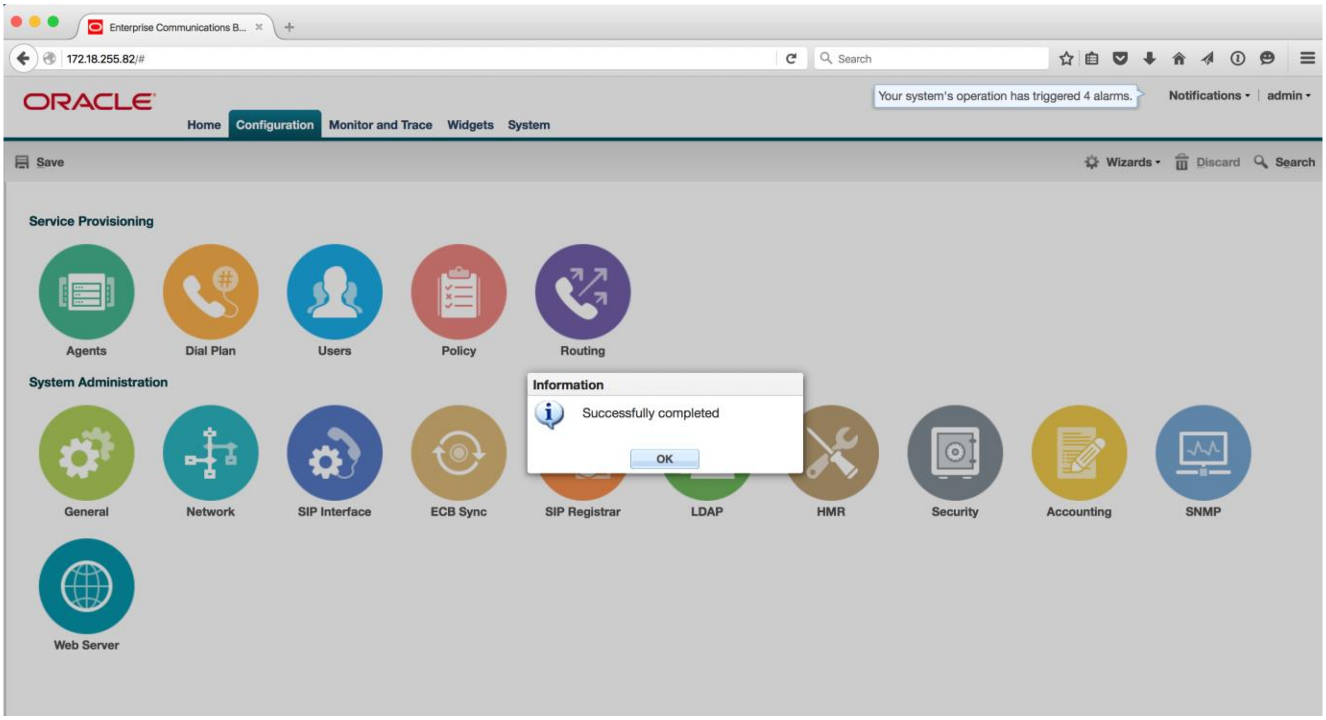


A progress dialog box will appear showing that the configuration in being saved.

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 Oracle Enterprise SBC

In this section we describe the steps for configuring an Oracle Enterprise SBC (E-SBC) for use with the Oracle ECB, Microsoft Lync & Skype for Business, Cisco CUCM, and Avaya Aura. The E-SBC will connect the Enterprise network to the Service Provider network in a SIP trunking scenario.

In Scope

The following guide for configuring the Oracle SBC assumes that this is a newly deployed device dedicated to a single customer. Please see the ACLI Configuration Guide on http://docs.oracle.com/cd/E61547_01/index.html for a better understanding of the Command Line Interface (CLI).

Note that Oracle offers several models of the SBC. This document covers the setup for the 1100, 3820, 4500, 4600, and 6300 platforms running OS ECZ7.3.0 MR-1 or later. If instructions are needed for other Oracle SBC models, please contact your Oracle representative.

Out of Scope

- Configuration of Network management including SNMP and RADIUS
- Configuration of Distributed Denial of Service (DDoS) protection parameters as these are based on individual customer requirements.

What will you need

- RJ45/DB9 serial adapter provided with the SBC, along with a straight-through Ethernet cable to go from the adapter to the SBC's console port (on the rear of the 1100, 4600, and 6300, and the front of the 3820 and 4500).
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the Oracle SBC
- IP address to be assigned to the management interface (eth0, labeled Mgmt0 on the SBC chassis) of the SBC - the eth0 management interface must be connected and configured to a management network separate from the service interfaces. Otherwise the SBC is subject to ARP overlap issues, loss of system access when the network is down, and compromised DDoS protection. Oracle does not support SBC configurations with management and media/service interfaces on the same subnet.
- IP address of the Oracle ECB.
- IP addresses to be used for the SBC internal and external facing ports (Service Interfaces)

SBC- Getting Started

Once the Oracle SBC is racked and the power cable connected, you are ready to set up physical network connectivity. **Note: use the console port on the front of the SBC, not the one on the back, on platforms such as the 3820 and 4500 that have two console ports.**

Plug the slot 0 port 0 (s0p0) interface into your outside (SIP Trunk-facing) network and the slot 1 port 0 (s1p0) interface into your inside (ECB-facing) network. Once connected, you are ready to power on and perform the following steps.

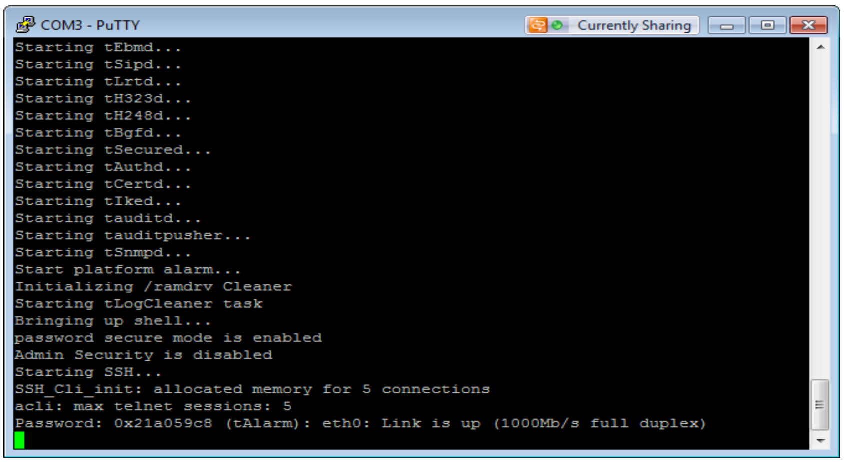
All commands are in bold, such as **`configure terminal`**; parameters in bold red such as **oraclesbc1** are parameters which are specific to an individual deployment. **Note:** The CLI is case sensitive.

Establish the serial connection and logging in the SBC

Confirm the SBC is powered off and connect one end of a straight-through Ethernet cable to the console port on the SBC and the other end to console adapter that ships with the SBC, connect the console adapter (a DB9 adapter) to the DB9 port on a workstation, running a terminal emulator application such as PuTTY. Start the terminal emulation application using the following settings:

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

Power on the SBC and confirm that you see the following output from the bootup sequence.



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

```
Password: acme
oraclesbc1> enable
Password: packet
oraclesbc1# configure terminal
oraclesbc1(configure)#
```

You are now in the global configuration mode.

Initial Configuration – Assigning the management Interface an IP address

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

oraclesbc1# 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. **Note these example boot parameters are specific to the 4600 platform. Other platforms will have different boot parameters. Use nnECZ730m1.64.bz for the 1100, 4500, 4600, and the 6300. Use nnECZ730m1.32.bz for the 3820.**

```
ORACLESBC1(configure)# bootparam

'.' = clear field;  '-' = go to previous field;  q
= quit
```

```
Boot File               : /boot/nnECZ730m1.64.bz
IP Address              : 192.168.79.44
VLAN                    :
Netmask                 : 255.255.255.224
Gateway                 : 192.168.79.33
IPv6 Address            :
IPv6 Gateway            :
Host IP                 : 0.0.0.0
FTP username            : vxftp
FTP password            : vxftp123
Flags                   :
Target Name             : oraclesbc1
Console Device          : COM1
Console Baudrate        : 115200
Other                   :

NOTE: These changed parameters will not go into
effect until reboot.
Also, be aware that some boot parameters may also
be changed through
PHY and Network Interface Configurations.
```

Configuring the SBC

The following section walks you through configuring the Oracle Enterprise SBC required to work with the Oracle Enterprise Communications Broker (ECB) in an environment with Microsoft Lync, Skype for Business, Cisco CUCM, and Avaya Aura.

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

High Availability

The Mgmt1 and Mgmt2 (labeled wancom1 and wancom2 in the configuration) ports which are on the rear panel of the SBC are used for the purpose of High Availability on the E-SBC. Crossover cables must be connected between these ports on the SBCs, i.e. Mgmt1 to Mgmt1 and Mgmt2 to Mgmt2. Please refer to the "High Availability Nodes" in the ACLI configuration guide for ECZ730 for more details.

Local Policies

Path: **configure terminal > session-router > local-policy**

```
local-policy
        from-address                            *
        to-address                              *
        source-realm                            SIP-Trunk
        description
        activate-time
        deactivate-time
        state                                   enabled
        policy-priority                         none
        policy-attribute
                next-hop                                10.64.3.124
                realm                                   towards-ecb
                action                                  none
                terminate-recursion                     disabled
                carrier
                start-time                              0000
                end-time                                2400
```

```
                days-of-week                            U-S
                cost                                    0
                state                                   enabled
                app-protocol                            SIP
                methods
                media-profiles
                lookup                                  single
                next-key
                eloc-str-lkup                           disabled
                eloc-str-match
local-policy
        from-address                            *
        to-address                              *
        source-realm                            towards-ecb
        description
        activate-time
        deactivate-time
        state                                   enabled
        policy-priority                         none
        policy-attribute
                next-hop                                192.168.147.48
                realm                                   SIP-Trunk
                action                                  none
                terminate-recursion                     disabled
                carrier
                start-time                              0000
                end-time                                2400
                days-of-week                            U-S
                cost                                    0
                state                                   enabled
                app-protocol                            SIP
                methods
                media-profiles
                lookup                                  single
                next-key
                eloc-str-lkup                           disabled
                eloc-str-match
```

Media Manager

Path: **configure terminal > media-manager > media-manager > select > 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
```

```
        options
        red-flow-port                           1985
        red-mgcp-port                           1986
        red-max-trans                           10000
        red-sync-start-time                     5000
        red-sync-comp-time                      1000
        media-policing                          enabled
        max-signaling-bandwidth                 10000000
        max-untrusted-signaling                 100
        min-untrusted-signaling                 30
        tolerance-window                        30
        trap-on-demote-to-deny                  disabled
        trap-on-demote-to-untrusted             disabled
        syslog-on-demote-to-deny                disabled
        syslog-on-demote-to-untrusted           disabled
        rtcp-rate-limit                         0
        anonymous-sdp                           disabled
        arp-msg-bandwidth                       32000
        rfc2833-timestamp                       disabled
        default-2833-duration                   100
        rfc2833-end-pkts-only-for-non-sig       enabled
        translate-non-rfc2833-event             disabled
        media-supervision-traps                 disabled
        dnsalg-server-failover                  disabled
        syslog-on-call-reject                   disabled
```

Network Interfaces

Path: **configure terminal > system > network-interface**

```
network-interface
        name                                    s0p0
        sub-port-id                             0
        description                             For SIP-Trunk
        hostname
        ip-address                              192.168.79.126 (virtual IP)
        pri-utility-addr                        192.168.79.127 (for HA only)
        sec-utility-addr                        192.168.79.128 (for HA only)
        netmask                                 255.255.255.224
        gateway                                 192.168.79.97
        sec-gateway
        gw-heartbeat
                state                                   disabled
                heartbeat                               0
                retry-count                             0
                retry-timeout                           1
                health-score                            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                             11
        signaling-mtu                           0
        hip-ip-list                             192.168.79.126 (add-hip-ip command)
        ftp-address
```

```
        icmp-address                        192.168.79.126 (add-icmp-ip command)
        snmp-address
        telnet-address
        ssh-address
network-interface
        name                                s1p0
        sub-port-id                         0
        description                         Facing Oracle ECB
        hostname
        ip-address                          10.64.3.122 (virtual IP)
        pri-utility-addr                    10.64.3.120 (for HA only)
        sec-utility-addr                    10.64.3.121 (for HA only)
        netmask                             255.255.0.0
        gateway                             10.64.1.1
        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
        signaling-mtu                       0
        hip-ip-list                         10.64.3.122 (add-hip-ip command)
        ftp-address
        icmp-address                        10.64.3.122 (add-icmp-ip command)
        snmp-address
        telnet-address
        ssh-address                         10.64.3.122 (add-ssh-ip command)
network-interface
        name                                wancom1
        sub-port-id                         0
        description
        hostname
        ip-address
        pri-utility-addr                    169.254.1.1
        sec-utility-addr                    169.254.1.2
        netmask                             255.255.255.252
        gateway
        sec-gateway
        gw-heartbeat
                state                               disabled
                heartbeat                           0
                retry-count                         0
                retry-timeout                       1
                health-score                        0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                         11
        signaling-mtu                       0
```

```
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
network-interface
        name                                    wancom2
        sub-port-id                             0
        description
        hostname
        ip-address
        pri-utility-addr                        169.254.2.1
        sec-utility-addr                        169.254.2.2
        netmask                                 255.255.255.252
        gateway
        sec-gateway
        gw-heartbeat
                state                                   disabled
                heartbeat                               0
                retry-count                             0
                retry-timeout                           1
                health-score                            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                             11
        signaling-mtu                           0
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
```

Physical Interfaces

Path: **configure terminal > system > phy-interface**

```
phy-interface
        name                                    s0p0
        operation-type                          Media
        port                                    0
        slot                                    0
        virtual-mac                             00:08:25:04:0d:1e <- determine by
issuing the "show prom-info main" command from the # prompt, noting the starting
MAC address, and replacing the last character with "e".  For HA only.
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode                             FULL
        speed                                   100
        wancom-health-score                     50
        overload-protection                     disabled
phy-interface
        name                                    s1p0
        operation-type                          Media
        port                                    0
        slot                                    1
        virtual-mac                             00:08:25:04:0d:1f <- determine by
issuing the "show prom-info main" command from the # prompt, noting the starting
MAC address, and replacing the last character with "f".  For HA only.
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode                             FULL
        speed                                   100
        wancom-health-score                     50
        overload-protection                     disabled
phy-interface
        name                                    wancom1
        operation-type                          Control
        port                                    1
        slot                                    0
        virtual-mac
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode
        speed
        wancom-health-score                     8
        overload-protection                     disabled
phy-interface
        name                                    wancom2
        operation-type                          Control
        port                                    2
        slot                                    0
        virtual-mac
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode
        speed
        wancom-health-score                     9
        overload-protection                     disabled
```

Realm Configs

Path: **configure terminal > media-manager > realm-config**

```
realm-config
        identifier                              SIP-Trunk
        description
        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
        max-bandwidth                           0
        fallback-bandwidth                      0
        max-priority-bandwidth                  0
        max-latency                             0
        max-jitter                              0
        max-packet-loss                         0
        observ-window-size                      0
        parent-realm
        dns-realm
        media-policy
        media-sec-policy
        srtp-msm-passthrough                    disabled
        class-profile
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        average-rate-limit                      0
        access-control-trust-level              none
        invalid-signal-threshold                0
        maximum-signal-threshold                0
        untrusted-signal-threshold              0
        nat-trust-threshold                     0
        max-endpoints-per-nat                   0
        nat-invalid-message-threshold           0
        wait-time-for-invalid-register          0
        deny-period                             30
        cac-failure-threshold                   0
        untrust-cac-failure-threshold           0
        ext-policy-svr
        diam-e2-address-realm
        subscription-id-type                    END_USER_NONE
        symmetric-latching                      disabled
        pai-strip                               disabled
        trunk-context
        device-id
        early-media-allow
        enforcement-profile
```

```
        additional-prefixes
        restricted-latching                    none
        restriction-mask                       32
        user-cac-mode                          none
        user-cac-bandwidth                     0
        user-cac-sessions                      0
        icmp-detect-multiplier                 0
        icmp-advertisement-interval            0
        icmp-target-ip
        monthly-minutes                        0
        options
        spl-options
        accounting-enable                      enabled
        net-management-control                 disabled
        delay-media-update                     disabled
        refer-call-transfer                    disabled
        hold-refer-reinvite                    disabled
        refer-notify-provisional               none
        dyn-refer-term                         disabled
        codec-policy                           to-trunk
        codec-manip-in-realm                   disabled
        codec-manip-in-network                 enabled
        rtcp-policy
        constraint-name
        session-recording-server
        session-recording-required             disabled
        manipulation-string
        manipulation-pattern
        stun-enable                            disabled
        stun-server-ip                         0.0.0.0
        stun-server-port                       3478
        stun-changed-ip                        0.0.0.0
        stun-changed-port                      3479
        sip-profile
        sip-isup-profile
        match-media-profiles
        qos-constraint
        block-rtcp                             disabled
        hide-egress-media-update               disabled
        tcp-media-profile
        monitoring-filters
        node-functionality
        default-location-string
        alt-family-realm
        pref-addr-type                         none
realm-config
        identifier                             towards-ecb
        description
        addr-prefix                            0.0.0.0
        network-interfaces                     s1p0:0
        mm-in-realm                            enabled
        mm-in-network                          enabled
        mm-same-ip                             enabled
        mm-in-system                           enabled
        bw-cac-non-mm                          disabled
        msm-release                            disabled
```

```
qos-enable                              disabled
max-bandwidth                           0
fallback-bandwidth                      0
max-priority-bandwidth                  0
max-latency                             0
max-jitter                              0
max-packet-loss                         0
observ-window-size                      0
parent-realm
dns-realm
media-policy
media-sec-policy
srtp-msm-passthrough                    disabled
class-profile
in-translationid
out-translationid
in-manipulationid
out-manipulationid
average-rate-limit                      0
access-control-trust-level              none
invalid-signal-threshold                0
maximum-signal-threshold                0
untrusted-signal-threshold              0
nat-trust-threshold                     0
max-endpoints-per-nat                   0
nat-invalid-message-threshold           0
wait-time-for-invalid-register          0
deny-period                             30
cac-failure-threshold                   0
untrust-cac-failure-threshold           0
ext-policy-svr
diam-e2-address-realm
subscription-id-type                    END_USER_NONE
symmetric-latching                      disabled
pai-strip                               disabled
trunk-context
device-id
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching                     none
restriction-mask                        32
user-cac-mode                           none
user-cac-bandwidth                      0
user-cac-sessions                       0
icmp-detect-multiplier                  0
icmp-advertisement-interval             0
icmp-target-ip
monthly-minutes                         0
options
spl-options
accounting-enable                       enabled
net-management-control                  disabled
delay-media-update                      disabled
refer-call-transfer                     enabled
hold-refer-reinvite                     disabled
```

```
        refer-notify-provisional               none
        dyn-refer-term                         disabled
        codec-policy
        codec-manip-in-realm                   disabled
        codec-manip-in-network                 enabled
        rtcp-policy
        constraint-name
        session-recording-server
        session-recording-required             disabled
        manipulation-string
        manipulation-pattern
        stun-enable                            disabled
        stun-server-ip                         0.0.0.0
        stun-server-port                       3478
        stun-changed-ip                        0.0.0.0
        stun-changed-port                      3479
        sip-profile
        sip-isup-profile
        match-media-profiles
        qos-constraint
        block-rtcp                             disabled
        hide-egress-media-update               disabled
        tcp-media-profile
        monitoring-filters
        node-functionality
        default-location-string
        alt-family-realm
        pref-addr-type                         none
```

Redundancy Config (HA Pairs Only)

Path: **configure terminal > system > redundancy > select**

```
redundancy-config
        state                                   enabled
        log-level                               INFO
        health-threshold                        75
        emergency-threshold                     50
        port                                    9090
        advertisement-time                      500
        percent-drift                           210
        initial-time                            1250
        becoming-standby-time                   180000
        becoming-active-time                    100
        cfg-port                                1987
        cfg-max-trans                           10000
        cfg-sync-start-time                     5000
        cfg-sync-comp-time                      1000
        gateway-heartbeat-interval              0
        gateway-heartbeat-retry                 0
        gateway-heartbeat-timeout               1
        gateway-heartbeat-health                0
        media-if-peercheck-time                 0
        peer
                name                                    oraclesbc1 <- must match
Primary SBC's target name in boot parameters
                state                                   enabled
                type                                    Primary
                destination
                        address                                 169.254.1.1:9090
                        network-interface                       wancom1:0
                destination
                        address                                 169.254.2.1:9090
                        network-interface                       wancom2:0
        peer
                name                                    oraclesbc2 <- must match
Secondary SBC's target name in boot parameters
                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
```

Session Agents

Path: **configure terminal > session-router > session-agent**

```
session-agent
        hostname                                10.64.3.124
        ip-address                              10.64.3.124
        port                                    5060
        state                                   enabled
        app-protocol                            SIP
        app-type
        transport-method                        StaticTCP
        realm-id                                towards-ecb
        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
        load-balance-dns-query                  hunt
        options
        spl-options
        media-profiles
        in-translationid
        out-translationid
        trust-me                                disabled
        request-uri-headers
        stop-recurse
        local-response-map
```

```
        ping-to-user-part
        ping-from-user-part
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate               0
        early-media-allow
        invalidate-registrations                disabled
        rfc2833-mode                            none
        rfc2833-payload                         0
        codec-policy
        enforcement-profile
        refer-call-transfer                     disabled
        refer-notify-provisional                none
        reuse-connections                       NONE
        tcp-keepalive                           none
        tcp-reconn-interval                     0
        max-register-burst-rate                 0
        register-burst-window                   0
        sip-profile
        sip-isup-profile
        kpml-interworking                       inherit
        monitoring-filters
        session-recording-server
        session-recording-required              disabled
        hold-refer-reinvite                     disabled
        send-tcp-fin                            disabled
session-agent
        hostname                                192.168.147.48
        ip-address                              192.168.147.48
        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                             OPTIONS
ping-interval                           30
ping-send-mode                          keep-alive
ping-all-addresses                      disabled
ping-in-service-response-codes
out-service-response-codes
load-balance-dns-query                  hunt
options
spl-options
media-profiles
in-translationid
out-translationid                       stripplus1
trust-me                                disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate               0
early-media-allow
invalidate-registrations                disabled
rfc2833-mode                            none
rfc2833-payload                         0
codec-policy
enforcement-profile
refer-call-transfer                     disabled
refer-notify-provisional                none
reuse-connections                       NONE
tcp-keepalive                           none
tcp-reconn-interval                     0
max-register-burst-rate                 0
register-burst-window                   0
sip-profile
sip-isup-profile
kpml-interworking                       inherit
monitoring-filters
session-recording-server
session-recording-required              disabled
hold-refer-reinvite                     disabled
send-tcp-fin                            disabled
```

Session Translation

Path: **configure terminal > session-router > session-translation**

```
session-translation
        id                                      stripplus1
        rules-calling                           stripplus1
        rules-called                            stripplus1
```

SIP Config

Path: **configure terminal > session-router > sip-config > select**

```
sip-config
        state                                   enabled
        operation-mode                          dialog
        dialog-transparency                     enabled
        home-realm-id                           towards-ecb
        egress-realm-id
        auto-realm-id
        nat-mode                                None
        registrar-domain                        *
        registrar-host                          *
        registrar-port                          0
        register-service-route                  always
        init-timer                              500
        max-timer                               4000
        trans-expire                            32
        initial-inv-trans-expire                0
        invite-expire                           180
        inactive-dynamic-conn                   32
        enforcement-profile
        pac-method
        pac-interval                            10
        pac-strategy                            PropDist
        pac-load-weight                         1
        pac-session-weight                      1
        pac-route-weight                        1
        pac-callid-lifetime                     600
        pac-user-lifetime                       3600
        red-sip-port                            1988
        red-max-trans                           10000
        red-sync-start-time                     5000
        red-sync-comp-time                      1000
        options                                 max-udp-length=0
        add-reason-header                       disabled
        sip-message-len                         4096
        enum-sag-match                          disabled
        extra-method-stats                      disabled
        extra-enum-stats                        disabled
        rph-feature                             disabled
        nsep-user-sessions-rate                 0
        nsep-sa-sessions-rate                   0
        registration-cache-limit                0
        register-use-to-for-lp                  disabled
        refer-src-routing                       disabled
```

```
        add-ucid-header                         disabled
        proxy-sub-events
        allow-pani-for-trusted-only             disabled
        atcf-stn-sr
        atcf-psi-dn
        atcf-route-to-sccas                     disabled
        eatf-stn-sr
        pass-gruu-contact                       disabled
        sag-lookup-on-redirect                  disabled
        set-disconnect-time-on-bye              disabled
        msrp-delayed-bye-timer                  15
        transcoding-realm
        transcoding-agents
        create-dynamic-sa                       disabled
        node-functionality                      P-CSCF
        match-sip-instance                      disabled
        sa-routes-stats                         disabled
        sa-routes-traps                         disabled
        rx-sip-reason-mapping                   disabled
        add-ue-location-in-pani                 disabled
        hold-emergency-calls-for-loc-info       0
```

SIP Feature

Path: **configure terminal > session-router > sip-feature**

```
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
```

SIP Interfaces

Path: **configure terminal > session-router > sip-interface**

```
sip-interface
        state                                   enabled
        realm-id                                SIP-Trunk
        description
        sip-port
                address                                 192.168.79.126
                port                                    5060
                transport-protocol                      UDP
                tls-profile
                allow-anonymous                         all
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                            0
```

```
        initial-inv-trans-expire                0
        invite-expire                           0
        max-redirect-contacts                   0
        proxy-mode
        redirect-action
        contact-mode                            none
        nat-traversal                           none
        nat-interval                            30
        tcp-nat-interval                        90
        registration-caching                    disabled
        min-reg-expire                          300
        registration-interval                   3600
        route-to-registrar                      disabled
        secured-network                         disabled
        teluri-scheme                           disabled
        uri-fqdn-domain
        options
        spl-options
        trust-mode                              all
        max-nat-interval                        3600
        nat-int-increment                       10
        nat-test-increment                      30
        sip-dynamic-hnt                         disabled
        stop-recurse                            401,407
        port-map-start                          0
        port-map-end                            0
        in-manipulationid
        out-manipulationid                      NAT_IP
        sip-ims-feature                         disabled
        sip-atcf-feature                        disabled
        subscribe-reg-event                     disabled
        operator-identifier
        anonymous-priority                      none
        max-incoming-conns                      0
        per-src-ip-max-incoming-conns           0
        inactive-conn-timeout                   0
        untrusted-conn-timeout                  0
        network-id
        ext-policy-server
        ldap-policy-server
        default-location-string
        term-tgrp-mode                          none
        charging-vector-mode                    pass
        charging-function-address-mode          pass
        ccf-address
        ecf-address
        implicit-service-route                  disabled
        rfc2833-payload                         101
        rfc2833-mode                            transparent
        constraint-name
        response-map
        local-response-map
        sec-agree-feature                       disabled
        sec-agree-pref                          ipsec3gpp
        enforcement-profile
        route-unauthorized-calls
```

```
        tcp-keepalive                           none
        add-sdp-invite                          disabled
        p-early-media-header                    disabled
        p-early-media-direction
        add-sdp-profiles
        manipulation-string
        manipulation-pattern
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                          0
        tunnel-name
        register-keep-alive                     none
        kpml-interworking                       disabled
        msrp-delay-egress-bye                   disabled
        send-380-response
        pcscf-restoration
        session-timer-profile
        session-recording-server
        session-recording-required              disabled
        service-tag
        reg-cache-route                         disabled
sip-interface
        state                                   enabled
        realm-id                                towards-ecb
        description
        sip-port
                address                                 10.64.3.122
                port                                    5060
                transport-protocol                      TCP
                tls-profile
                allow-anonymous                         all
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                            0
        initial-inv-trans-expire                0
        invite-expire                           0
        max-redirect-contacts                   0
        proxy-mode
        redirect-action
        contact-mode                            none
        nat-traversal                           none
        nat-interval                            30
        tcp-nat-interval                        90
        registration-caching                    disabled
        min-reg-expire                          300
        registration-interval                   3600
        route-to-registrar                      disabled
        secured-network                         disabled
        teluri-scheme                           disabled
        uri-fqdn-domain
        options                                 100rel-interworking
        spl-options
        trust-mode                              all
        max-nat-interval                        3600
        nat-int-increment                       10
```

```
        nat-test-increment                        30
        sip-dynamic-hnt                           disabled
        stop-recurse                              401,407
        port-map-start                            0
        port-map-end                              0
        in-manipulationid
        out-manipulationid                        NAT_IP
        sip-ims-feature                           disabled
        sip-atcf-feature                          disabled
        subscribe-reg-event                       disabled
        operator-identifier
        anonymous-priority                        none
        max-incoming-conns                        0
        per-src-ip-max-incoming-conns             0
        inactive-conn-timeout                     0
        untrusted-conn-timeout                    0
        network-id
        ext-policy-server
        ldap-policy-server
        default-location-string
        term-tgrp-mode                            none
        charging-vector-mode                      pass
        charging-function-address-mode            pass
        ccf-address
        ecf-address
        implicit-service-route                    disabled
        rfc2833-payload                           101
        rfc2833-mode                              transparent
        constraint-name
        response-map
        local-response-map
        sec-agree-feature                         disabled
        sec-agree-pref                            ipsec3gpp
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                             none
        add-sdp-invite                            disabled
        p-early-media-header                      disabled
        p-early-media-direction
        add-sdp-profiles
        manipulation-string
        manipulation-pattern
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                            0
        tunnel-name
        register-keep-alive                       none
        kpml-interworking                         disabled
        msrp-delay-egress-bye                     disabled
        send-380-response
        pcscf-restoration
        session-timer-profile
        session-recording-server
        session-recording-required                disabled
        service-tag
        reg-cache-route                           disabled
```

SIP Manipulations (Header Manipulation Rules – HMR)

Path: **configure terminal > session-router > sip-manipulation**

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

SIP Monitoring

Path: **configure terminal > session-router > sip-monitoring > select**

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

Steering Pools

Path: **configure terminal > media-manager > steering-pool**

```
steering-pool
        ip-address                             10.64.3.122
        start-port                             49152
        end-port                               65535
        realm-id                               towards-ecb
        network-interface
steering-pool
        ip-address                             192.168.79.126
        start-port                             49152
        end-port                               65535
        realm-id                               SIP-Trunk
        network-interface
```

System Config

Path: **configure terminal > system > system-config > select**

```
system-config
        hostname                               ORACLESBC
        description                            4600 for ECB Testing
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled                           enabled
        enable-snmp-auth-traps                 disabled
        enable-snmp-syslog-notify              disabled
        enable-snmp-monitor-traps              disabled
        enable-env-monitor-traps               disabled
        enable-mblk_tracking                   disabled
        snmp-syslog-his-table-length           1
        snmp-syslog-level                      WARNING
        system-log-level                       WARNING
        process-log-level                      DEBUG (change to NOTICE after
testing is complete)
        process-log-ip-address                 0.0.0.0
        process-log-port                       0
        collect
                sample-interval                        5
                push-interval                          15
```

```
            boot-state                                  disabled
            start-time                                  now
            end-time                                    never
            red-collect-state                           disabled
            red-max-trans                               1000
            red-sync-start-time                         5000
            red-sync-comp-time                          1000
            push-success-trap-state                     disabled
    comm-monitor
            state                                       disabled
            sbc-grp-id                                  0
            tls-profile
            qos-enable                                  enabled
            interim-qos-update                          disabled
    call-trace                                  disabled
    internal-trace                              disabled
    log-filter                                  all
    default-gateway                             192.168.79.33
    restart                                     enabled
    exceptions
    telnet-timeout                              0
    console-timeout                             0
    remote-control                              enabled
    cli-audit-trail                             enabled
    link-redundancy-state                       disabled
    source-routing                              enabled
    cli-more                                    disabled
    terminal-height                             24
    debug-timeout                               0
    trap-event-lifetime                         0
    ids-syslog-facility                         -1
    options
    default-v6-gateway                          ::
    ipv6-signaling-mtu                          1500
    ipv4-signaling-mtu                          1500
    cleanup-time-of-day                         00:00
    snmp-engine-id-suffix
    snmp-agent-mode                             v1v2
```

Translation Rules

Path: **configure terminal > session-router > translation-rule**

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

Web Server Config

Path: **configure terminal > system > web-server-config > select**

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

Save, Activate, and Reboot

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. Some config elements are not Real-Time Configuration (RTC) supported, so a reboot is required after the initial configuration.

```
oraclesbc1# 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'.
oraclesbc1# 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
oraclesbc1# reboot force
```

The E-SBC configuration is now complete.

Phase 3 – Configuring Active Directory for LDAP Integration with the ECB

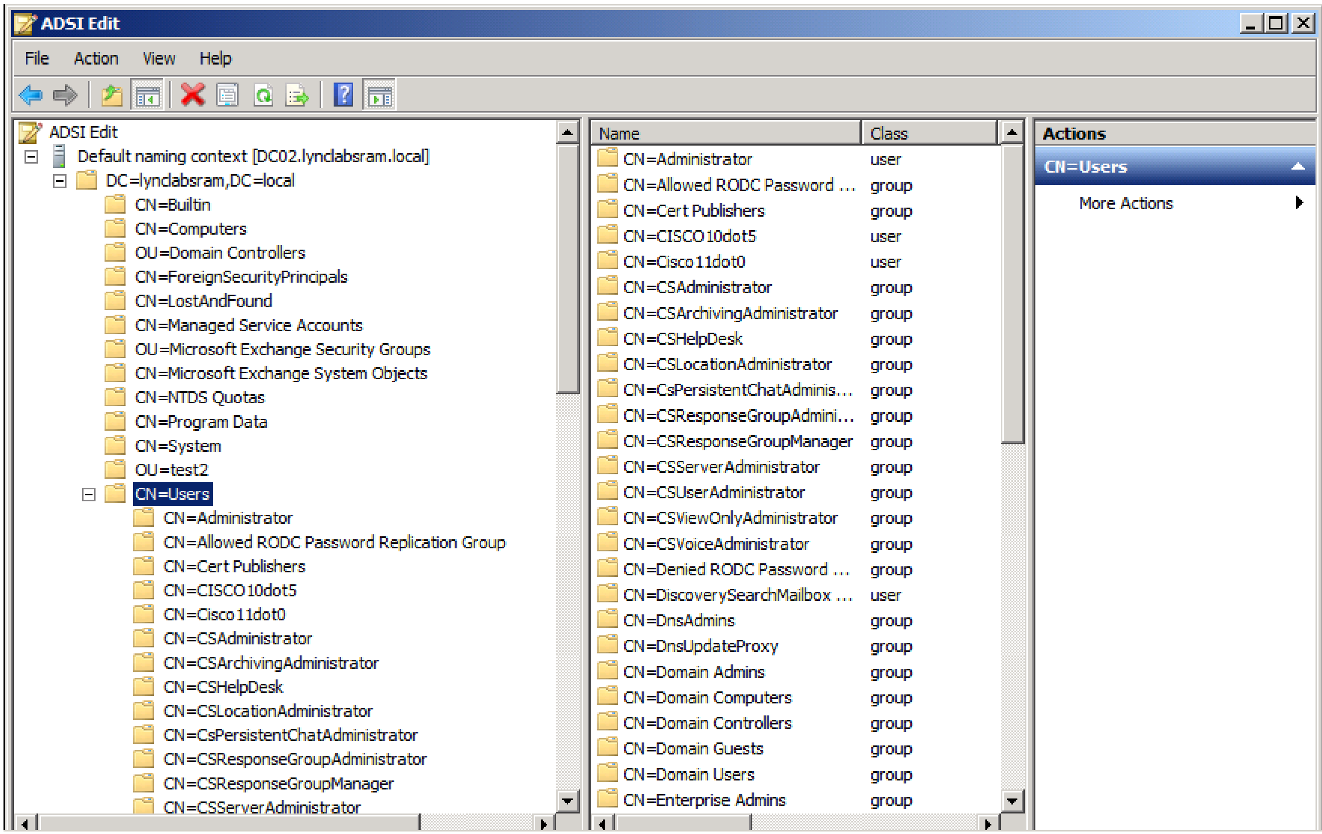
In this section we describe the steps for configuring Active Directory (AD) for LDAP integration with the ECB. **This step is optional.** It allows the ECB to receive routing information from AD for users on a per-call basis and can be used for parallel or serial call forking, such as having a user with the same number ring on both Skype for Business and Cisco CUCM. If a user is already defined on the Lync or Skype for Business server, then the user will already exist with the "msRTCSIP-Line" parameter in AD.

If the ECB and AD are configured for LDAP integration, then it is NOT necessary to define users in the User database on the ECB.

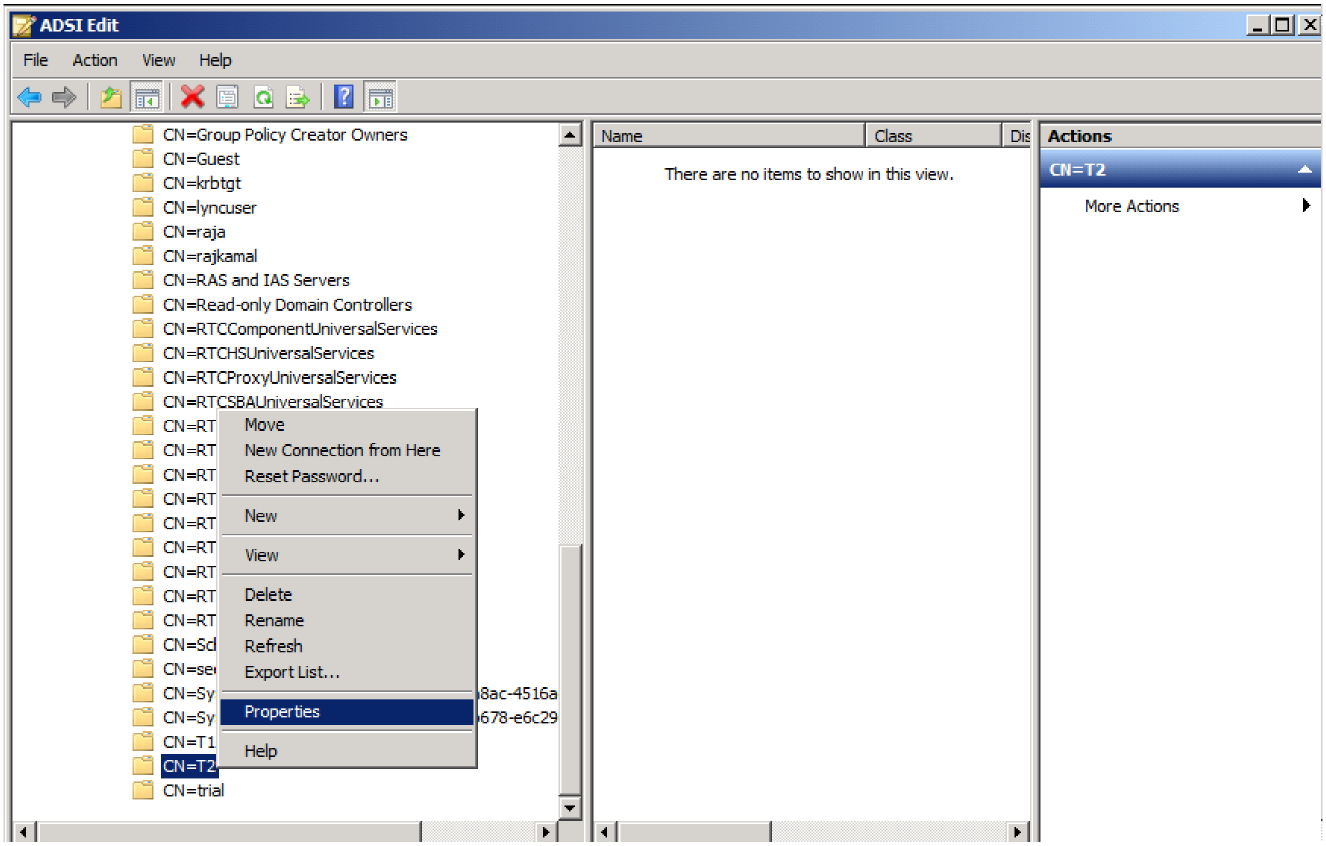
Adding a User's Phone Number(s) to Active Directory

1. On the Active Directory (AD) server, click the **Start** menu, then click on **ADSI Edit**.

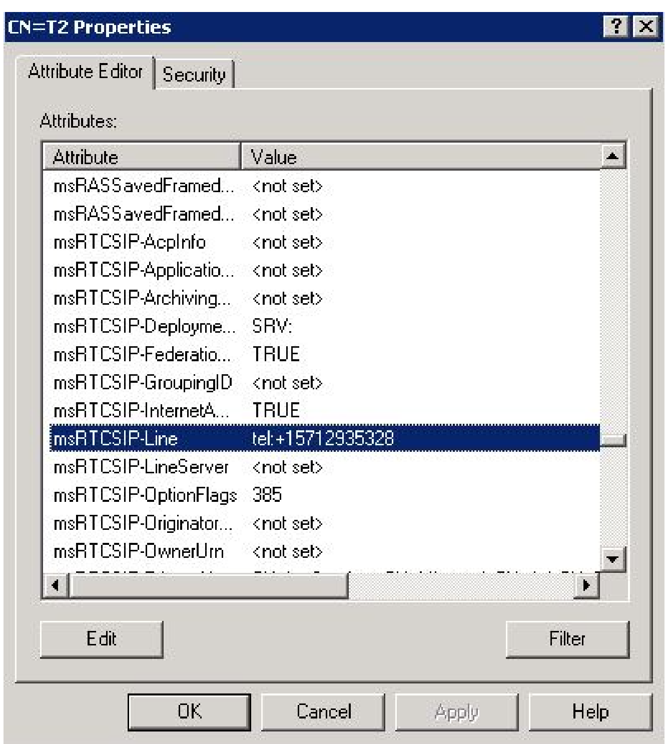
2. Expand the **Default running context** (DC02.lynclabsram.local in our example), expand **DC=lynclabsram,DC=local**, then expand **CN=Users**.



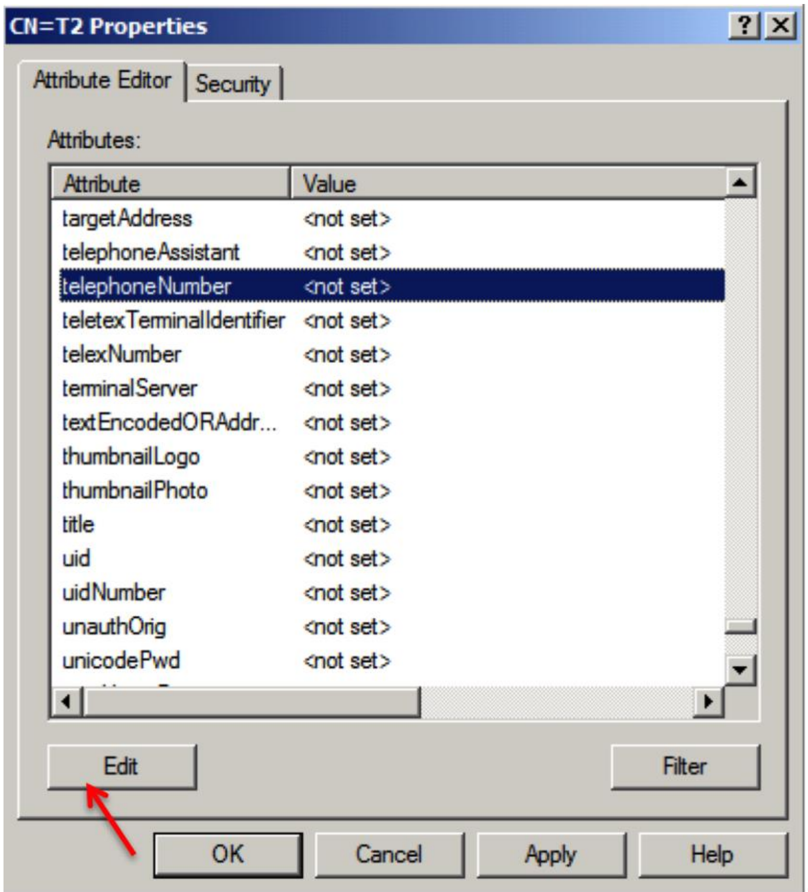
3. Select the user to be modified (T2 in the this example). Right click on the user and select **Properties**.



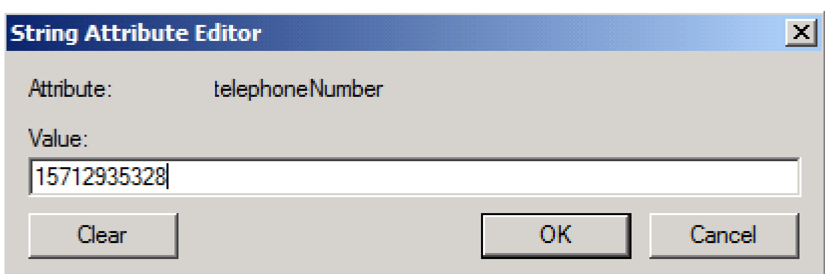
4. If a user is already defined on the Lync or Skype for Business server, then the user will already exist with the "msRTCSIP-Line" parameter in AD. Scroll down to msRTCSIP-Line and verify the telephone number is the correct value.



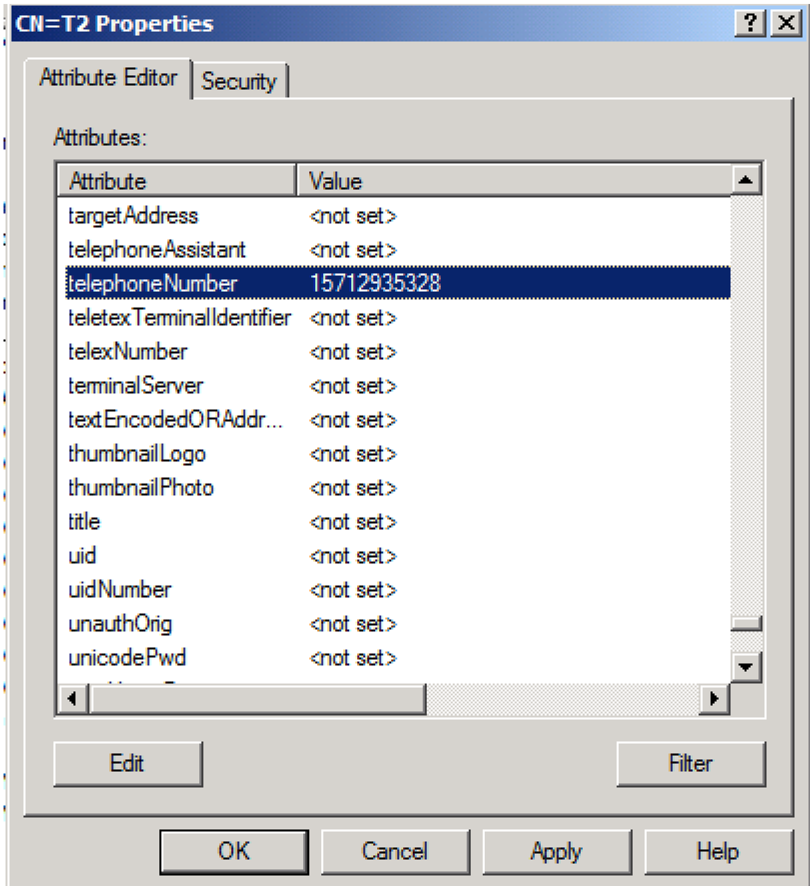
5. Scroll down to telephoneNumber and click Edit.



6. Enter the user’s telephone number and click OK.



7. Click OK. You can also scroll to other attributes to define the user's telephone number on other systems, such as Avaya Aura. **Ensure that these attributes match those defined on the ECB under LDAP integration.** In the examples in this document, msRTCSIP-Line represents a Lync 2013 user, and telephoneNumber represents a CUCM user.



The Active Directory configuration is now complete.

Phase 4 – 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.

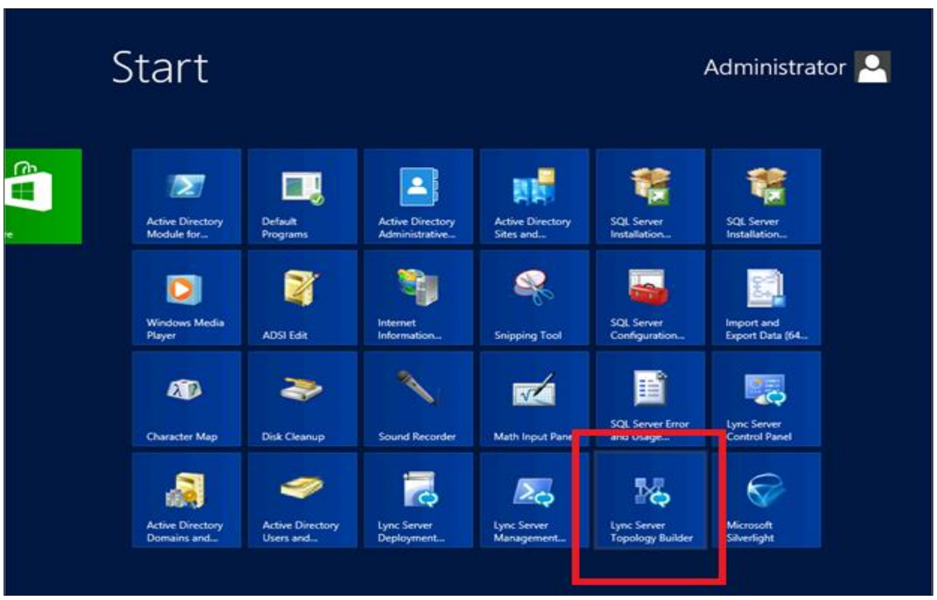
To add the PSTN gateway, we will need:

- 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

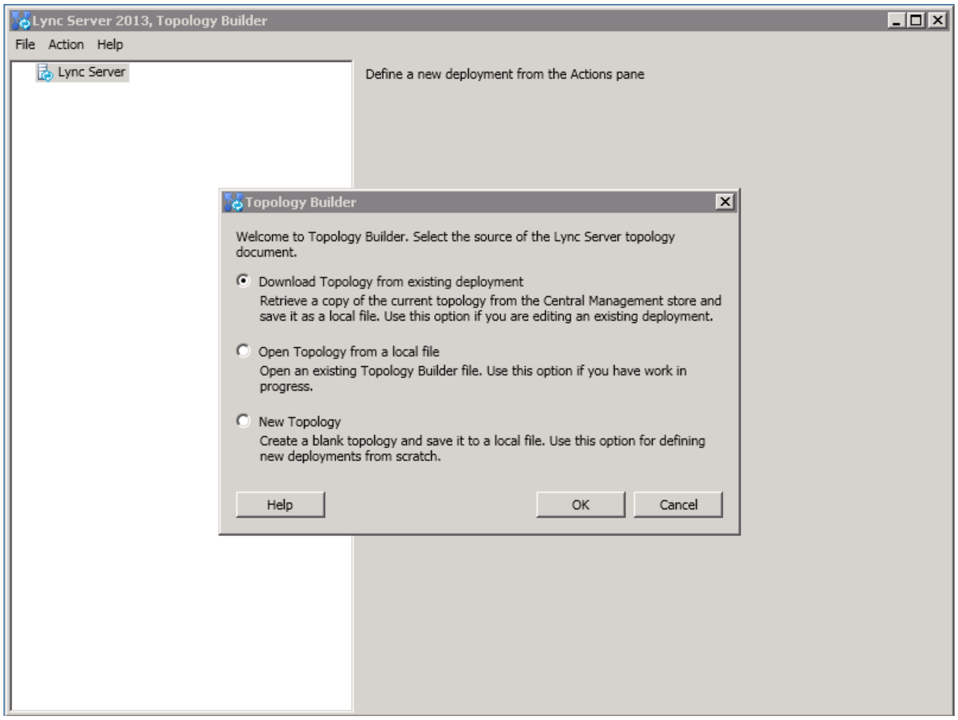
Adding the ECB as a PSTN gateway

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

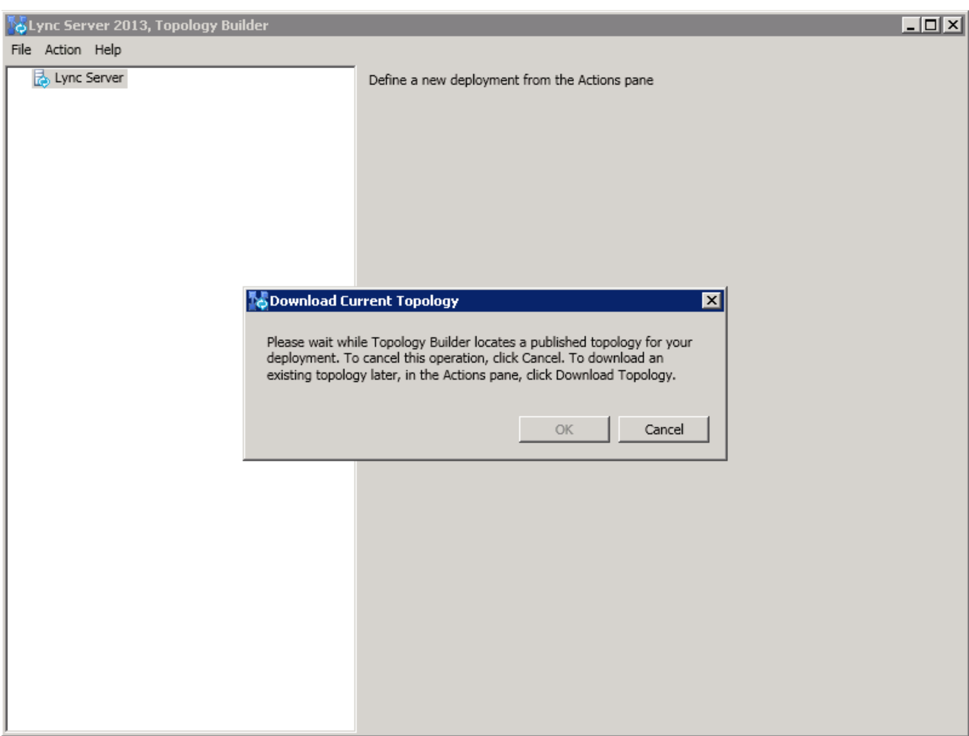
1. On the server where the Topology Builder is located start the console.
2. From the Start bar, select Lync Server Topology Builder.



3. The Topology Builder window will now be displayed. Select **Download Topology from existing deployment**.

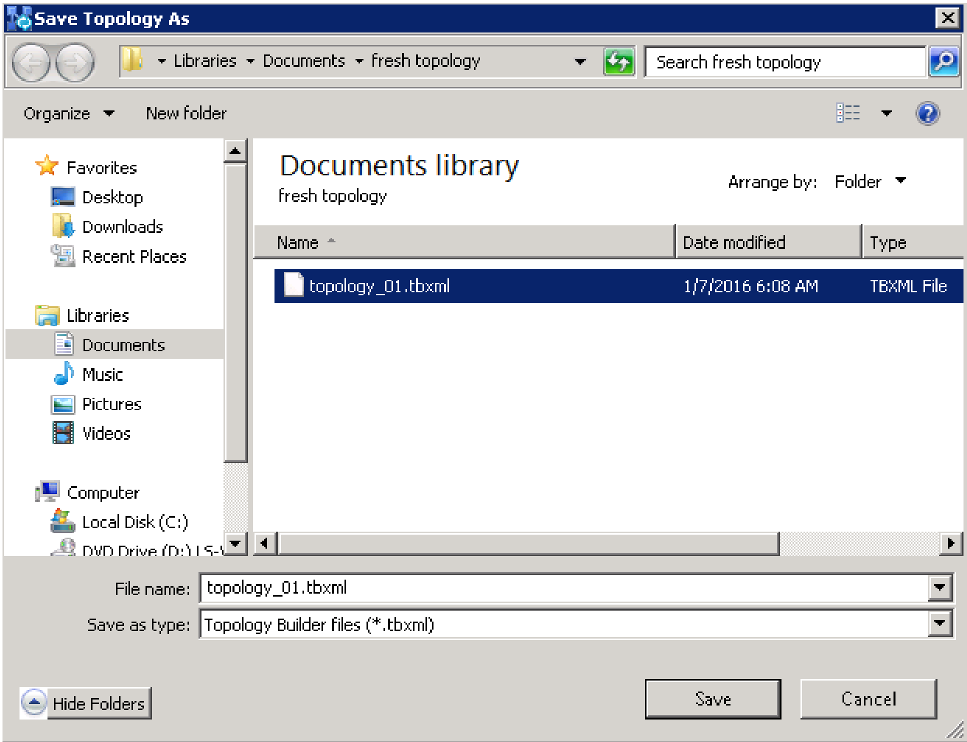


4. You will then see a screen showing that the current toplogy is being downloaded. Click the **Ok** button.



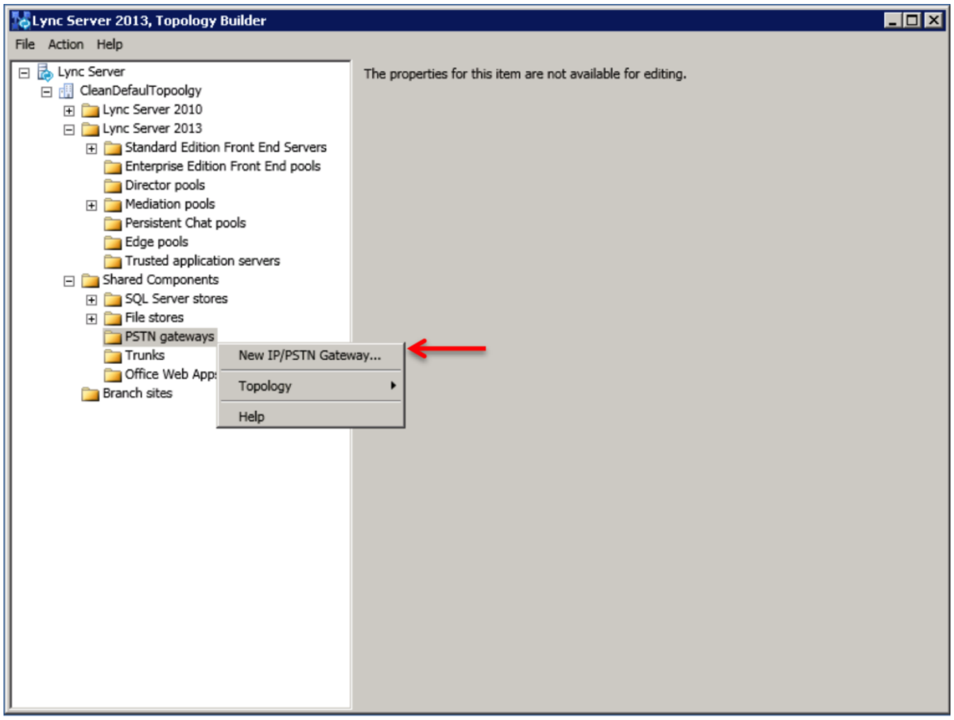
5. Next you will be prompted to save the topology which you have imported. You should revision the name or number of the topology according to the standards used within the enterprise. Click the **Save** button

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

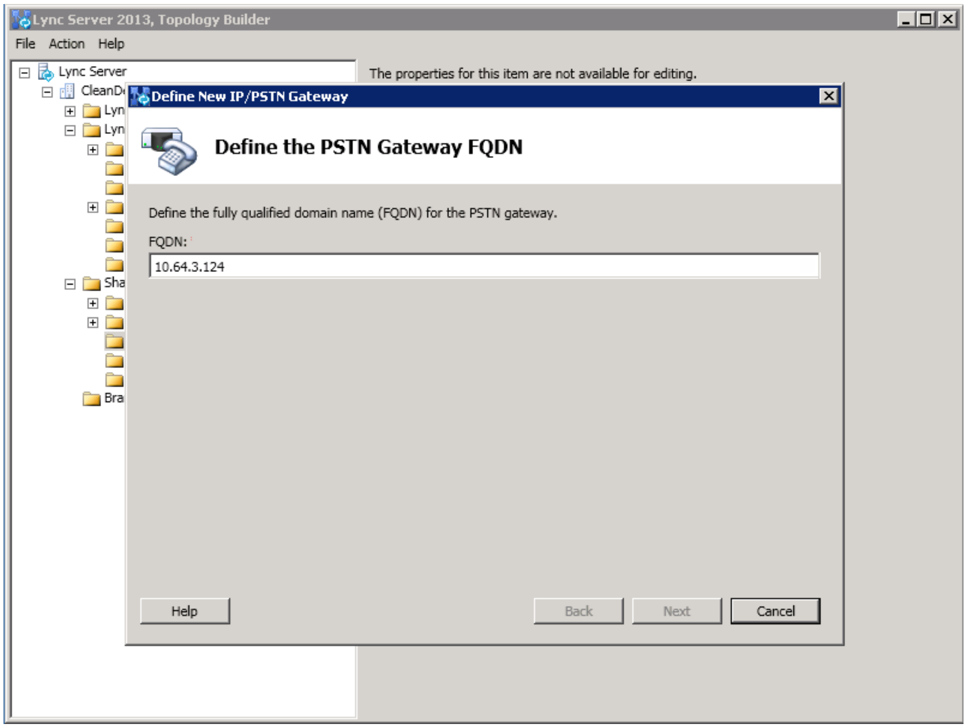


6. You will now see the topology builder screen with the enterprise's topology imported.

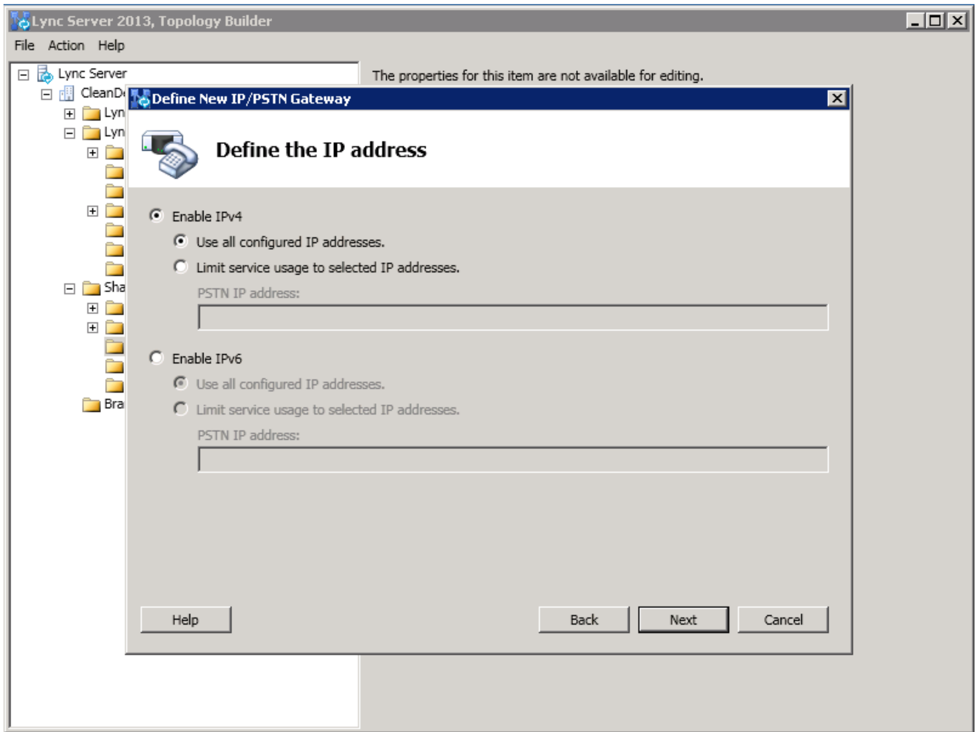
7. In the upper left hand corner, expand the site in which the PSTN gateway will be added. In our case, the site is labeled **CleanDefaultTopology**. Expand **Shared Components.** Then click on the **PSTN Gateways.** Right click on **PSTN gateways** and select **New IP/PSTN Gateway**.



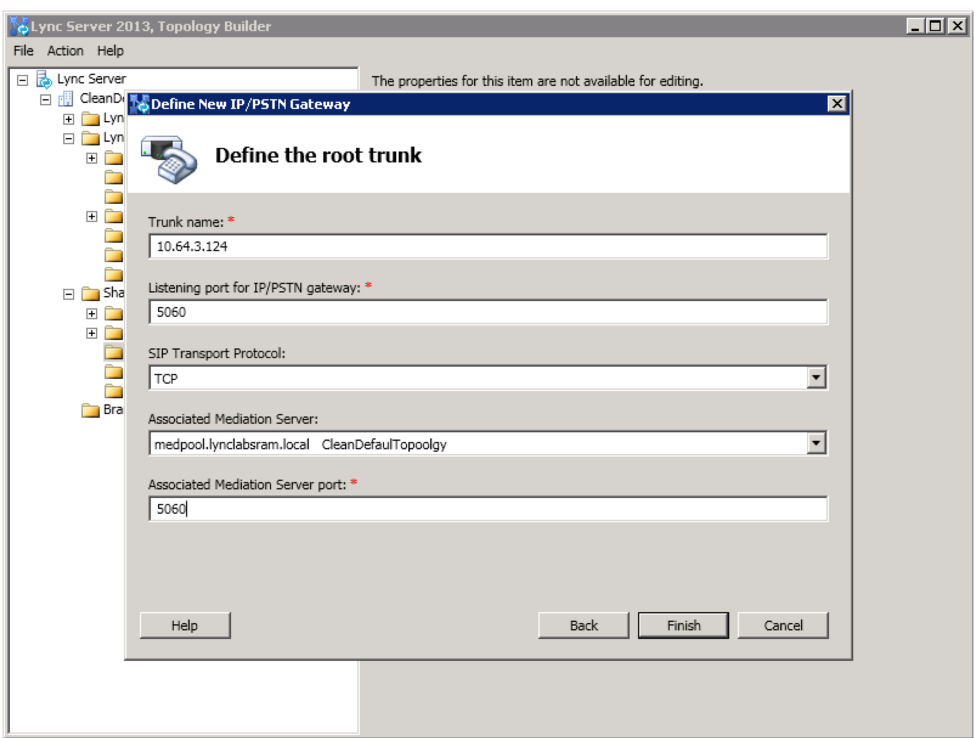
8. In the **Define New IP/PSTN Gateway** window, enter the IP address of the SIP interface of the ECB in the **FQDN** text box and click **Next**.



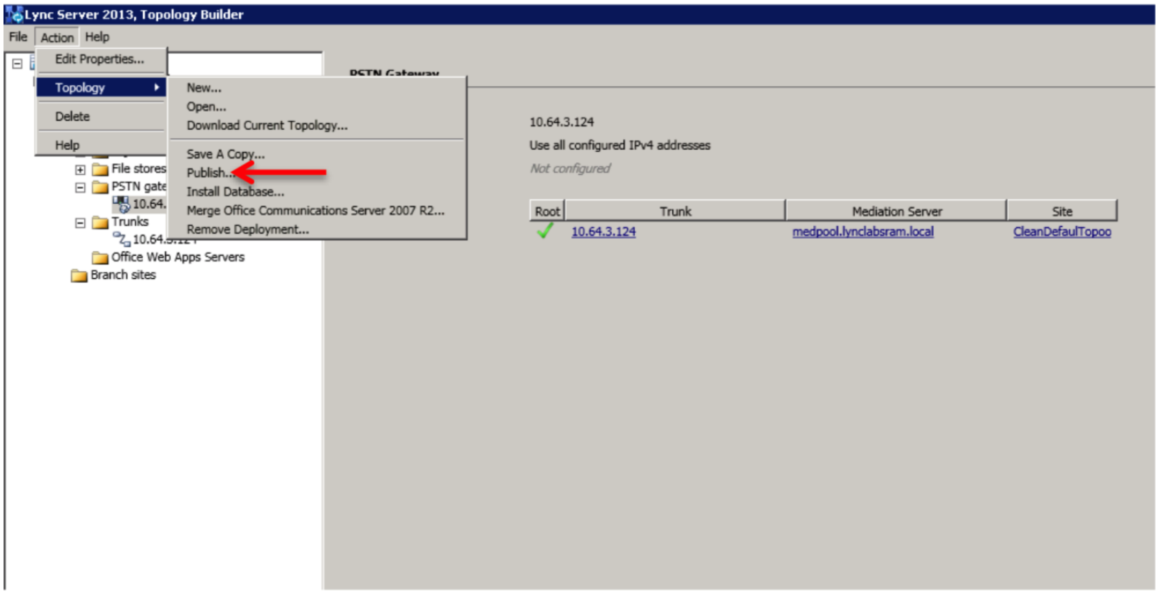
9. Select **Enable IPv4** in the **Define the IP address** section and click **Next**.



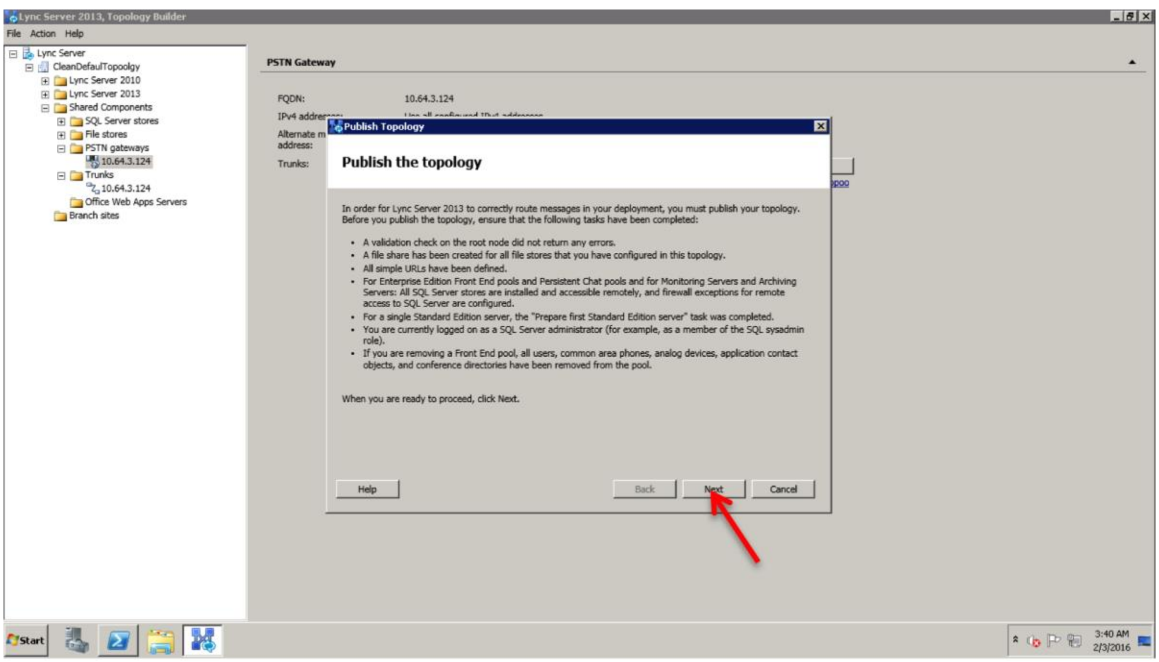
10. In the next section, enter the IP address of the ECB's SIP interface under **Trunk name**. Configure the **Listening port for IP/PSTN gateway** as 5060, TCP as the **SIP Transport Protocol**, and 5060 as the **Associated Mediation Server port,** and click **Finish**.



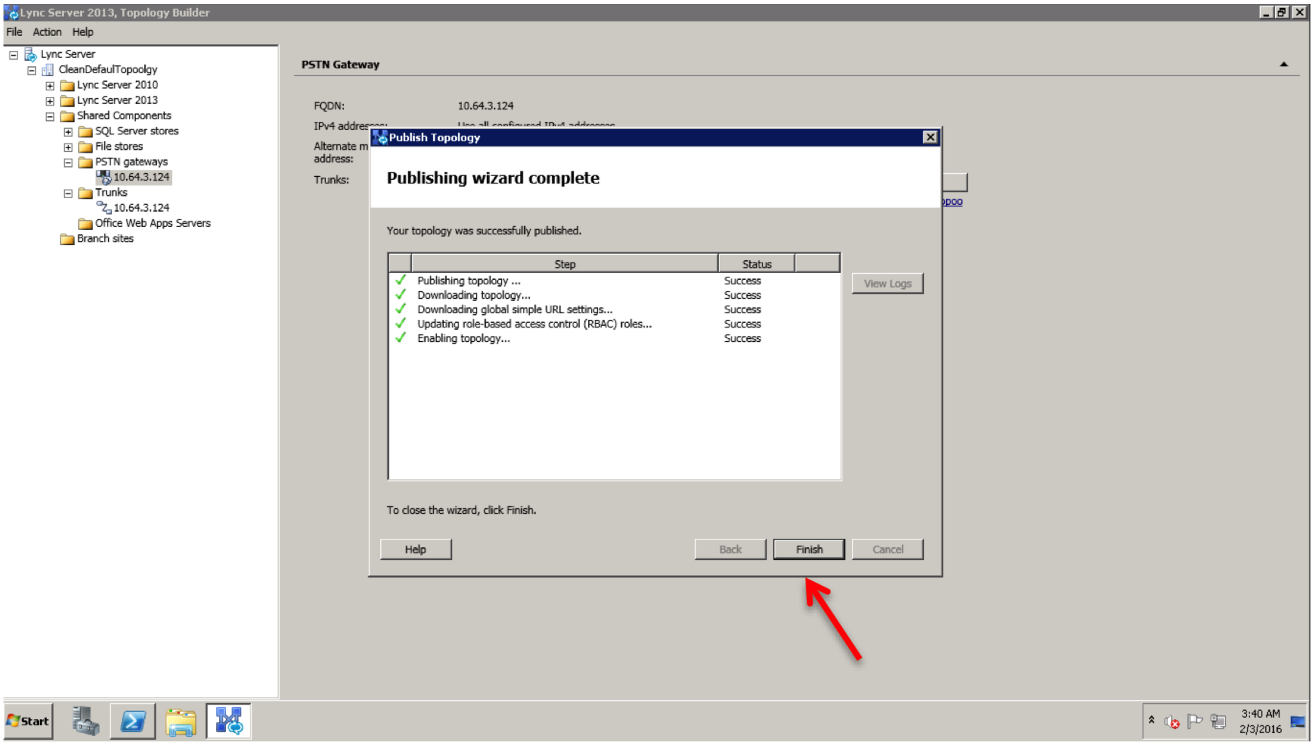
11. In the upper right hand corner of your screen under **Actions** select **Topology** then select **Publish**.



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



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



14. 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

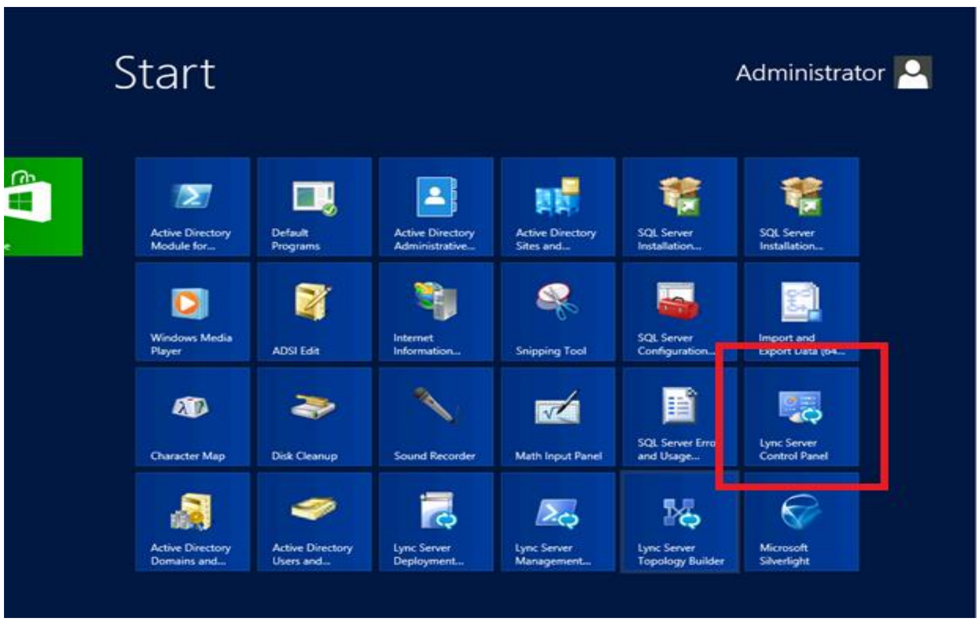
In order for the Lync Server Enterprise Voice clients to utilize the SIP trunking infrastructure that has been put in place, a route will need to be created to allow direction to this egress. Routes specify how Lync Server handles calls placed by enterprise voice users. When a user places a call, the 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

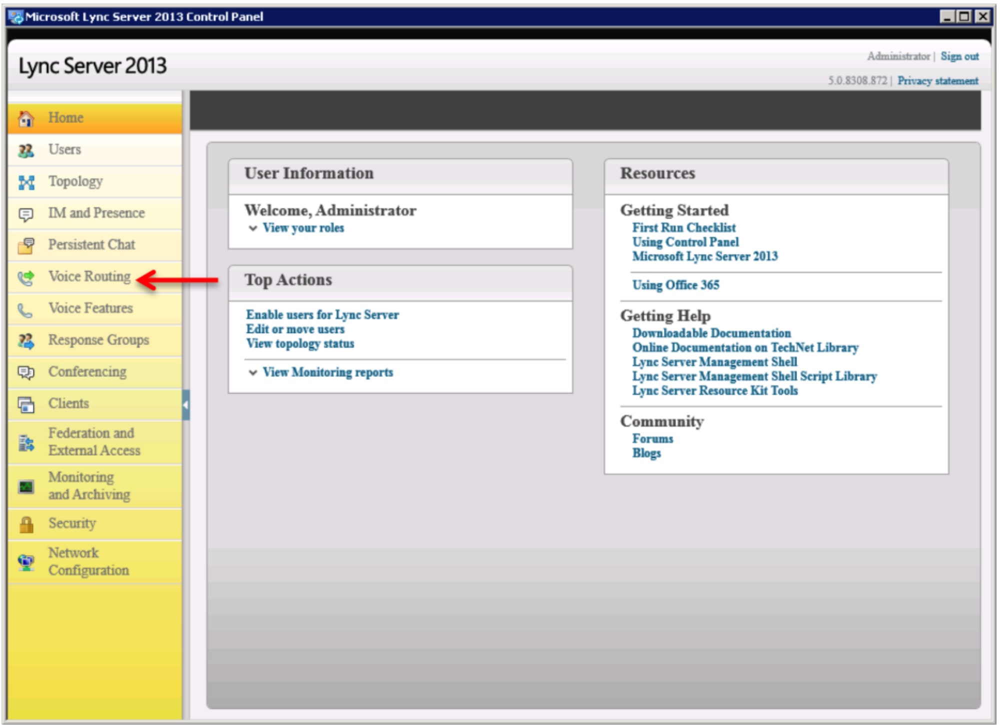
The following process details the steps to create the route:

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

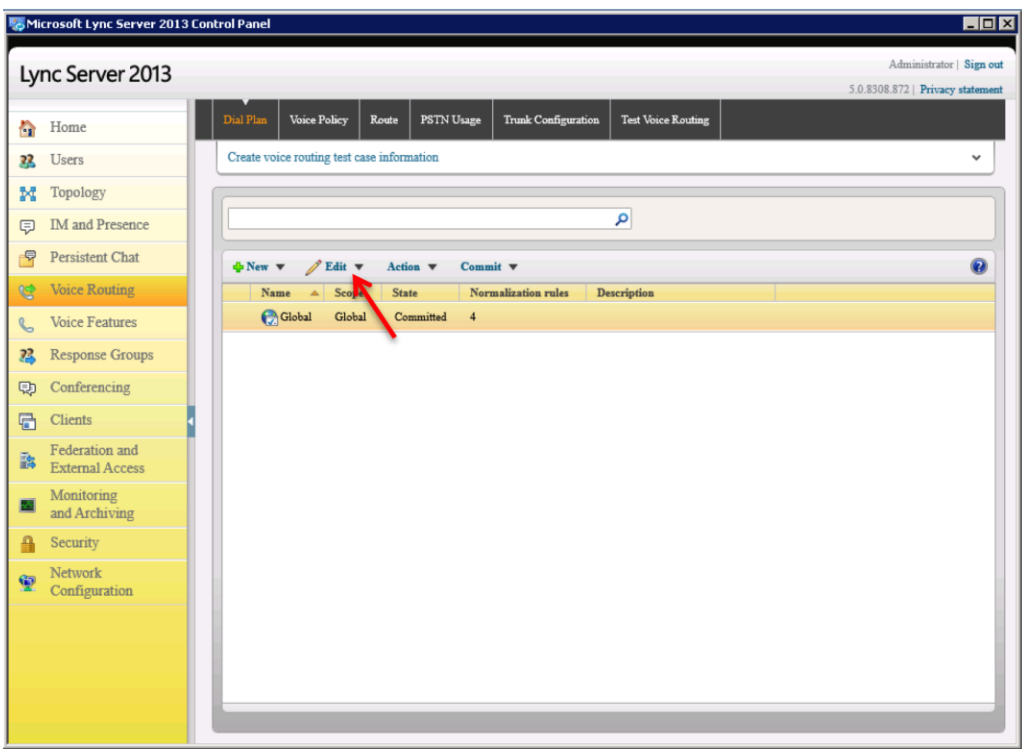


You will be prompted for credentials, enter your domain username and password.

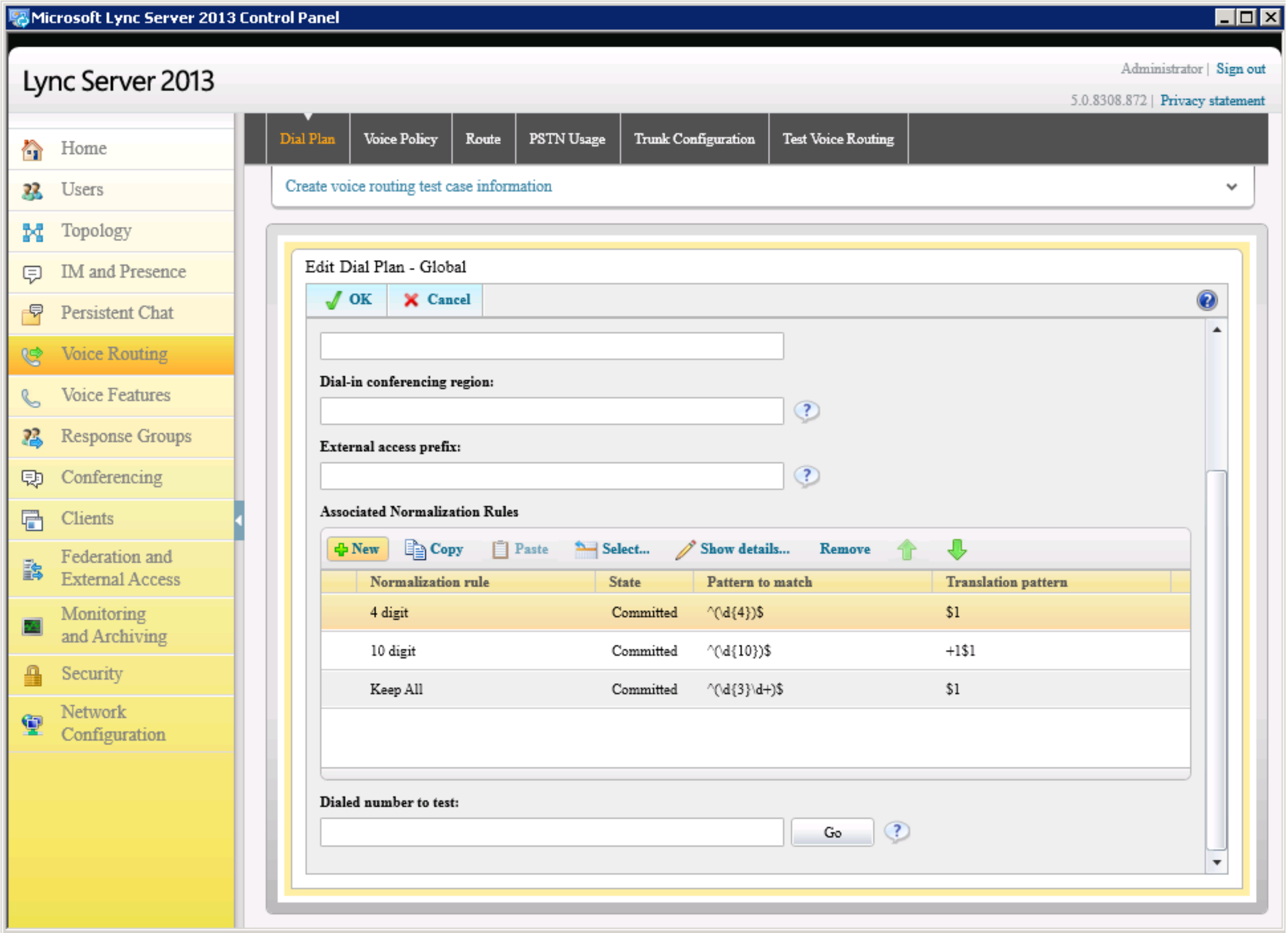
2. Once logged in, you will now be at the “Welcome Screen”. On the left hand side of the window, click on **Voice Routing**.



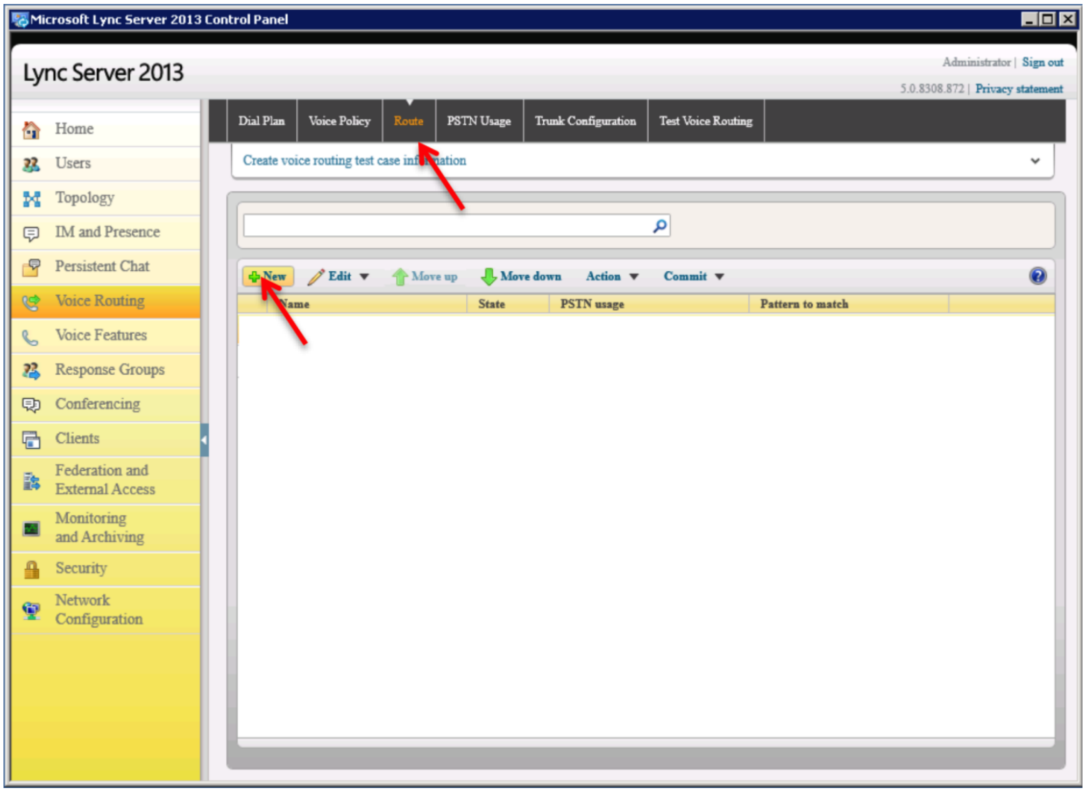
3. The Dial Plan tab in the Voice Routing section will be displayed. Select the Global dial plan. On the content area toolbar, click **Edit**



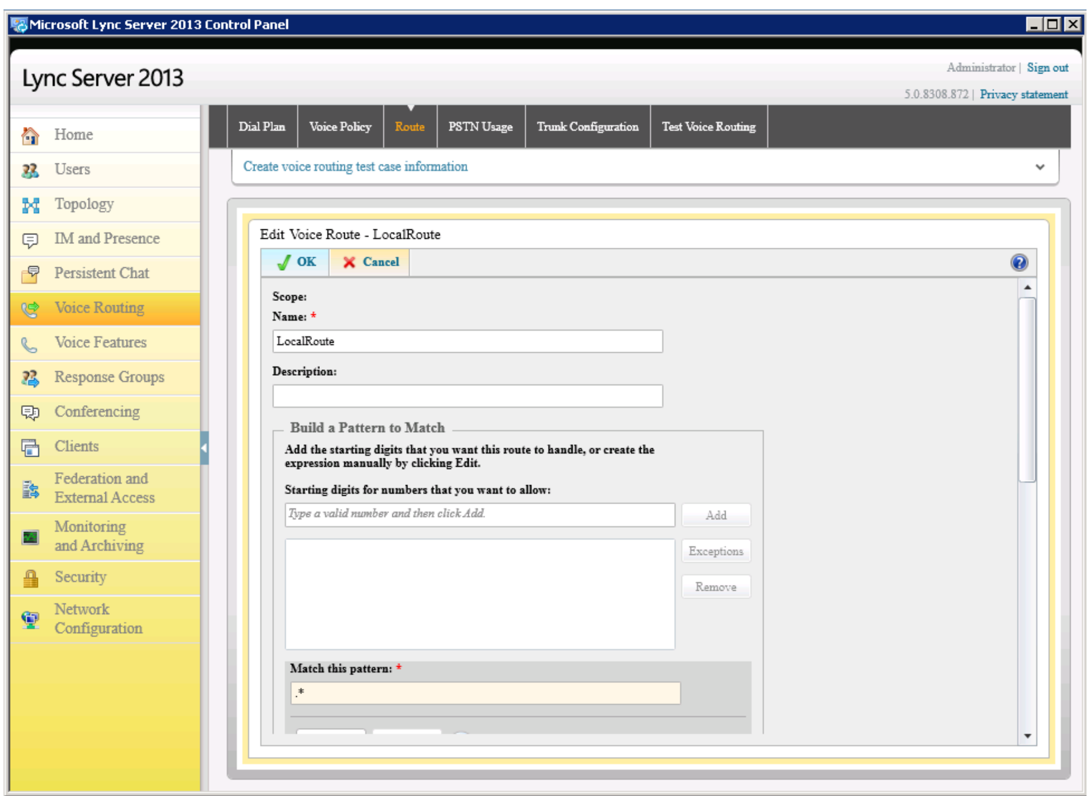
4. Next you build a Dial Plan and a translation rule for the phone numbers you want this route to handle.



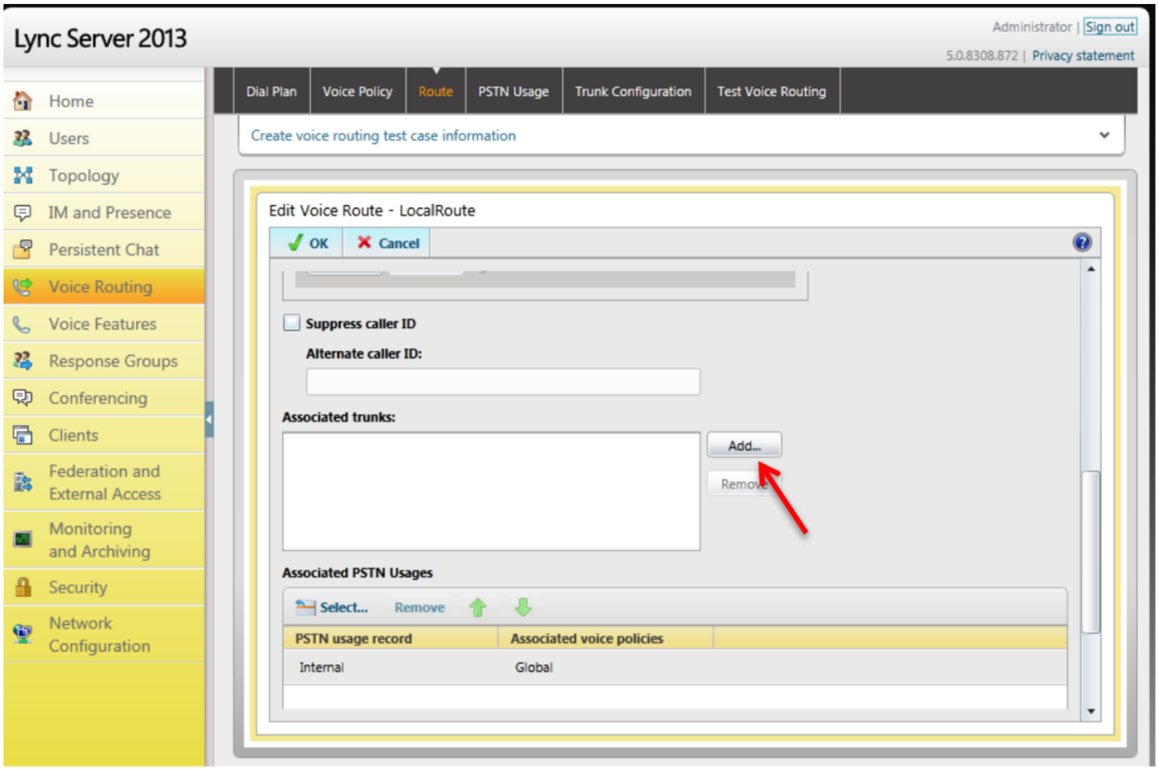
5. On the top row of the tabs, select **Route**. On the content area toolbar, click **+New**.



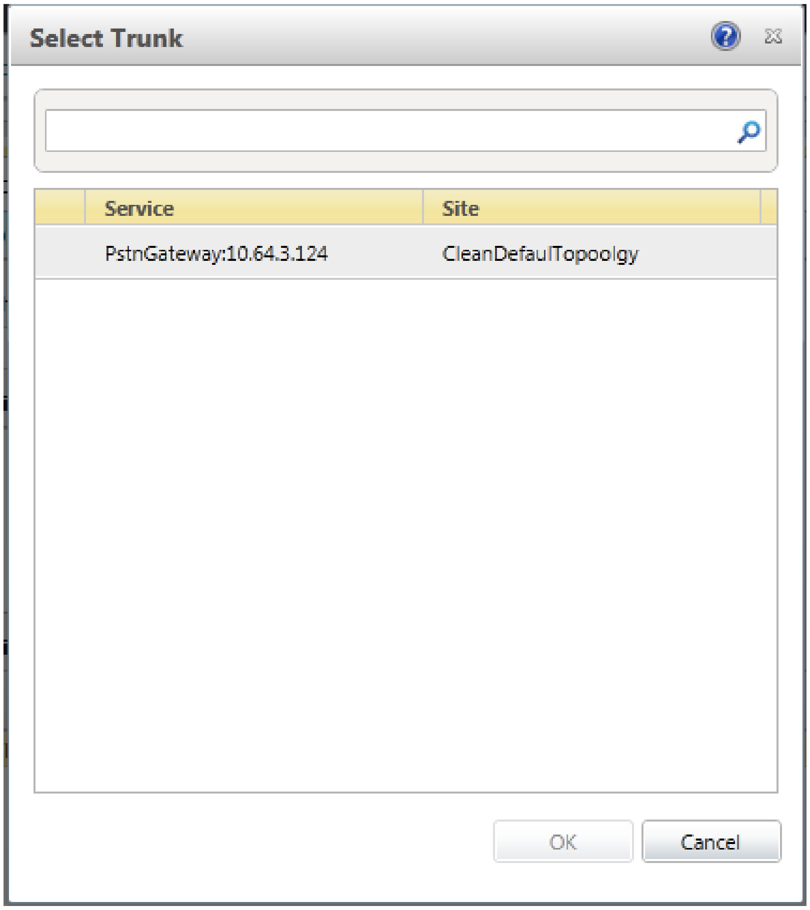
6. On the **New Voice Route** page, in the **Name** field, enter the name you have selected for the Route. In our example, it is labeled “LocalRoute”. Leave the **Match this pattern** field as .* so all numbers will be matched.



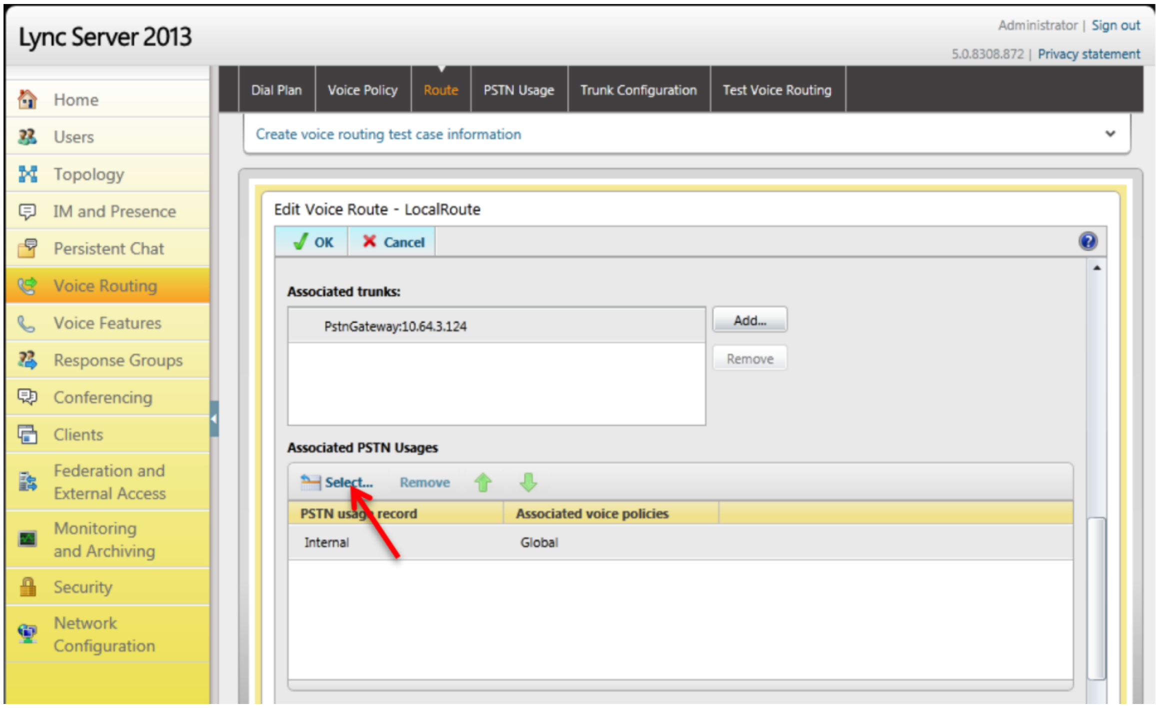
7. Next you want to associate the Voice Route with the **Trunk** you have just created. Scroll down to **Associated Trunks**, click on the **Add** button.



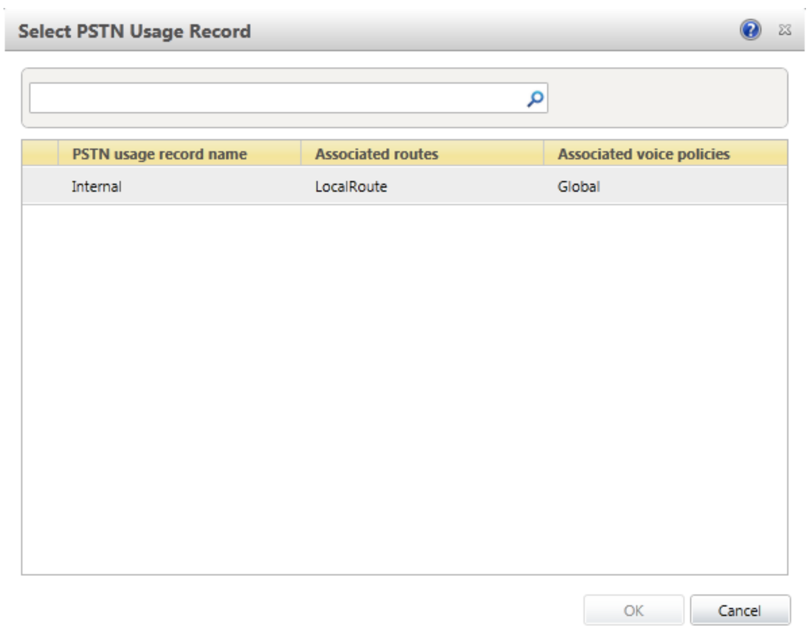
8. You will now be at a window showing available Trunks to associate your Voice Route. Click on the PSTN gateway that you just created and then click the **OK** button.



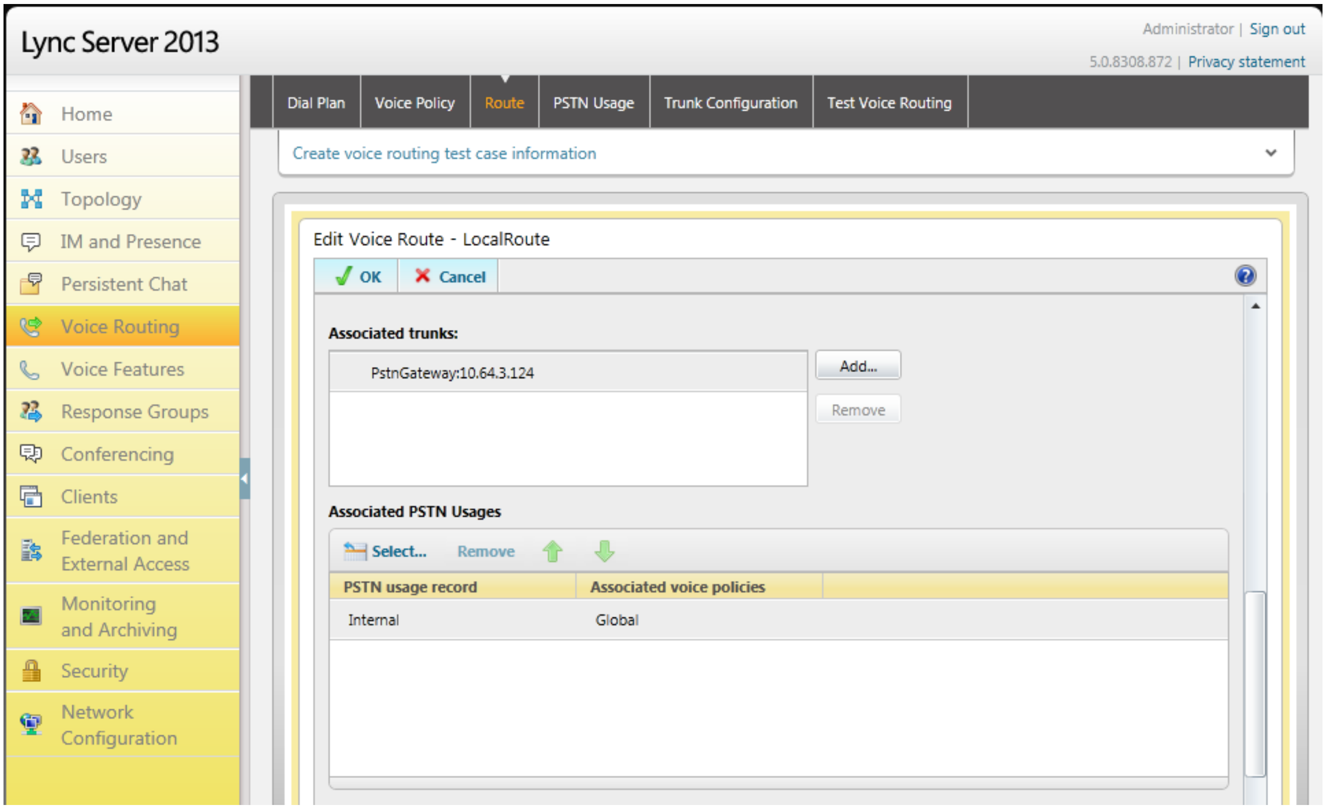
9. You can now see that you have associated your trunk with the route you created. An appropriate PSTN usage record will need to be assigned as well. In our example, we use one that was already created in the enterprise. Click on the **Select** button under **Associated PSTN Usages**



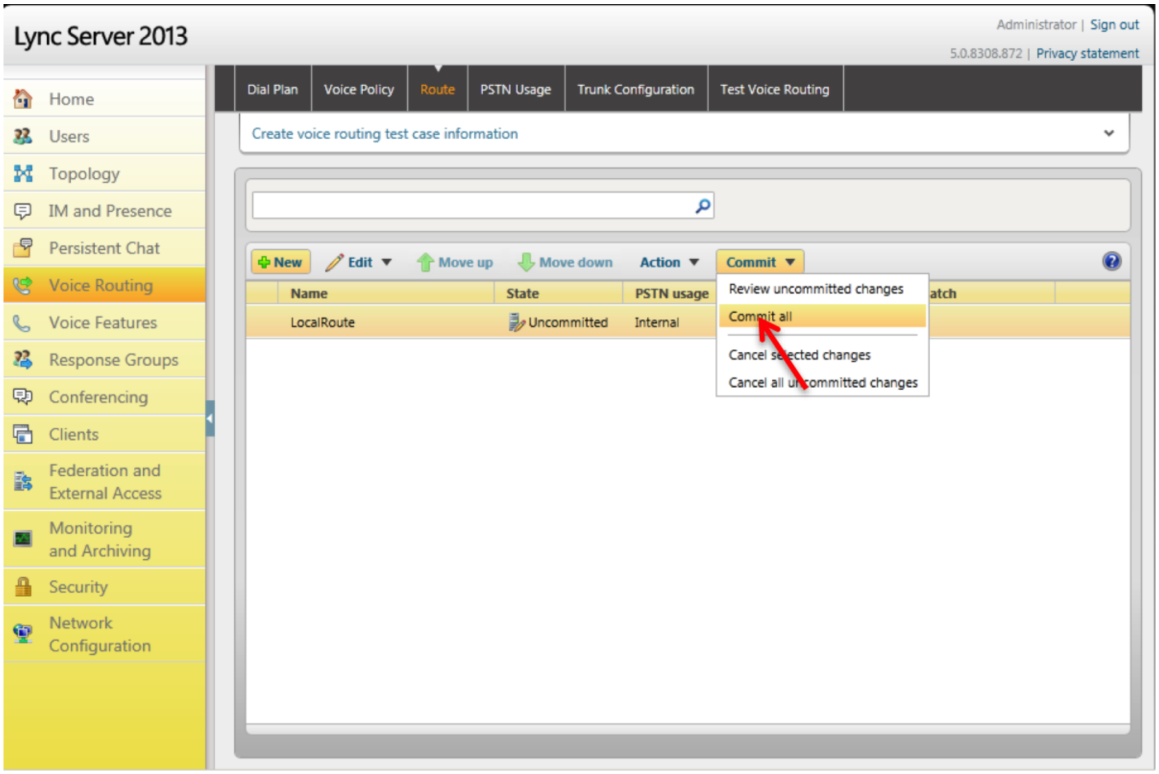
10. In the **Select PSTN Usage Record** window displayed, select the appropriate PSTN Usage Record and click **OK**.



11. 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.



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



Phase 5 – Configuring the Skype for Business Server

The enterprise will have a fully functioning Skype for Business (SFB) 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 SFB Server to operate with the Oracle ECB:

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

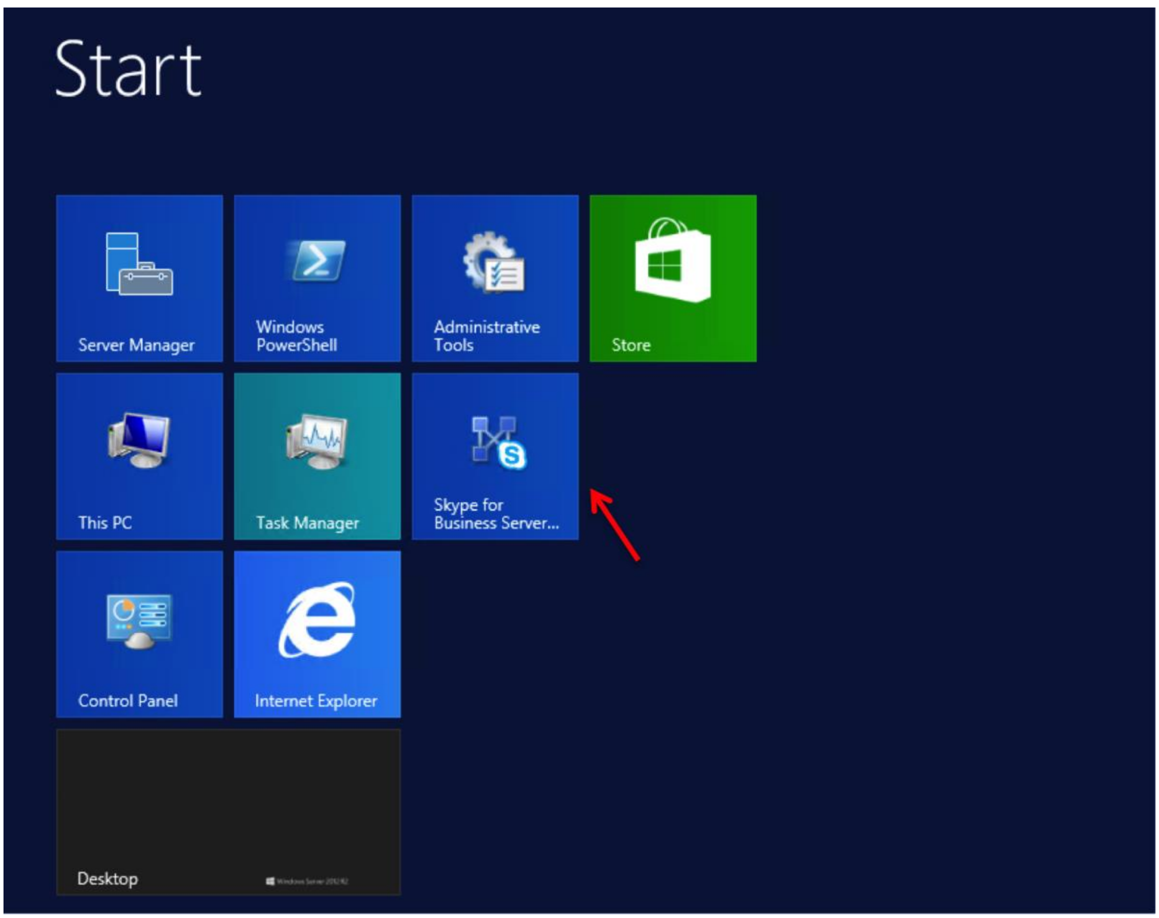
To add the PSTN gateway, we will need:

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

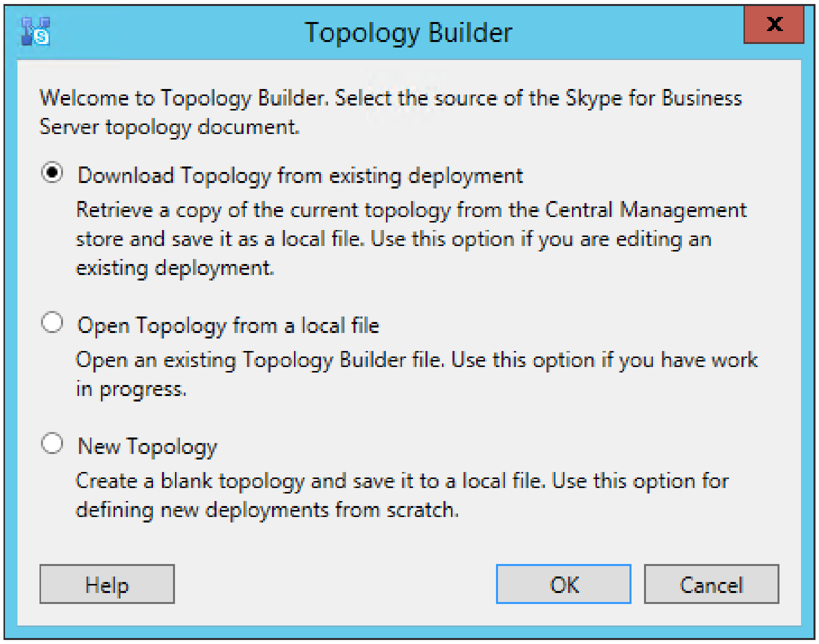
Adding the ECB as a PSTN gateway

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 SFB Server Topology Builder.

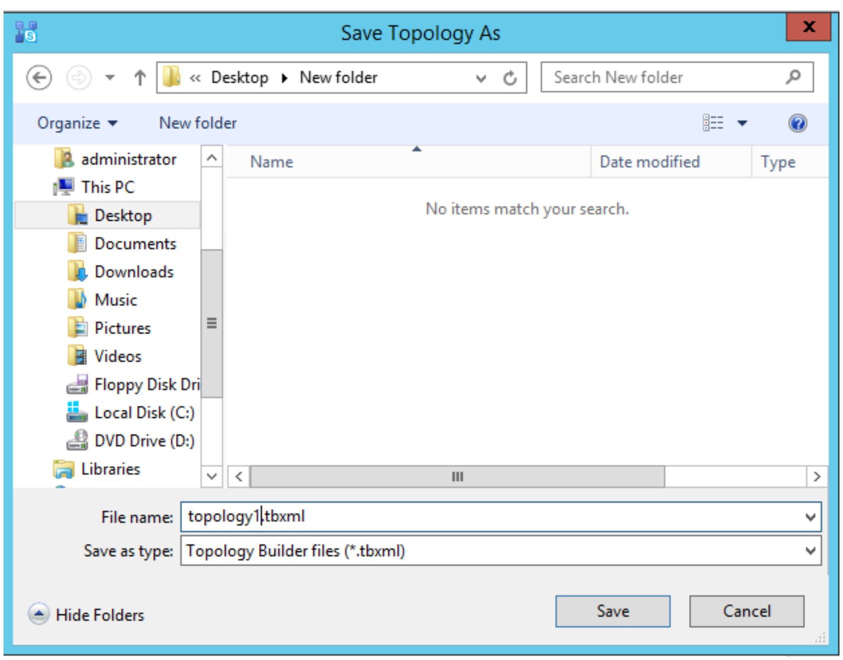


1. The Topology Builder window will now be displayed. Select **Download Topology from existing deployment**.

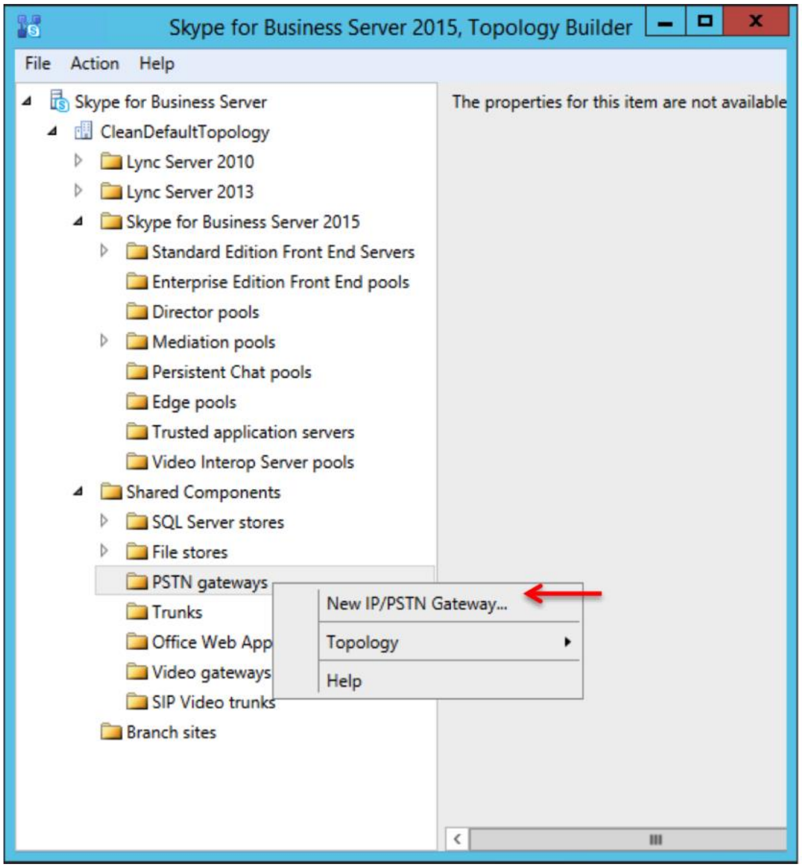


2. You will then see a screen showing that the current toplogy is being downloaded. Click the **OK** button.
3. Next you will be prompted to save the topology which you have imported. You should revision the name or number of the topology according to the standards used within the enterprise. Click the **Save** button

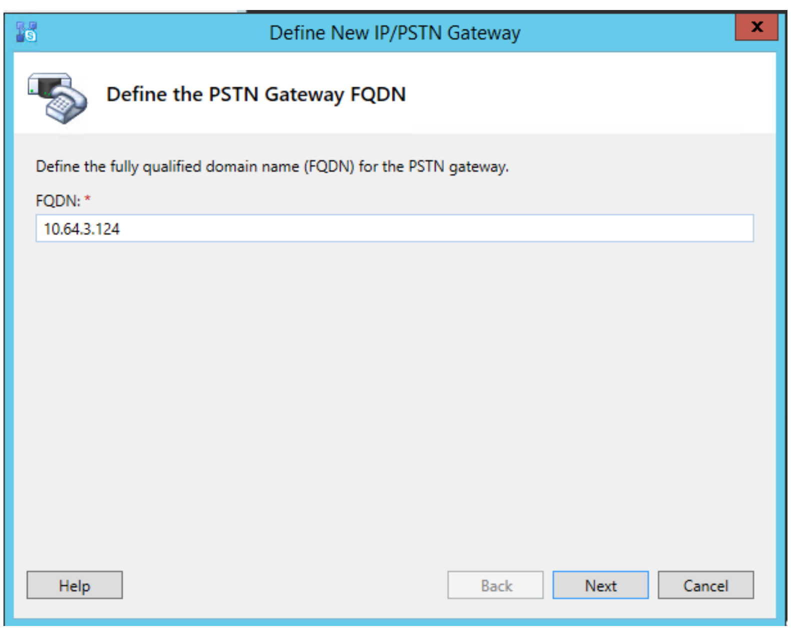
Note: This keeps track of topology changes and, if desired, will allow you to fall back from any changes you make during this installation



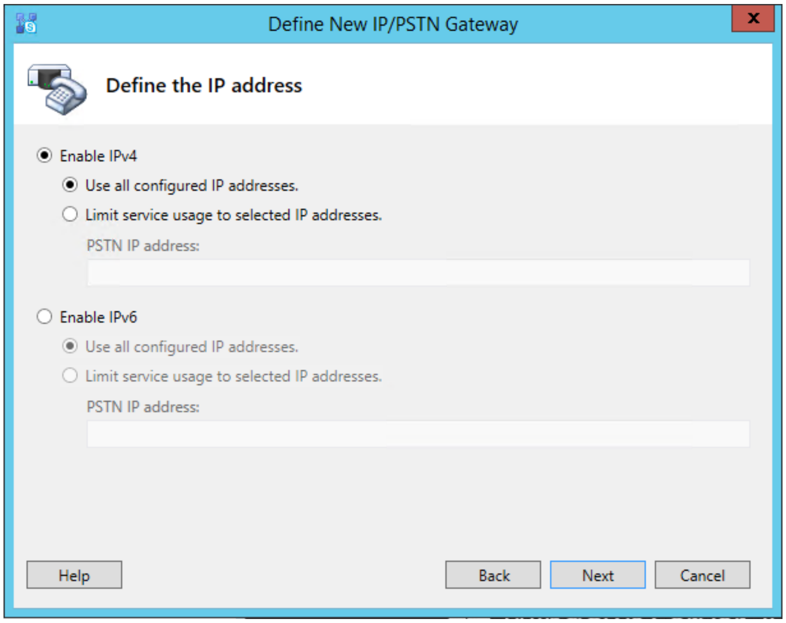
4. In the upper left hand corner, expand the site in which the PSTN gateway will be added. In our case, the site is labeled **CleanDefaultTopology**. Expand **Shared Components.** Then click on the **PSTN Gateways.** Right click on **PSTN gateways** and select **New IP/PSTN Gateway**.



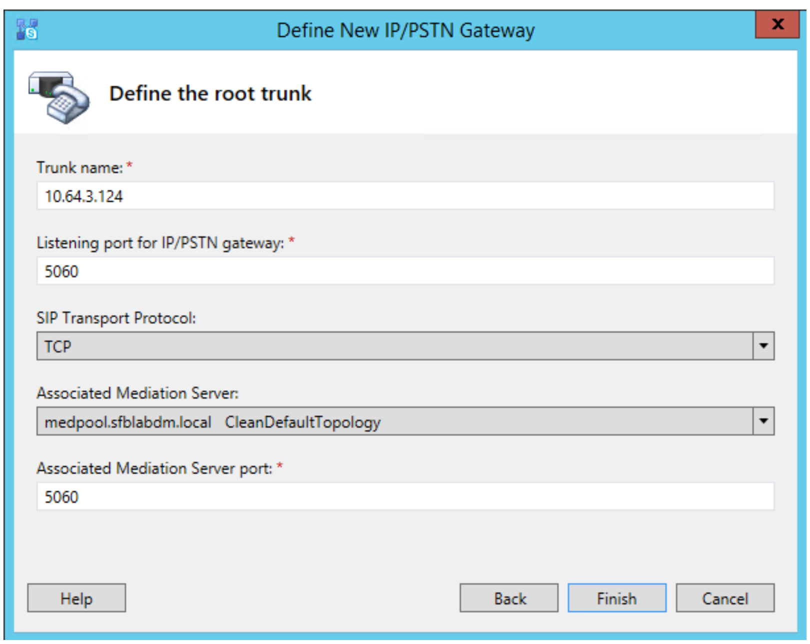
5. In the **Define New IP/PSTN Gateway** window, enter the IP address of the SIP interface of the ECB in the **FQDN** text box and click **Next**.



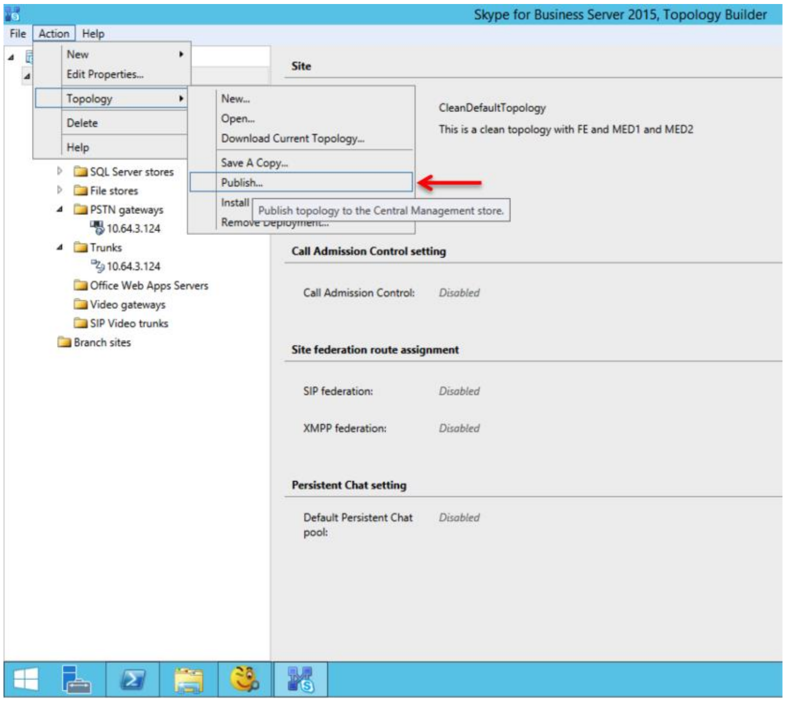
6. Select **Enable IPv4** in the **Define the IP address** section and click **Next**.



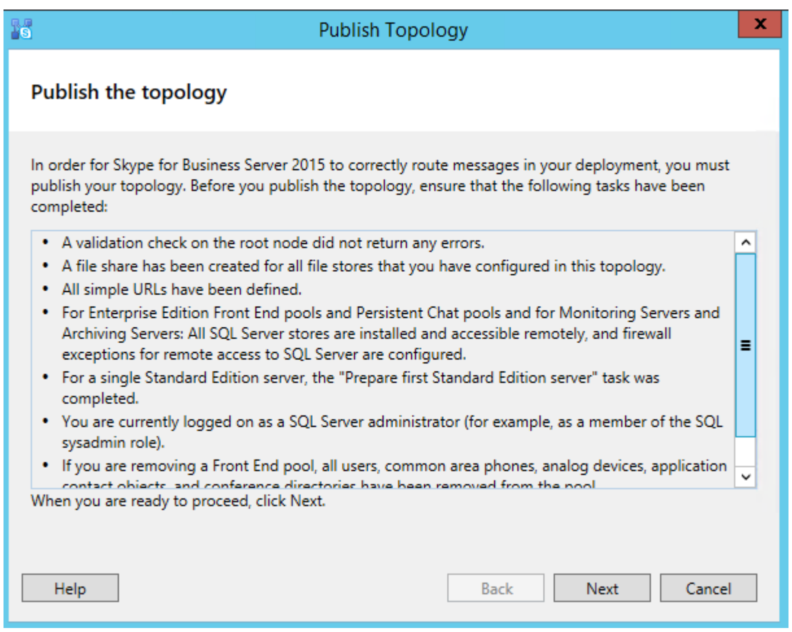
7. In the next section, enter the IP address of the ECB's SIP interface under **Trunk name**. Configure the **Listening port for IP/PSTN gateway** as 5060, TCP as the **SIP Transport Protocol**, and 5060 as the **Associated Mediation Server port,** and click **Finish**.



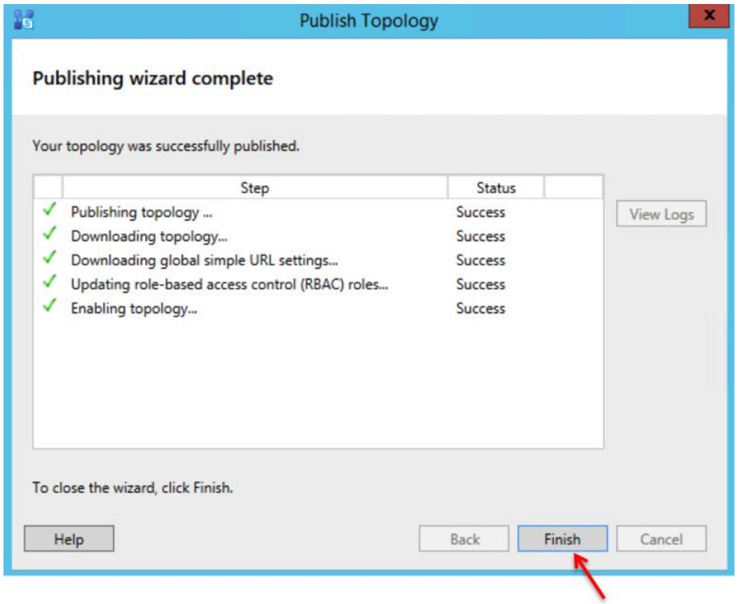
8. In the upper right hand corner of your screen under **Actions** select **Topology** then select **Publish**.



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



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



Creating a route within the Skype for Business infrastructure

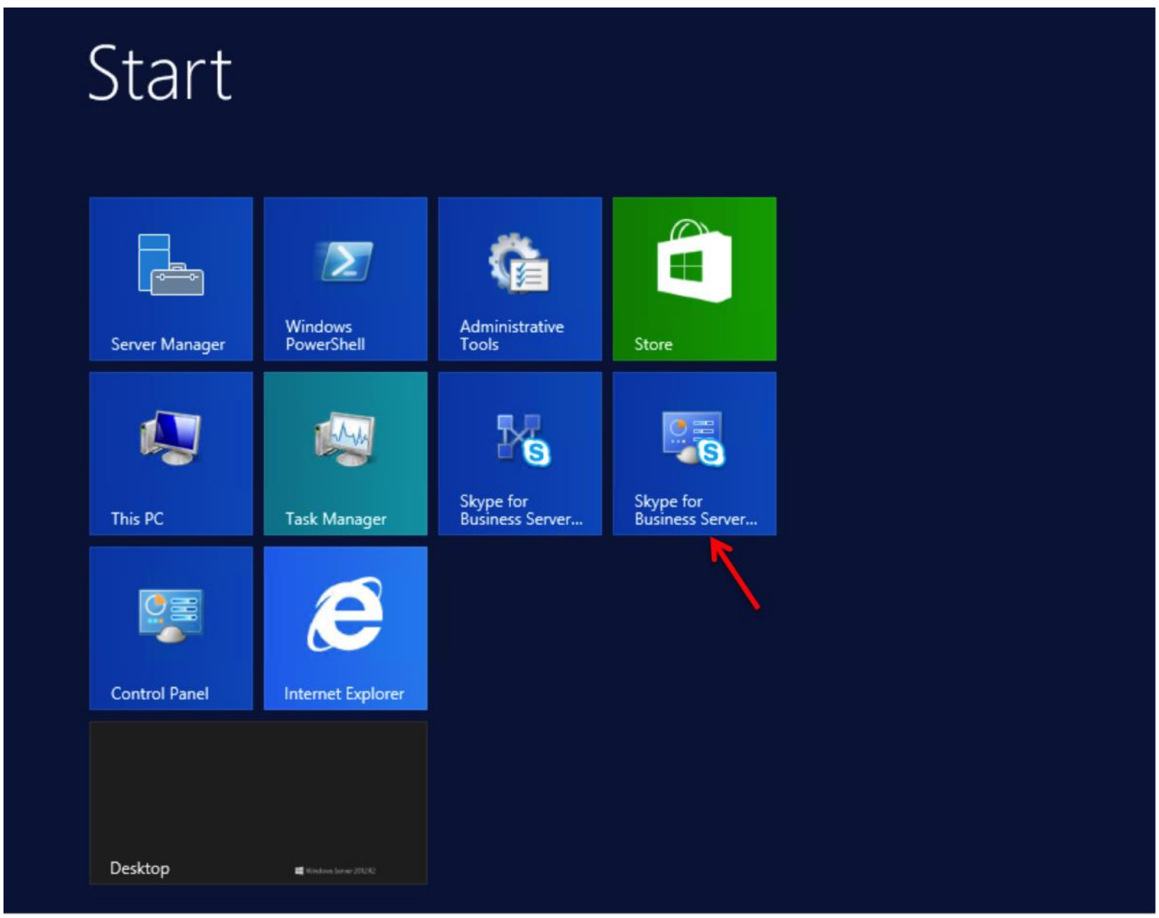
In order for the Skype for Business (SFB) clients to utilize the SIP trunking infrastructure that has been put in place, a route will need to be created to allow direction to this egress. Routes specify how SFB handles calls placed by enterprise voice users. When a user places a call, the server, if 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 SFB deployments such as dial plans, voice policies, and PSTN usages are not covered.

To add the route we will need:

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

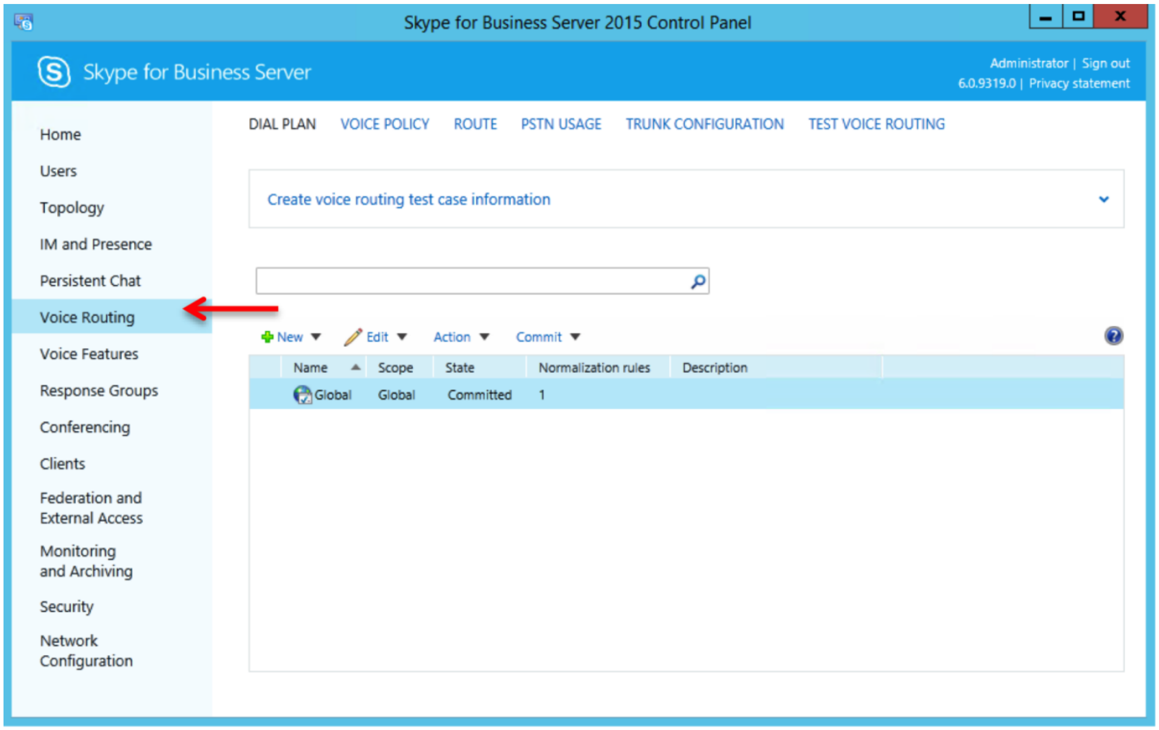
The following process details the steps to create the route:

1. From the Start bar, select SFB Control Panel.

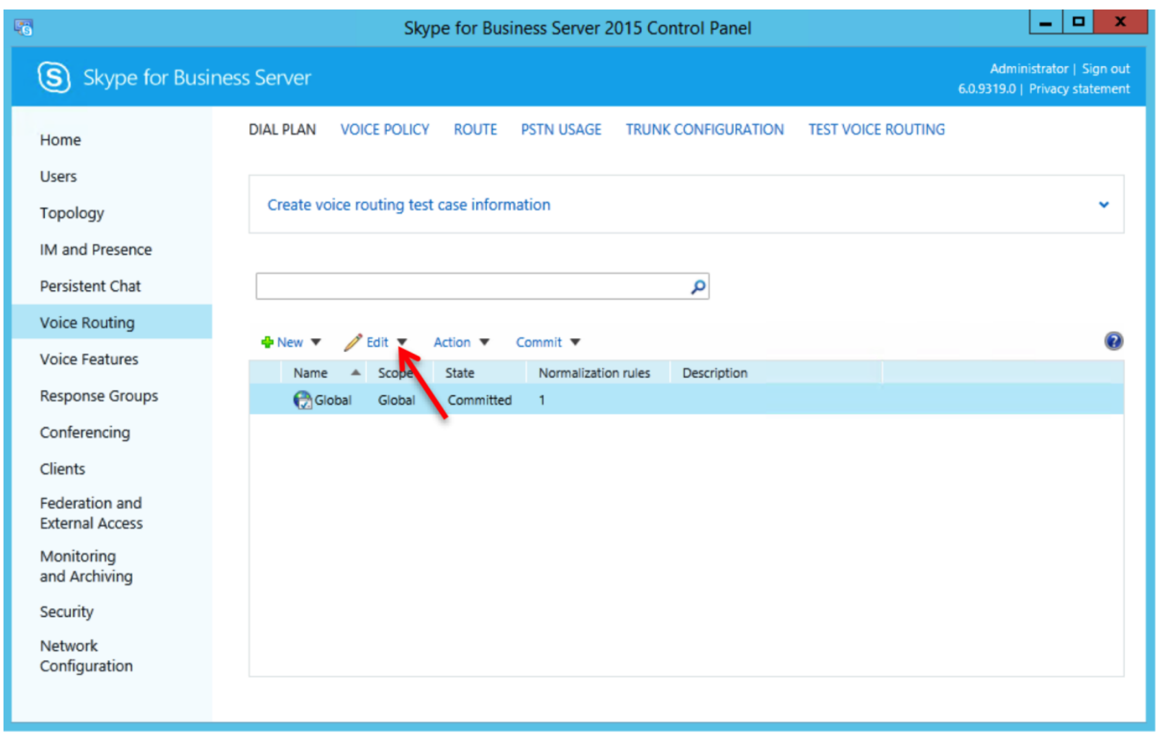


You will be prompted for credentials, enter your domain username and password.

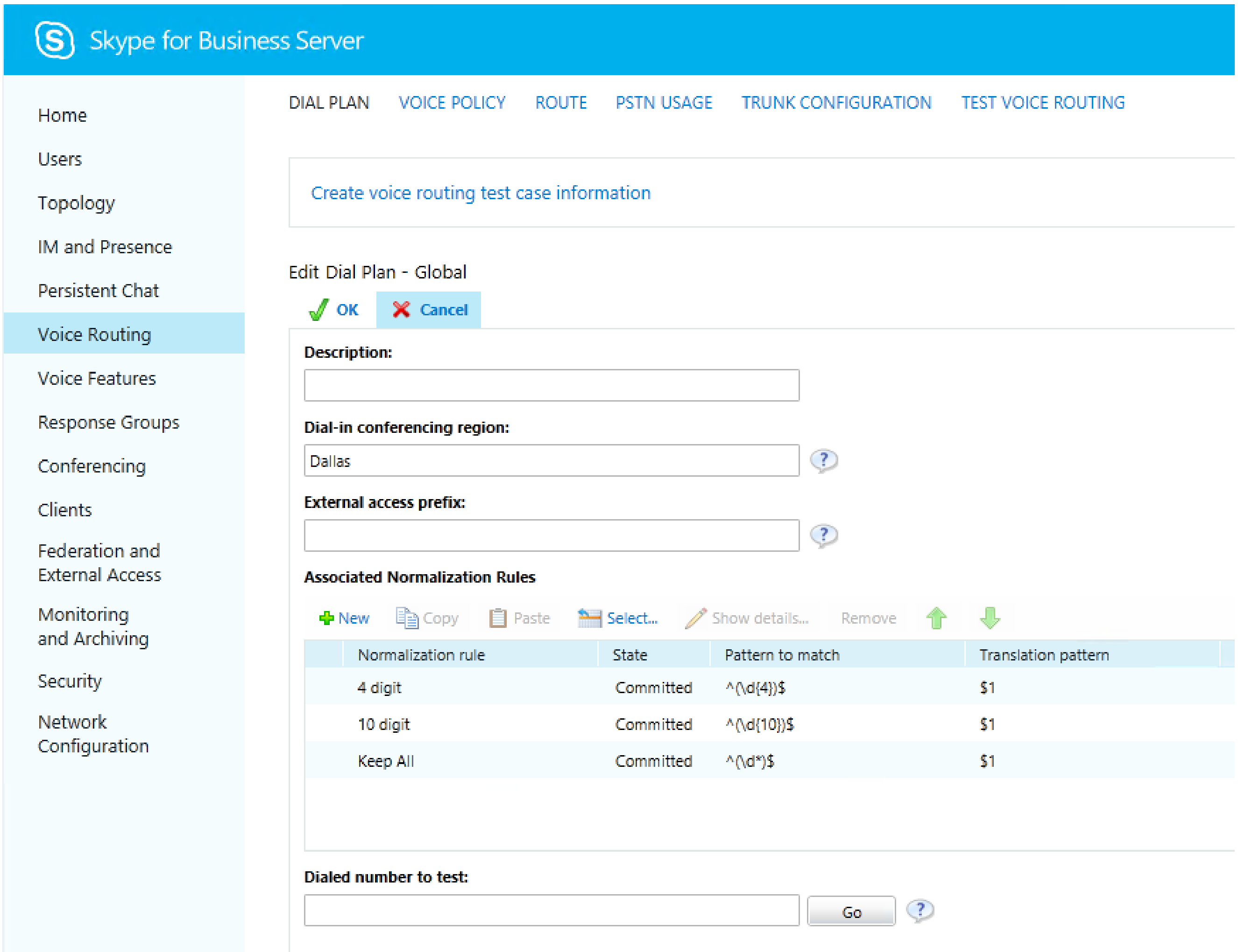
2. Once logged in, you will now be at the “Welcome Screen”. On the left hand side of the window, click on **Voice Routing**.



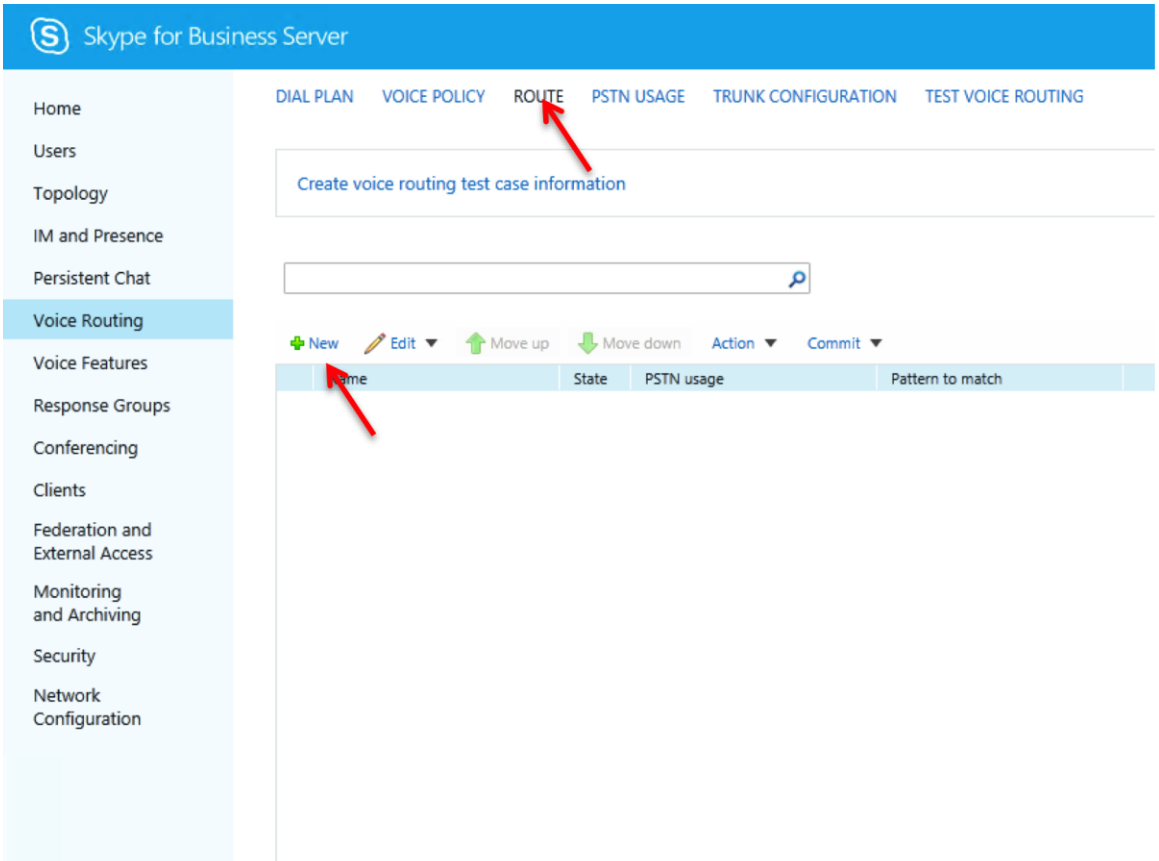
3. The Dial Plan tab in the Voice Routing section will be displayed. Select the Global dial plan. On the content area toolbar, click **Edit**



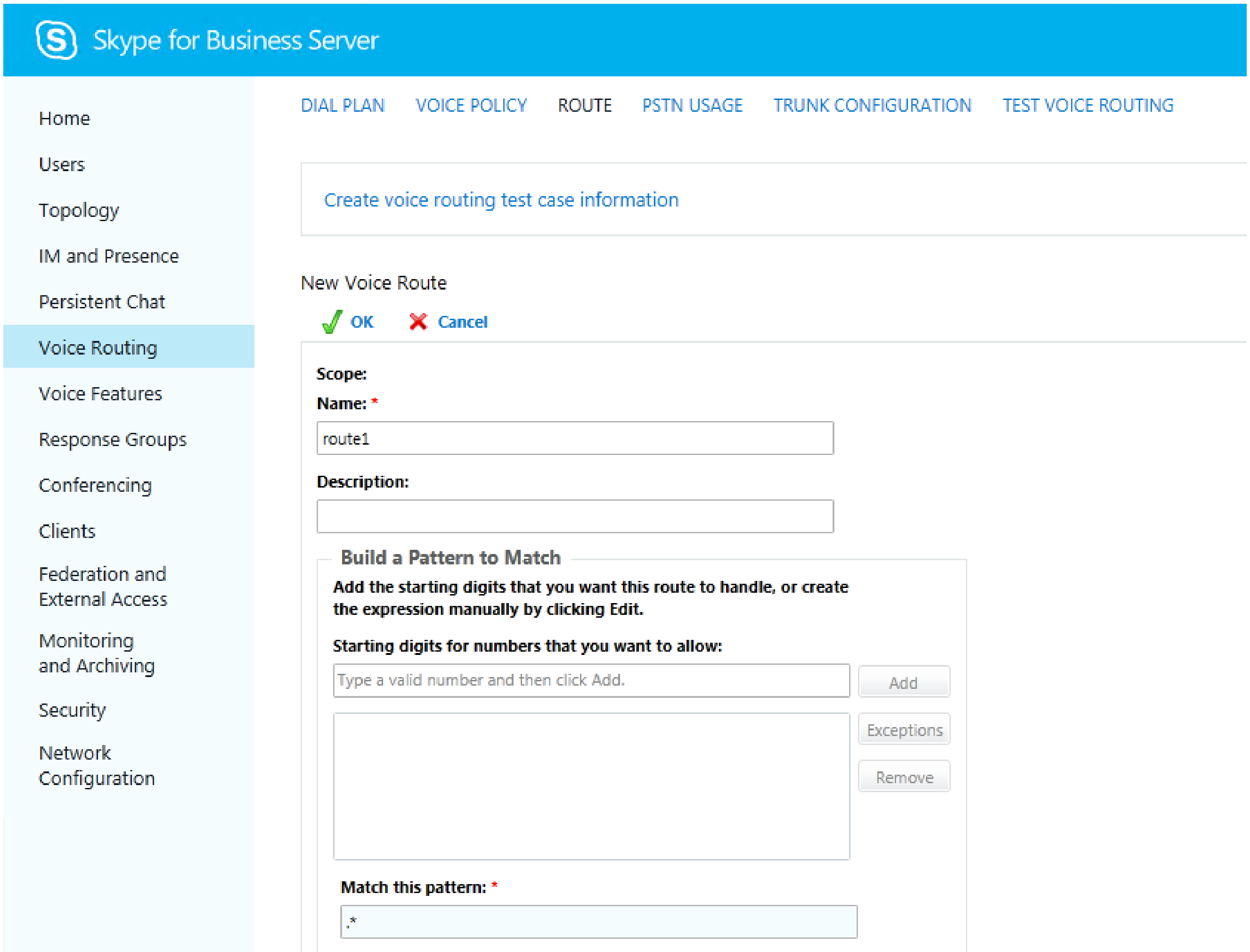
4. Next you build a Dial Plan and a translation rule for the phone numbers you want this route to handle.



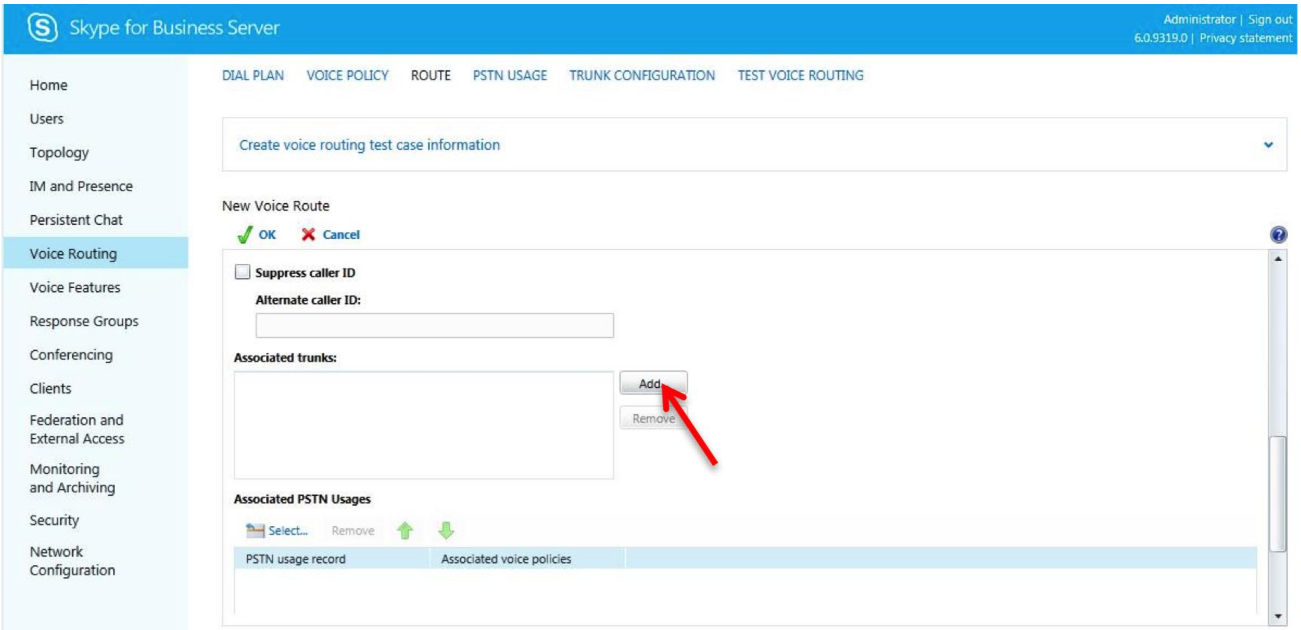
5. On the top row of the tabs, select **Route**. On the content area toolbar, click **+New**.



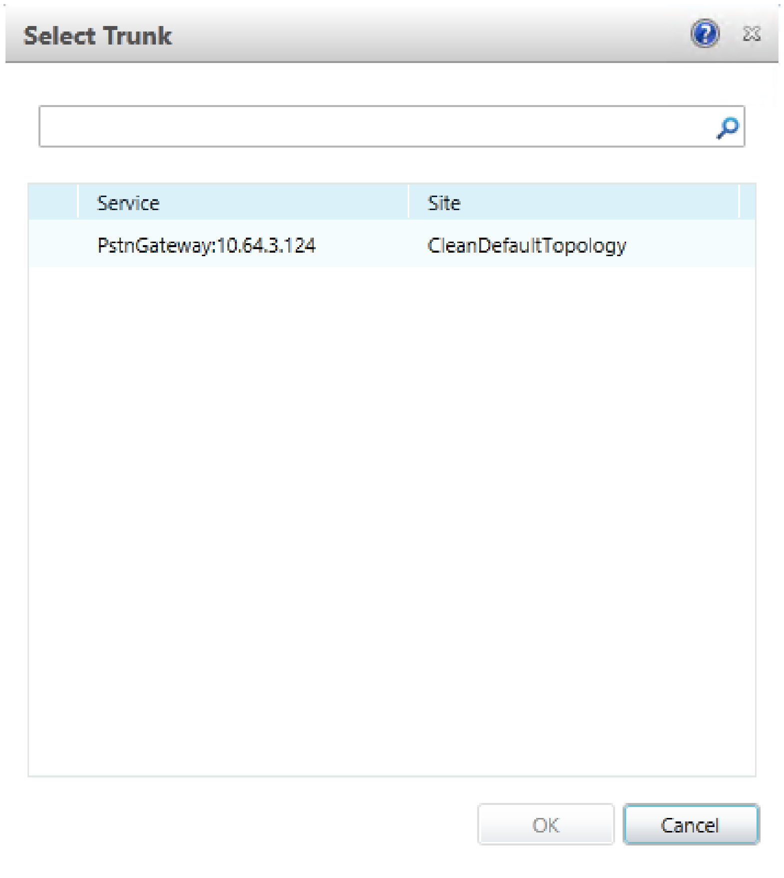
6. On the **New Voice Route** page, in the **Name** field, enter the name you have selected for the Route. In our example, it is labeled “route1”. Leave the **Match this pattern** field as .* so all numbers will be matched.



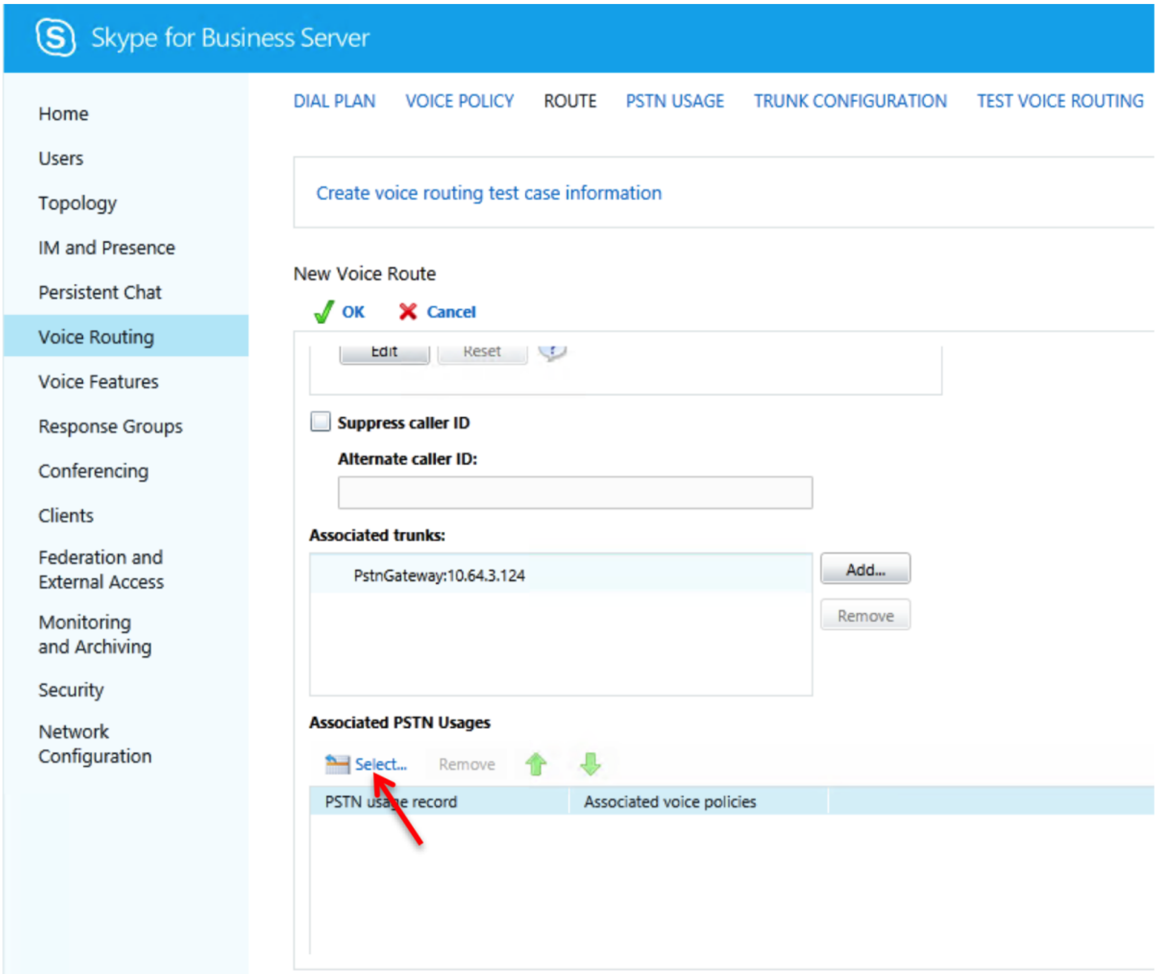
7. Next you want to associate the Voice Route with the **Trunk** you have just created. Scroll down to **Associated Trunks**, click on the **Add** button.



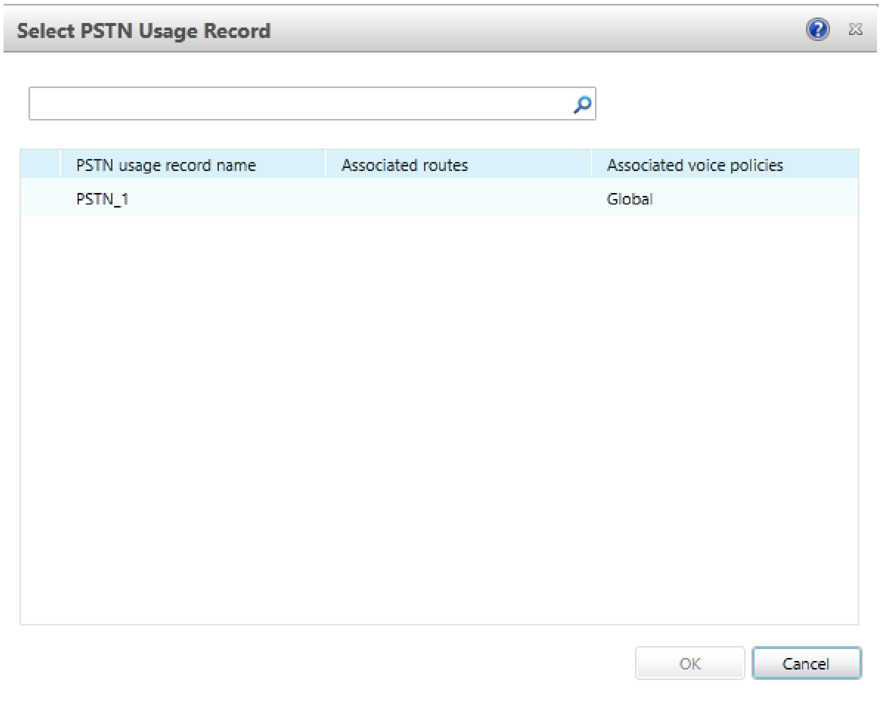
8. You will now be at a window showing available Trunks to associate your Voice Route. Click on the PSTN gateway that you just created and then click the **OK** button.



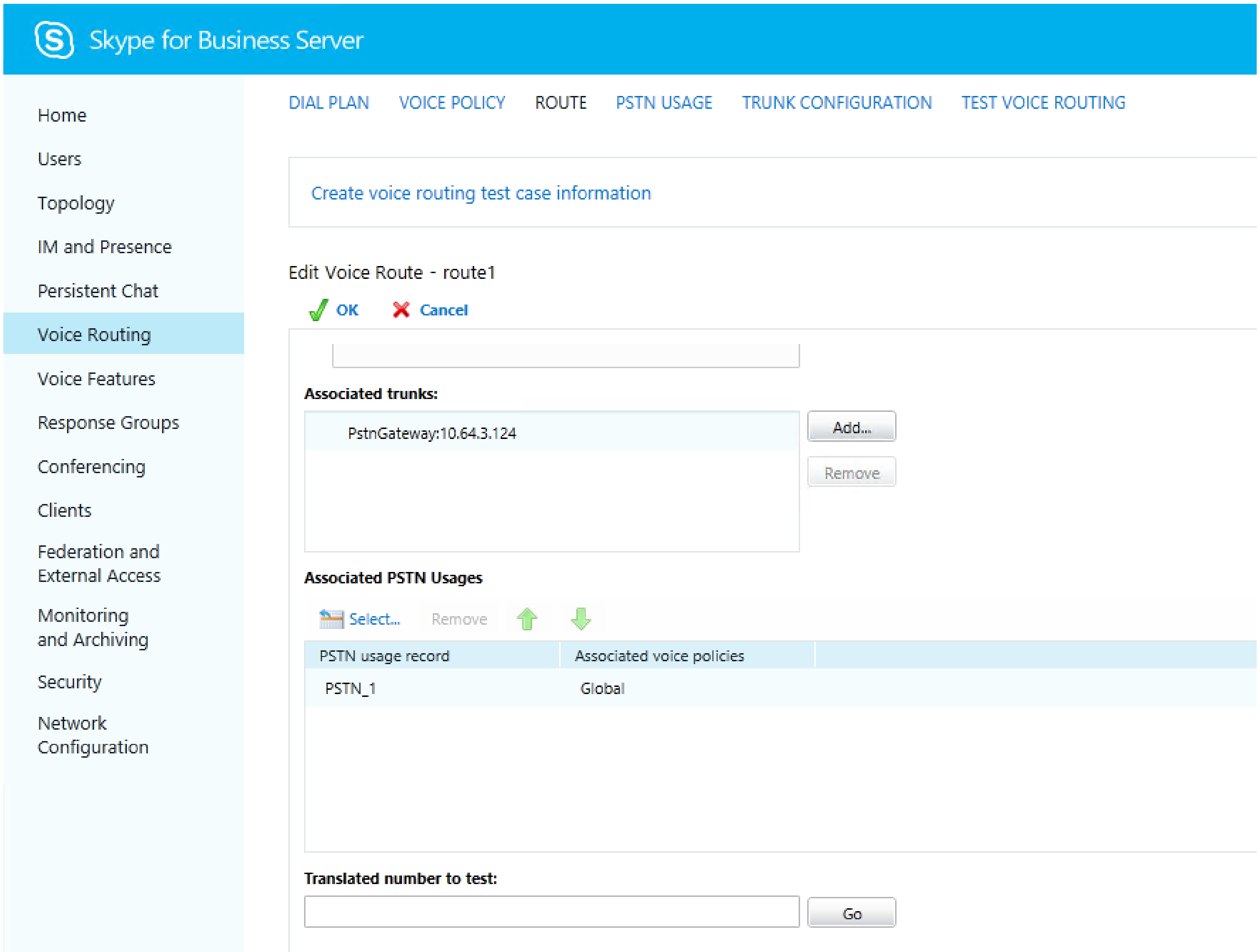
9. You can now see that you have associated your trunk with the route you created. An appropriate PSTN usage record will need to be assigned as well. In our example, we use one that was already created in the enterprise. Click on the **Select** button under **Associated PSTN Usages**



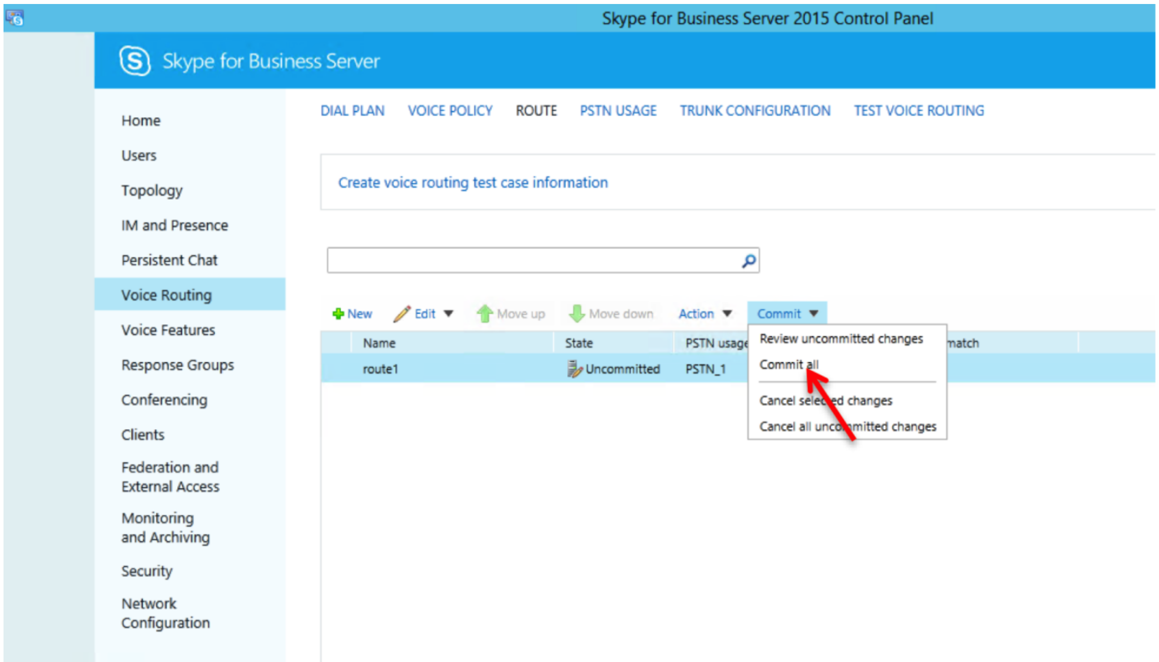
10. In the **Select PSTN Usage Record** window displayed, select the appropriate PSTN Usage Record and click **OK**.



11. 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.



12. You will now be at the Routes page showing route1. Click the **Commit** drop-down menu, and then **Commit All**.



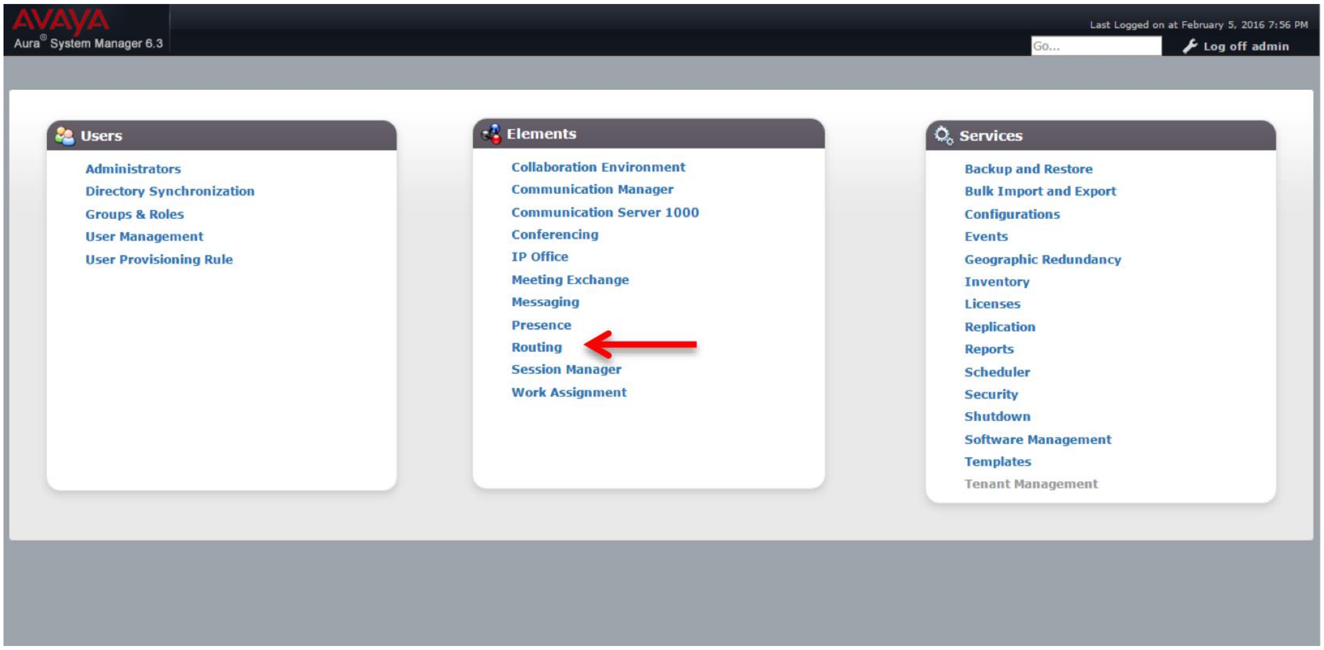
Phase 6 – Configuring the Avaya Session Manager 6.3

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

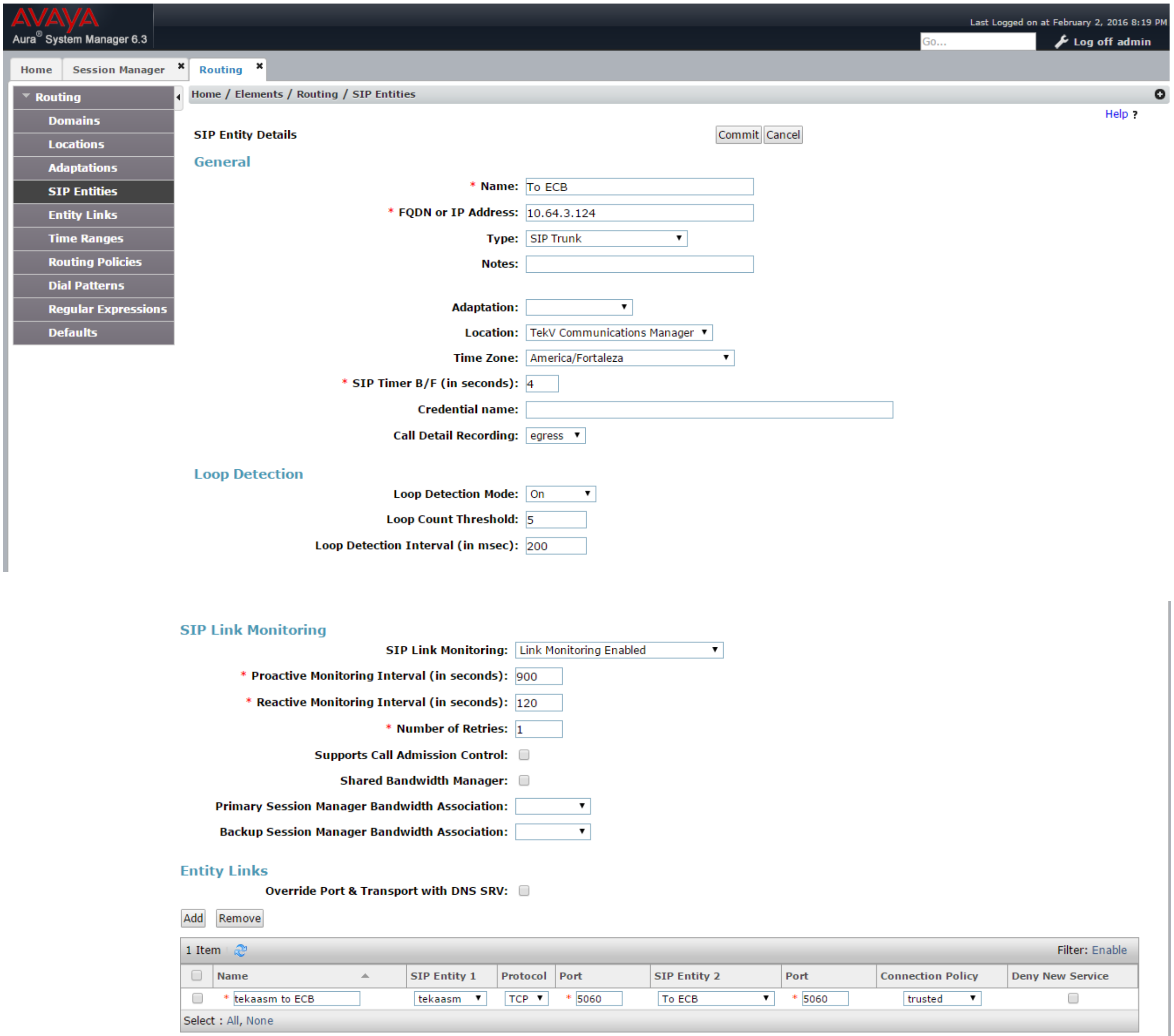
- Adding the ECB as a SIP Entity
- Configuring an Entity link between ECB and Session Manager
- Creating a Routing policy to assign the appropriate routing destination.

Adding the ECB as a SIP Entity

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

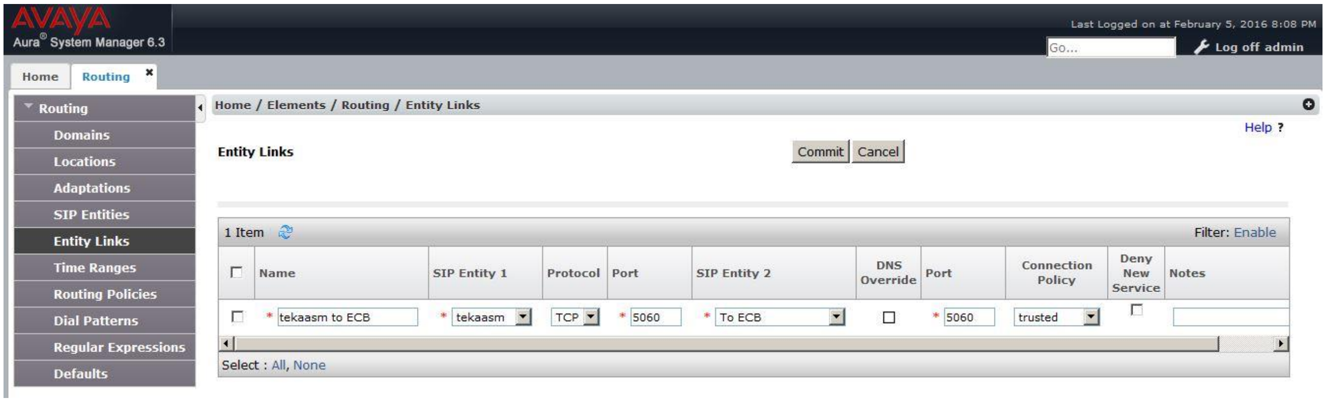


On the **Routing** tab, select **SIP Entities** from the menu on the left side of the screen. Click **New** to add ECB as a SIP entity as shown below and click **Commit**.



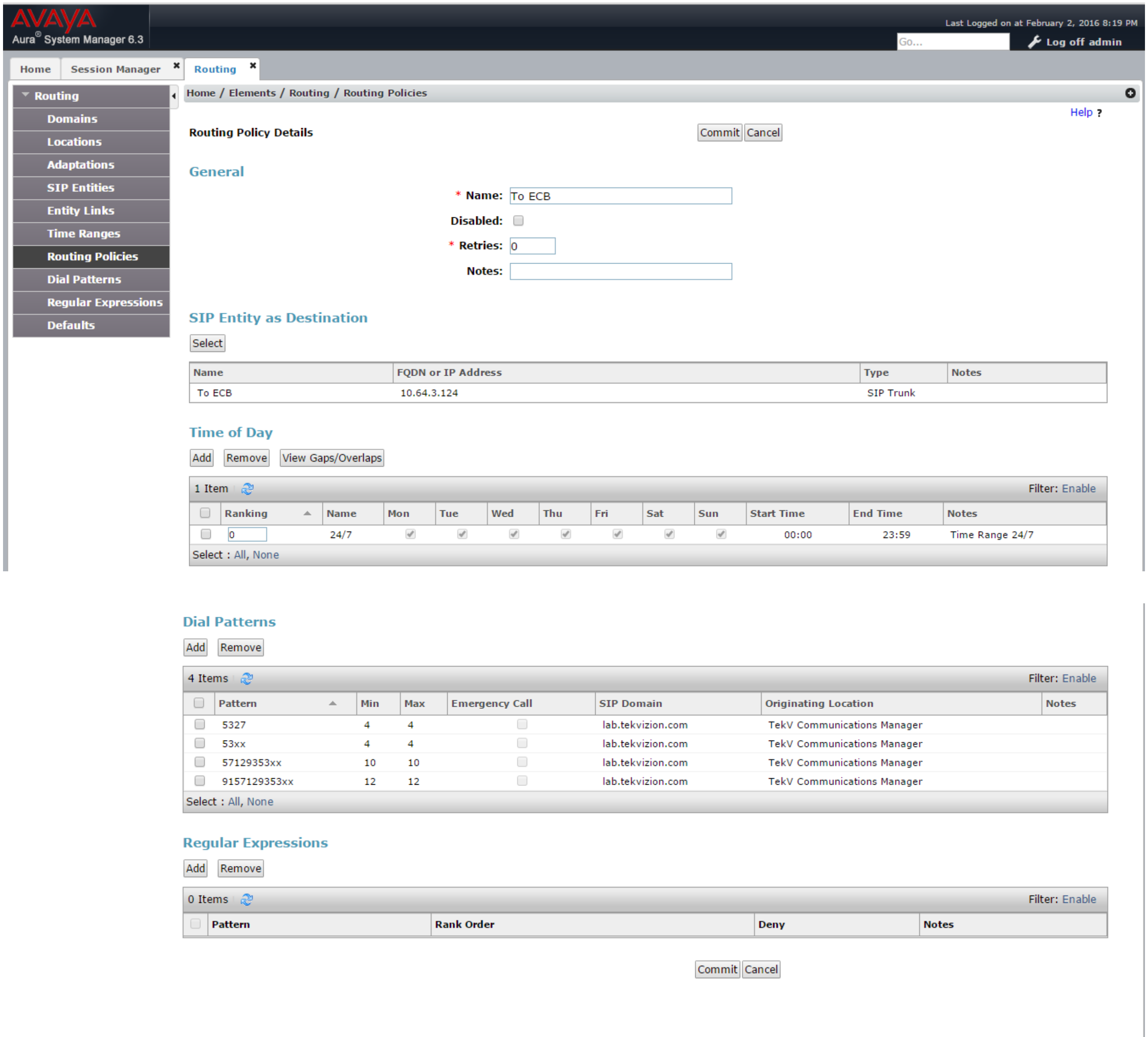
Configuring an Entity link between ECB and Session Manager

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.**



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 Avaya System Manager 6.3 is now configured to operate with the ECB.

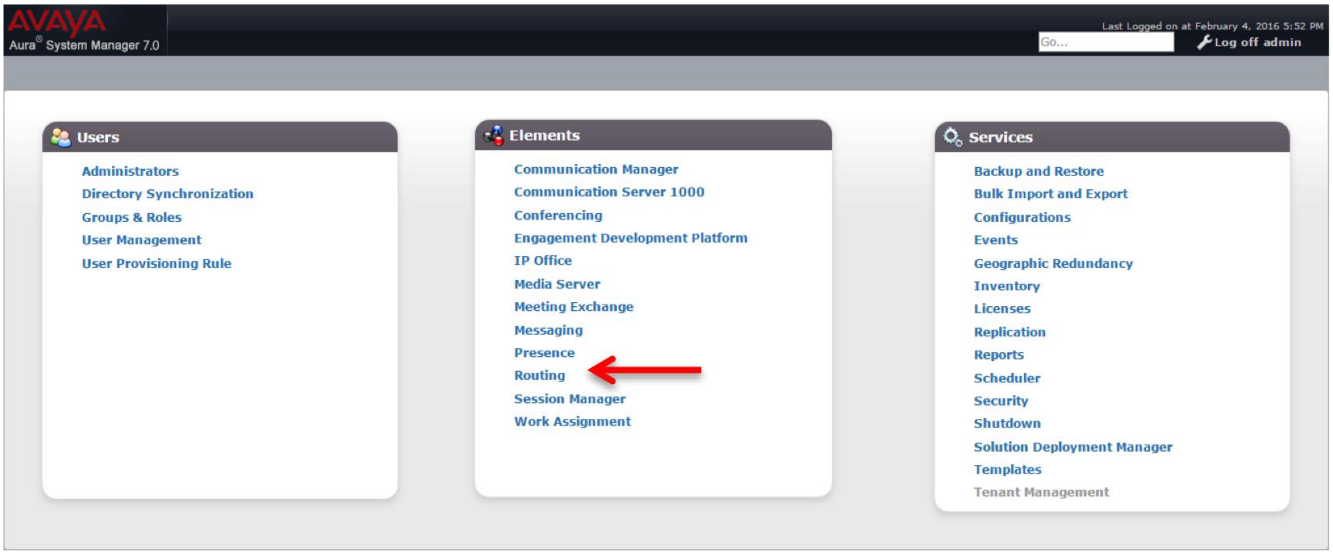
Phase 7 – Configuring the Avaya Session Manager 7.0

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

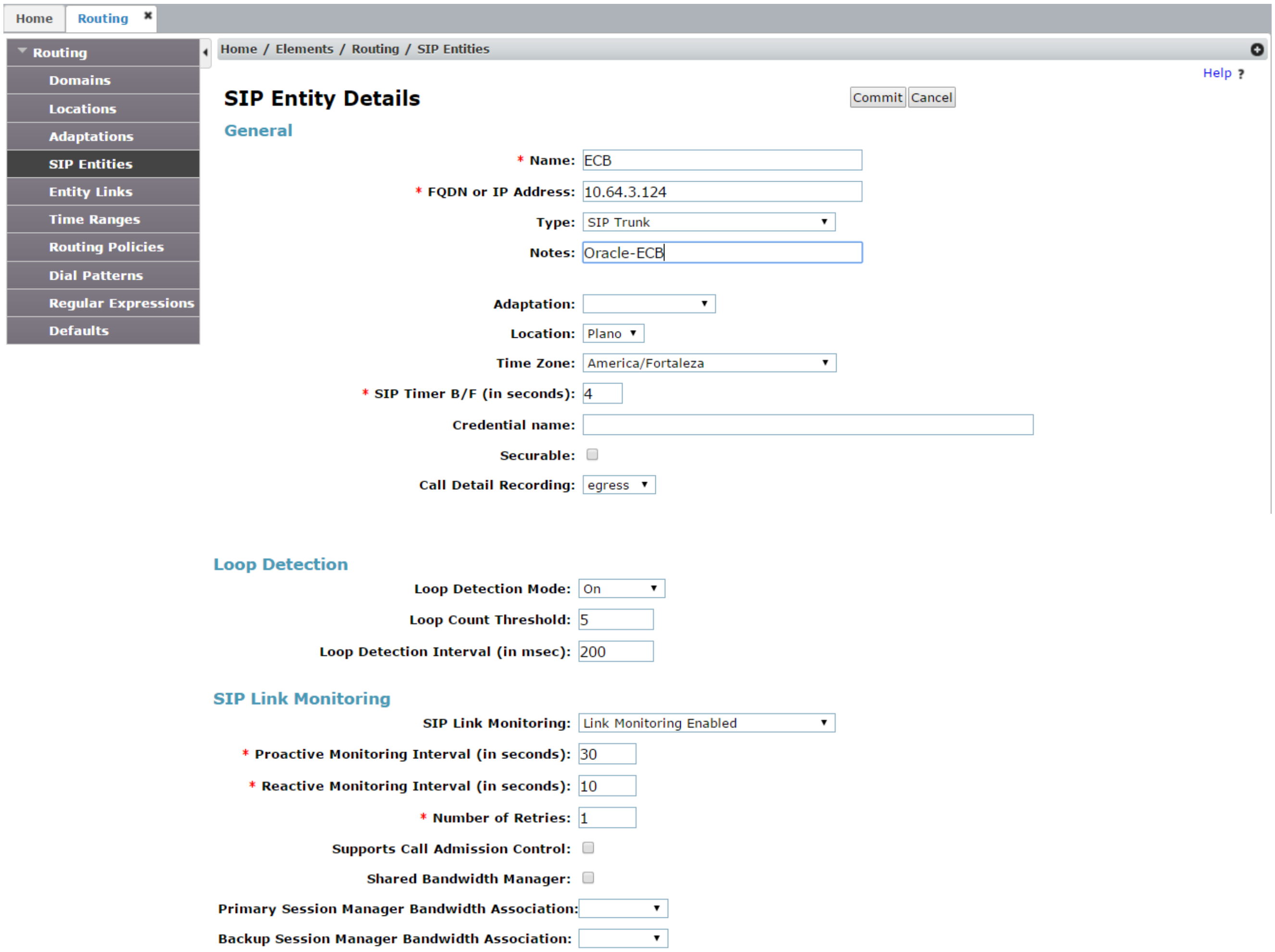
- Adding the ECB as a SIP Entity
- Configuring an Entity link between ECB and Session Manager
- Creating a Routing policy to assign the appropriate routing destination.

Adding the ECB as a SIP Entity

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

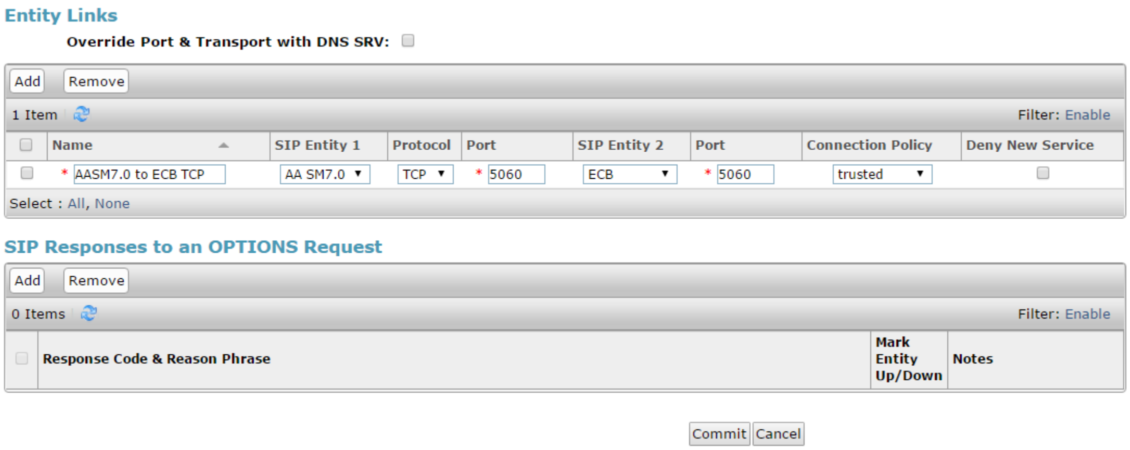


On the **Routing** tab, select **SIP Entities** from the menu on the left side of the screen. Click **New** to add ECB as a SIP entity as shown below and click **Commit**.



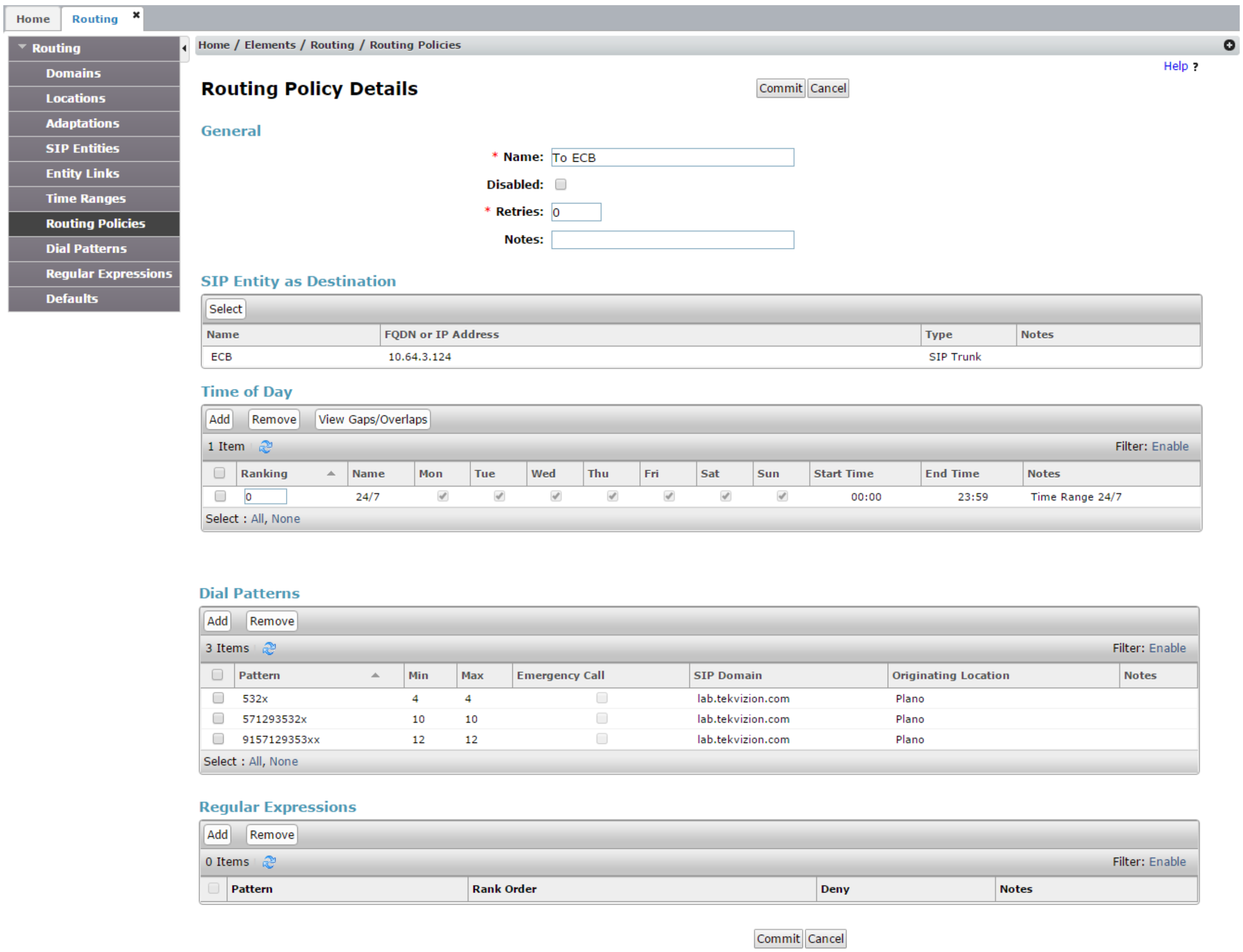
Configuring an Entity link between ECB and Session Manager

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.**



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 Avaya System Manager 7.0 is now configured to operate with the ECB.

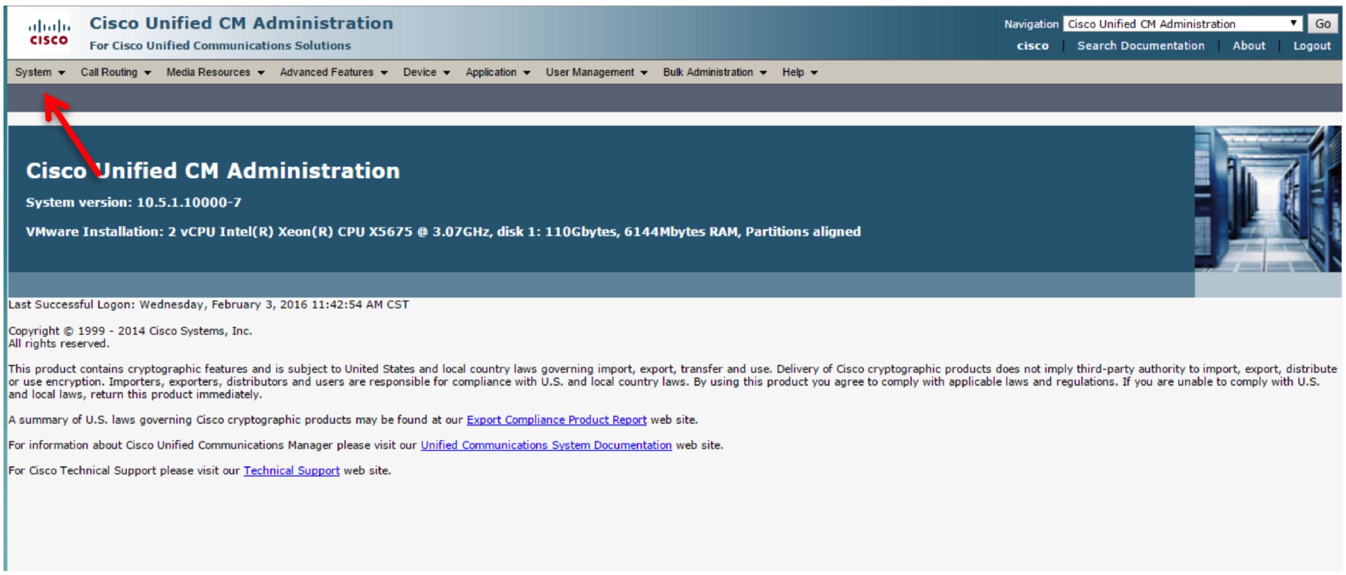
Phase 8 – Configuring Cisco Unified Communications Manager 10.5

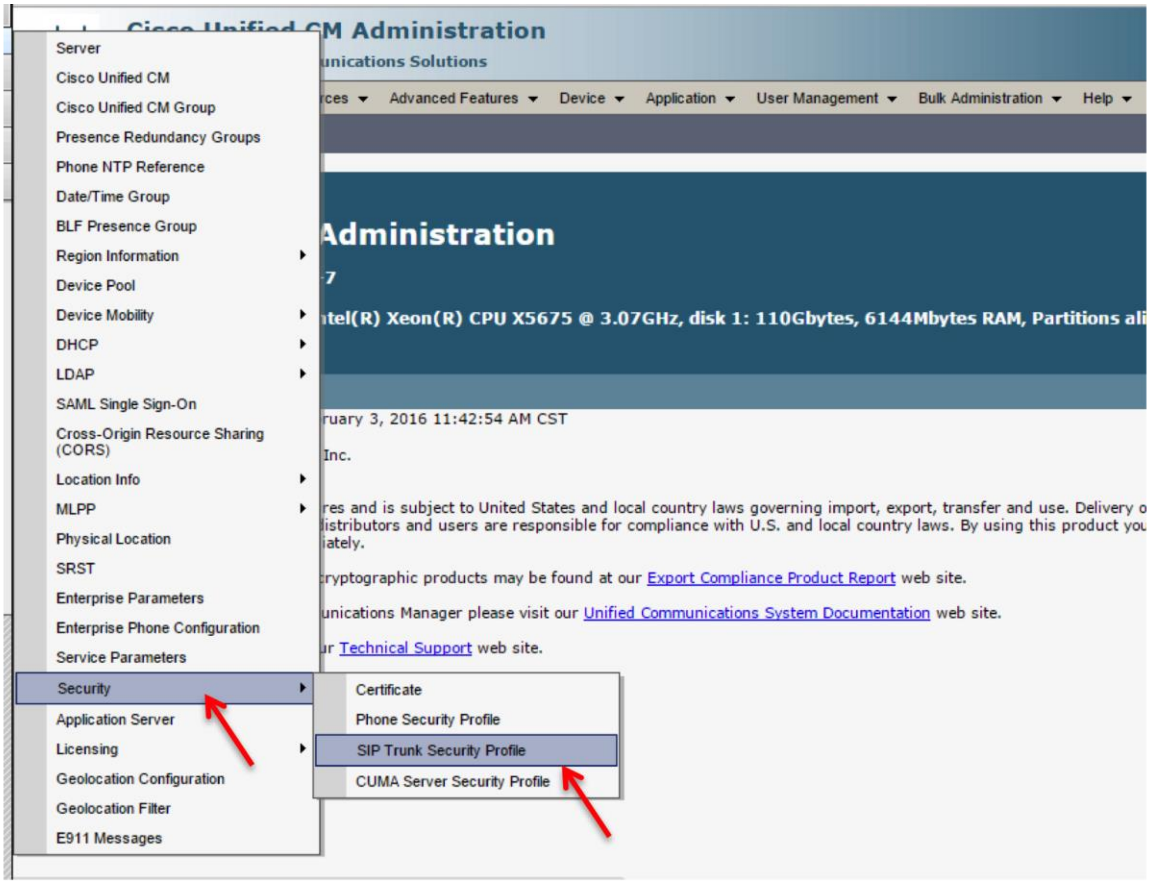
The enterprise will have a fully functioning Cisco Unified Communications Manager deployed. We will now configure it to operate with the ECB. This consists of the following steps

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

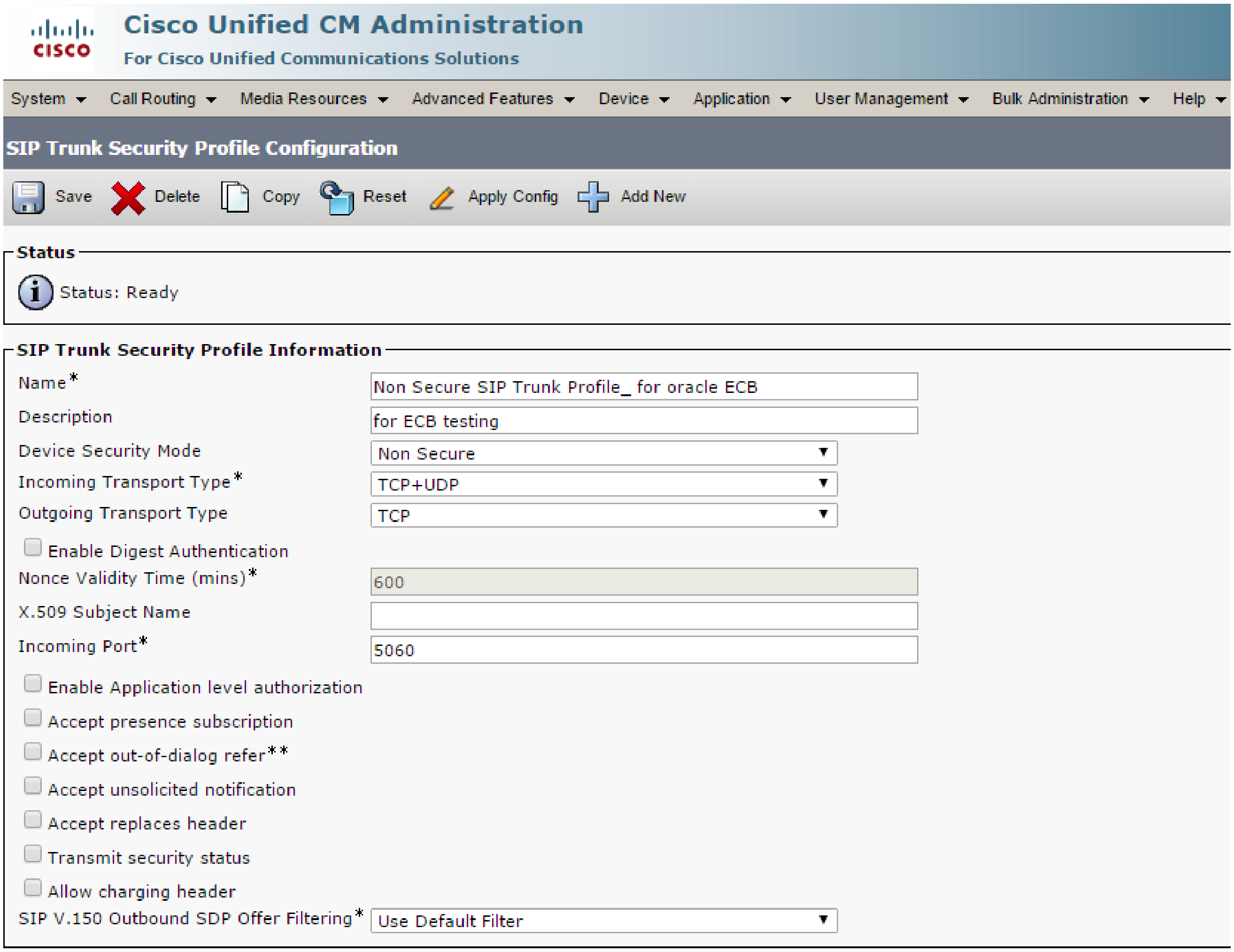
Configuring the SIP Trunk Security Profile

1. Log into the Cisco Unified CM administration page using https://server-ip/ and then click on **Cisco Unified Communications Manager** under **Installed Applications**.
2. To go to the **SIP trunk security profile** page, expand the **System** drop down menu, select **SIP Trunk Security Profile** under **Security**



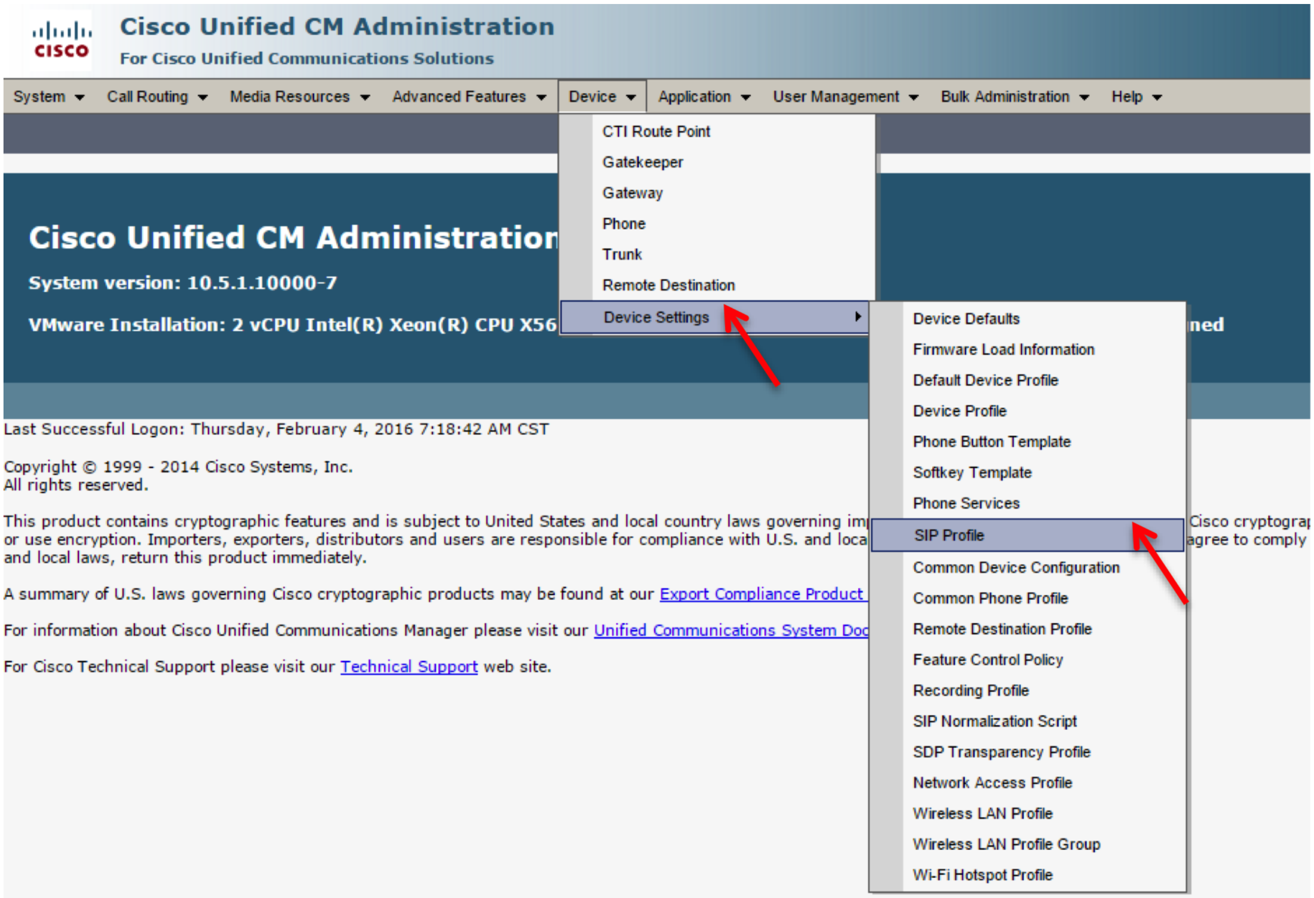

Cisco Unified CM Administration
unications Solutions
rces
Advanced Features
Device
Application
User Management
Bulk Administration
Help
Server
Cisco Unified CM
Cisco Unified CM Group
Presence Redundancy Groups
Phone NTP Reference
Date/Time Group
BLF Presence Group
Region Information
Device Pool
Device Mobility
DHCP
LDAP
SAML Single Sign-On
Cross-Origin Resource Sharing (CORS)
Location Info
MLPP
Physical Location
SRST
Enterprise Parameters
Enterprise Phone Configuration
Service Parameters
Security
Application Server
Licensing
Geolocation Configuration
Geolocation Filter
E911 Messages
Certificate
Phone Security Profile
SIP Trunk Security Profile
CUMA Server Security Profile
Administration
tel(R) Xeon(R) CPU X5675 @ 3.07GHz, disk 1: 110Gbytes, 6144Mbytes RAM, Partitions ali
ruary 3, 2016 11:42:54 AM CST
Inc.
res and is subject to United States and local country laws governing import, export, transfer and use. Delivery o
listributors and users are responsible for compliance with U.S. and local country laws. By using this product you
iately.
ryptographic products may be found at our Export Compliance Product Report web site.
unications Manager please visit our Unified Communications System Documentation web site.
r Technical Support web site.

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



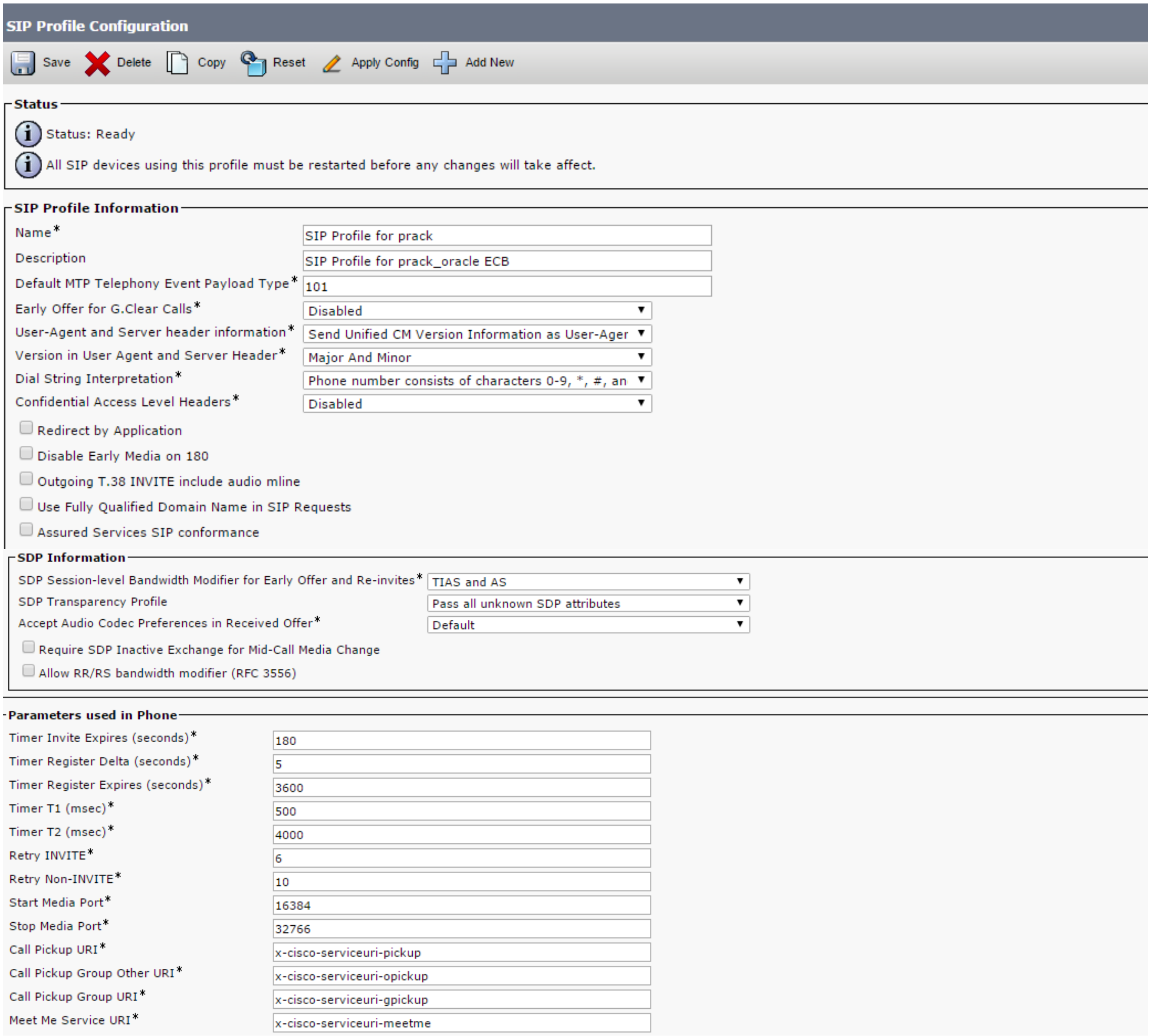
Configuring the SIP Profile

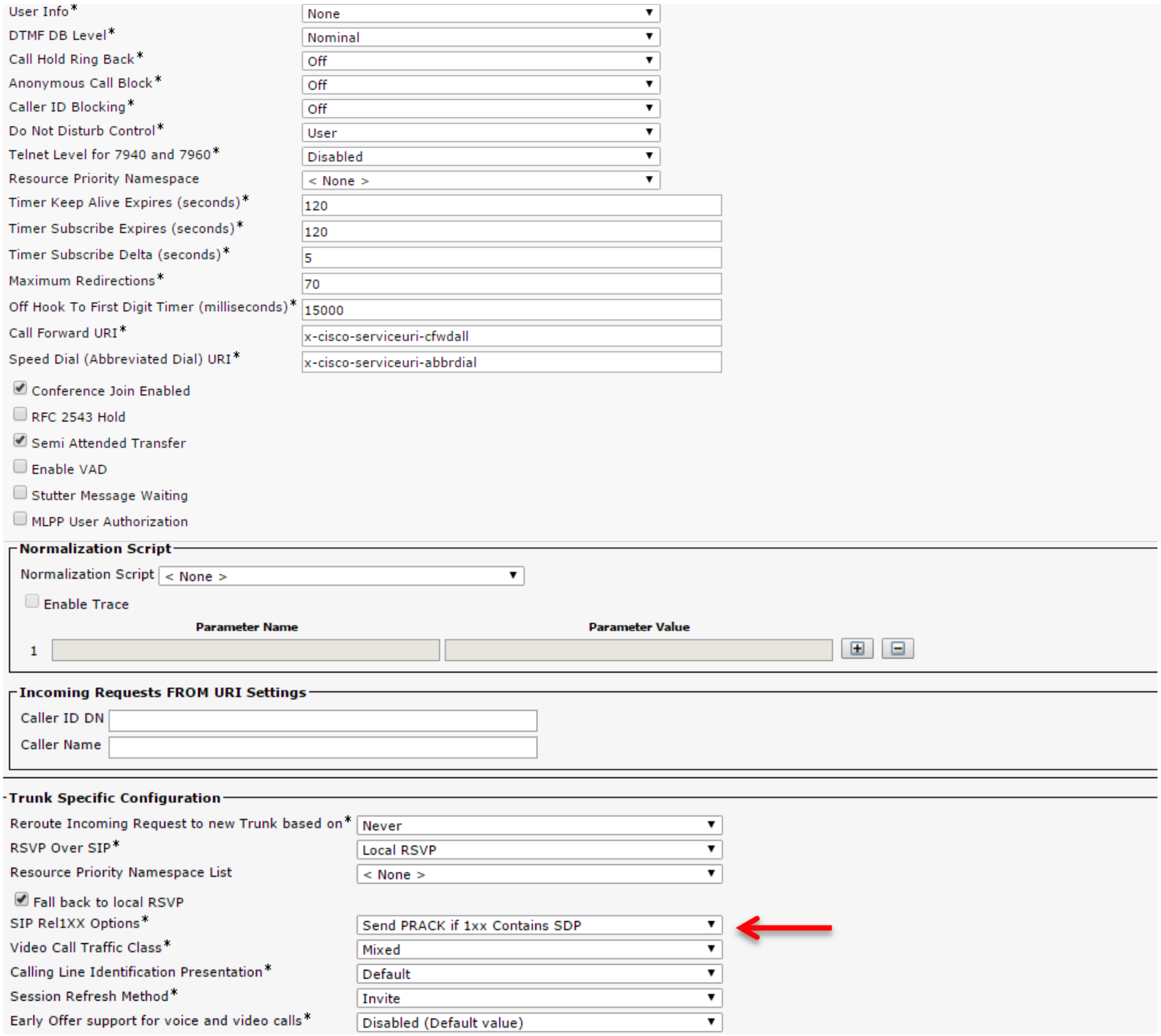
1. To go to the SIP Profile page, expand the **Device** drop down menu and select **SIP Profile** from **Device Settings**.



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.

3. Add a new SIP profile with the following settings. It is same as the default profile but includes PRACK support. Click **Save** when finished.





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
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

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

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

	Parameter Name	Parameter Value
1		

Incoming Requests FROM URI Settings

Caller ID DN

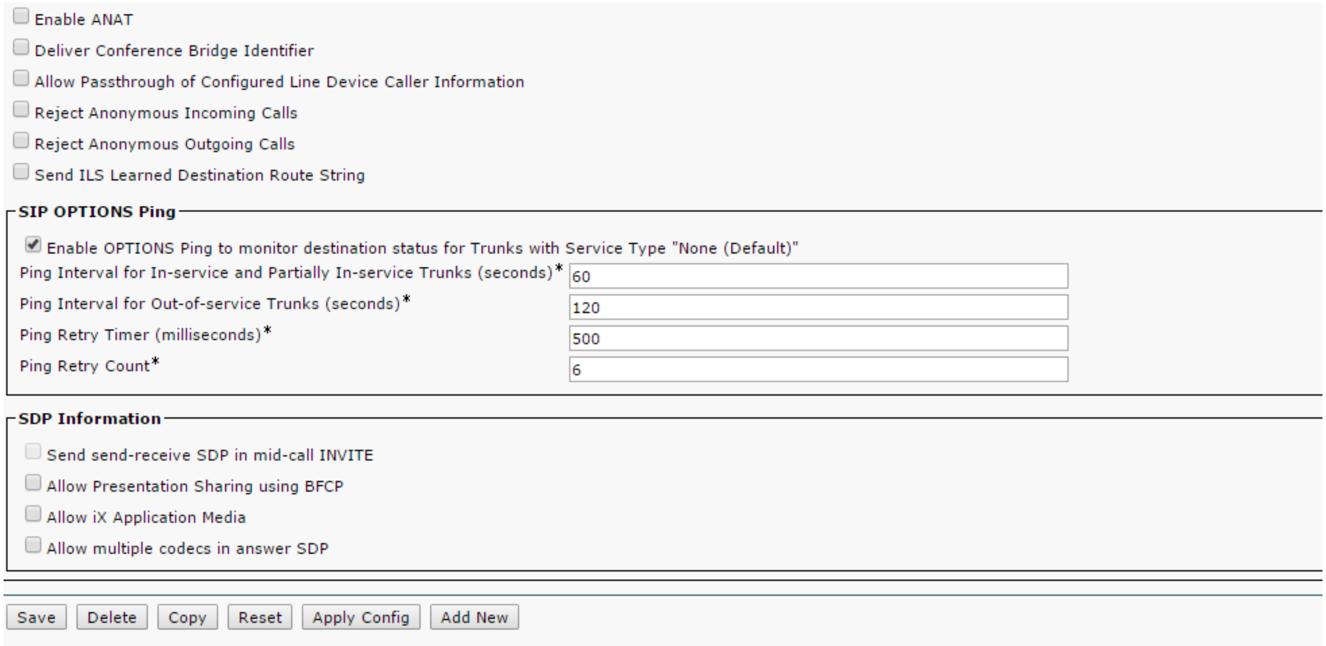
Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >

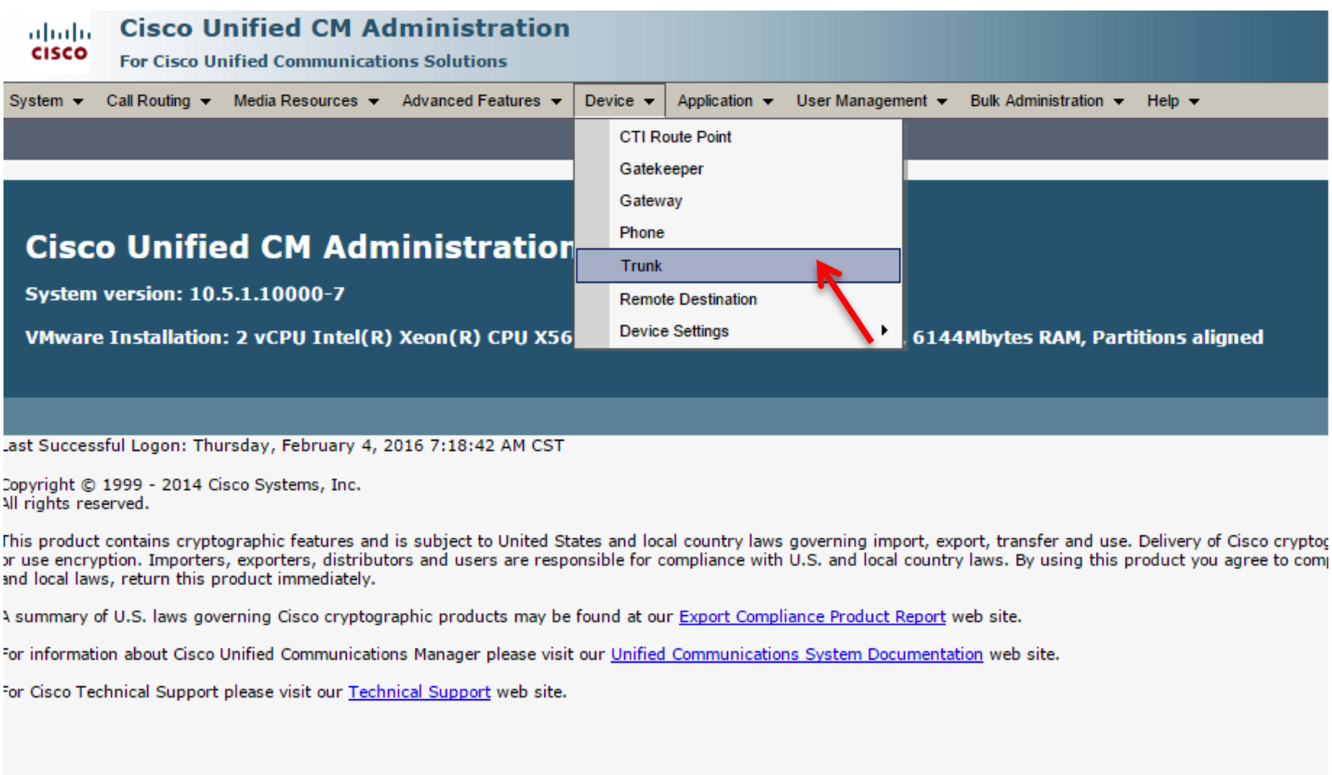
- [x] Fall back to local RSVP

SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Disabled (Default value)

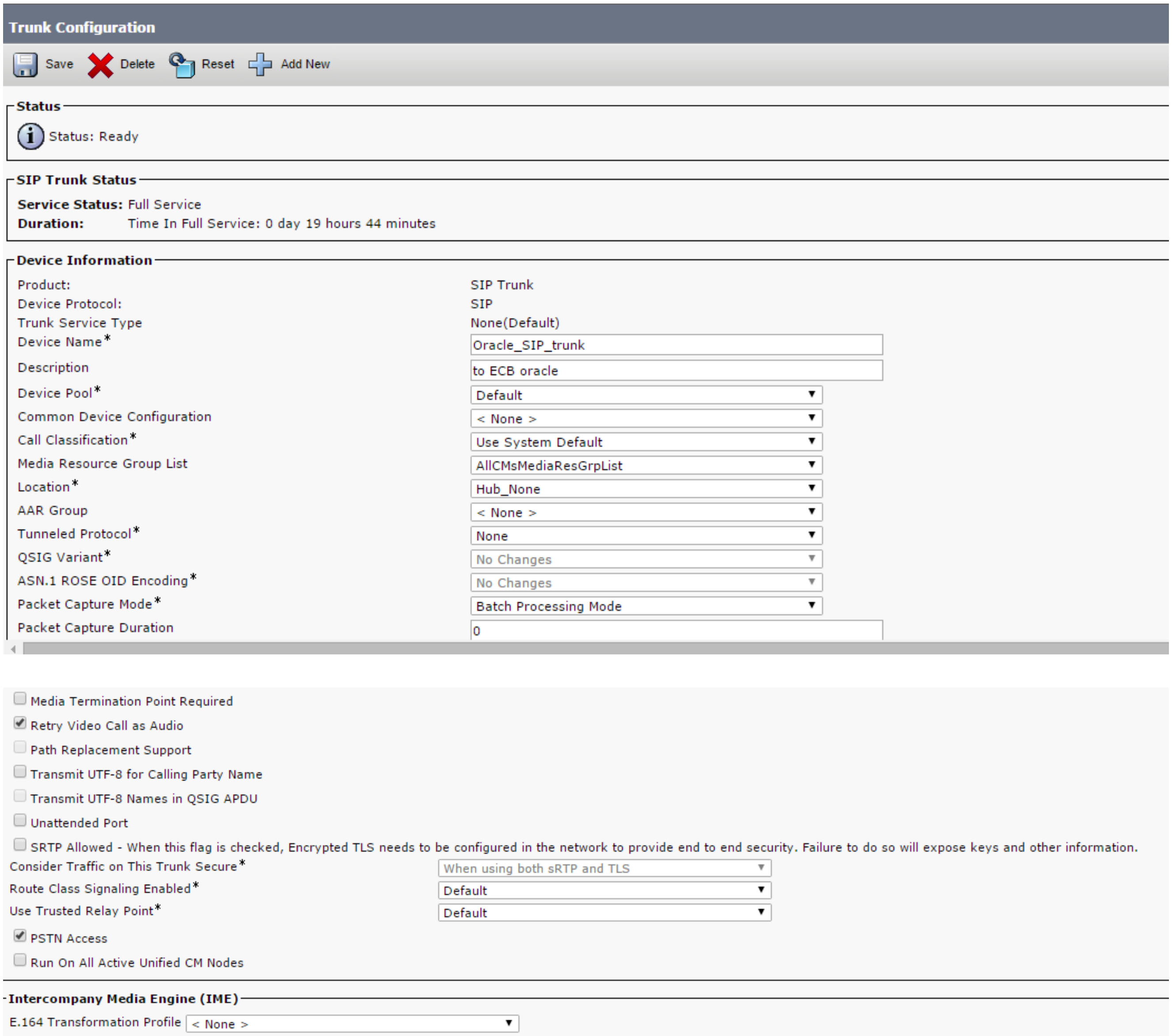


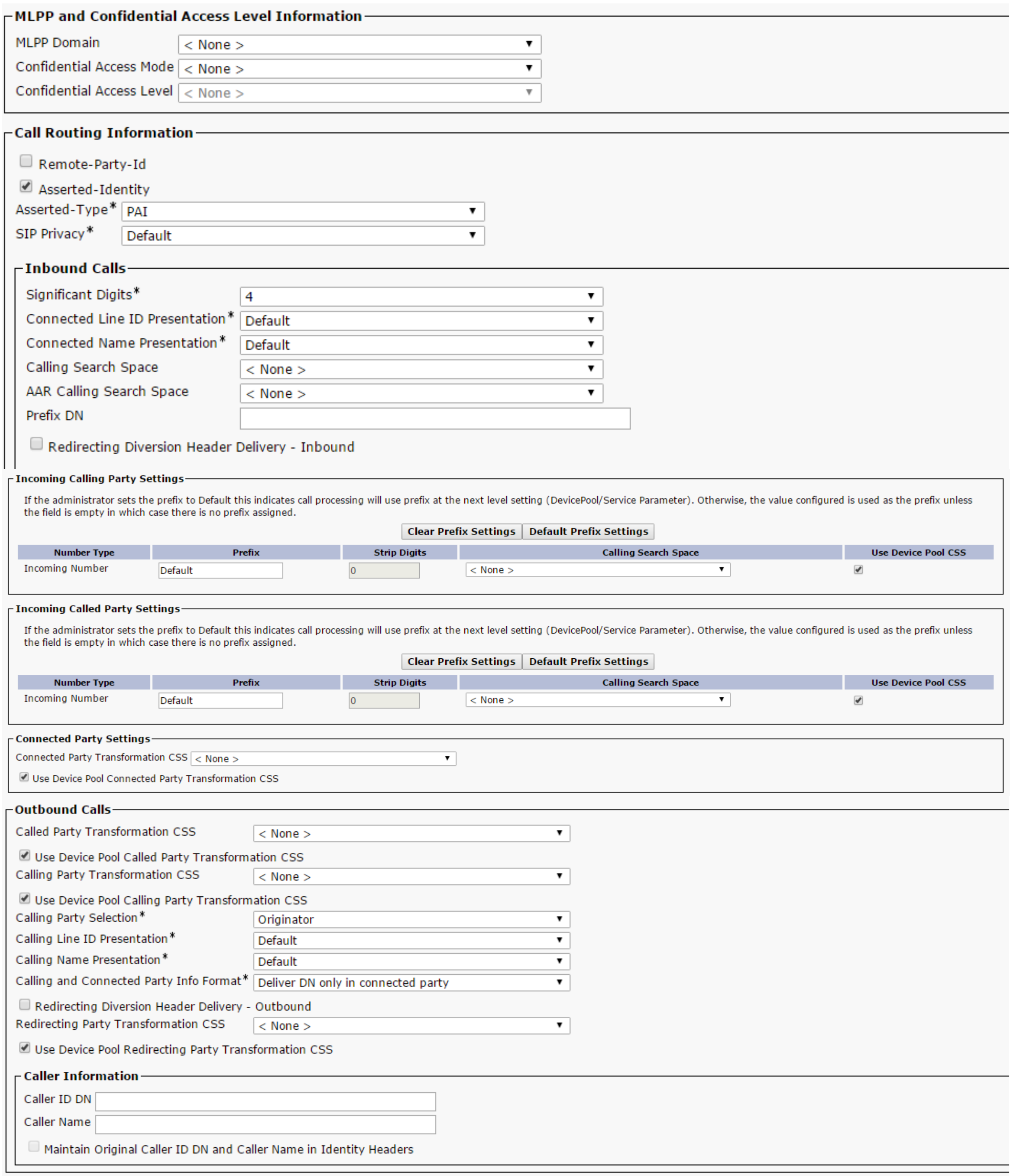
Configuring the Trunk

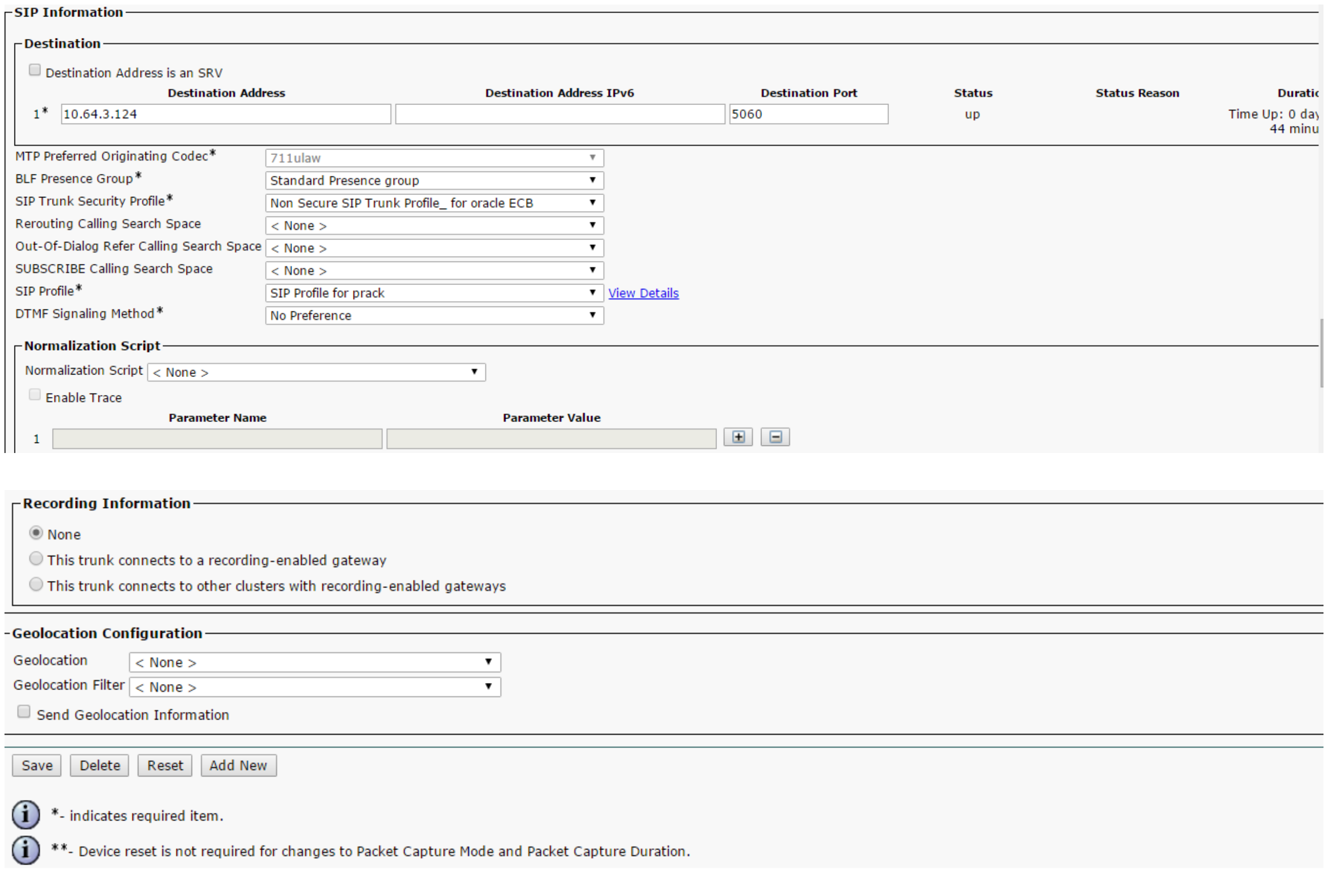
1. To go to the Trunks page, select **Trunk** from the **Device** drop down menu.



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

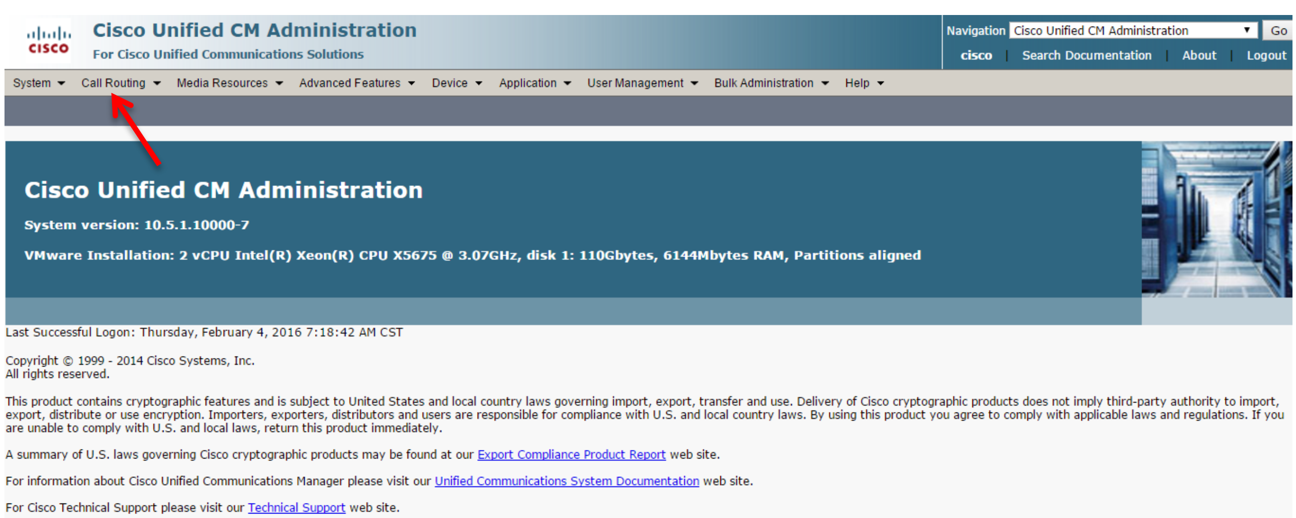


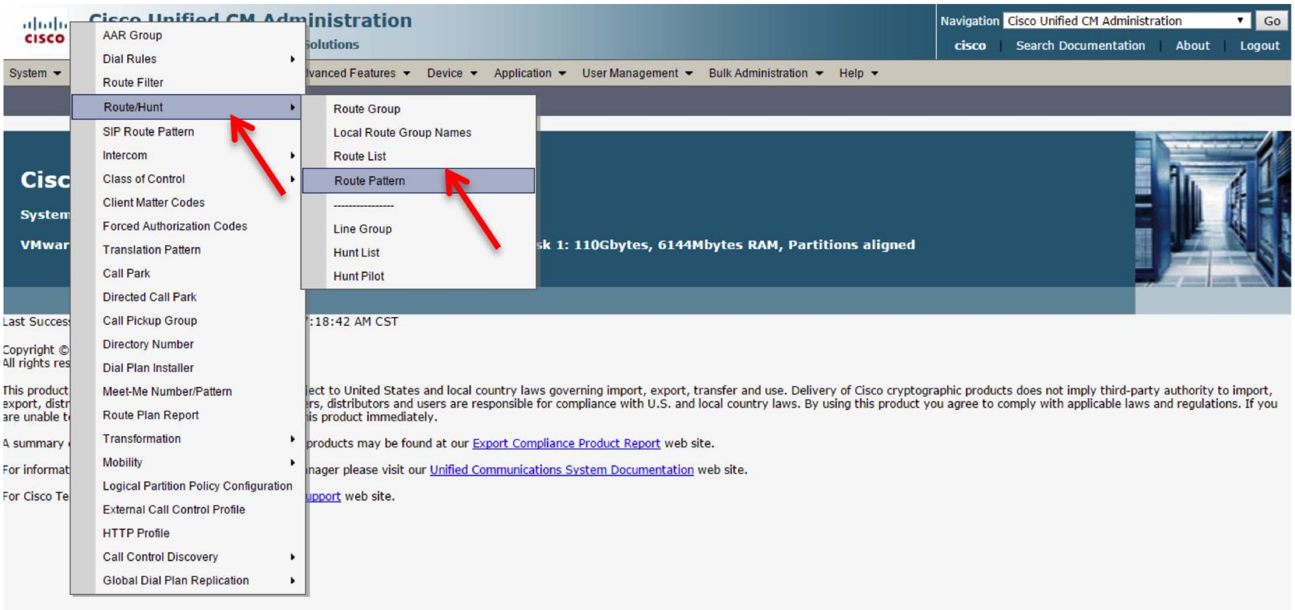

MLPP and Confidential Access Level Information
MLPP Domain
< None >
Confidential Access Mode
< None >
Confidential Access Level
< None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
PAI
SIP Privacy*
Default
Inbound Calls
Significant Digits*
4
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Calling Search Space
< None >
AAR Calling Search Space
< None >
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.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Incoming Called 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.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling and Connected Party Info Format*
Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
< None >
Use Device Pool Redirecting Party Transformation CSS
Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers



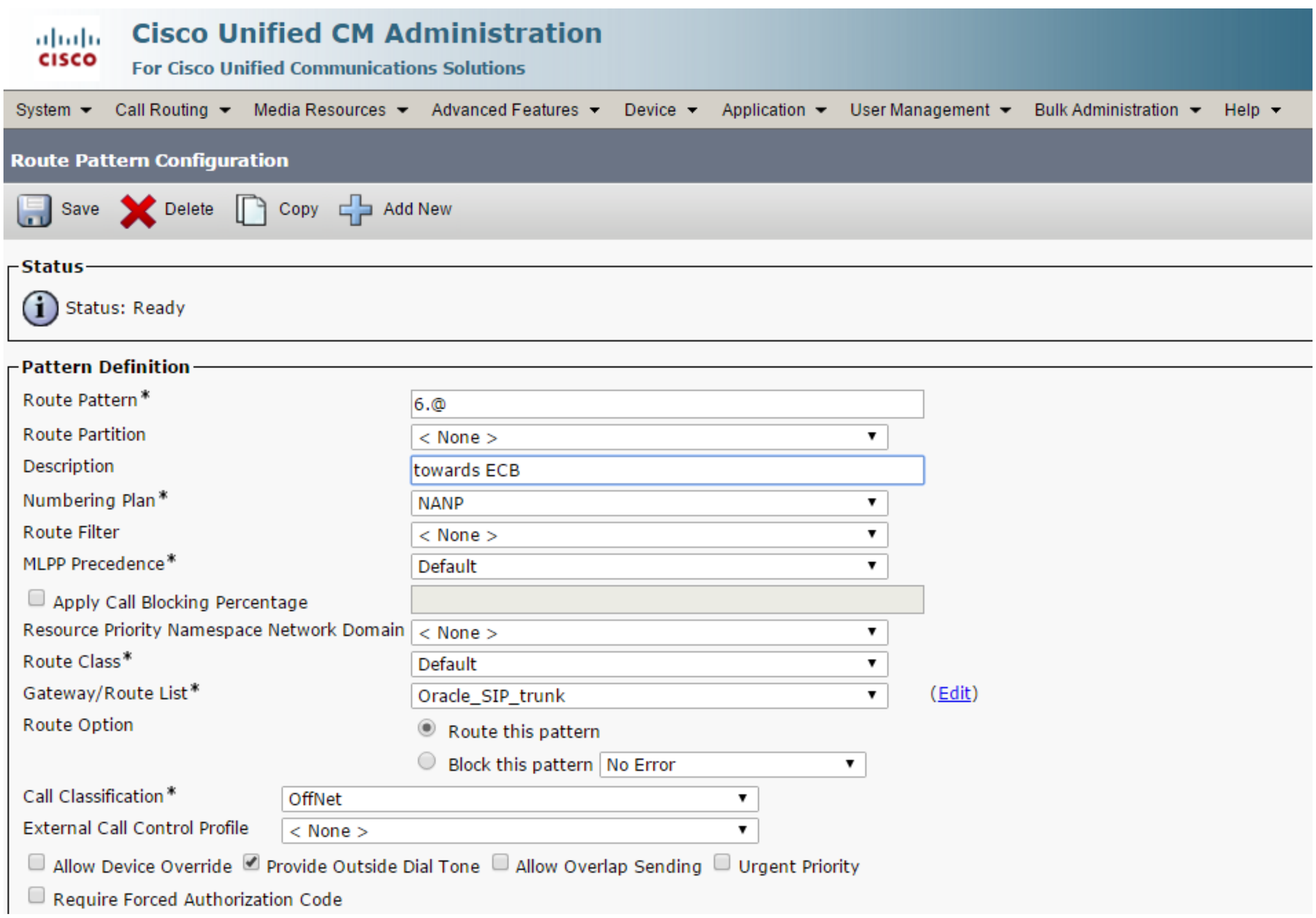
Configuring the Route Pattern

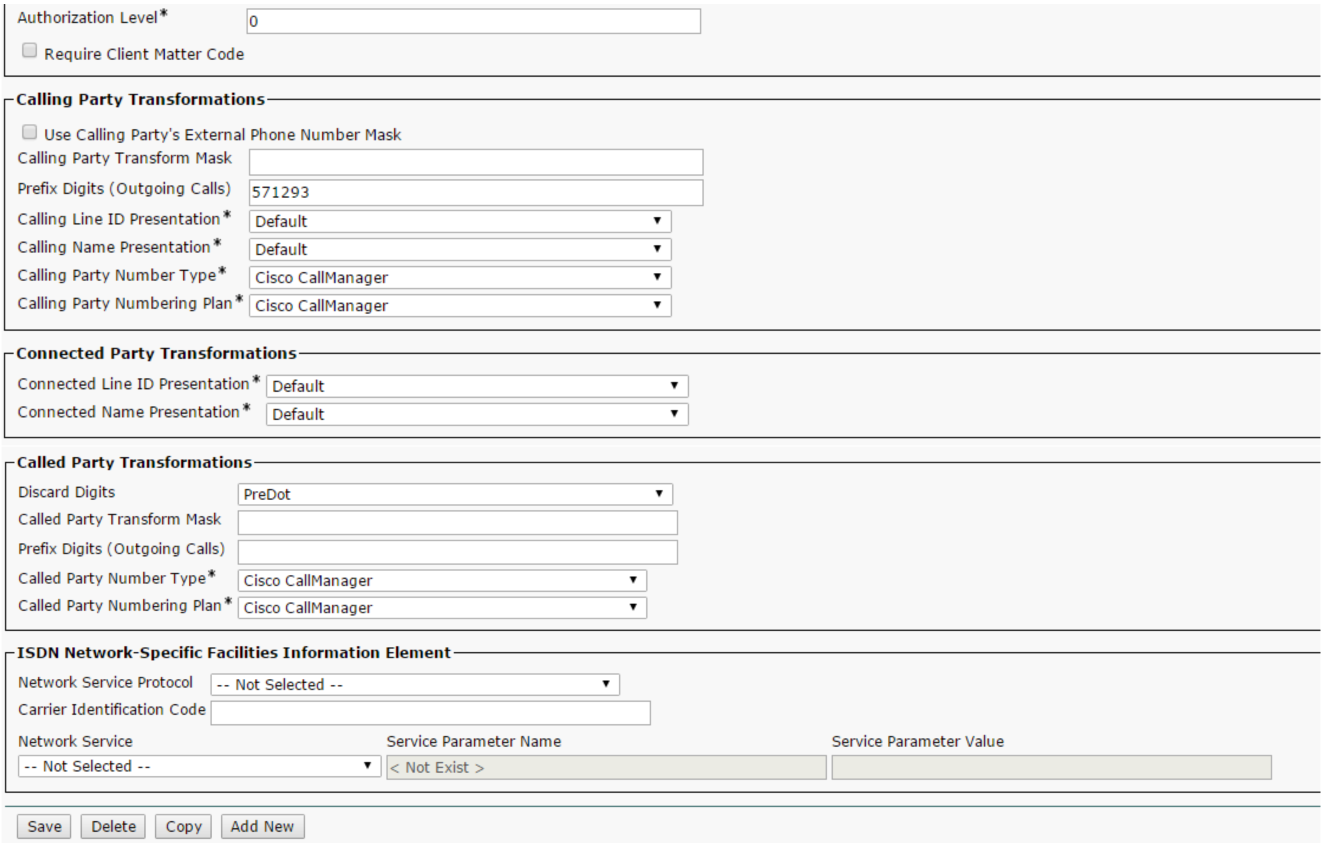
1. To go to the Route pattern page, click on **Call Routing** and select **Route Pattern** from the **Route/Hunt** drop down menu.





2. In our setup, users dial 6 to dial out. Add a route pattern with the following settings and associate it with the trunk configured in the previous step, then click **Save**.





The CUCM 10.5 configuration is now complete.

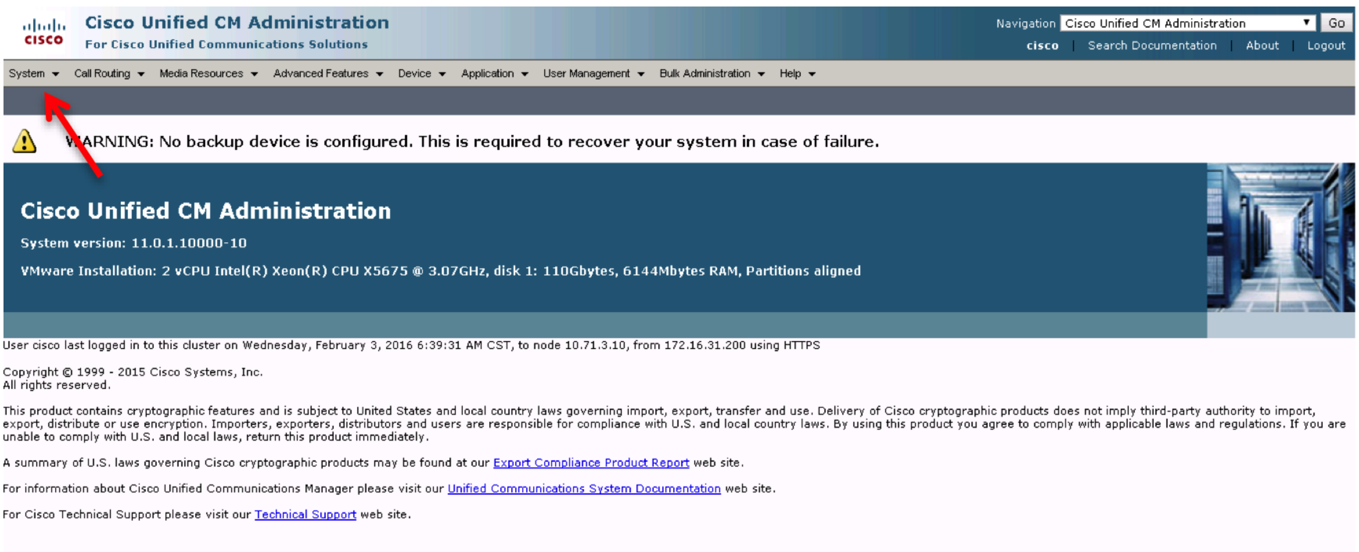
Phase 9 – Configuring Cisco Unified Communications Manager 11.0

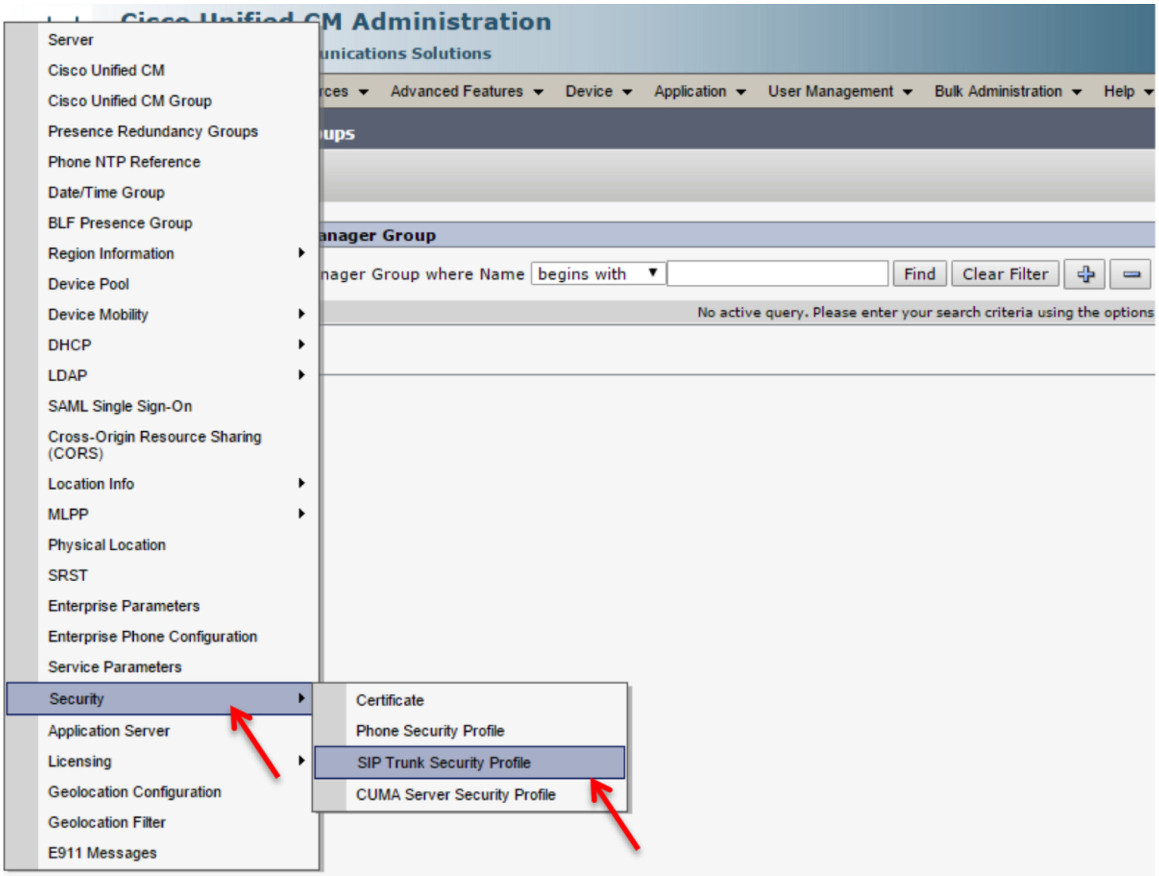
The enterprise will have a fully functioning Cisco Unified Communications Manager deployed. We will now configure it to operate with the ECB. This consists of the following steps

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

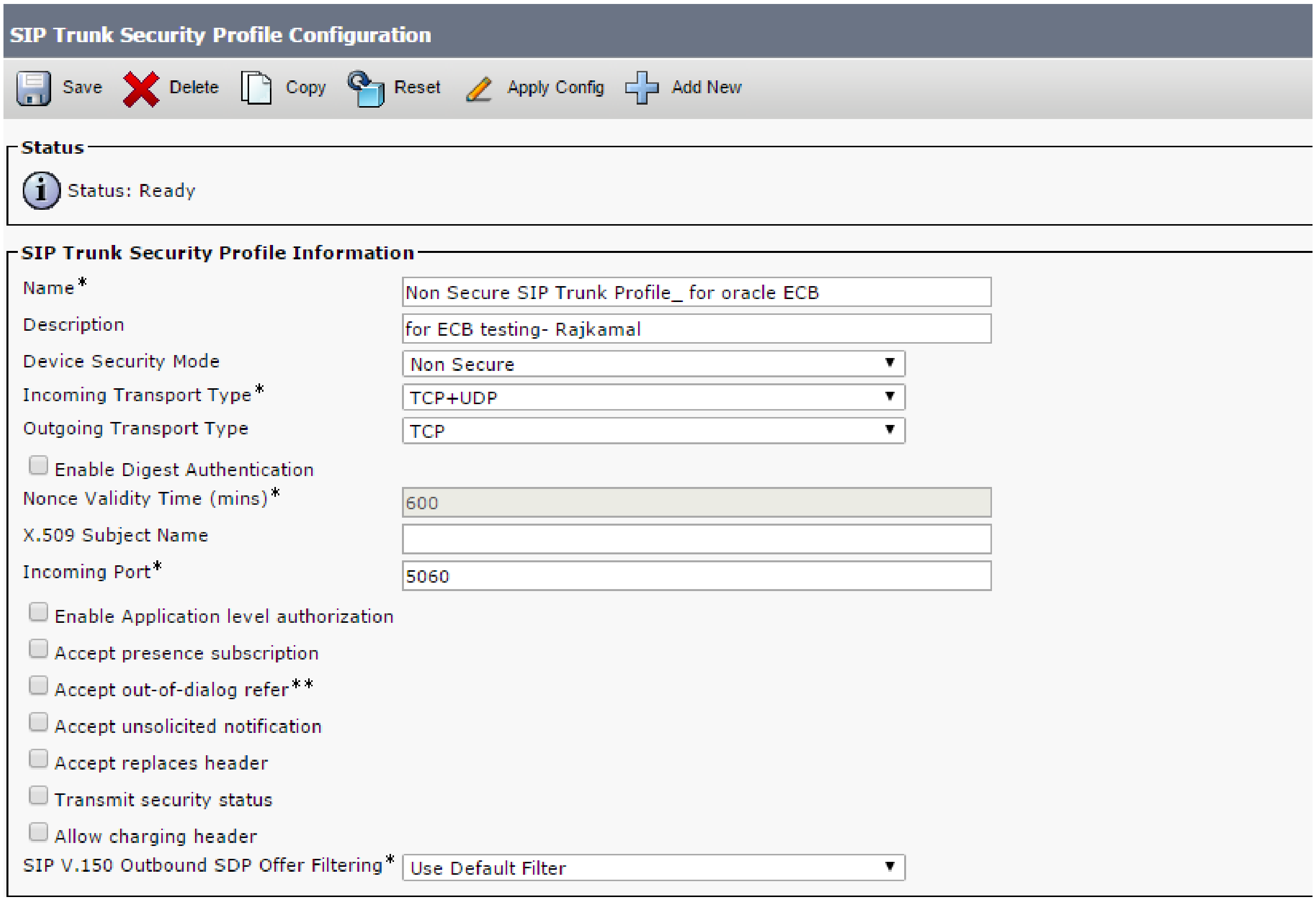
Configuring the SIP Trunk Security Profile

1. Log into the Cisco Unified CM administration page using https://server_ip/ and then click on Cisco Unified **Communications Manager** under **Installed Applications**.
2. To go to the **SIP trunk security profile** page, expand the **System** drop down menu, select **SIP Trunk Security Profile** under **Security**



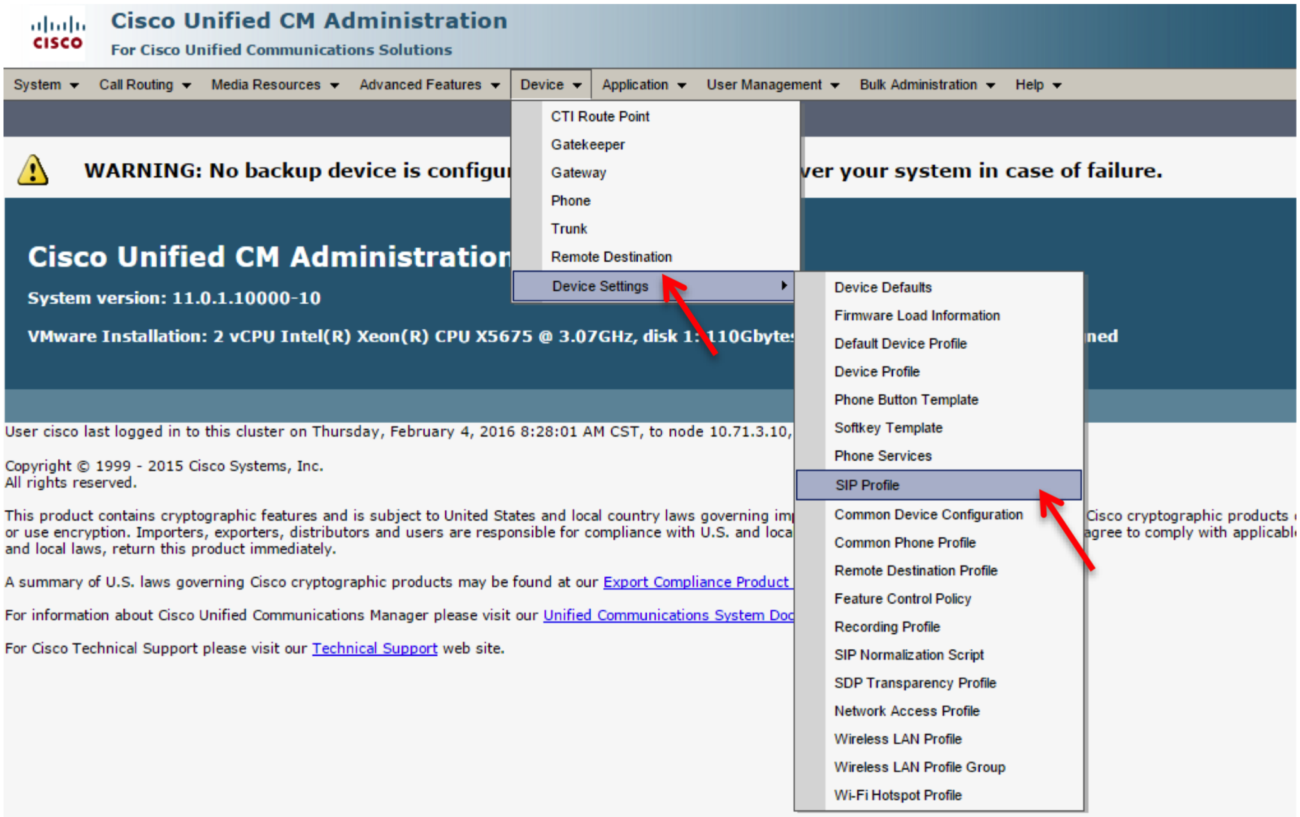

Cisco Unified CM Administration
unications Solutions
rces
Advanced Features
Device
Application
User Management
Bulk Administration
Help
ups
anager Group
nager Group where Name
begins with
Find
Clear Filter
No active query. Please enter your search criteria using the options
Server
Cisco Unified CM
Cisco Unified CM Group
Presence Redundancy Groups
Phone NTP Reference
Date/Time Group
BLF Presence Group
Region Information
Device Pool
Device Mobility
DHCP
LDAP
SAML Single Sign-On
Cross-Origin Resource Sharing (CORS)
Location Info
MLPP
Physical Location
SRST
Enterprise Parameters
Enterprise Phone Configuration
Service Parameters
Security
Application Server
Licensing
Geolocation Configuration
Geolocation Filter
E911 Messages
Certificate
Phone Security Profile
SIP Trunk Security Profile
CUMA Server Security Profile

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



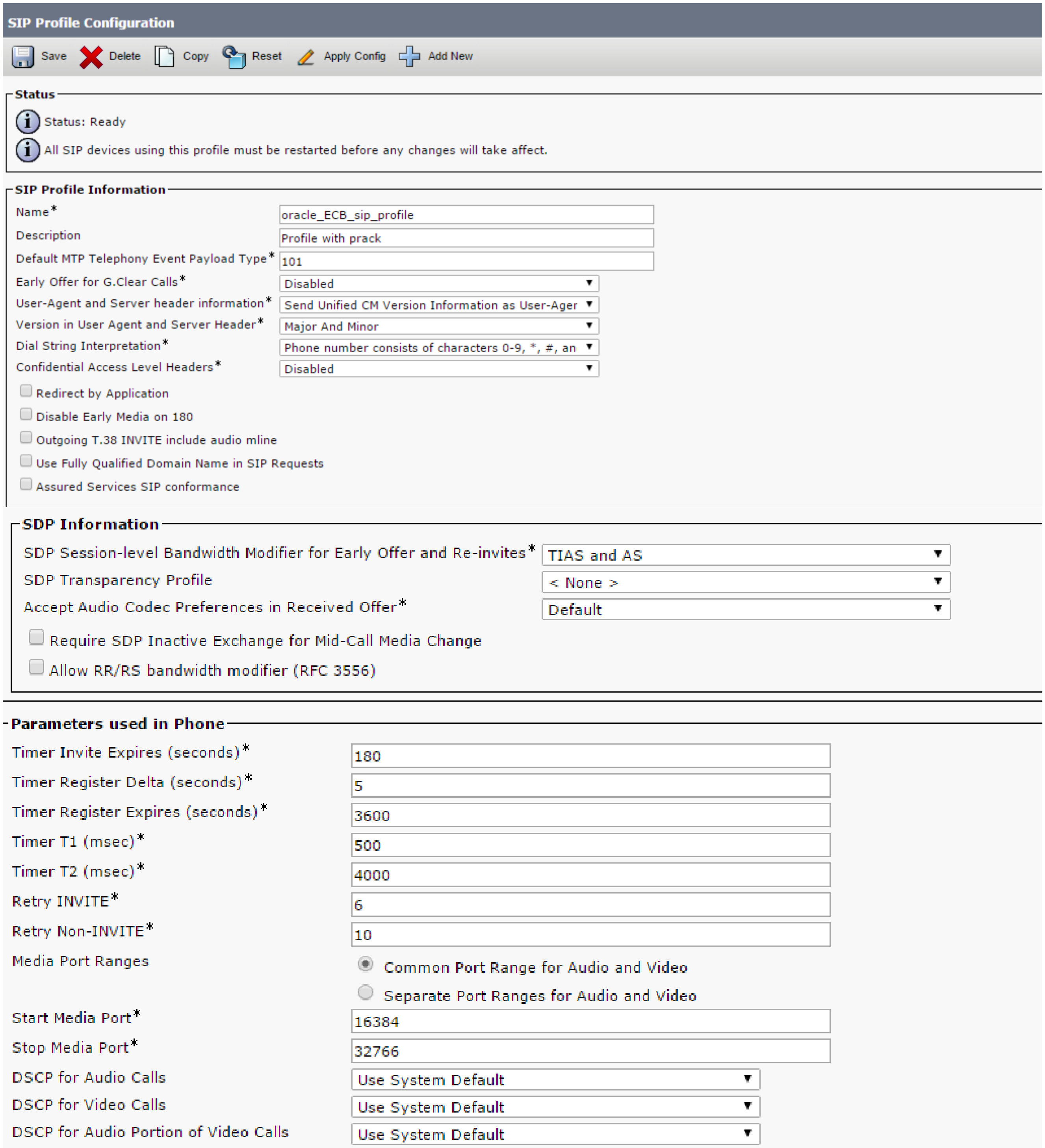
Configuring the SIP Profile

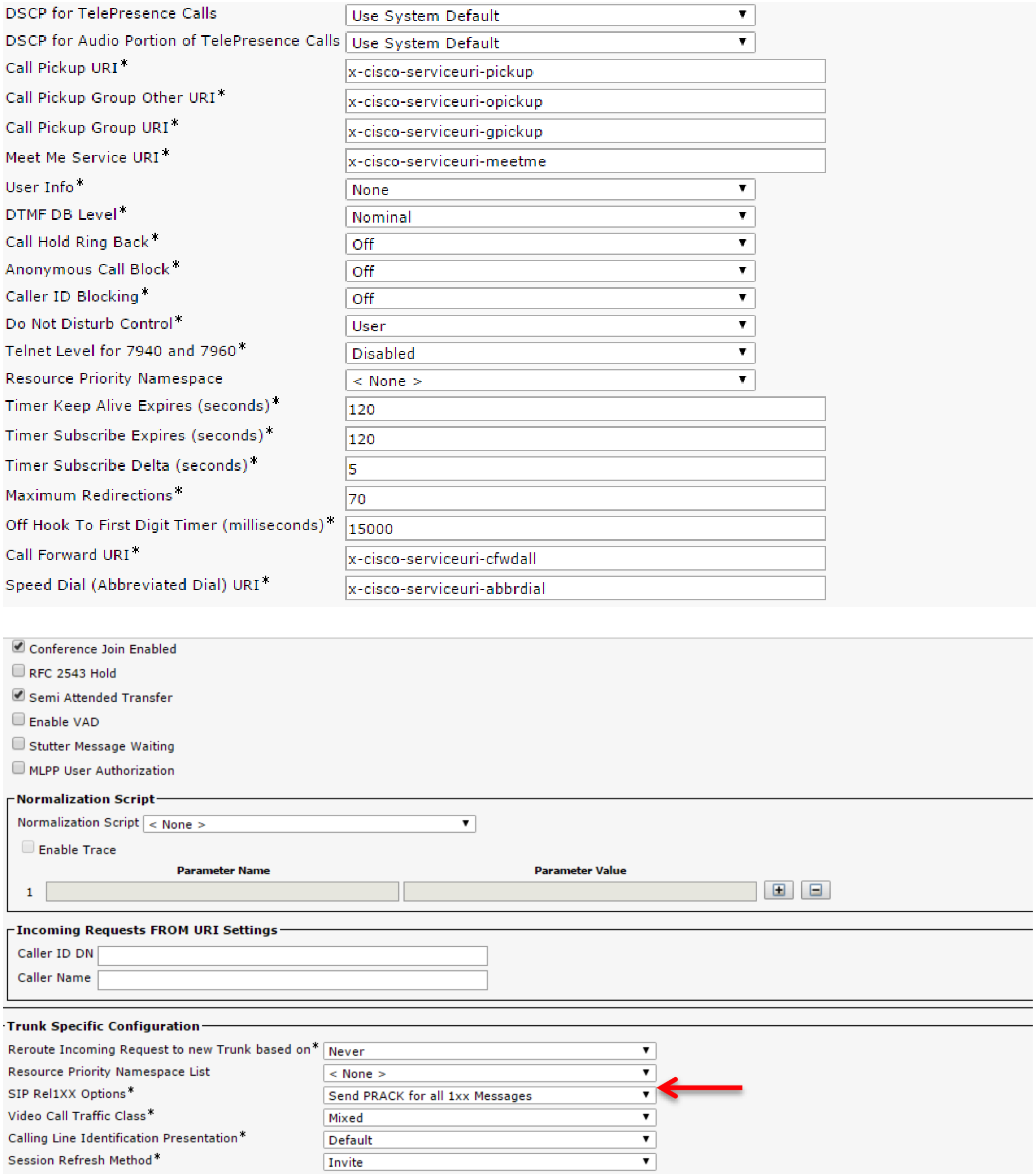
1. To go to the SIP Profile page, expand the **Device** drop down menu and select **SIP Profile** from **Device Settings**.

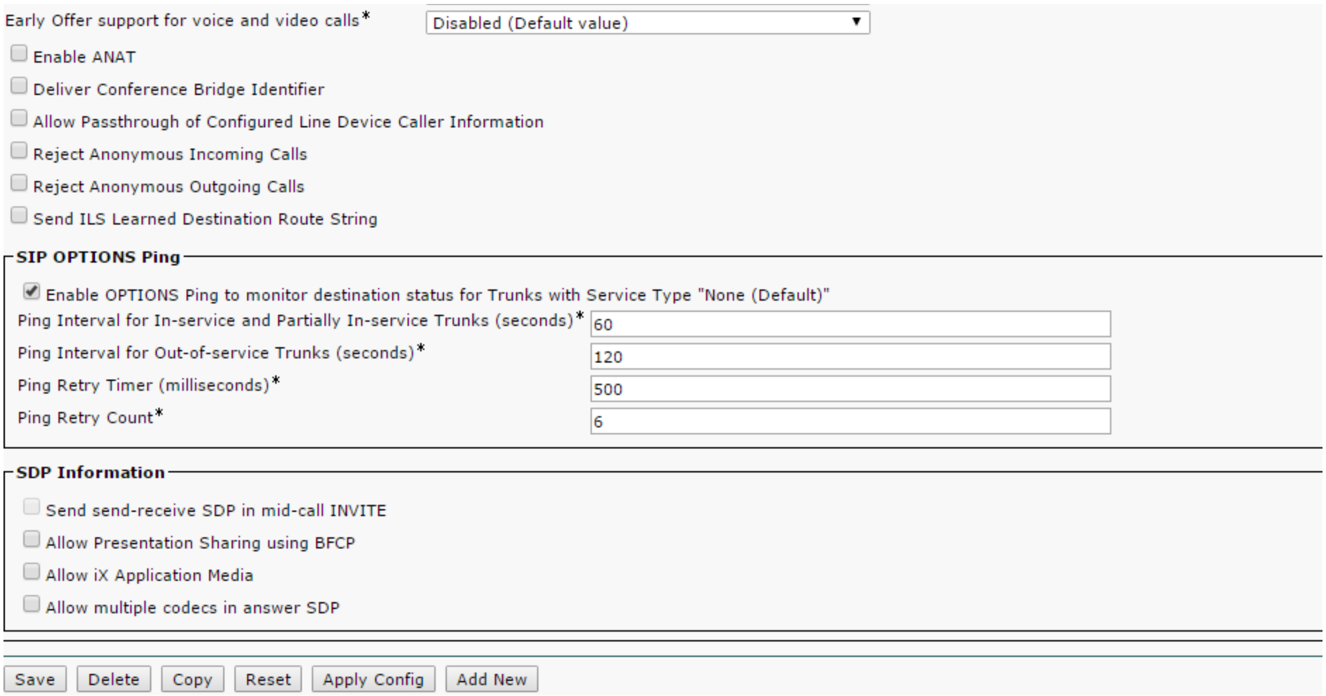


2. The **Find and List SIP Profiles** page will display the default SIP profile. Click on the **Copy** button to create a new SIP profile.

3. Add a new SIP profile with the following settings. It is same as the default profile but includes PRACK support. Click **Save** when finished.

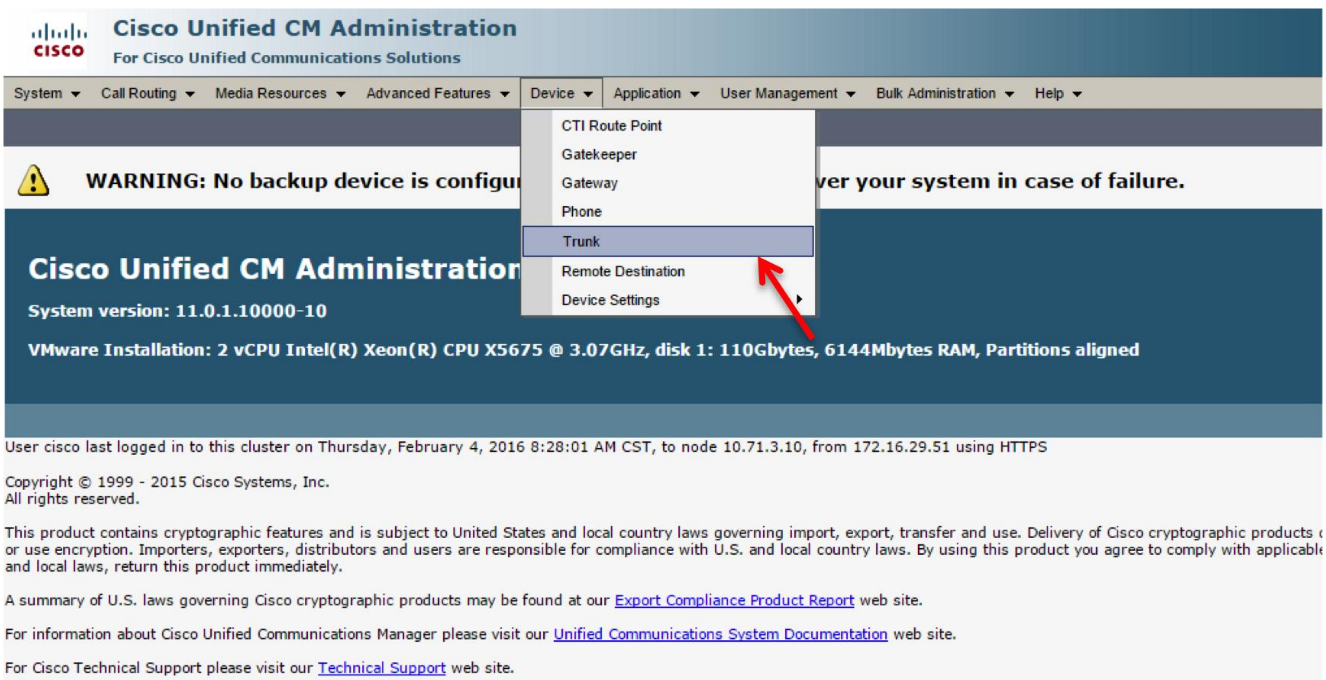



DSCP for TelePresence Calls
Use System Default
DSCP for Audio Portion of TelePresence Calls
Use System Default
Call Pickup URI*
x-cisco-serviceuri-pickup
Call Pickup Group Other URI*
x-cisco-serviceuri-opickup
Call Pickup Group URI*
x-cisco-serviceuri-gpickup
Meet Me Service URI*
x-cisco-serviceuri-meetme
User Info*
None
DTMF DB Level*
Nominal
Call Hold Ring Back*
Off
Anonymous Call Block*
Off
Caller ID Blocking*
Off
Do Not Disturb Control*
User
Telnet Level for 7940 and 7960*
Disabled
Resource Priority Namespace
< None >
Timer Keep Alive Expires (seconds)*
120
Timer Subscribe Expires (seconds)*
120
Timer Subscribe Delta (seconds)*
5
Maximum Redirections*
70
Off Hook To First Digit Timer (milliseconds)*
15000
Call Forward URI*
x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*
x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
Stutter Message Waiting
MLPP User Authorization
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1
Incoming Requests FROM URI Settings
Caller ID DN
Caller Name
Trunk Specific Configuration
Reroute Incoming Request to new Trunk based on*
Never
Resource Priority Namespace List
< None >
SIP Rel1XX Options*
Send PRACK for all 1xx Messages
Video Call Traffic Class*
Mixed
Calling Line Identification Presentation*
Default
Session Refresh Method*
Invite

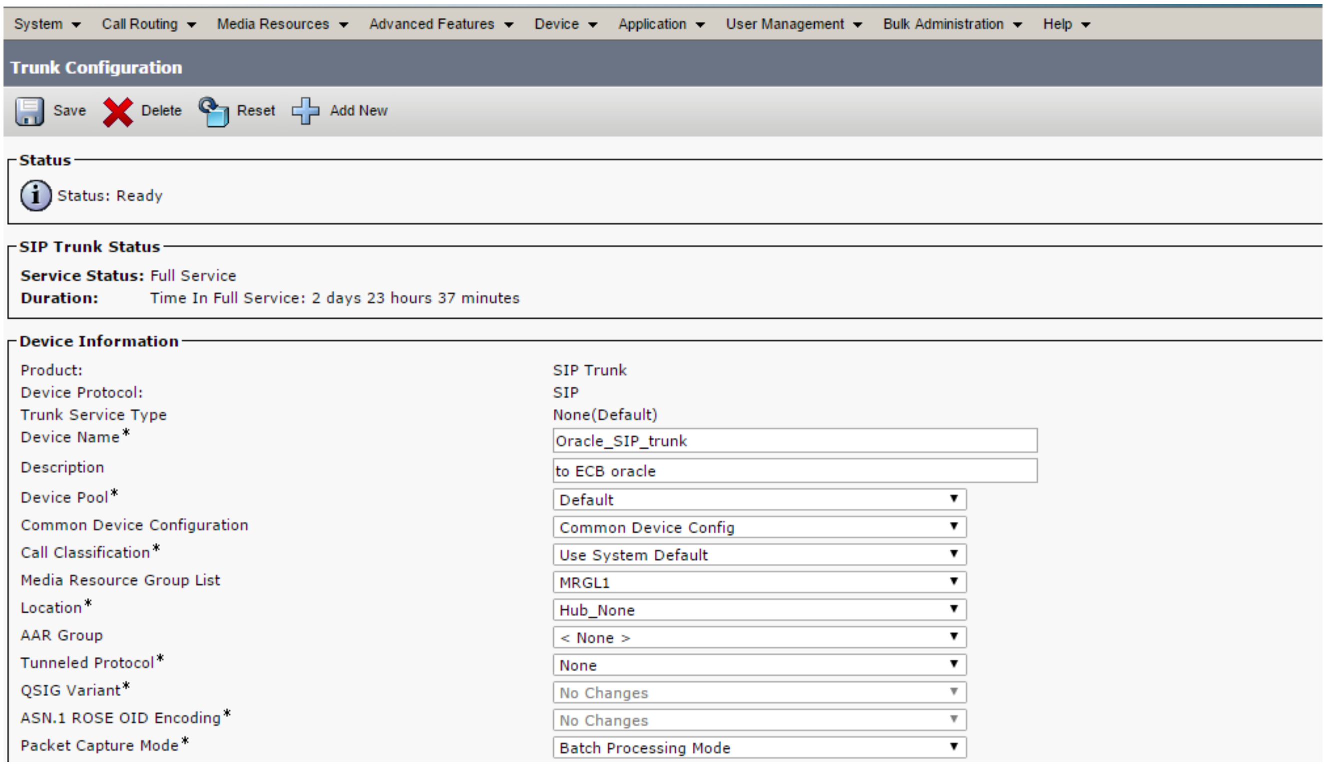


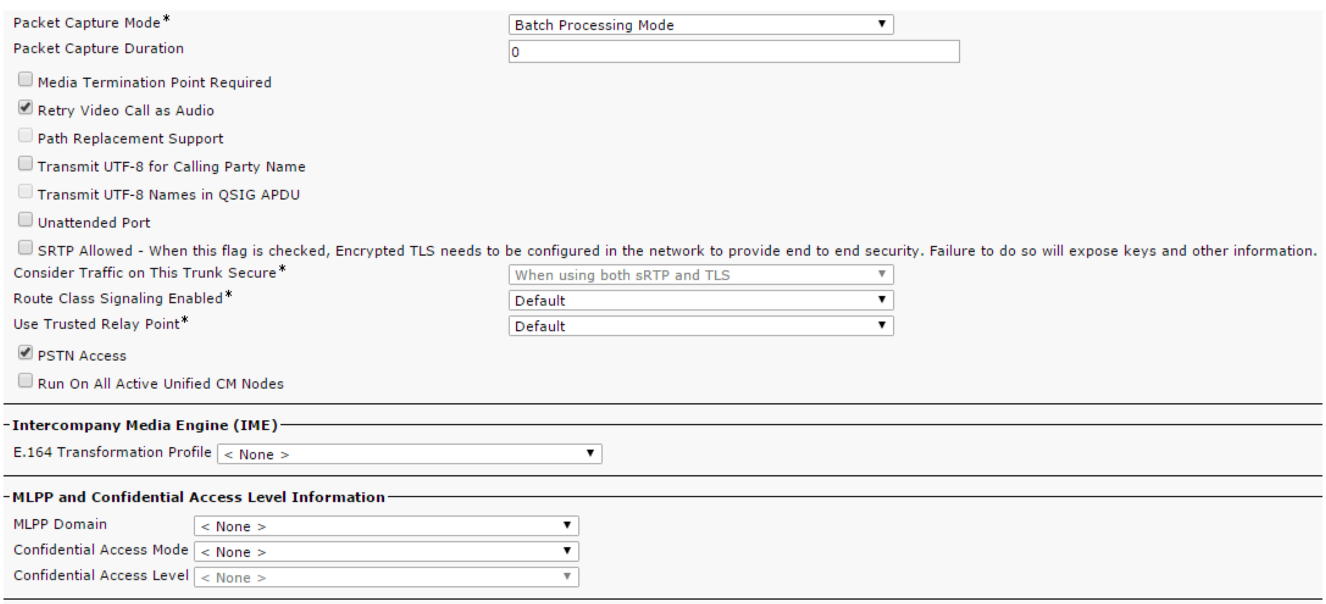
Configuring the Trunk

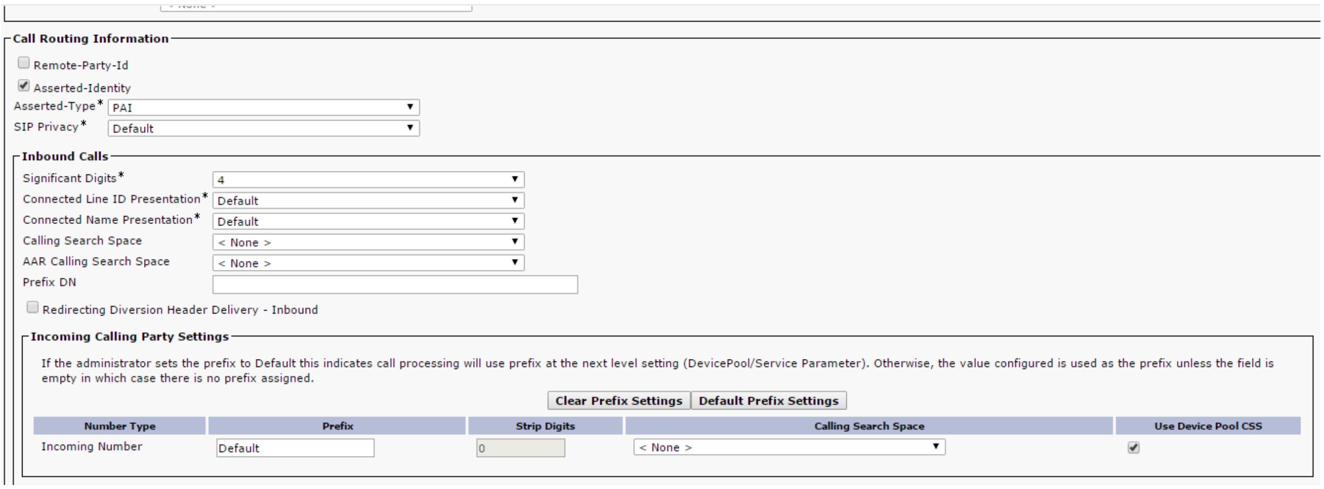
1. To go to the Trunks page, select **Trunk** from the **Device** drop down menu.

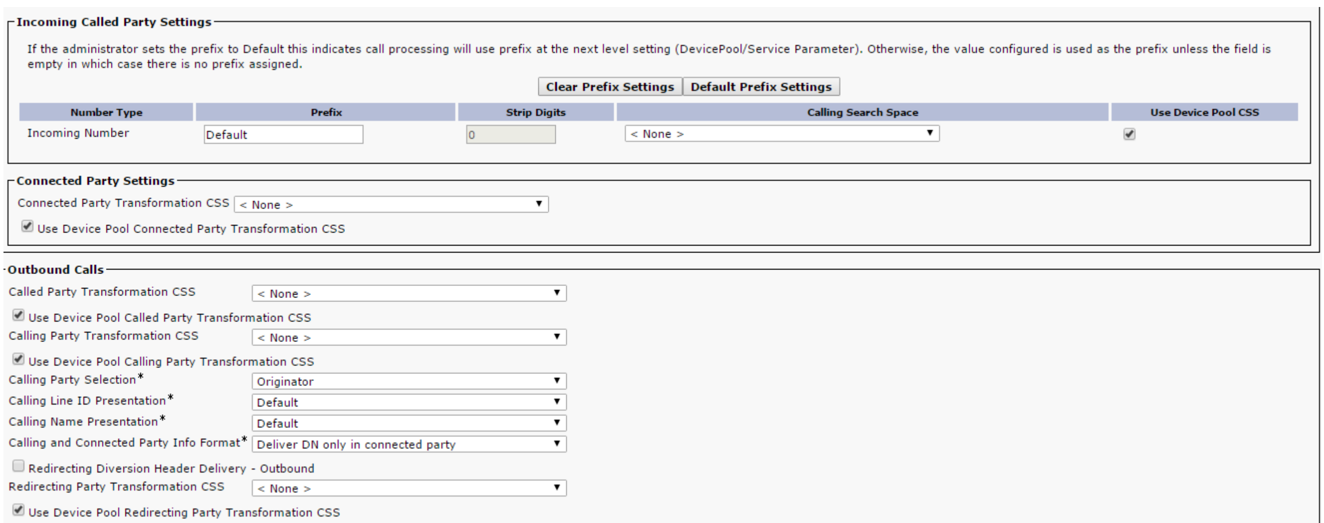


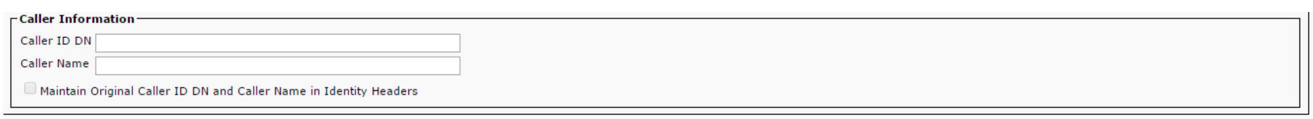
2. Add a trunk with the following settings and click **Save**.

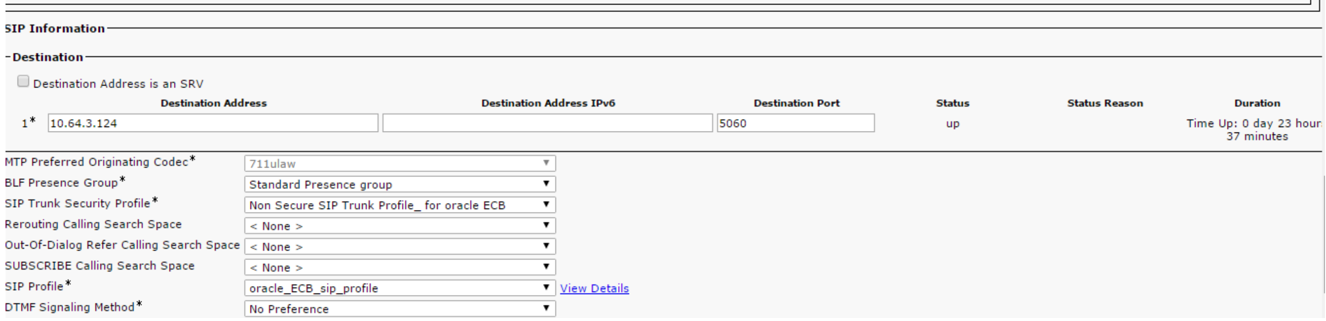


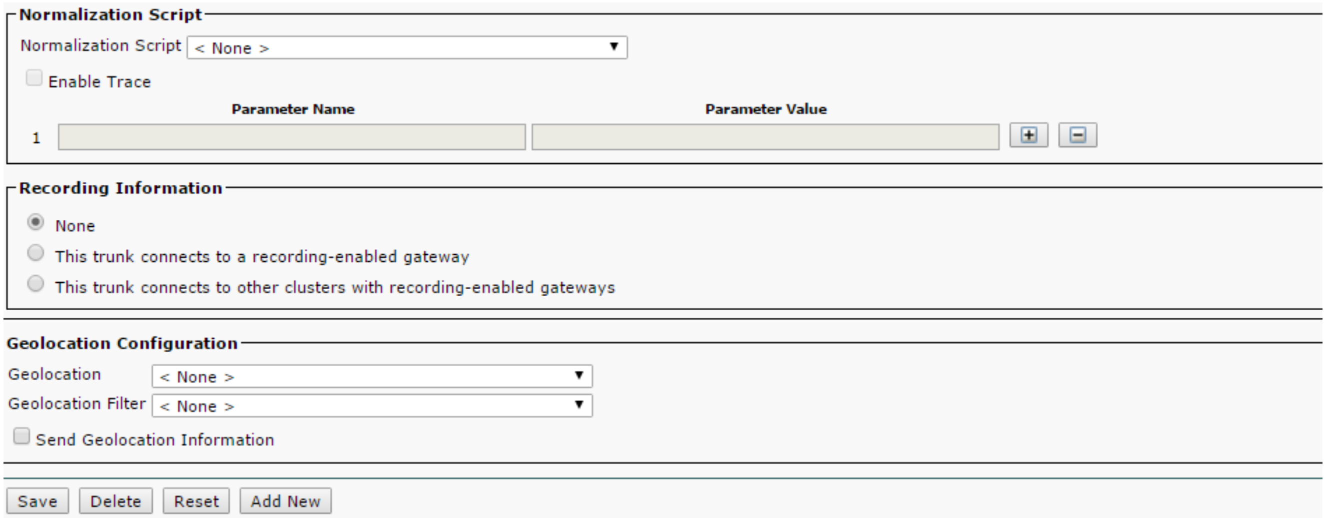



Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* PAI
SIP Privacy* Default
Inbound Calls
Significant Digits* 4
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion 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.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >


Incoming Called 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.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS < None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS < None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection* Originator
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling and Connected Party Info Format* Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS < None >
Use Device Pool Redirecting Party Transformation CSS

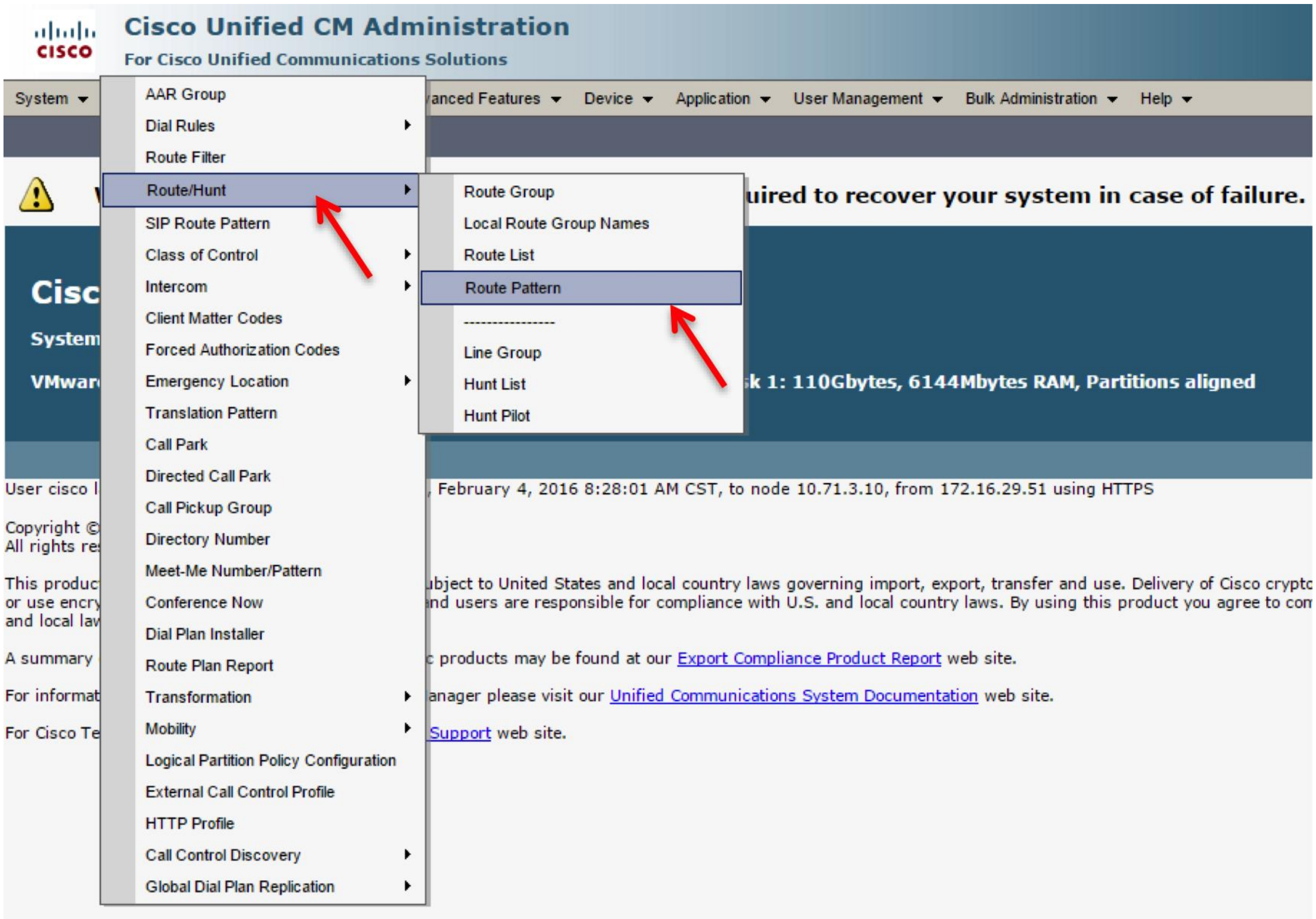

Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers


SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
Status
Status Reason
Duration
1* 10.64.3.124
5060
up
Time Up: 0 day 23 hour 37 minutes
MTP Preferred Originating Codec* 711ulaw
BLF Presence Group* Standard Presence group
SIP Trunk Security Profile* Non Secure SIP Trunk Profile_ for oracle ECB
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* oracle_ECB_sip_profile
View Details
DTMF Signaling Method* No Preference



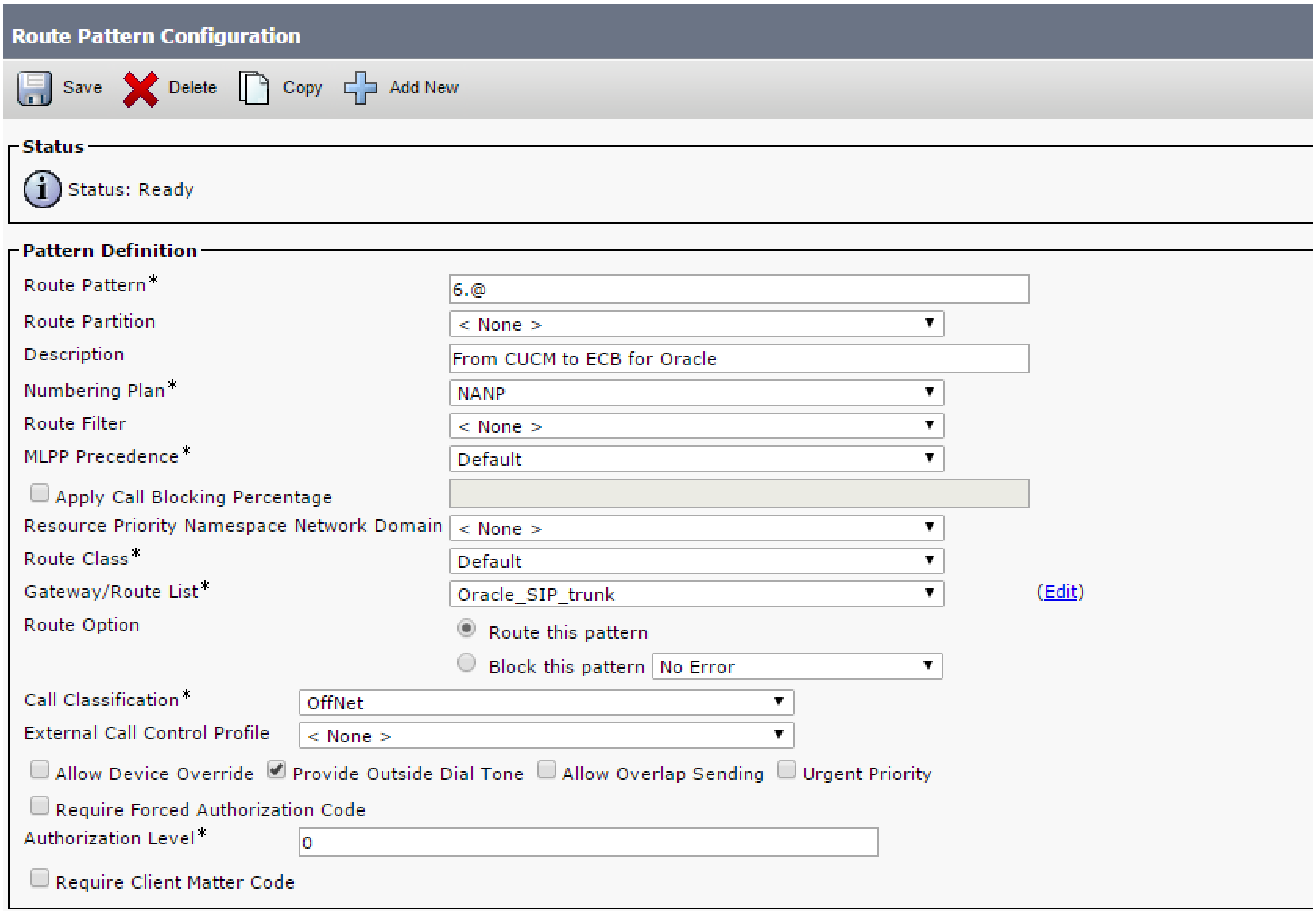
Configuring the Route Pattern

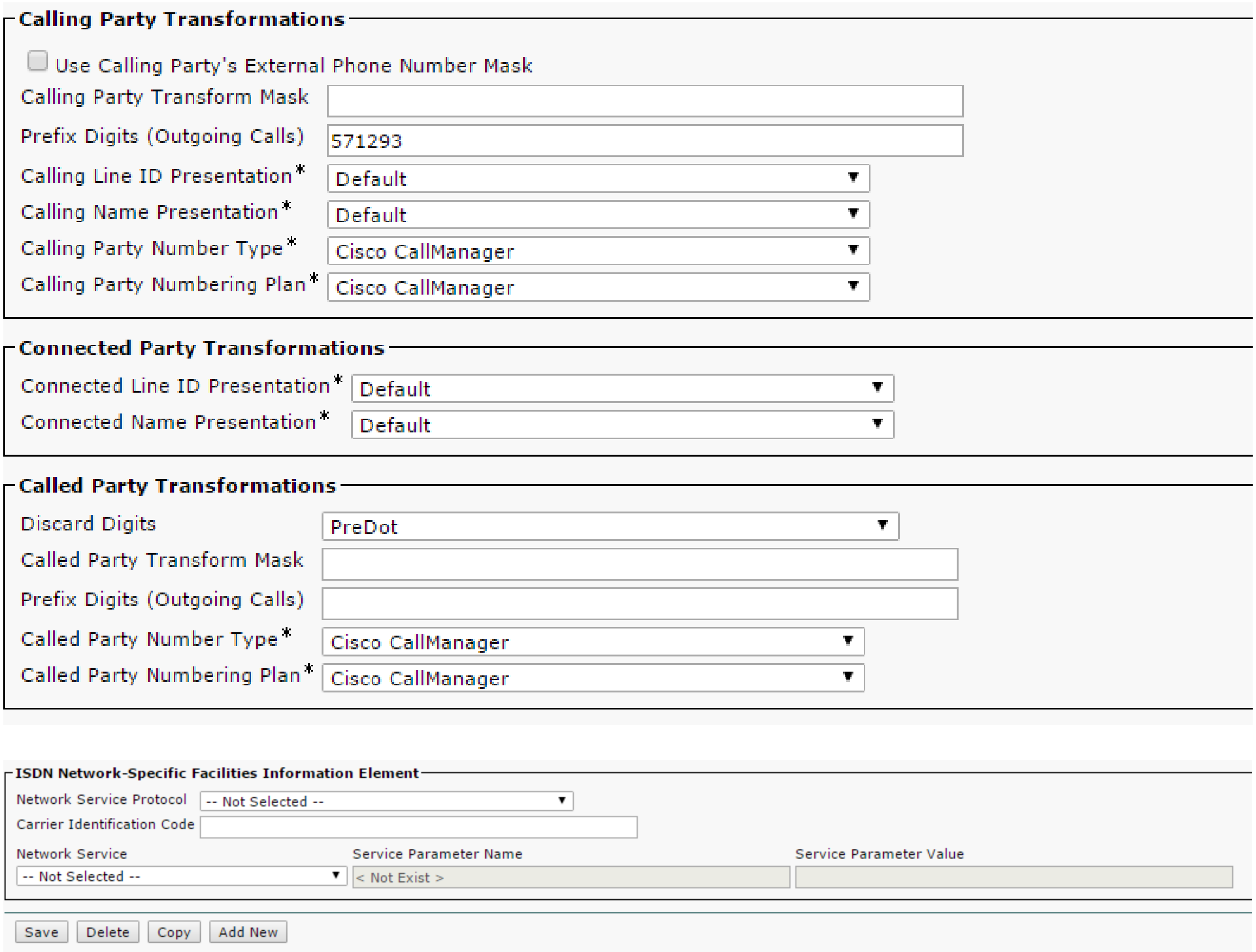
1. To go to the Route pattern page, click on **Call Routing** and select **Route Pattern** from the **Route/Hunt** drop down menu.


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System
anced Features
Device
Application
User Management
Bulk Administration
Help
AAR Group
Dial Rules
Route Filter
Route/Hunt
SIP Route Pattern
Class of Control
Intercom
Client Matter Codes
Forced Authorization Codes
Emergency Location
Translation Pattern
Call Park
Directed Call Park
Call Pickup Group
Directory Number
Meet-Me Number/Pattern
Conference Now
Dial Plan Installer
Route Plan Report
Transformation
Mobility
Logical Partition Policy Configuration
External Call Control Profile
HTTP Profile
Call Control Discovery
Global Dial Plan Replication
Route Group
Local Route Group Names
Route List
Route Pattern

Line Group
Hunt List
Hunt Pilot
uired to recover your system in case of failure.
k 1: 110Gbytes, 6144Mbytes RAM, Partitions aligned
, February 4, 2016 8:28:01 AM CST, to node 10.71.3.10, from 172.16.29.51 using HTTPS
ubject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco crypto
nd users are responsible for compliance with U.S. and local country laws. By using this product you agree to com
c products may be found at our Export Compliance Product Report web site.
anager please visit our Unified Communications System Documentation web site.
Support web site.

2. In our setup, users dial 6 to dial out. Add a route pattern with the following settings and associate it with the trunk configured in the previous step, then click **Save**.





The CUCM 11.0 configuration is now complete.

Test Plan & Results

Test Plan

The testing was done with SIP/TCP and RTP and was performed by tekVizion.

The test plan consisted of the following test cases. All tests passed.

External ID	Title	Status (Pass or Fail)	Comments
Inbound / Outbound / Extension Dialing			
1	Avaya 6.3 calls Avaya 7.0 using extension dialing	Pass	
2	Avaya 6.3 calls Avaya 7.0 using 10 digit dialing	Pass	
3	Avaya 6.3 calls Avaya 7.0 using PSTN dialing	Pass	
4	Avaya 6.3 calls Cisco 10.5 using extension dialing	Pass	
5	Avaya 6.3 calls Cisco 10.5 using 10 digit dialing	Pass	
6	Avaya 6.3 calls Cisco 10.5 using PSTN dialing	Pass	
7	Avaya 6.3 calls Cisco 11.0 using extension dialing	Pass	
8	Avaya 6.3 calls Cisco 11.0 using 10 digit dialing	Pass	
9	Avaya 6.3 calls Cisco 11.0 using PSTN dialing	Pass	
10	Avaya 6.3 calls Microsoft Lync 2013 using extension dialing	Pass	
11	Avaya 6.3 calls Microsoft Lync 2013 using 10 digit dialing	Pass	
12	Avaya 6.3 calls Microsoft Lync 2013 using PSTN dialing	Pass	
13	Avaya 6.3 calls Skype for Business using extension dialing	Pass	
14	Avaya 6.3 calls Skype for Business using 10 digit dialing	Pass	
15	Avaya 6.3 calls Skype for Business using PSTN dialing	Pass	
16	Avaya 7.0 calls Avaya 6.3 using extension dialing	Pass	
17	Avaya 7.0 calls Avaya 6.3 using 10 digit dialing	Pass	
18	Avaya 7.0 calls Avaya 6.3 using PSTN dialing	Pass	
19	Avaya 7.0 calls Cisco 10.5 using extension dialing	Pass	
20	Avaya 7.0 calls Cisco 10.5 using 10 digit dialing	Pass	

21	Avaya 7.0 calls Cisco 10.5 using PSTN dialing	Pass	
22	Avaya 7.0 calls Cisco 11.0 using extension dialing	Pass	
23	Avaya 7.0 calls Cisco 11.0 using 10 digit dialing	Pass	
24	Avaya 7.0 calls Cisco 11.0 using PSTN dialing	Pass	
25	Avaya 7.0 calls Microsoft Lync 2013 using extension dialing	Pass	
26	Avaya 7.0 calls Microsoft Lync 2013 using 10 digit dialing	Pass	
27	Avaya 7.0 calls Microsoft Lync 2013 using PSTN dialing	Pass	
28	Avaya 7.0 calls Skype for Business using extension dialing	Pass	
29	Avaya 7.0 calls Skype for Business using 10 digit dialing	Pass	
30	Avaya 7.0 calls Skype for Business using PSTN dialing	Pass	
31	Cisco 10.5 calls Avaya 6.3 using extension dialing	Pass	
32	Cisco 10.5 calls Avaya 6.3 using 10 digit dialing	Pass	
33	Cisco 10.5 calls Avaya 6.3 using PSTN dialing	Pass	
34	Cisco 10.5 calls Avaya 7.0 using extension dialing	Pass	
35	Cisco 10.5 calls Avaya 7.0 using 10 digit dialing	Pass	
36	Cisco 10.5 calls Avaya 7.0 using PSTN dialing	Pass	
37	Cisco 10.5 calls Cisco 11.0 using extension dialing	Pass	
38	Cisco 10.5 calls Cisco 11.0 using 10 digit dialing	Pass	
39	Cisco 10.5 calls Cisco 11.0 using PSTN dialing	Pass	
40	Cisco 10.5 calls Microsoft Lync 2013 using extension	Pass	
41	Cisco 10.5 calls Microsoft Lync 2013 using 10 digit dialing	Pass	
42	Cisco 10.5 calls Microsoft Lync 2013 using PSTN dialing	Pass	
43	Cisco 10.5 calls Skype for Business using extension dialing	Pass	
44	Cisco 10.5 calls Skype for Business using 10 digit dialing	Pass	
45	Cisco 10.5 calls Skype for Business using PSTN dialing	Pass	
46	Cisco 11.0 calls Avaya 6.3 using extension dialing	Pass	
47	Cisco 11.0 calls Avaya 6.3 using 10 digit dialing	Pass	
48	Cisco 11.0 calls Avaya 6.3 using	Pass	

	PSTN dialing		
49	Cisco 11.0 calls Avaya 7.0 using extension dialing	Pass	
50	Cisco 11.0 calls Avaya 7.0 using 10 digit dialing	Pass	
51	Cisco 11.0 calls Avaya 7.0 using PSTN dialing	Pass	
52	Cisco 11.0 calls Cisco 10.5 using extension dialing	Pass	
53	Cisco 11.0 calls Cisco 10.5 using 10 digit dialing	Pass	
54	Cisco 11.0 calls Cisco 10.5 using PSTN dialing	Pass	
55	Cisco 11.0 calls Microsoft Lync 2013 using extension dialing	Pass	
56	Cisco 11.0 calls Microsoft Lync 2013 using 10 digit dialing	Pass	
57	Cisco 11.0 calls Microsoft Lync 2013 using PSTN dialing	Pass	
58	Cisco 11.0 calls Skype for Business using extension dialing	Pass	
59	Cisco 11.0 calls Skype for Business using 10 digit dialing	Pass	
60	Cisco 11.0 calls Skype for Business using PSTN dialing	Pass	
61	Microsoft Lync 2013 calls Avaya 6.3 using extension dialing	Pass	
62	Microsoft Lync 2013 calls Avaya 6.3 using 10 digit dialing	Pass	
63	Microsoft Lync 2013 calls Avaya 6.3 using PSTN dialing	Pass	
64	Microsoft Lync 2013 calls Avaya 7.0 using extension dialing	Pass	
65	Microsoft Lync 2013 calls Avaya 7.0 using 10 digit dialing	Pass	
66	Microsoft Lync 2013 calls Avaya 7.0 using PSTN dialing	Pass	
67	Microsoft Lync 2013 calls Cisco 10.5 using extension dialing	Pass	
68	Microsoft Lync 2013 calls Cisco 10.5 using 10 digit dialing	Pass	
69	Microsoft Lync 2013 calls Cisco 10.5 using PSTN dialing	Pass	
70	Microsoft Lync 2013 calls Cisco 11.0 using extension dialing	Pass	
71	Microsoft Lync 2013 calls Cisco 11.0 using 10 digit dialing	Pass	
72	Microsoft Lync 2013 calls Cisco 11.0 using PSTN dialing	Pass	
73	Microsoft Lync 2013 calls Skype for Business using extension dialing	Pass	
74	Microsoft Lync 2013 calls Skype for Business using 10 digit dialing	Pass	

75	Microsoft Lync 2013 calls Skype for Business using PSTN dialing	Pass	
76	Skype for Business calls Avaya 6.3 using extension dialing	Pass	
77	Skype for Business calls Avaya 6.3 using 10 digit dialing	Pass	
78	Skype for Business calls Avaya 6.3 using PSTN dialing	Pass	
79	Skype for Business calls Avaya 7.0 using extension dialing	Pass	
80	Skype for Business calls Avaya 7.0 using 10 digit dialing	Pass	
81	Skype for Business calls Avaya 7.0using PSTN dialing	Pass	
82	Skype for Business calls Cisco 10.5 using extension dialing	Pass	
83	Skype for Business calls Cisco 10.5 using 10 digit dialing	Pass	
84	Skype for Business calls Cisco 10.5 using PSTN dialing	Pass	
85	Skype for Business calls Cisco 11.0 using extension dialing	Pass	
86	Skype for Business calls Cisco 11.0 using 10 digit dialing	Pass	
87	Skype for Business calls Cisco 11.0 using PSTN dialing	Pass	
88	Skype for Business calls Microsoft Lync 2013 using extension dialing	Pass	
89	Skype for Business calls Microsoft Lync 2013 using 10 digit dialing	Pass	
90	Skype for Business calls Microsoft Lync 2013 using PSTN dialing	Pass	
Transfer Functionality			
91	PSTN calls into Avaya 6.3 and transfers to Avaya 7.0 using 10 digit dialing	Pass	
92	PSTN calls into Avaya 6.3 and transfers to Avaya 7.0 using extension dialing	Pass	
93	PSTN calls into Avaya 6.3 and transfers to Cisco 10.5 using 10 digit dialing	Pass	
94	PSTN calls into Avaya 6.3 and transfers to Cisco 10.5 using extension dialing	Pass	
95	PSTN calls into Avaya 6.3 and transfers to Cisco 11.0 using 10 digit dialing	Pass	
96	PSTN calls into Avaya 6.3 and transfers to Cisco 11.0 using extension dialing	Pass	
97	PSTN calls into Avaya 6.3 and transfers to Microsoft Lync 2013 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Avaya phone and not the original PSTN party

98	PSTN calls into Avaya 6.3 and transfers to Microsoft Lync 2013 using extension dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Avaya phone and not the original PSTN party
99	PSTN calls into Avaya 6.3 and transfers to Skype for Business using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Avaya phone and not the Original PSTN party
100	PSTN calls into Avaya 6.3 and transfers to Skype for Business using extension dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Avaya phone and not the original PSTN party
101	PSTN calls into Avaya 7.0 and transfers to Avaya 6.3 using 10 digit dialing	Pass	
102	PSTN calls into Avaya 7.0 and transfers to Avaya 6.3 using extension dialing	Pass	
103	PSTN calls into Avaya 7.0 and transfers to Cisco 10.5 using 10 digit dialing	Pass	
104	PSTN calls into Avaya 7.0 and transfers to Cisco 10.5 using extension dialing	Pass	
105	PSTN calls into Avaya 7.0 and transfers to Cisco 11.0 using 10 digit dialing	Pass	
106	PSTN calls into Avaya 7.0 and transfers to Cisco 11.0 using extension dialing	Pass	
107	PSTN calls into Avaya 7.0 and transfers to Microsoft Lync 2013 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Avaya phone and not the original PSTN party
108	PSTN calls into Avaya 7.0 and transfers to Microsoft Lync 2013 using extension dialing	Pass	The transfer was successful but the calling number displayed on Lync Client is of Avaya phone and not the Original PSTN party
109	PSTN calls into Avaya 7.0 and transfers to Skype for Business using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Avaya phone and not the original PSTN party
110	PSTN calls into Avaya 7.0 and transfers to Skype for Business using extension dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Avaya phone and not the original PSTN party
111	PSTN calls into Cisco 10.5 and transfers to Avaya 6.3 using 10 digit dialing	Pass	
112	PSTN calls into Cisco 10.5 and transfers to Avaya 6.3 using extension dialing	Pass	
113	PSTN calls into Cisco 10.5 and transfers to Avaya 7.0 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Avaya phone is of Cisco phone and not the original PSTN party. To resolve this HMR is added in ECB to add PAI to the UPDATE header

114	PSTN calls into Cisco 10.5 and transfers to Avaya 7.0 using extension dialing	Pass	The transfer was successful but the calling number displayed on Avaya phone is of Cisco phone and not the original PSTN party. To resolve this HMR is added in ECB to add PAI to the UPDATE header
115	PSTN calls into Cisco 10.5 and transfers to Cisco 11.0 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on the second PSTN party is of Cisco 10.5 phone and not the original PSTN party
116	PSTN calls into Cisco 10.5 and transfers to Cisco 11.0 using extension dialing	Pass	The transfer was successful but the calling number displayed on the second PSTN party is of Cisco 10.5 phone and not the original PSTN party
117	PSTN calls into Cisco 10.5 and transfers to Microsoft Lync 2013 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Cisco phone and not the original PSTN party
118	PSTN calls into Cisco 10.5 and transfers to Microsoft Lync 2013 using extension dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Cisco phone and not the original PSTN party
119	PSTN calls into Cisco 10.5 and transfers to Skype for Business using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Cisco phone and not the original PSTN party
120	PSTN calls into Cisco 10.5 and transfers to Skype for Business using extension dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Cisco phone and not the original PSTN party
121	PSTN calls into Cisco 11.0 and transfers to Avaya 6.3 using 10 digit dialing	Pass	
122	PSTN calls into Cisco 11.0 and transfers to Avaya 6.3 using extension dialing	Pass	
123	PSTN calls into Cisco 11.0 and transfers to Avaya 7.0 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Avaya phone is of Cisco phone and not the original PSTN party. To resolve this HMR is added in ECB to add PAI to the UPDATE header
124	PSTN calls into Cisco 11.0 and transfers to Avaya 7.0 using extension dialing	Pass	The transfer was successful but the calling number displayed on Avaya phone is of Cisco phone and not the original PSTN party. To resolve this HMR is added in ECB to add PAI to the UPDATE header
125	PSTN calls into Cisco 11.0 and transfers to Cisco 10.5 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on the second PSTN party is of Cisco 11.0 phone and not the original PSTN party
126	PSTN calls into Cisco 11.0 and transfers to Cisco 10.5 using extension dialing	Pass	The transfer was successful but the calling number displayed on the second PSTN party is of Cisco 11.0 phone and not the original PSTN party
127	PSTN calls into Cisco 11.0 and transfers to Microsoft Lync 2013 using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Cisco phone and not the original PSTN party

128	PSTN calls into Cisco 11.0 and transfers to Microsoft Lync 2013 using extension dialing	Pass	The transfer was successful but the calling number displayed on Lync client is of Cisco phone and not the original PSTN party
129	PSTN calls into Cisco 11.0 and transfers to Skype for Business using 10 digit dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Cisco phone and not the original PSTN party
130	PSTN calls into Cisco 11.0 and transfers to Skype for Business using extension dialing	Pass	The transfer was successful but the calling number displayed on Skype for Business client is of Cisco phone and not the original PSTN party
131	PSTN calls into Microsoft Lync 2013 and transfers to Avaya 6.3 using 10 digit dialing	Pass	
132	PSTN calls into Microsoft Lync 2013 and transfers to Avaya 6.3 using extension dialing	Pass	
133	PSTN calls into Microsoft Lync 2013 and transfers to Avaya 7.0 using 10 digit dialing	Pass	
134	PSTN calls into Microsoft Lync 2013 and transfers to Avaya 7.0 using extension dialing	Pass	
135	PSTN calls into Microsoft Lync 2013 and transfers to Cisco 10.5 using 10 digit dialing	Pass	
136	PSTN calls into Microsoft Lync 2013 and transfers to Cisco 10.5 using extension dialing	Pass	
137	PSTN calls into Microsoft Lync 2013 and transfers to Cisco 11.0 using 10 digit dialing	Pass	
138	PSTN calls into Microsoft Lync 2013 and transfers to Cisco 11.0 using extension dialing	Pass	
139	PSTN calls into Microsoft Lync 2013 and transfers to Skype for Business using 10 digit dialing	Pass	
140	PSTN calls into Microsoft Lync 2013 and transfers to Skype for Business using extension	Pass	
141	PSTN calls into Skype for Business and transfers to Avaya 6.3 using 10 digit dialing	Pass	
142	PSTN calls into Skype for Business and transfers to Avaya 6.3 using extension dialing	Pass	
143	PSTN calls into Skype for Business and transfers to Avaya 7.0 using 10 digit dialing	Pass	
144	PSTN calls into Skype for Business and transfers to Avaya 7.0 using extension dialing	Pass	
145	PSTN calls into Skype for Business and transfers to Cisco 10.5 using 10 digit dialing	Pass	

146	PSTN calls into Skype for Business and transfers to Cisco 10.5 using extension dialing	Pass	
147	PSTN calls into Skype for Business and transfers to Cisco 11.0 using 10 digit dialing	Pass	
148	PSTN calls into Skype for Business and transfers to Cisco 11.0 using extension dialing	Pass	
149	PSTN calls into Skype for Business and transfers to Microsoft Lync 2013 using 10 digit dialing	Pass	
150	PSTN calls into Skype for Business and transfers to Microsoft Lync 2013 using extension dialing	Pass	
Call Hold / Resume			
151	Avaya 6.3 calls Avaya 7.0 using PSTN dialing and places the call on hold & reconnects	Pass	
152	Avaya 6.3 calls Avaya 7.0 using extension dialing and places the call on hold & reconnects	Pass	
153	Avaya 6.3 calls Cisco 10.5 using PSTN dialing and places the call on hold & reconnects	Pass	
154	Avaya 6.3 calls Cisco 10.5 using extension dialing and places the call on hold & reconnects	Pass	
155	Avaya 6.3 calls Cisco 11.0 using PSTN dialing and places the call on hold & reconnects	Pass	
156	Avaya 6.3 calls Cisco 11.0 using extension dialing and places the call on hold & reconnects	Pass	
157	Avaya 6.3 calls Microsoft Lync 2013 using PSTN dialing and places the call on hold & reconnects	Pass	
158	Avaya 6.3 calls Microsoft Lync 2013 using extension dialing and places the call on hold & reconnects	Pass	
159	Avaya 6.3 calls Skype for Business using PSTN dialing and places the call on hold & reconnects	Pass	
160	Avaya 6.3 calls Skype for Business using extension dialing and places the call on hold & reconnects	Pass	
161	Avaya 7.0 calls Avaya 6.3 using PSTN dialing and places the call on hold & reconnects	Pass	
162	Avaya 7.0 calls Avaya 6.3 using extension dialing and places the call on hold & reconnects	Pass	
163	Avaya 7.0 calls Cisco 10.5 using PSTN dialing and places the call on hold & reconnects	Pass	

164	Avaya 7.0 calls Cisco 10.5 using extension dialing and places the call on hold & reconnects	Pass	
165	Avaya 7.0 calls Cisco 11.0 using PSTN dialing and places the call on hold & reconnects	Pass	
166	Avaya 7.0 calls Cisco 11.0 using extension dialing and places the call on hold & reconnects	Pass	
167	Avaya 7.0 calls Microsoft Lync 2013 using PSTN dialing and places the call on hold & reconnects	Pass	
168	Avaya 7.0 calls Microsoft Lync 2013 using extension dialing and places the call on hold & reconnects	Pass	
169	Avaya 7.0 calls Skype for Business using PSTN dialing and places the call on hold & reconnects	Pass	
170	Avaya 7.0 calls Skype for Business using extension dialing and places the call on hold & reconnects	Pass	
171	Cisco 10.5 calls Avaya 6.3 using PSTN dialing and places the call on hold & reconnects	Pass	
172	Cisco 10.5 calls Avaya 6.3 using extension dialing and places the call on hold & reconnects	Pass	
173	Cisco 10.5 calls Avaya 7.0 using PSTN dialing and places the call on hold & reconnects	Pass	
174	Cisco 10.5 calls Avaya 7.0 using extension dialing and places the call on hold & reconnects	Pass	
175	Cisco 10.5 calls Cisco 11.0 using PSTN dialing and places the call on hold & reconnects	Pass	
176	Cisco 10.5 calls Cisco 11.0 using extension dialing and places the call on hold & reconnects	Pass	
177	Cisco 10.5 calls Microsoft Lync 2013 using PSTN dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Lync responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Lync to add the DTMF event 101 when the re-invite does not offer it.
178	Cisco 10.5 calls Microsoft Lync 2013 using extension dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Lync responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Lync to add the DTMF event 101 when the re-invite does not offer it.

179	Cisco 10.5 calls Skype for Business using PSTN dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Skype for Business responds with a 488 DTMF not supported and call is dropped. Issue resolved by adding a HMR in ECB towards Skype for Business to add the DTMF event 101 when the re-invite does not offer it.
180	Cisco 10.5 calls Skype for Business using extension dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Skype for Business responds with a 488 DTMF not supported and call is dropped. Issue resolved by adding a HMR in ECB towards Skype for Business to add the DTMF event when the re-invite does not offer it.
181	Cisco 11.0 calls Avaya 6.3 using PSTN dialing and places the call on hold & reconnects	Pass	
182	Cisco 11.0 calls Avaya 6.3 using extension dialing and places the call on hold & reconnects	Pass	
183	Cisco 11.0 calls Avaya 7.0 using PSTN dialing and places the call on hold & reconnects	Pass	
184	Cisco 11.0 calls Avaya 7.0 using extension dialing and places the call on hold & reconnects	Pass	
185	Cisco 11.0 calls Cisco 10.5 using PSTN dialing and places the call on hold & reconnects	Pass	
186	Cisco 11.0 calls Cisco 10.5 using extension dialing and places the call on hold & reconnects	Pass	
187	Cisco 11.0 calls Microsoft Lync 2013 using PSTN dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Lync responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Lync to add the DTMF event 101 when the re-invite does not offer it.
188	Cisco 11.0 calls Microsoft Lync 2013 using extension and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Lync responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Lync to add the DTMF event 101 when the re-invite does not offer it.

189	Cisco 11.0 calls Skype for Business using PSTN dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Skype for Business responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Skype for Business to add the DTMF event 101 when the re-invite does not offer it.
190	Cisco 11.0 calls Skype for Business using extension dialing and places the call on hold & reconnects	Pass	The re-invite with SDP sent from the CUCM to resume the call on hold does not support DTMF. Due to this Skype for Business responds with a “488 DTMF not supported” and call is dropped. Issue resolved by adding a HMR in ECB towards Skype for Business to add the DTMF event 101 when the re-invite does not offer it.
191	Microsoft Lync 2013 calls Avaya 6.3 using PSTN dialing and places the call on hold & reconnects	Pass	
192	Microsoft Lync 2013 calls Avaya 6.3 using extension dialing and places the call on hold & reconnects	Pass	
193	Microsoft Lync 2013 calls Avaya 7.0 using PSTN dialing and places the call on hold & reconnects	Pass	
194	Microsoft Lync 2013 calls Avaya 7.0 using extension dialing and places the call on hold & reconnects	Pass	
195	Microsoft Lync 2013 calls Cisco 10.5 using PSTN dialing and places the call on hold & reconnects	Pass	
196	Microsoft Lync 2013 calls Cisco 10.5 using extension dialing and places the call on hold & reconnects	Pass	
197	Microsoft Lync 2013 calls Cisco 11.0 using PSTN dialing and places the call on hold & reconnects	Pass	
198	Microsoft Lync 2013 calls Cisco 11.0 using extension dialing and places the call on hold & reconnects	Pass	
199	Microsoft Lync 2013 calls Skype for Business using PSTN dialing and places the call on hold & reconnects	Pass	
200	Microsoft Lync 2013 calls Skype for Business using extension dialing and places the call on hold & reconnects	Pass	
201	Skype for Business calls Avaya 6.3 using PSTN dialing and places the call on hold & reconnects	Pass	
202	Skype for Business calls Avaya 6.3 using extension dialing and places the call on hold & reconnects	Pass	
203	Skype for Business calls Avaya 7.0 using PSTN dialing and places the call on hold & reconnects	Pass	

204	Skype for Business calls Avaya 7.0 using extension dialing and places the call on hold & reconnects	Pass	
205	Skype for Business calls Cisco 10.5 using PSTN dialing and places the call on hold & reconnects	Pass	
206	Skype for Business calls Cisco 10.5 using extension dialing and places the call on hold & reconnects	Pass	
207	Skype for Business calls Cisco 11.0 using PSTN dialing and places the call on hold & reconnects	Pass	
208	Skype for Business calls Cisco 11.0 using extension dialing and places the call on hold & reconnects	Pass	
209	Skype for Business calls Microsoft Lync 2013 using PSTN dialing and places the call on hold & reconnects	Pass	
210	Skype for Business calls Microsoft Lync 2013 using extension dialing and places the call on hold & reconnects	Pass	

Parallel Forking w/ LDAP Integration			
211	Avaya 6.3 calls a user (via 10-digit dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in parallel and either can be answered.	Pass	Ring back is not heard on Avaya during this test case execution. Added a HMR in ECB to convert “183 with SDP” from Lync to “180 RINGING with SDP” to produce the ring back.
212	Avaya 6.3 calls a user (via extension dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in parallel and either can be answered.	Pass	Ring back is not heard on Avaya during this test case execution. Added a HMR in ECB to convert “183 with SDP” from Lync to “180 RINGING with SDP” to produce the ring back.
213	Avaya 6.3 calls a user (via 10 digit dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in parallel and either can be answered.	Pass	Ring back is not heard on Avaya during this test case execution. Added a HMR in ECB to convert “183 with SDP” from Lync to “180 RINGING with SDP” to produce the ring back.
214	Avaya 6.3 calls a user (via extension dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in parallel and either can be answered.	Pass	Ring back is not heard on Avaya during this test case execution. Added a HMR in ECB to convert “183 with SDP” from Lync to “180 RINGING with SDP” to produce the ring back.
215	Avaya 7.0 calls a user (via 10-digit dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in parallel and either can be answered.	Pass	
216	Avaya 7.0 calls a user (via extension dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in parallel and either can be answered.	Pass	
217	Avaya 7.0 calls a user (via 10 digit dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in parallel and the call is unanswered and forwarded to voice mail	Pass	
218	Avaya 7.0 calls a user (via extension dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in parallel and either can be answered.	Pass	
Serial Forking w/ LDAP Integration			
219	Avaya 6.3 calls a user (via 10 digit dial) which is configured on both	Pass	

	Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in Serially and either can be answered.		
220	Avaya 6.3 calls a user (via extension dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in serially .This call is unanswered and forwarded to Voice Mail	Pass	
221	Avaya 6.3 calls a user (via 10 digit dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in serially and either can be answered.	Pass	
222	Avaya 6.3 calls a user (via extension dial) which is configured on both Cisco 10.5 and Lync 2013. Both Cisco 10.5 and Lync 2013 instances should ring in serially and either can be answered.	Pass	
223	Avaya 7.0 calls a user (via 10 digit dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in serially and either can be answered.	Pass	
224	Avaya 7.0 calls a user (via extension dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in serially and either can be answered.	Pass	
225	Avaya 7.0 calls a user (via 10 digit dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in serially and either can be answered.	Pass	
226	Avaya 7.0 calls a user (via extension dial) which is configured on both Cisco 11.0 and Skype for Business. Both Cisco 11.0 and Skype for Business instances should ring in serially and either can be answered.	Pass	
From Header Replacement w/ Active Directory			
227	User A (x4444) configured in ECB for endpoints on cisco 10.5 and Lync 2013. Call is initiated from Avaya 6.3 system to x4444. Result = both endpoints ring and display CID on cisco 10.5 and Lync 2013 end points.	Pass	

228	Call initiated by Avaya 6.3 and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013.Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	
229	User A (x4444) configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Call is initiated from Avaya 6.3 system to x4444. Result = both endpoints ring and display CID on Cisco 10.5 and Lync 2013 end points.	Pass	
230	Call initiated by Avaya 6.3 and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	
231	User A (x5555) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Avaya 7.0 system to x5555. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business end points.	Pass	
232	Call initiated by Avaya 7.0 and calls User A (x5555) with full 10 digit number (312-777-5555). Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	
233	User A (x5555) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Avaya 7.0 system to x5555. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business end points.	Pass	
234	Call initiated by Avaya 7.0 and calls User A (x5555) with full 10 digit number (312-777-5555). Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	
235	User A (x4444) configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Call is initiated from Cisco 10.5 system to x4444. Result = both endpoints ring and display CID on Cisco 10.5 and Lync 2013	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.XXXX)

	endpoints.		
236	Call initiated by Cisco 10.5 and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.312777XXXX).
237	User A (x4444) configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Call is initiated from Cisco 10.5 system to x4444. Result = both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.XXXX)
238	Call initiated by Cisco 10.5 and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.312777XXXX).
239	User A (x5555) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Cisco 11.0 system to x5555. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business end points.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.XXXX)
240	Call initiated by Cisco 11.0 and calls User A (x5555) with full 10 digit number (312-777-5555). Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.312777XXXX).
241	User A (x5555) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Cisco 11.0 system to x5555. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business end points.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.XXXX)
242	Call initiated by Cisco 11.0 and calls User A (x5555) with full 10 digit number (312-777-5555). Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route pattern in CUCM is required (ex: 9.312777XXXX).

243	User A (x4444) configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Call is initiated from Lync 2013 system to x4444.Result = both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	
244	Call initiated by Lync 2013 and calls User A (x4444) with full 10 digit number (312-777-4444).User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013.Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route in Lync is required (ex: 9.312777XXXX).
245	User A (x4444) configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Call is initiated from Lync 2013 system to x4444. Result = both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	
246	Call initiated by Lync 2013 and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 10.5 and Lync 2013. Result = Both endpoints ring and display CID on Cisco 10.5 and Lync 2013 endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route in Lync is required (ex: 9.312777XXXX).
247	User A (x4444) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Skype for Business system to x4444. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	
248	Call initiated by Skype for Business and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Result = Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route in Skype for Business is required (ex: 9.312777XXXX).
249	User A (x4444) configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Call is initiated from Skype for Business system to x4444. Result = both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	

250	Call initiated by Skype for Business and calls User A (x4444) with full 10 digit number (312-777-4444). User A configured in ECB for endpoints on Cisco 11.0 and Skype for Business. Result = Both endpoints ring and display CID on Cisco 11.0 and Skype for Business endpoints.	Pass	Call does not reach the ECB since the extension dialed belongs to the PBX from which the call is originated. To route the call towards ECB a separate route in Skype for Business is required (ex: 9.312777XXXX).
Active Directory Failover (Test querying LDAP and the query failing / falling back to backup LDAP / User DB / Default mode)			
251	Cisco 10.5 calls Avaya 6.3	Pass	
252	Cisco 10.5 calls Avaya 7.0	Pass	
253	Cisco 10.5 calls cisco 11.0	Pass	
254	Cisco 10.5 calls Lync 2013	Pass	
255	Cisco 10.5 calls Skype for Business	Pass	
256	Lync 2013 calls Avaya 6.3	Pass	
257	Lync 2013 calls Avaya 7.0	Pass	
258	Lync 2013 calls Cisco 10.5	Pass	
259	Lync 2013 calls Cisco 11.0	Pass	
260	Lync 2013 calls Skype for Business	Pass	

Software Versions Used

The following are the software versions used in this testing by tekVizion.

Component	Version
ECB	PCZ2.0.0 MR-2 Patch 1 (Build 209)
E-SBC	ECZ7.3.0 MR-1 GA (Build 104)
Microsoft Lync 2013 Server	5.0.8308.0
Microsoft Skype for Business Server 2015	6.0.9319.0
Cisco Unified Communication Manager	10.5.1 & 11.0.1
Avaya Aura	6.3.14 & 7.0.0.1

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/SFB Server, and the Oracle ECB and 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.

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/SFB Server mediation server, the Lync/SFB 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 and/or SFB 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/SFB Server Front End server;
- A Lync/SFB Server Standard Edition Server; and
- A Lync/SFB Server Mediation Server.

On the Oracle ECB and E-SBC

The Oracle SBC and ECB provide a rich set of statistical counters available from the CLI, 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 console:

```
oraclesbc1# reset sipd
oraclesbc1# notify sipd debug
oraclesbc1#
enabled SIP Debugging
oraclesbc1# notify all rotate-logs
```

Examining the log files

Note: You will FTP to the management interface of the ECB or 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 oraclesbc1FTP server (VxWorks 6.4) ready.
User (192.168.85.55:(none)): user
```

```
331 Password required for user.
Password: acme
230 User user logged in.
ftp> cd /ramdrv/logs
250 CWD command successful.
ftp> get sipmsg.log
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/sipmsg.log' (3353
bytes).
226 Transfer complete.
ftp: 3447 bytes received in 0.00Seconds 3447000.00Kbytes/sec.
ftp> get log.sipd
200 PORT command successful.
150 Opening ASCII mode data connection for '/ramdrv/logs/log.sipd' (204681
bytes).
226 Transfer complete.
ftp: 206823 bytes received in 0.11Seconds 1897.46Kbytes/sec.
ftp> bye
221 Goodbye.
```

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

Through the Web GUI

You can also check the display results of filtered SIP session data from the Oracle E-SBC and ECB, and provide traces in a common log format for local viewing or for exporting to your PC. Please check the “Monitor and Trace SIP Messages” section (page 140) of the E-SBC Web GUI User Guide available at http://docs.oracle.com/cd/E56581_01/index.htm. For the ECB, see the “Monitor and Trace” section (page 95) of the User’s Guide available at http://docs.oracle.com/cd/E55725_01/index.htm.

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 the there is nothing blocking them such as firewalls. Ensure that you have a TELNET client available on a workstation.

All devices tested in this document will listen on TCP port 5060 for SIP signaling. In our example we are listening on 5060 on the PSTN facing NIC. Tests may include:

- Client to pool server: **`telnet <servername> 5060`**
- Pool server to Mediation Server: **`telnet <servername> 5060`**

Cisco Real-Time Monitoring Tool (RTMT)

The Cisco Real-Time Monitoring Tool (RTMT) is a tool that can be downloaded from CUCM to a Windows or Linux computer. See https://supportforums.cisco.com/document/93281/using-rtmt-monitor-cisco-unity-connection-and-cucm for details.

Appendix A

Accessing the ACLI

Access to the ACLI is provided by:

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

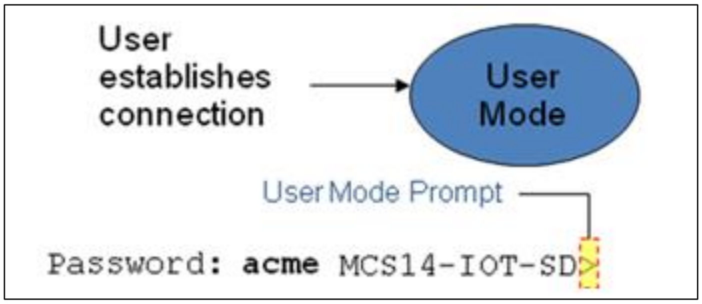
Initial connectivity will be through the serial console port. At a minimum, this is how to configure the management (eth0) interface on the SBC.

ACLI Basics

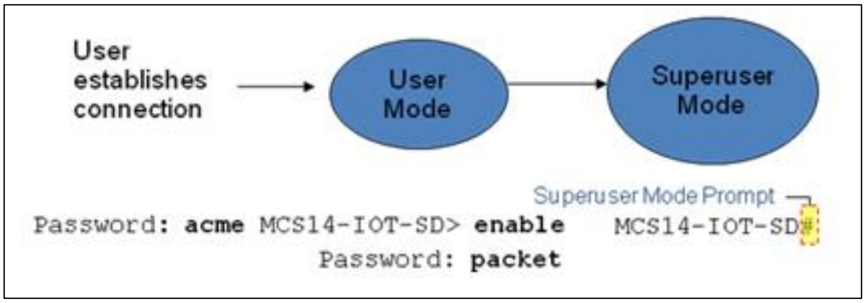
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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

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



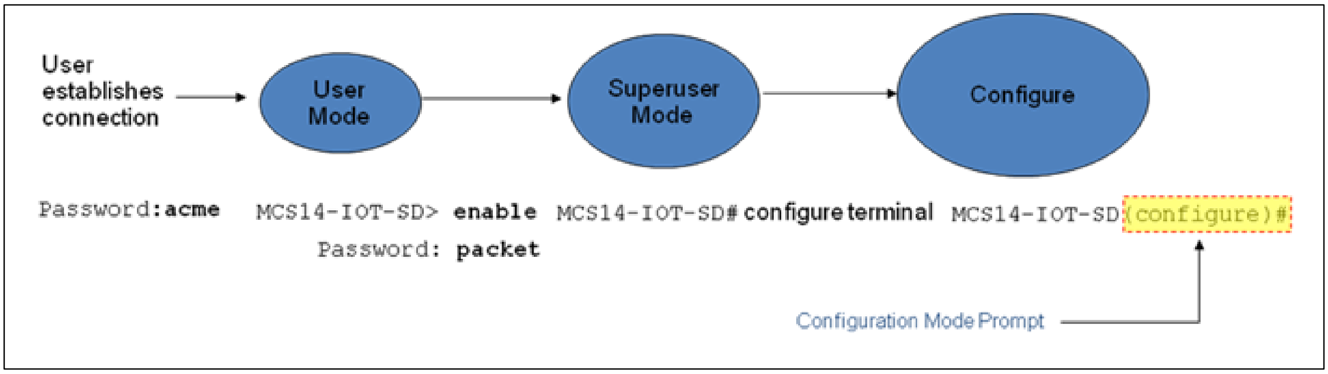
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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

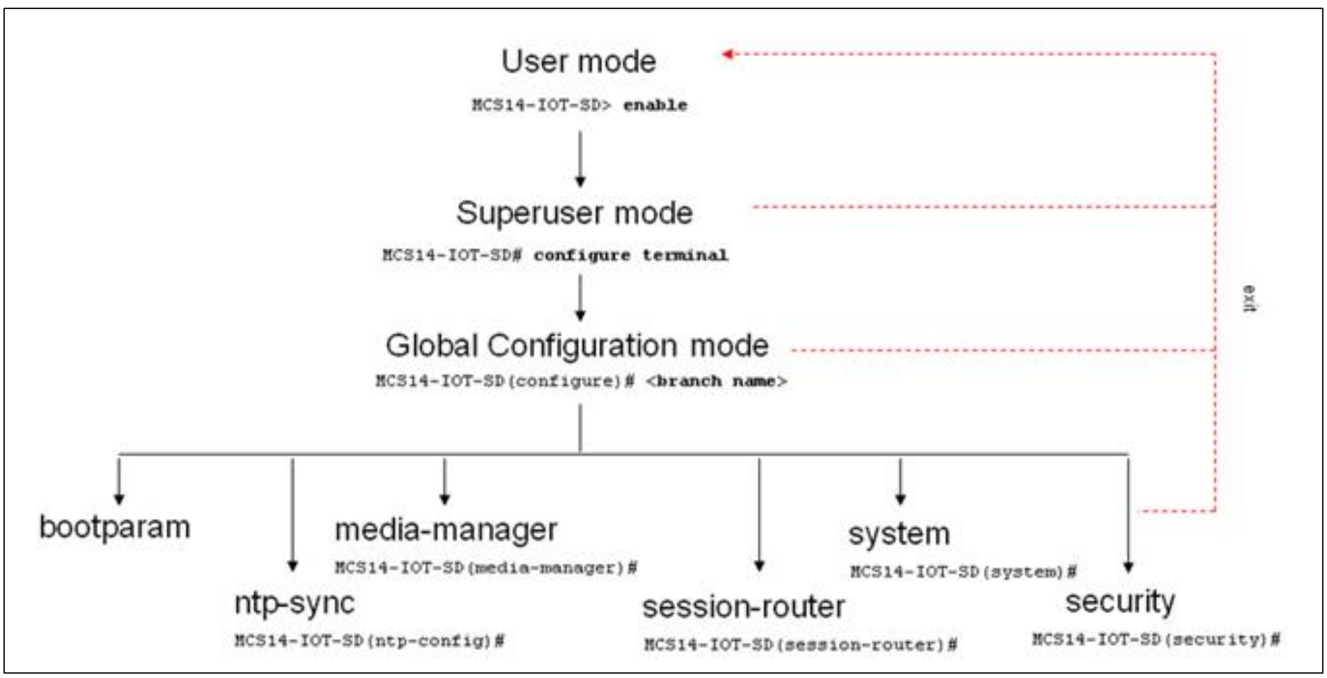
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the `configure terminal` command in the Superuser mode.

Configuration mode is identified by the word configure in parenthesis followed by the pound sign (#) in the prompt after the target name, for example, **oraclesbc1(configure)#**. To return to the Superuser mode, issue the `exit` command.



In the configuration mode, there are six configuration branches:

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



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

The bootparam branch provides access to SBC boot parameters.

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

The security branch provides access to security configuration.

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

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

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

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

Configuration Elements

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

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

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

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

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

Creating an Element

1. To create a single-instance element, you go to the appropriate level in the ACLI path and enter its parameters. There is no need to specify a unique identifier property because a single-instance element is a global element and there is only one instance of this element.
2. When creating a multiple-instance element, you must specify a unique identifier for each instance of the element.
3. It is important to check the parameters of the element you are configuring before committing the changes. You do this by issuing the **`show`** command before issuing the **`done`** command. The parameters that you did not configure are filled with either default values or left empty.
4. On completion, you must issue the **`done`** command. The done command causes the configuration to be echoed to the screen and commits the changes to the volatile memory. It is a good idea to review this output to ensure that your configurations are correct.
5. Issue the **`exit`** command to exit the selected element.

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

Editing an Element

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

1. Enter the element that you will edit at the correct level of the ACLI path.
2. Select the element that you will edit, and view it before editing it.
   The **`select`** command loads the element to the volatile memory for editing. The **`show`** command allows you to view the element to ensure that it is the right one that you want to edit.
3. Once you are sure that the element you selected is the right one for editing, edit the parameter one by one. The new value you provide will overwrite the old value.

4. It is important to check the properties of the element you are configuring before committing it to the volatile memory. You do this by issuing the **`show`** command before issuing the **`done`** command.
5. On completion, you must issue the **`done`** command.
6. Issue the **`exit`** command to exit the selected element.

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

Deleting an Element

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

To delete a single-instance element,

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

To delete a multiple-instance element,

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

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

Configuration Versions

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

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

Saving the Configuration

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

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

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

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

```
oraclesbc1 # save-config
Save-Config received, processing.
waiting 1200 for request to finish
Request to 'SAVE-CONFIG' has Finished,
Save complete
Currently active and saved configurations do not match!
To sync & activate, run 'activate-config' or 'reboot activate'.
oraclesbc1 #
```

Activating the Configuration

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

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

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

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