ORACLE
COMMUNICATIONS

ATT IP Flexible Reach Service Including MIS/PNT/AVPN Transports with Cisco UCM 10.5 & Oracle Enterprise Session Border Controller

Technical Application Note

Disclaimer

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

Table of Contents

INTENDED AUDIENCE 5
DOCUMENT OVERVIEW 5
INTRODUCTION 6
AUDIENCE 6
REQUIREMENTS 6
ARCHITECTURE 7
LAB CONFIGURATION 8
PHASE 1 – CONFIGURING THE ORACLE ENTERPRISE SBC 9
IN SCOPE 9
OUT OF SCOPE 9
WHAT YOU WILL NEED 9
CONFIGURING THE SBC 9
Establish the serial connection and logging in the SBC 10
Initial Configuration – Assigning the management Interface an IP address 10
Configuring the SBC 11
CONFIGURATION HIGHLIGHTS 11
SIP MANIPULATIONS 14
MEDIA-MANAGER CONFIGURATION FOR CALL/HOLD/RESUME SCENARIOS 21
PHASE 2 – CONFIGURING THE CISCO UNIFIED CALL MANAGER V10.5 23
CREATING A SIP PROFILE IN CUCM 24
ADDING THE E-SBC AS A TRUNK IN CUCM 25
CREATING A ROUTE PATTERN IN CUCM 26
ADDING DHCP SERVER AND SUBNET IN CUCM 28
ADDING DEVICES/PHONES AND CONFIGURING DIRECTORY NUMBERS 30
CREATING & ASSIGNING A REGION AND SPECIFYING USE OF G.729 CODEC IN CUCM 31
PHASE 3 – CONFIGURING THE CISCO UNITY CONNECTION (CUC) SERVER AND INTEGRATING IT WITH CUCM 34
IN SCOPE 34
OUT OF SCOPE 34
WHAT YOU WILL NEED 34
CONFIGURING THE CISCO UNITY CONNECTION SERVER FOR VOICEMAIL 34
CREATE TRUNK TO CONNECT TO CUC 35
CREATE VOICEMAIL PROFILE, PORTS, MWI 36
CREATE ROUTE PATTERN TO ROUTE LOCALLY FROM CUCM TO CUC 38
CREATE PHONE SYSTEM IN CUC, ADD PORTS AND REFERENCE VOICEMAIL PORT 39
CREATE MAILBOX AND USERS IN CUC 41
CREATING SYSTEM CALL HANDLERS IN CUC 42
DEFINE CTI ROUTE POINT IN CUCM FOR AA 43
TEST SUMMARY 44
TROUBLESHOOTING TOOLS 45
CISCO UNIFIED SERVICEABILITY IN CUCM 45
WIRESHARK 45

Oracle E-SBC 3820 45
Resetting the statistical counters, enabling logging and restarting the log files 45
Examining the log files 45
Through the Web GUI 46

APPENDIX A 47
Accessing the ACLI 47
ACLI Basics 47
Configuration Elements 50
Creating an Element 50
Editing an Element 51
Deleting an Element 51
Configuration Versions 51
Saving the Configuration 52
Activating the Configuration 53

APPENDIX B – SBC CONFIGURATION 54

Intended Audience

This document is intended for use by Oracle Systems Engineers, third party Systems Integrators, Oracle Enterprise customers and partners and end users of the Oracle Enterprise Session Border Controller (E-SBC). It assumes that the reader is familiar with basic operations of the Oracle Enterprise Session Border Controller 3820/4500 platforms.

Document Overview

This Oracle technical application note outlines the recommended configurations for the Oracle enterprise session border controller 3820 series for connecting AT&T's IP Flexible Reach service to Cisco Unified Call Manager (CUCM) 10.5 customers. The solution contained within this document has been certified on Oracle's Acme Packet OS SCZ 7.3p2.

Cisco Unified Call Manager provides industry-leading reliability, security, scalability, efficiency, and enterprise call and session management and is the core call control application of the collaboration portfolio. This reduces the cost and complexity of extending an enterprise's telephony system outside its network borders. Oracle Enterprise Session Border Controllers (SBCs) play an important role in SIP trunking as they are used by many ITSPs and some enterprises as part of their SIP trunking infrastructure.

This application note has been prepared as a means of ensuring that SIP trunking between CUCM 10.5, Oracle E-SBCs and IP Trunking services are configured in the optimal manner.

It should be noted that while this application note focuses on the optimal configurations for the Oracle ESBC in an enterprise CUCM 10.5 environment, the same SBC configuration model can also be used for other enterprise SIP trunking applications with a few tweaks to the configuration for required features. In addition, it should be noted that the SBC configuration provided in this guide focuses strictly on the CUCM Server associated parameters. Many SBC applications may have additional configuration requirements that are specific to individual customer requirements. These configuration items are not covered in this guide. Please contact your Oracle representative with any questions pertaining to this topic.

For additional information on CUCM 10.5, please visit http://www.cisco.com/c/en/us/products/unified-communications/unified-communications-manager-version-10-5/index.html

Introduction

Audience

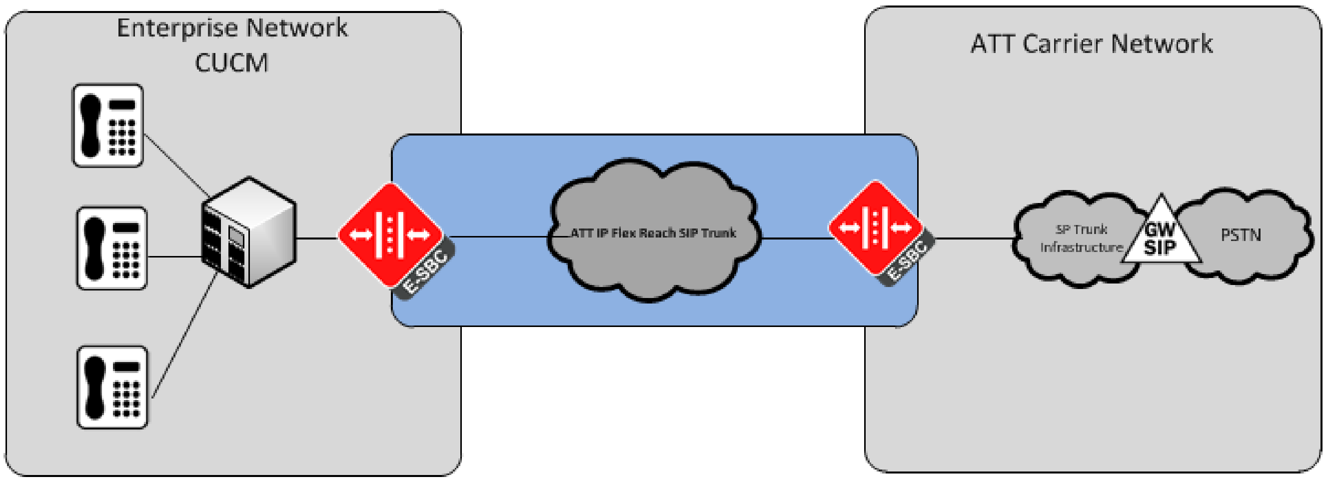
This is a technical document intended for telecommunications engineers with the purpose of configuring the Oracle Enterprise SBC and CUCM 10.5. There will be steps that require navigating the CUCM 10.5 server configuration as well as the Acme Packet Command Line Interface (ACLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

Requirements

- Working configuration and functioning Cisco UCM 10.5
- CUCM configuration would include Cisco Unity Connection for Voicemail and Auto Attendant based enterprise functions
- Cisco soft phone and hard phones connected/registered to the CUCM server
- Oracle Enterprise Session Border Controller (hereafter Oracle E-SBC) 3820 series running ECZ7.3.0p2. Note: the configuration running on the SBC is backward/forward compatible with any release in the 7.2.0 stream against any platform including VME, 1100, 3820, 4500, 4600, 6300
- Oracle E-SBC having established SIP connectivity with CUCM on CPE side and ATT IP FR SIP trunk on PSTN side.

Architecture

The following reference architecture shows a logical view of the connectivity between CUCM and the SBC.

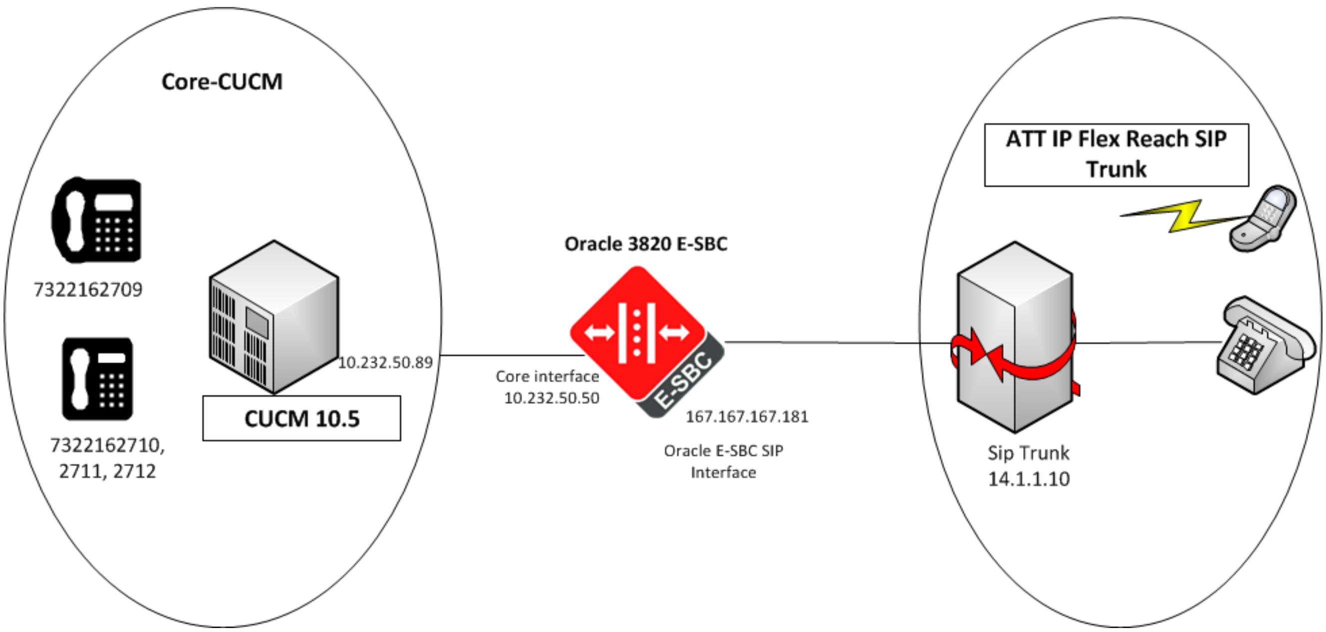


Area on left of the Oracle SBC brown box is the customer's on premise infrastructure, which includes the CUCM, DNS and CUC with the enterprise phones systems. Area on right of the SBC represents the service provider infrastructure which provides PSTN service via the SIP trunk. The SBC provides integration of these two environments over an IP network and provides security, service reachability, interoperability/normalization of SIP messages over the IP network. The CUCM and SBC are the edge components that form the boundary of the SIP trunk. The configuration, validation and troubleshooting of these two is the focus of this document and will be described in three phases:

- Phase 1 – Configuring the Oracle SBC
- Phase 2 – Configuring the Cisco Unified Call Manager v10.5
- Phase 3 - Configuring the Cisco Unity Connection server and integrating it with CUCM

Lab Configuration

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



Phase 1 – Configuring the Oracle Enterprise SBC

In this section we describe the steps for configuring an Oracle Enterprise SBC, formally known as an Acme Packet Net-Net Session Director ("SBC"), for use with Cisco Unified Call manager v10.5 in an ATT IP Flex Reach SIP Trunk service.

In Scope

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

Note that Oracle offers several models of SBC. This document covers the setup for the 3820 platform series running Net-Net OS ECZ7.3.0 or newer. If instructions are needed for other Oracle SBC models, please contact your Oracle representative.

Out of Scope

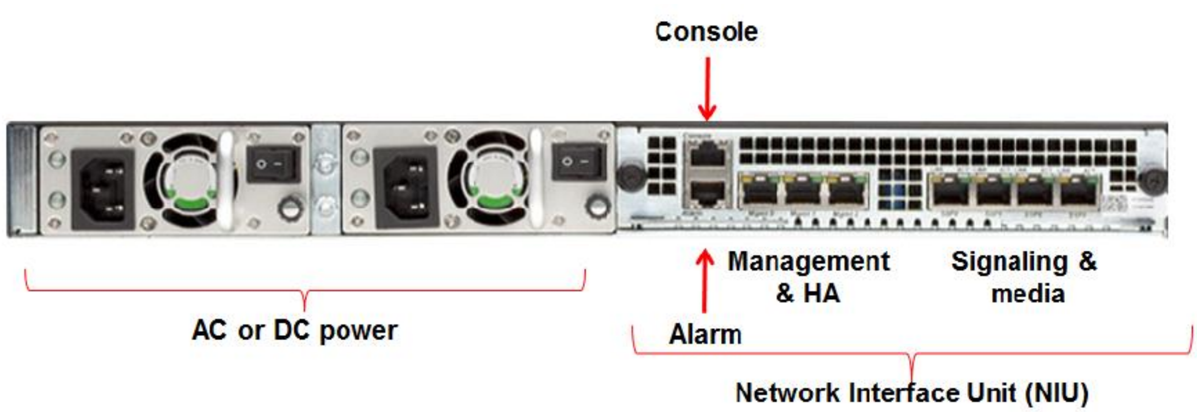
- Configuration of Network management including SNMP and RADIUS

What you will need

- Serial Console cross over cable with RJ-45 connector
- Terminal emulation application such as PuTTY or HyperTerm
- Passwords for the User and Superuser modes on the Oracle SBC
- IP address to be assigned to management interface (Wancom0) of the SBC - the Wancom0 management interface must be connected and configured to a management network separate from the service interfaces. Otherwise the SBC is subject to ARP overlap issues, loss of system access when the network is down, and compromising DDoS protection. Oracle does not support SBC configurations with management and media/service interfaces on the same subnet.
- IP address of the CUCM SIP interface facing the SBC
- IP addresses to be used for the SBC internal (CUCM facing) and external facing ports (Service Interfaces)
- IP address of the next hop gateway in the ATT IP Flex Reach network
- IP address of the enterprise DNS server

Configuring the SBC

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



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

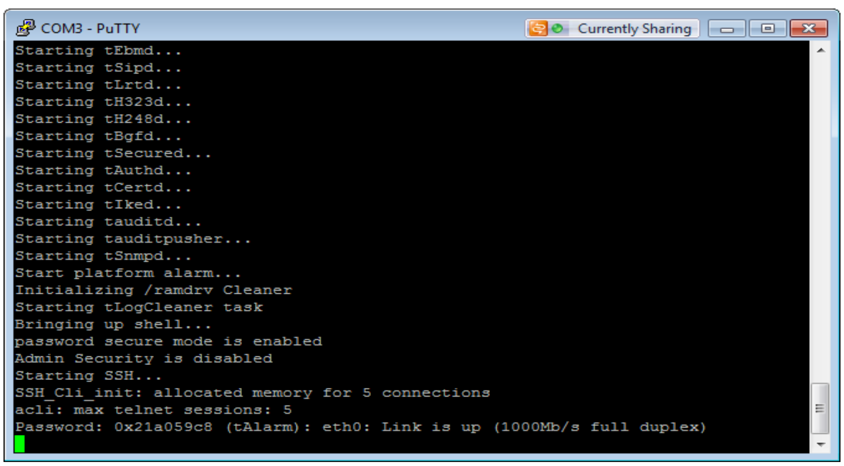
All commands are in bold, such as **`configure terminal`**; parameters in bold red such as CUCM-ATTFlexreach are parameters which are specific to an individual deployment. **Note:** The ACLI is case sensitive.

Establish the serial connection and logging in the SBC

Confirm the SBC is powered off and connect one end of a straight-through Ethernet cable to the front console port (which is active by default) on the SBC and the other end to console adapter that ships with the SBC, connect the console adapter (a DB-9 adapter) to the DB-9 port on a workstation, running a terminal emulator application such as PuTTY. Start the terminal emulation application using the following settings:

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

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



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

CUCM-ATTFlexreach#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

```
CUCM-ATTFlexreach#(configure)bootparam
'.' = clear field;  '-' = go to previous field;  q = quit
boot device              : eth0
processor number         : 0
host name                : acmesystem
file name                : /boot/ EZ730p2.32.bz --- >location where the
software is loaded on the SBC
inet on ethernet (e)     : 172.18.255.104:ffffff80 --- > This is the ip
address of the management interface of the SBC, type the IP address and
mask in hex
inet on backplane (b)    :
host inet (h)            :
gateway inet (g)         : 172.18.0.1 --- > gateway address here
user (u)                 : vxftp
ftp password (pw) (blank = use rsh)      : vxftp
flags (f)                :
target name (tn)         : CUCM-ATTFlexreach
startup script (s)       :
other (o)                :
```

Configuring the SBC

The following section walks you through configuring the Oracle Communications Enterprise SBC configuration required to work with CUCM v10.5 and ATT's IP Flex Reach SIP Trunk server. In the configuration, the transport protocol used between the SBC and CUCM server is TCP and the SIP trunk is configured for UDP. The test plan requires G.729 on trunk therefore configuration describing that is included as well.

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

High Availability

The wancom1 and wancom 2 port which is on the rear panel of the SBC is used for the purpose of High Availability in the 3820. Please refer to the Oracle Enterprise Session Border Controller ECZ7.3.0 ACLI Configuration guide for more detailed update on High availability configuration. (http://docs.oracle.com/cd/E61547_01/index.htm)

The following section entails notable configuration highlights that pertain to an enterprise environment that deploys an Oracle E-SBC and CUCM v10.5 to work with ATT IP Flex Reach SIP trunk service. A full copy of the configuration that was used for this certification follows the section as well.

Configuration Highlights

The SBC configuration in general follows enterprise SIP trunk configuration, with a few additional elements specific to interworking with CUCM. These are outlined below. We have also configured some options and sip-manipulations which are specific to any Cisco deployment with the Oracle Enterprise SBC. They are explained below

Add SDP and Media-Profile

CUCM does not offer any SDP in outbound INVITEs, however, Oracle E-SBC has capability to add SDP to an INVITE and this can be achieve by setting the add-sdp-invite in sip-interface to invite.

```
sip-interface
        state                                   enabled
        realm-id                                ATT-Trunk
        description
        sip-port
                address                                 167.167.167.181
```

```
                port                                        5060
                transport-protocol                          UDP
                tls-profile
                allow-anonymous                             agents-only
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                                0
….
add-sdp-invite                              invite
```

Together with adding SDP, since ATT IP Flex Reach SIP Trunk server requires calls from CPE to offer G.729 codec as first choice on trunk side, therefore the SBC offers these when constructing the SDP. One can configure this by creating media-profiles in SBC

```
media-profile
        name                                        G729
        subname
        media-type                                  audio
        payload-type                                18
        transport                                   RTP/AVP
        clock-rate                                  0
        req-bandwidth                               0
        frames-per-packet                           0
        parameters                                  annexb=no
        average-rate-limit                          0
        peak-rate-limit                             0
        max-burst-size                              0
        sdp-rate-limit-headroom                     0
        sdp-bandwidth                               disabled
        police-rate                                 0
        standard-pkt-rate                           0
media-profile
        name                                        g729wAnnexB
        subname                                     annexb-yes
        media-type                                  audio
        payload-type                                18
        transport                                   RTP/AVP
        clock-rate                                  0
        req-bandwidth                               0
        frames-per-packet                           0
        parameters                                  annexb=yes
        average-rate-limit                          0
        peak-rate-limit                             0
        max-burst-size                              0
        sdp-rate-limit-headroom                     0
        sdp-bandwidth                               disabled
media-profile
        name                                        PCMA
        subname
        media-type                                  audio
        payload-type                                8
        transport                                   RTP/AVP
        clock-rate                                  0
```

```
        req-bandwidth                           0
        frames-per-packet                       0
        parameters
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 16:34:01
media-profile
        name                                    PCMU
        subname
        media-type                              audio
        payload-type                            0
        transport                               RTP/AVP
        clock-rate                              0
        req-bandwidth                           0
        frames-per-packet                       0
        parameters
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
```

Once configured, media-profiles can be referenced in the ATT trunk facing SIP interface on the SBC

```
sip-interface
        state                                   enabled
        realm-id                                ATT-Trunk
        description
        sip-port
                address                                 167.167.167.181
                port                                    5060
                transport-protocol                      UDP
                tls-profile
                allow-anonymous                         agents-only
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                            0
…
add-sdp-profiles                                G729
                                                g729wAnnexB
                                                PCMU
                                                PCMA
```

SIP Manipulations

The Oracle E-SBC helps resolve certain SIP interoperability issues and preferences on ATT trunk side by invoking one of its most strongest and robust features SIP Header Manipulation Rules (HMR). Below is a summary of the SIP manipulations used and their use cases in this project.

SIP HMR	Description
ChangePAI	Fixes P-Asserted-Identity header and Remote-Pary-ID header towards ATT Trunk
ChangeforPAIandNAT	PAI HMR, adds diversion header for CFU and provides topology hiding
NAT_IP_rel	Strips PCMU codec towards CUCM
ForPrivacyandcodec_rel	Outbound HMR on CUCM sip-interface - Strips PCMU codec towards CUCM and provides topology hiding
changetoanonymous	Modify From for Anonymous calls
AddDiversion	Add Diversion header for Call forwarding Unconditional scenario

The header manipulation rules listed in the above table are further elaborated:

Setting the correct P-Asserted-Identity and ensuring topology hiding in place

This sip-manipulation is applied as out-manipulationid on the ATT trunk facing sip-interface in E-SBC. ATT IP Flex Reach SIP trunk service requires the CPE to set P-Asserted-Identity and Remote-party-ID headers correctly to reflect SBC IP.

```
sip-manipulation
        name                                       ChangePAI
        description
        split-headers
        join-headers
        header-rule
                name                                       Storecontacthost
                header-name                                Contact
                action                                     store
                comparison-type                            pattern-rule
                msg-type                                   any
                methods                                    INVITE
                match-value
                new-value
                element-rule
                        name                                       storehost
                        parameter-name
                        type                                       uri-host
                        action                                     store
                        match-val-type                             any
                      comparison-type                            pattern-rule
                        match-value
                        new-value
```

```
        header-rule
                name                                    ModPAI
               header-name                             P-Asserted-Identity
                action                                  manipulate
                comparison-type                         boolean
                msg-type                                any
                methods                                 INVITE
                match-value
$Storecontacthost.$storehost.$0
                new-value
                element-rule
                        name                                    modhost
                        parameter-name
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          any
                      comparison-type                         pattern-rule
                        match-value
                        new-value
$Storecontacthost.$storehost.$0
        header-rule
                name                                    StoreFromuser
                header-name                             From
                action                                  store
                comparison-type                         pattern-rule
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    Storeuser
                        parameter-name
                        type                                    uri-user
                        action                                  store
                        match-val-type                          any
                      comparison-type                         pattern-rule
                        match-value
                        new-value
        header-rule
                name                                    ChangePAIuser
               header-name                             P-Asserted-Identity
                action                                  manipulate
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    moduser
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                    comparison-type                         case-sensitive
```

```
                        match-value
              new-value                             $StoreFromuser.$Storeuser.$0
        header-rule
                name                                        RPI_Header
                header-name                                 Remote-Party-ID
                action                                      manipulate
                comparison-type                             case-sensitive
                msg-type                                    any
                methods
                match-value
                new-value
                element-rule
                        name                                     changehostRPI
                        parameter-name
                        type                                        uri-host
                        action                                      replace
                        match-val-type                              any
                    comparison-type                             case-sensitive
                        match-value
                        new-value                                   $LOCAL_IP
```

One can use the ACME_TO_FROM_NAT_IP variable to ensure SBC is doing topology hiding and replacing host-portions in SIP URIs of From and To headers or outbound messages from the CPE.

In cases where a Diversion header needs to be added for inbound calls to CPE with call forwarding unconditional set to a 1-800 number on the PSTN, an additional sip manipulation needs to be added.

```
sip-manipulation
        name                                        AddDiversion
        description
        split-headers
        join-headers
header-rule
                name                                        checkfor800
                header-name                                 To
                action                                      manipulate
                comparison-type                             case-sensitive
                msg-type                                    request
                methods                                     INVITE
                match-value
                new-value
                element-rule
                        name
checuriuser
                        parameter-name
                        type                                        uri-user
                        action                                      store
                        match-val-type                              any
                        comparison-type                             pattern-
rule
                        match-value                                 8772888362
                        new-value
header-rule
                name                                        AddDiv
```

```
                header-name                                 Diversion
                action                                      add
                comparison-type                             case-sensitive
                msg-type                                    request
                methods                                     INVITE
                match-value
                new-value
sip:7322162709@167.167.167.181
                          (This value is added as an example only, actual
TN will be provided by the customer)
```

In that case, the full sip manipulation on the ATT trunk interface will be

```
sip-manipulation
        name                                        ChangeforPAIandNAT
        description
        split-headers
        join-headers
        header-rule
                name                                        changePAI
                header-name                                 From
                action                                      sip-manip
                comparison-type                             case-sensitive
                msg-type                                    any
                methods
                match-value
                new-value                                   ChangePAI
        header-rule
                name                                        forprivacy
                header-name                                 From
                action                                      sip-manip
                comparison-type                             case-sensitive
                msg-type                                    any
                methods
                match-value
               new-value                                   ACME_NAT_TO_FROM_IP
         header-rule
                name                                        Foradinfiv
                header-name                                 From
                action                                      sip-manip
                comparison-type                             case-sensitive
                msg-type                                    any
                methods
                match-value
                new-value                                   AddDiversion
```

Once configured, this sip-manipulation needs to be applied as an out-manipulationid on the ATT trunk facing sip-interface

```
sip-interface
        state                                       enabled
        realm-id                                    ATT-Trunk
        description
        sip-port
                address                                     167.167.167.181
```

```
                port                                        5060
                transport-protocol                          UDP
                tls-profile
                allow-anonymous                             agents-only
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                                0
…
out-manipulationid                              ChangeforPAIandNAT
```

SIP Manipulation to strip G711ulaw codec towards CUCM and topology hiding

This sip-manipulation is applied as an out-manipulation on the sip-interface facing CUCM.

```
sip-manipulation
        name                                        NAT_IP_rel
        description
        split-headers
        join-headers
        header-rule
                name                                        checkG729Sdp
                header-name                                 Content-Type
                action                                      store
                comparison-type                             pattern-rule
                msg-type                                    request
                methods                                     INVITE
                match-value
                new-value
                element-rule
                        name                                        checksdp
                   parameter-name                              application/sdp
                        type                                        mime
                        action                                      store
                        match-val-type                              any
                      comparison-type                             pattern-rule
          match-value                                 (\Rm=audio [0-9]{1,5}
RTP/AVP ([0-9]{1,4})*)([0-9 ]*)\b
                        new-value
                element-rule
                        name                                        checkcodec
                   parameter-name                              application/sdp
                        type                                        mime
                        action                                      store
                        match-val-type                              any
                      comparison-type                             pattern-rule
                   match-value                                 a=rtpmap:18.*\R
                        new-value
        header-rule
                name                                        fixSdpRequest
                header-name                                 Content-Type
                action                                      manipulate
                comparison-type                             boolean
```

```
                 msg-type                                        request
                 methods                                         INVITE
          match-value                                   $checkG729Sdp.$checkcodec
                 new-value
                 element-rule
                         name                                         removeG711
                    parameter-name                               application/sdp
                         type                                         mime
                   action                                       find-replace-all
                         match-val-type                              any
                       comparison-type                              pattern-rule
                     match-value                                  a=rtpmap:0.*\R
                         new-value
                 element-rule
                        name                                         mod100mLine
                    parameter-name                               application/sdp
                         type                                         mime
                   action                                       find-replace-all
                         match-val-type                              any
                       comparison-type                              pattern-rule
                       match-value
(m=audio.*RTP/AVP).*( 18).*( 100)\R
                     new-value                                    $1+$2+$3+$CRLF
                 element-rule
                        name                                         mod101mLine
                    parameter-name                               application/sdp
                         type                                         mime
                   action                                       find-replace-all
                         match-val-type                              any
                       comparison-type                              pattern-rule
              match-value
(m=audio.*RTP/AVP).*( 18).*( 101)\R
                     new-value                                    $1+$2+$3+$CRLF
```

Combine the above rule with the ACME_NAT_FROM_TO_IP to build the full SIP manipulation

```
sip-manipulation
        name                                    ForPrivacyandcodec_rel
        description
        split-headers
        join-headers
        header-rule
                name                                    doNATforCUCM
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
               new-value                               ACME_NAT_TO_FROM_IP
        header-rule
              name                                    Forrel_711codecstrip
                header-name                             From
                action                                  sip-manip
```

```
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               NAT_IP_rel
```

Reference this rule as out-manipulationid on the SIP interface facing CUCM

```
sip-interface
        state                                   enabled
        realm-id                                CUCM
        description
        sip-port
                address                                 10.232.50.50
                port                                    5060
                transport-protocol                      TCP
                tls-profile
                allow-anonymous                         all
                multi-home-addrs
                ims-aka-profile
        carriers
        trans-expire                            0
…
out-manipulationid                      ForPrivacyandcodec_rel
```

Note:

Stripping codec in SDP can be achieved using codec-policy feature of the SBC as well. Following is an example configuration and codec-policy is referenced in the realm configuration. In this case, it is required to strip G711ulaw codec when SIP messages are sent to CUCM, therefore codec-policy can be defined and referenced in the CUCM realm.

Sample below:

```
codec-policy
        name                               stripPCMU
        allow-codecs                       *PCMU:no
        order-codecs
```

```
realm-config
        identifier                              CUCM
        description
        addr-prefix                             0.0.0.0
        network-interfaces                      s1p0:0
        mm-in-realm                             disabled
        mm-in-network                           enabled
        ...
        dyn-refer-term                          disabled
        codec-policy                            stripPCMU
        codec-manip-in-realm                    disabled
        codec-manip-in-network                  enabled
        rtcp-policy
```

Media-manager configuration for Call/Hold/Resume scenarios

For Re-Invites to be processed in hold/resume scenarios, one will need to set anonymous-sdp to enabled and add option unique-sdp-id in media-manager configuration as show below:

```
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                                 unique-sdp-id
        red-flow-port                           1985
        red-mgcp-port                           1986
        red-max-trans                           10000
        red-sync-start-time                     5000
        red-sync-comp-time                      1000
        media-policing                          enabled
        max-signaling-bandwidth                 10000000
        max-untrusted-signaling                 100
        min-untrusted-signaling                 30
        app-signaling-bandwidth                 0
        tolerance-window                        30
        trap-on-demote-to-deny                  disabled
        trap-on-demote-to-untrusted             disabled
        syslog-on-demote-to-deny                disabled
        syslog-on-demote-to-untrusted           disabled
        rtcp-rate-limit                         0
        anonymous-sdp                           enabled
…
```

Removing Plus 1 for inbound calls to CUCM

Following configuration is to strip +1 from the regular NANP based numbering in SIP messages towards CUCM

```
translation-rules
        id                                      removetheplus1
        type                                    delete
        add-string
        add-index                               0
        delete-string                           +1
        delete-index                            0
```

Reference the translation-rule as a called rule (for incoming calls into CUCM) in session-translation configuration

```
session-translation
        id                                      removeplus1
        rules-calling
        rules-called                            removetheplus1

```

Finally, reference the session-translation as an out-translationid in session-agent defined for CUCM as shown below:

```
session-agent
        hostname                                10.232.50.89
        ip-address
        port                                    5060
        state                                   enabled
        app-protocol                            SIP
        app-type
        transport-method                        StaticTCP
        realm-id                                CUCM
        egress-realm-id
        description
        carriers
…
        out-translationid                       removeplus1
```

This completes the major configuration highlights from the testing. A fully copy of the E-SBC configuration is elaborated in the Appendix Section of this document.

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

The enterprise will have a fully functioning CUCM v10.5 installed and deployed for this certification. In addition to that, a separate server Cisco Unity Connection server will need to be installed and configured pointing to the main CUCM v10.5 topology in order to facilitate Voicemail and Auto Attendant scenarios.

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

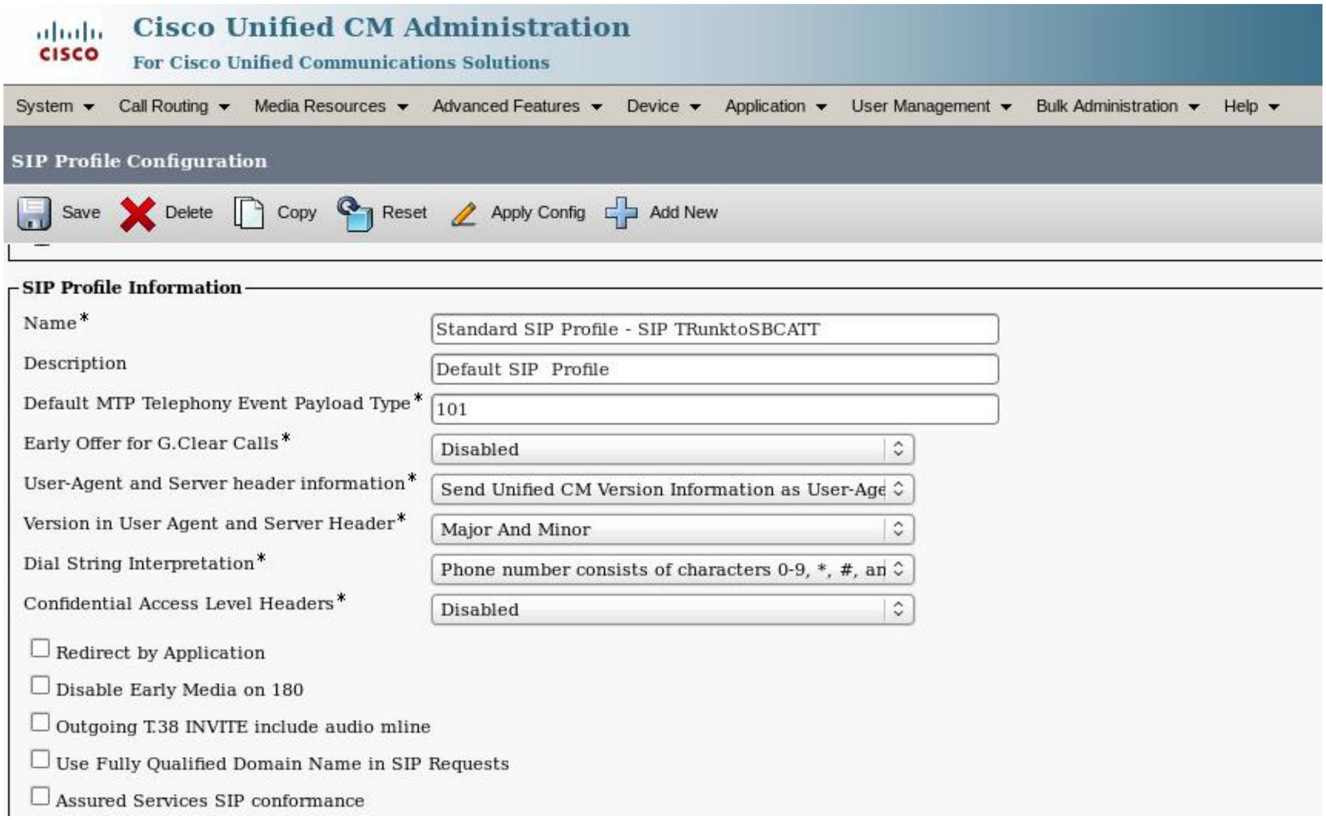
- Creating a SIP profile in CUCM and enabling OPTIONS ping to pro-actively monitor the SIP connectivity with the SBC
- Adding the SBC as a trunk to the CUCM infrastructure
- Creating a route pattern in the CUCM configuration to utilize the configured SBC trunk and route calls from CUCM to the SBC
- Additional configuration to add Directory Numbers, Phones to register to the CUCM and enabling a DHCP server for assigning IP addresses to Cisco phones
- ATT IP Flex reach requires G.729 for certain tests on the trunk side, configuration to support G.729 for the phones in CUCM
- Additional configuration including Cisco Unity Connection server installation and configuration for Voicemail and Auto Attendant scenarios which are covered in Phase 3 section of this document

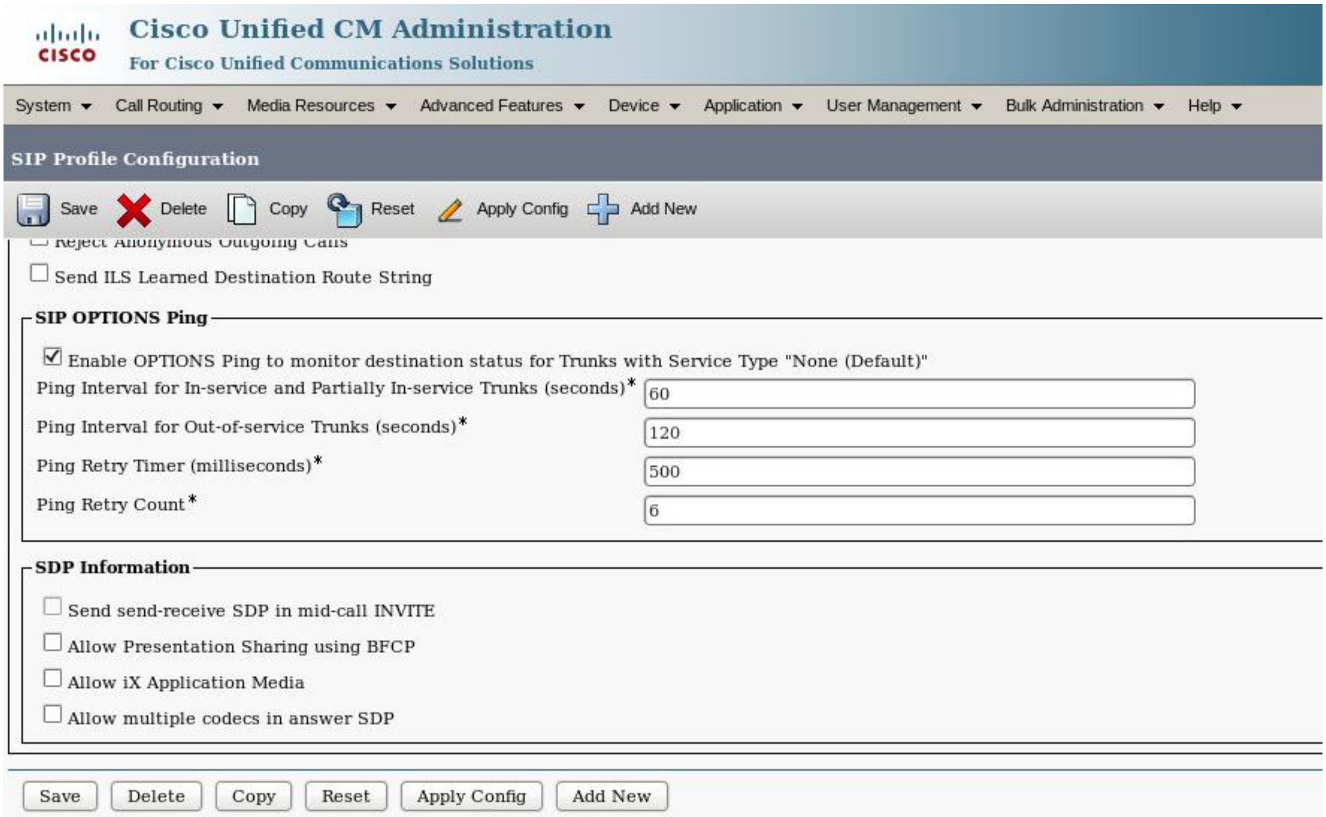
To add the SBC as a trunk in CUCM, we will need:

- IP address of the CUCM NIC facing SBC
- IP address and port of the sip interface of the SBC facing CUCM
- Access to the CUCM Web UI (http://ipaddress) and select Cisco Unified CM Administration from navigation drop down menu

Creating a SIP Profile in CUCM

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

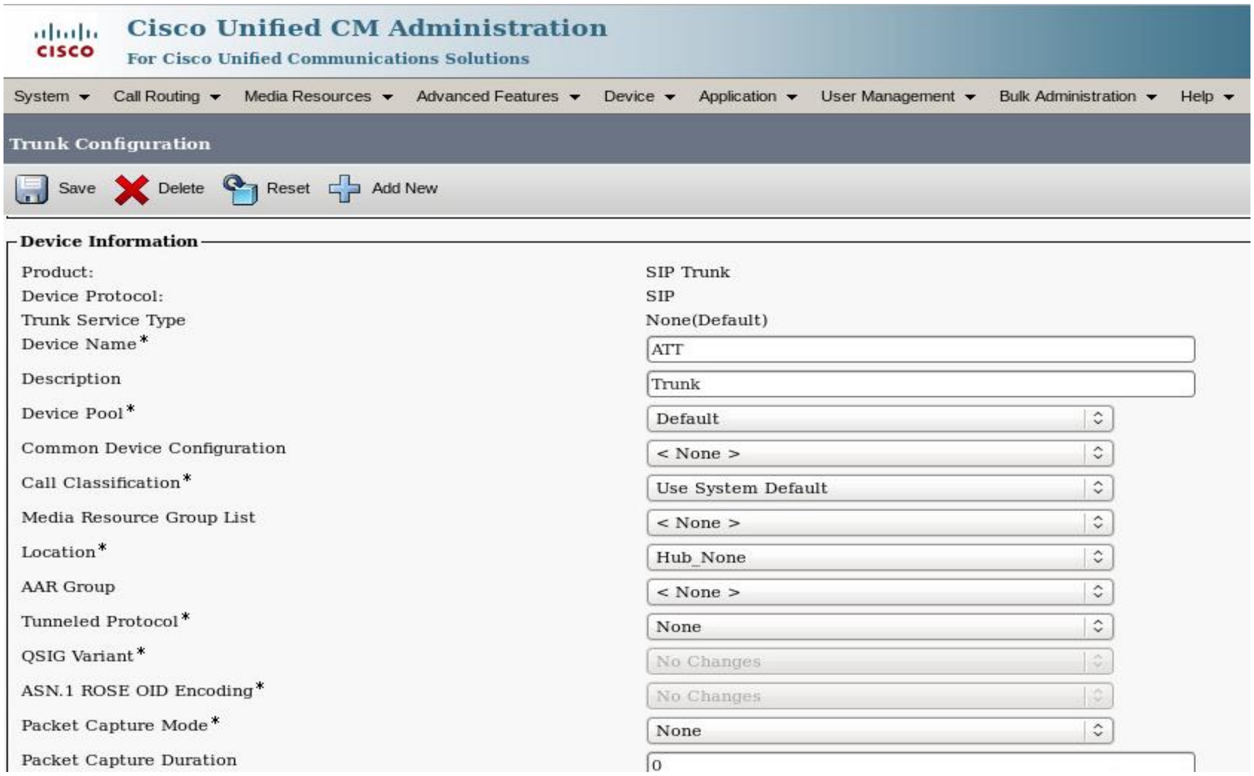


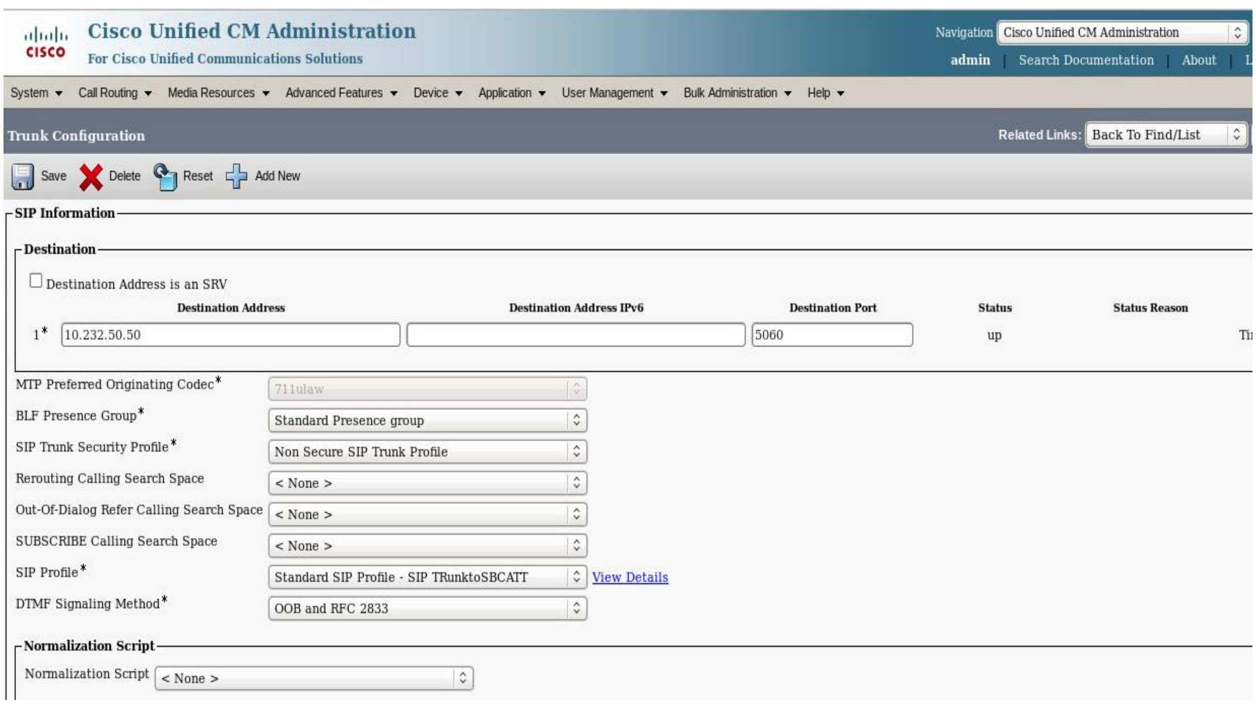


Adding the E-SBC as a trunk in CUCM

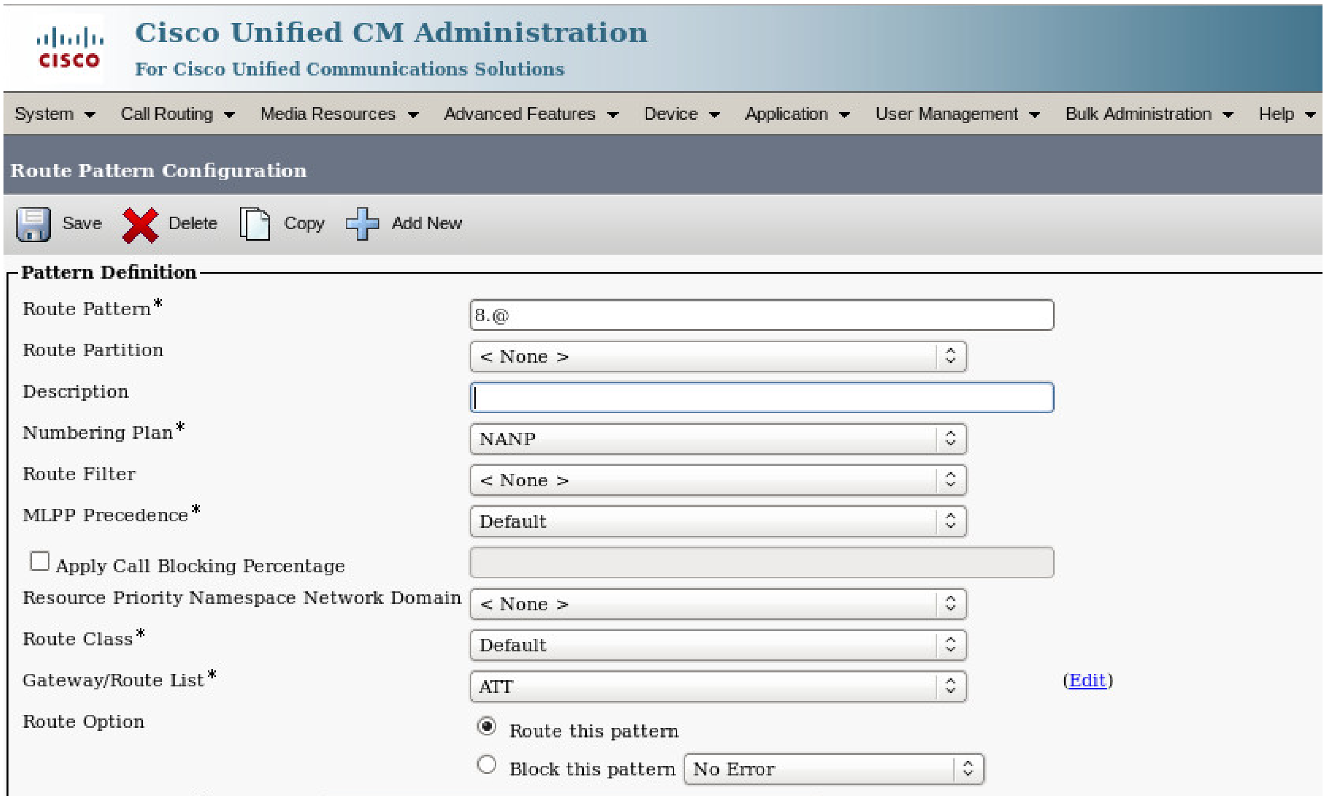
The following process details the steps to add the SBC as a trunk in CUCM Web UI

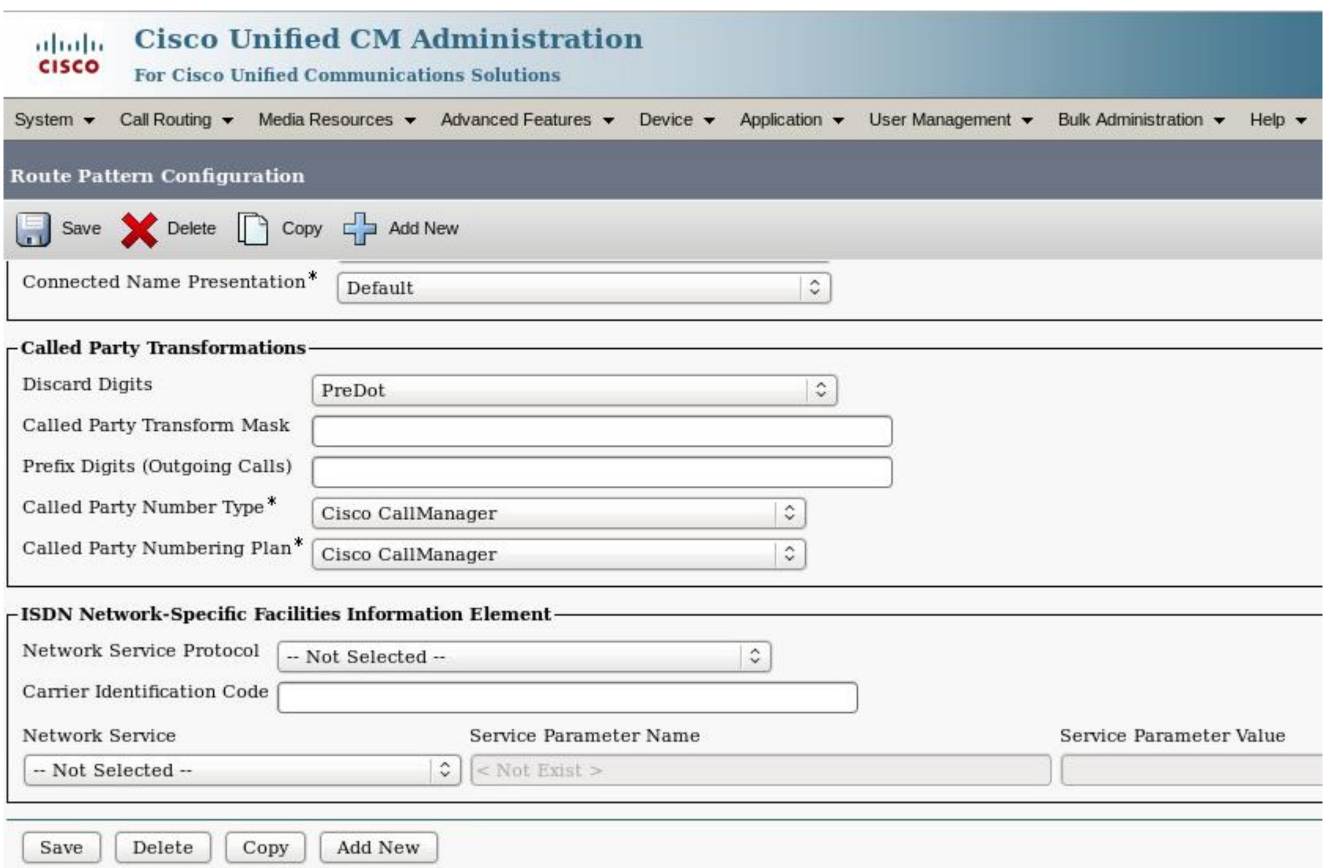
1. On the CUCM administration console (UI), maneuver to **Device --- > Trunk**. Click on New
2. Select SIP Trunk from the Trunk Type drop down menu and protocol will also be SIP
3. Let default of none be selected on the Trunk service type
4. Following 2 screenshots are the other settings to be configured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





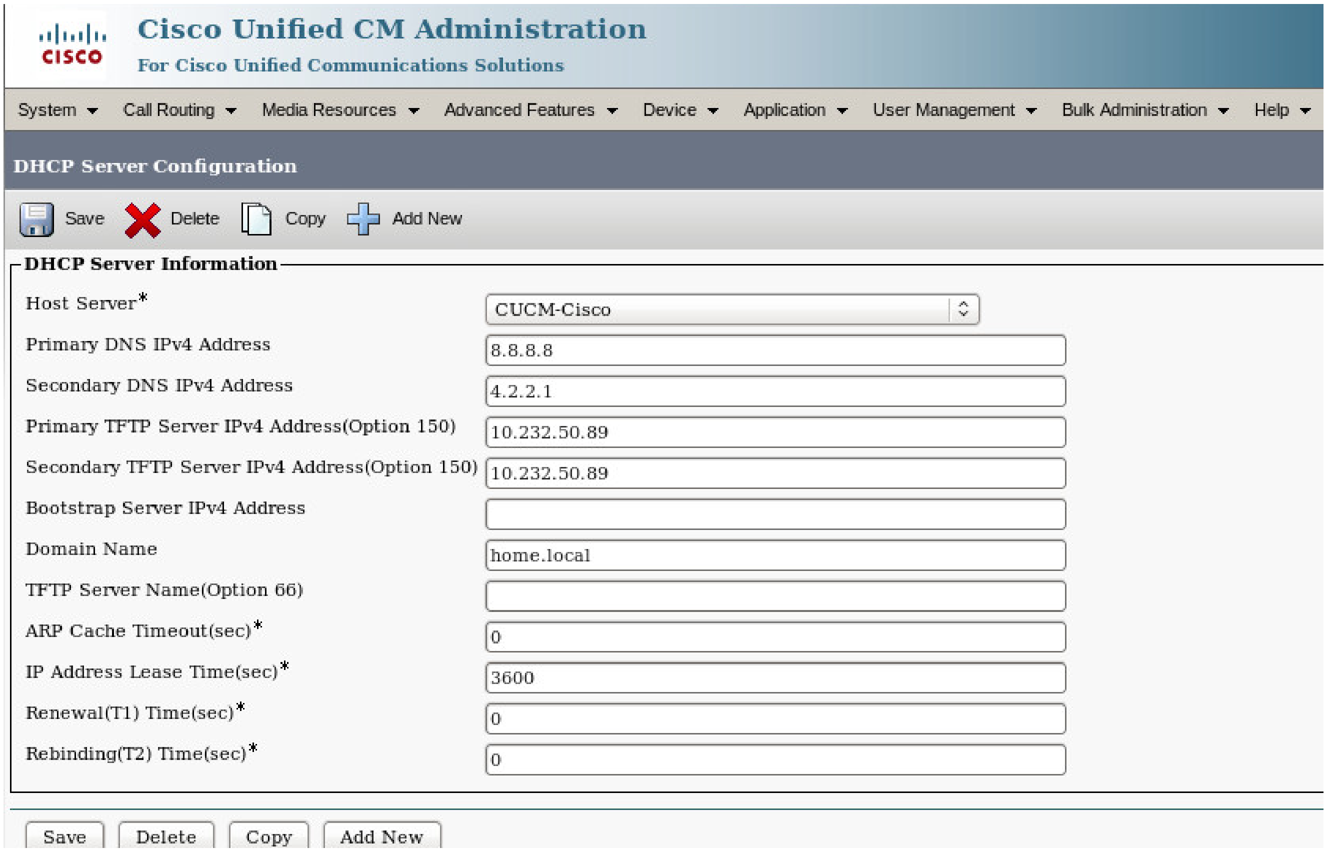
Route pattern in CUCM take the form of regular expressions to define specific routes and give flexibility in network design for dialing outbound calls from CUCM users to the PSTN via the E-SBC. A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that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


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Route Pattern Configuration
Save Delete Copy Add New
Pattern Definition
Route Pattern* 8.@
Route Partition < None >
Description
Numbering Plan* NANP
Route Filter < None >
MLPP Precedence* Default
Apply Call Blocking Percentage
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* ATT (Edit)
Route Option
Route this pattern
Block this pattern No Error

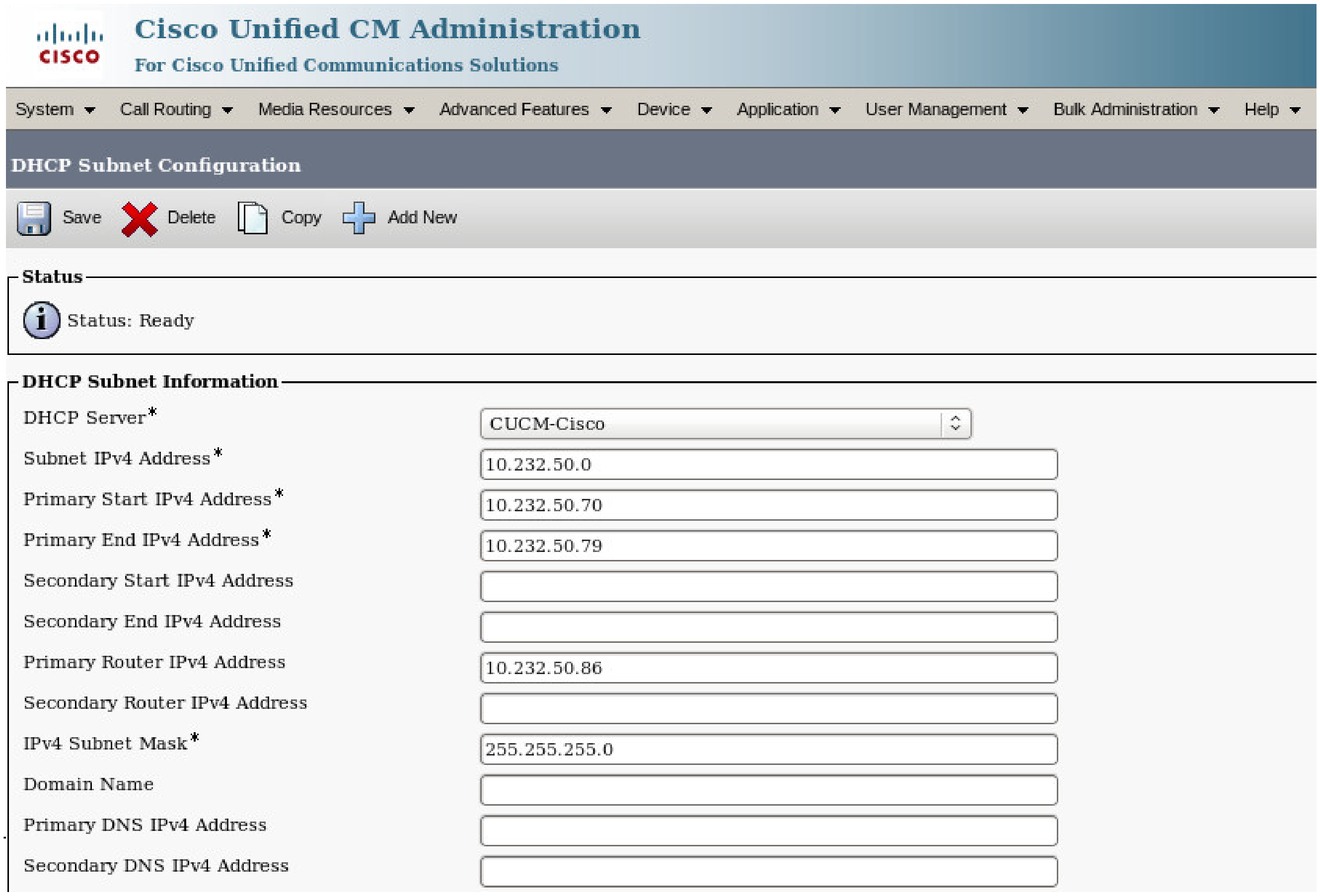

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Route Pattern Configuration
Save Delete Copy Add New
Connected Name Presentation* Default
Called Party Transformations
Discard Digits PreDot
Called Party Transform Mask
Prefix Digits (Outgoing Calls)
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager
ISDN Network-Specific Facilities Information Element
Network Service Protocol -- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save Delete Copy Add New

Adding DHCP server and subnet in CUCM

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

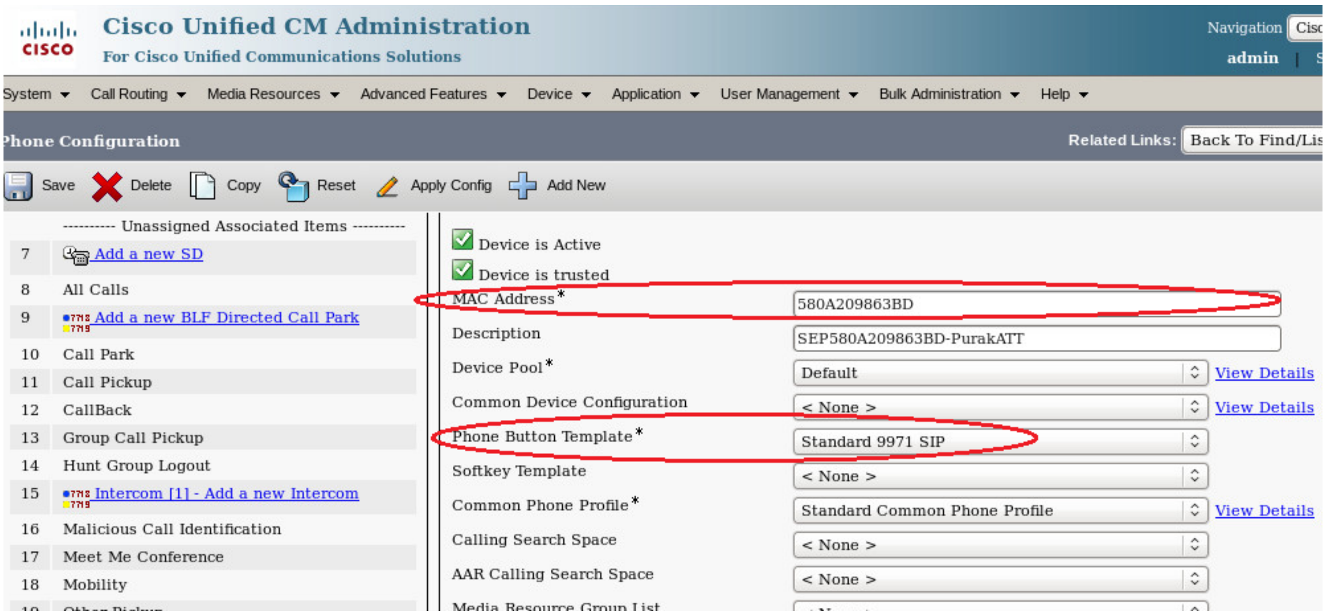


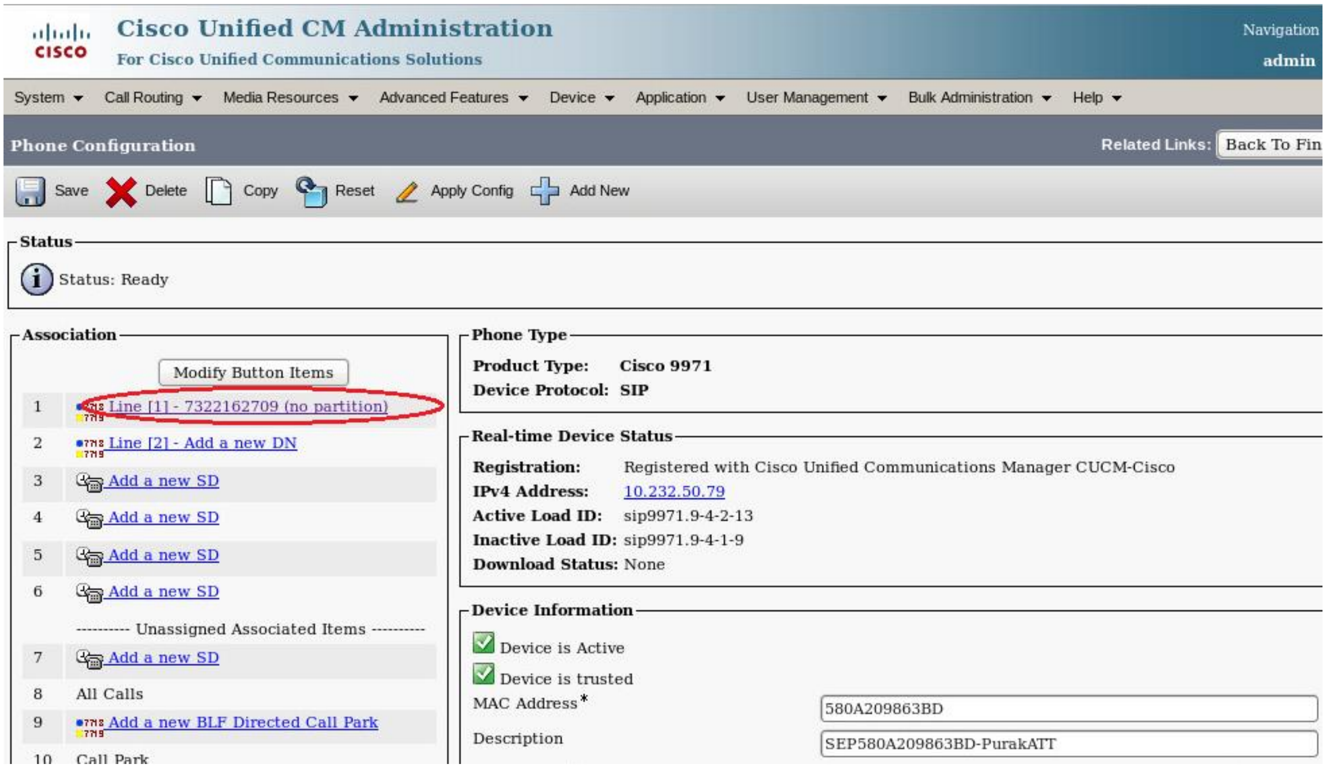
Add a DHCP subnet from the same menu path: System --- >DHCP --- > DHCP subnet



Adding Devices/Phones and configuring Directory numbers

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



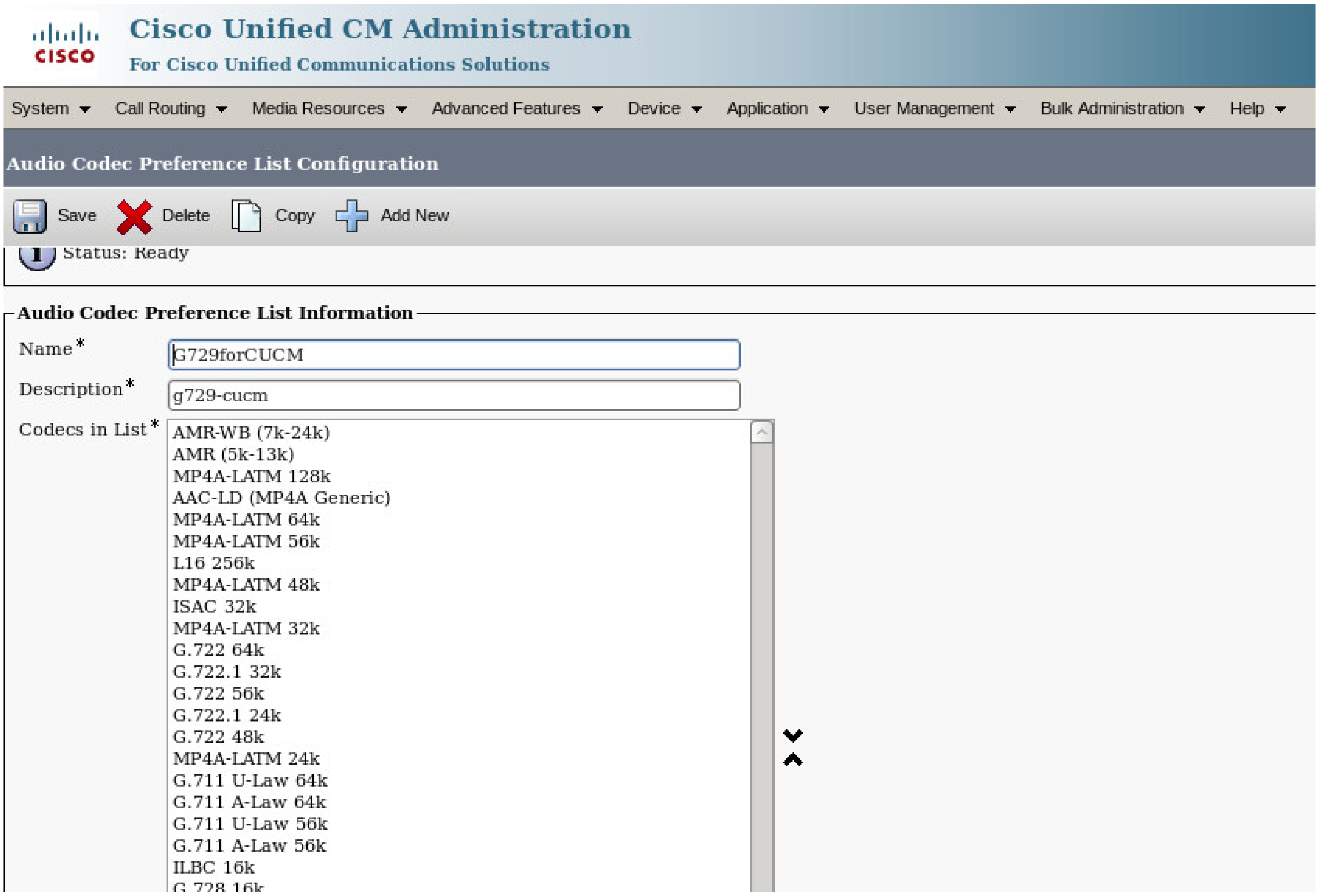


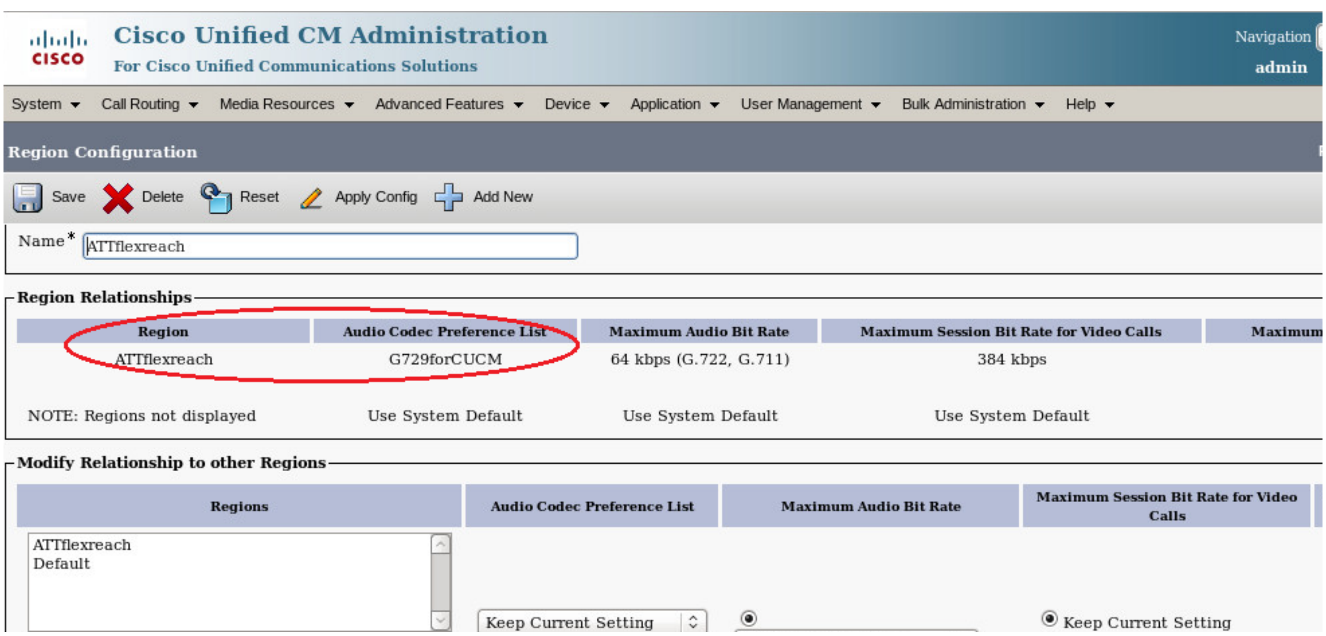
Creating & Assigning a Region and specifying use of G.729 codec in CUCM

ATT IP Flex Reach service certification covers AVPN transport and requires G.729 without annex b as choice of codec on the trunk side. CUCM 10.5 defaults to G.711 U law and therefore requires configuration to use G.729 codec. We can achieve this by the following steps:

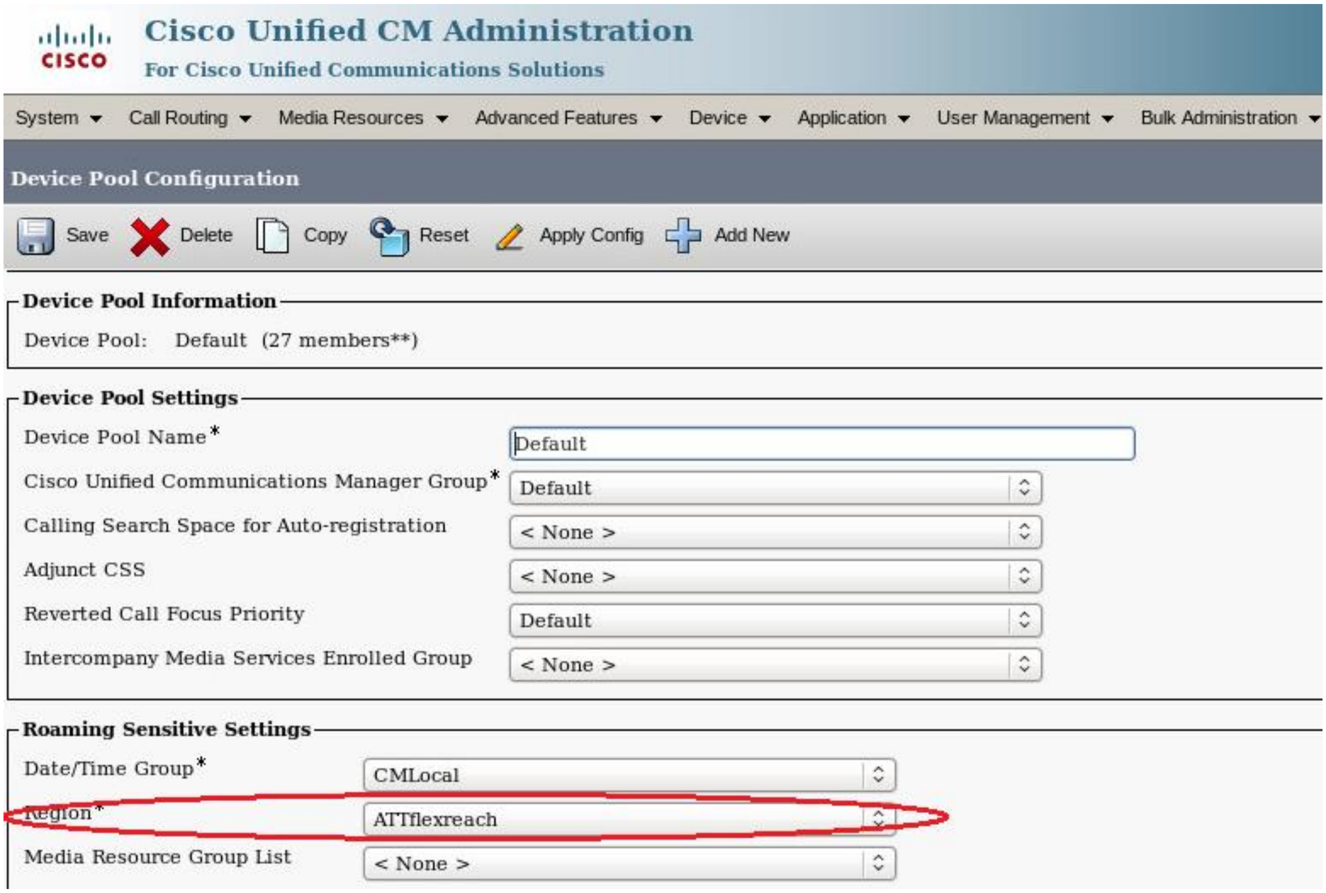
- Specifying Audio codec preference
- Assigning it to a defined Region and
- finally assigning the region to the default Device Pool

In CUCM administration console, use the System --- >Region Information --- > Audio Codec Preference List and System --- >Region Information --- > Region menu path to configure these (Add new). Below screenshots provide an overview of the same:





Next, assign the above defined region to the Device Pool to complete the configuration (System --- >Device Pool)



The CUCM is now ready to send/receive calls and establish SIP connectivity with the Oracle E-SBC. The document will cover additional configuration of CUCM and Cisco Unity Connection (CUC) server in Phase 3 of this document to elaborate and support Voicemail and Auto Attendant scenarios for an enterprise.

Phase 3 – Configuring the Cisco Unity Connection (CUC) Server and integrating it with CUCM

In this section we describe the steps for configuring Cisco Unity Connection server to support Voicemail and Auto Attendant scenarios and also integrate it with the existing CUCM topology to be fully functional.

In Scope

Configuring CUC and establishing connectivity with CUCM. Additional information on CUC is available at the following links:

http://docwiki.cisco.com/wiki/Virtualization_for_Cisco_Unity_Connection#Version_10.5.28x.29

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/integration/guide/cucm_sip/cucintcucmsip.pdf

If instructions are needed to install the physical server hosting the CUC application, please work with your Oracle representative.

Out of Scope

- Installing the CUC SW on a VM and network management to connect to CUCM topology

What you will need

- CUCM configured for inbound/outbound calls
- CUCM established connectivity with the E-SBC and trunk verified calls working
- CUCM VM built and installed the SW, ready to be configured
- IP address assigned to the CUC
- IP address of the CUCM administration console and knowledge of users/devices configured in CUCM

Configuring the Cisco Unity Connection Server for Voicemail

In addition to a fully functioning CUCM v10.5, a separate server known as CUC will be installed configured pointing to the main CUCM topology to support Voicemail and Auto attendant scenarios.

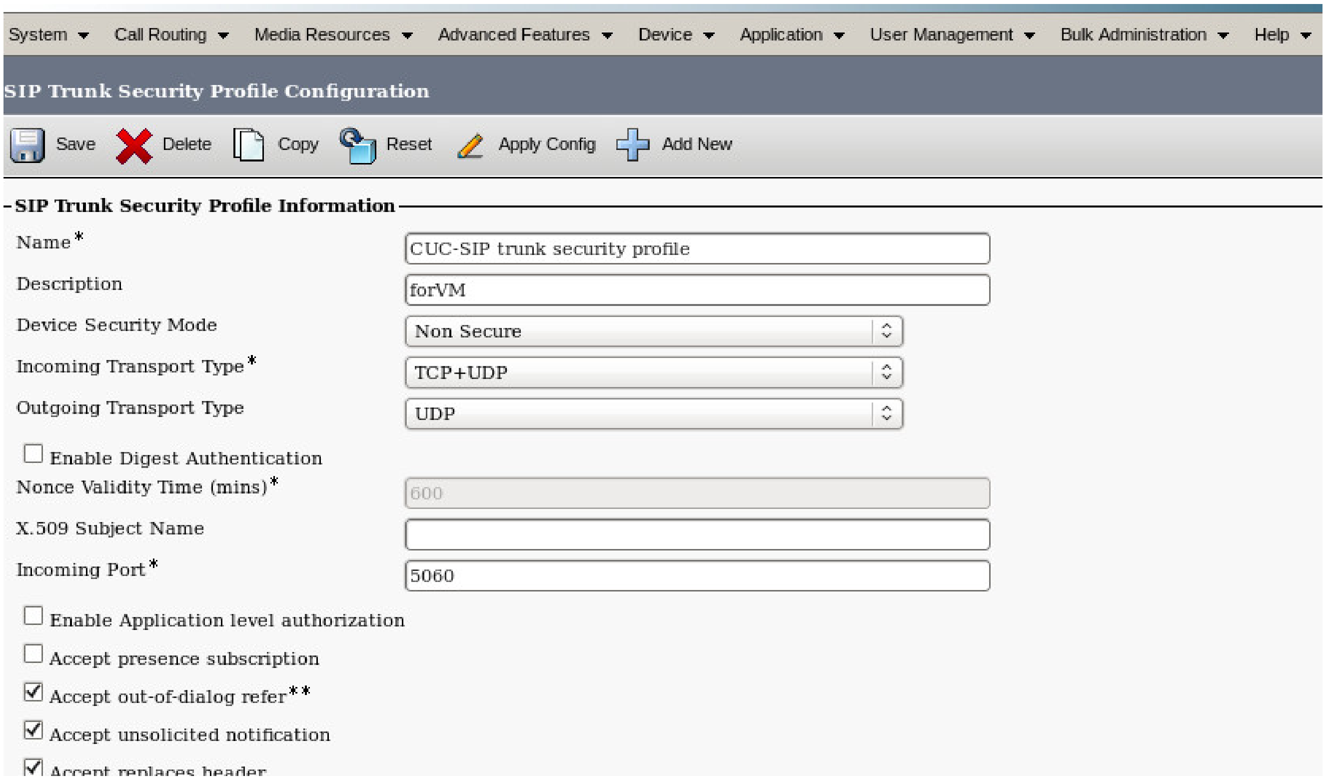
There are a few parts for configuring the CUC and pointing it to CUCM, these are enlisted below:

- Creating a SIP profile, SIP trunk security profile in CUCM and enabling OPTIONS ping to CUC
- Creating a separate trunk in CUCM and add CUC server address
- Create partition, calling search space for VM in CUCM
- Create Voicemail profile, voicemail ports, voicemail pilot, MWI in CUCM
- Create Route pattern to route locally from CUCM to CUC that hosts voicemail and Auto attendant services
- Create Phone system in CUC, add ports and reference Voicemail port
- Create mail box users in CUC

Create a SIP Profile and SIP trunk Security profile in CUCM

This will follow similar procedure as described for SBC integration in CUCM administration console under Device ---- > Device settings --- > SIP Profile. The SIP profile is named as CUC-SIP ProfileforVM.

The SIP trunk security profile is configured under System --- > SIP trunk security profile

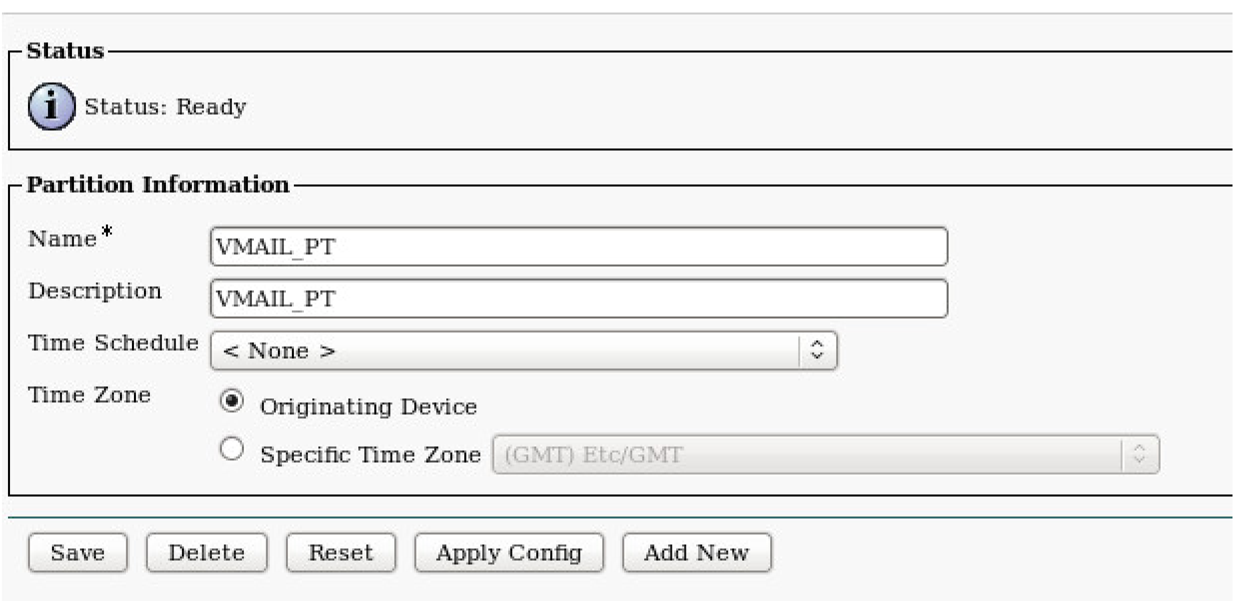


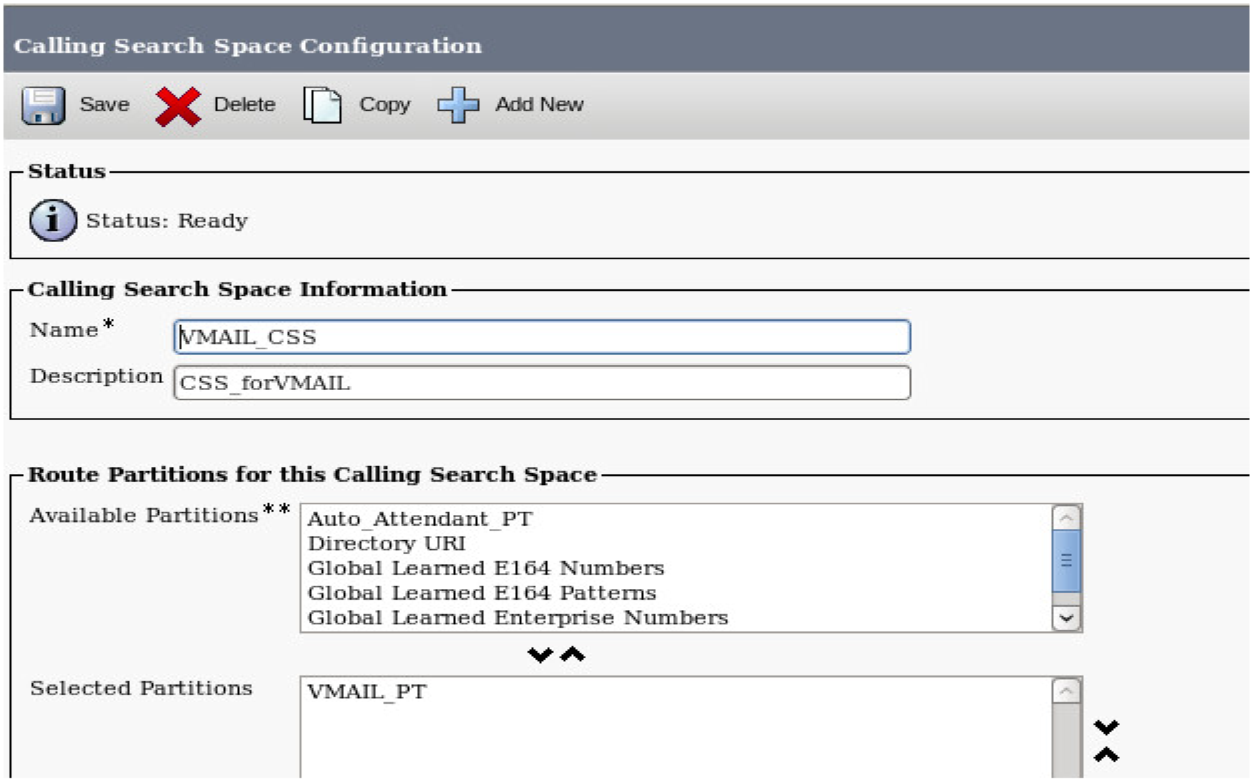
Create Trunk to connect to CUC

Similar to SBC integration in CUCM, we configure another trunk for CUCM to route Voicemail, auto attendant calls to CUC and named as CUC-VM-Trunk from Device ---- > Trunk

Create Calling Search space and Partition for VM in CUCM

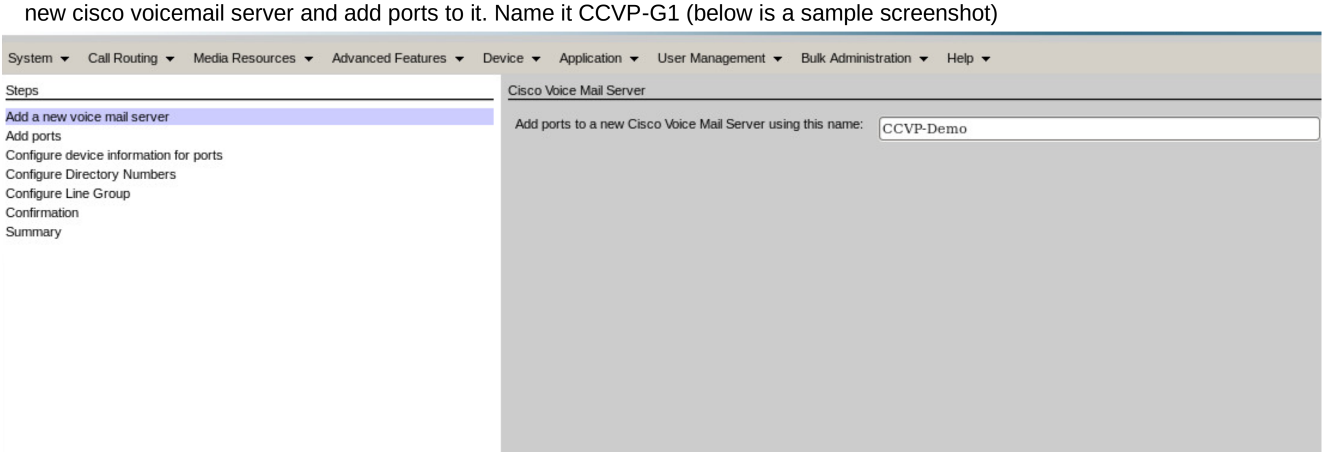
In CUCM administration console, use the Call Routing --- > Class of Control --- > Partition and Class of control --- > Calling Search space to define these for Voicemail and Auto attendant operation.





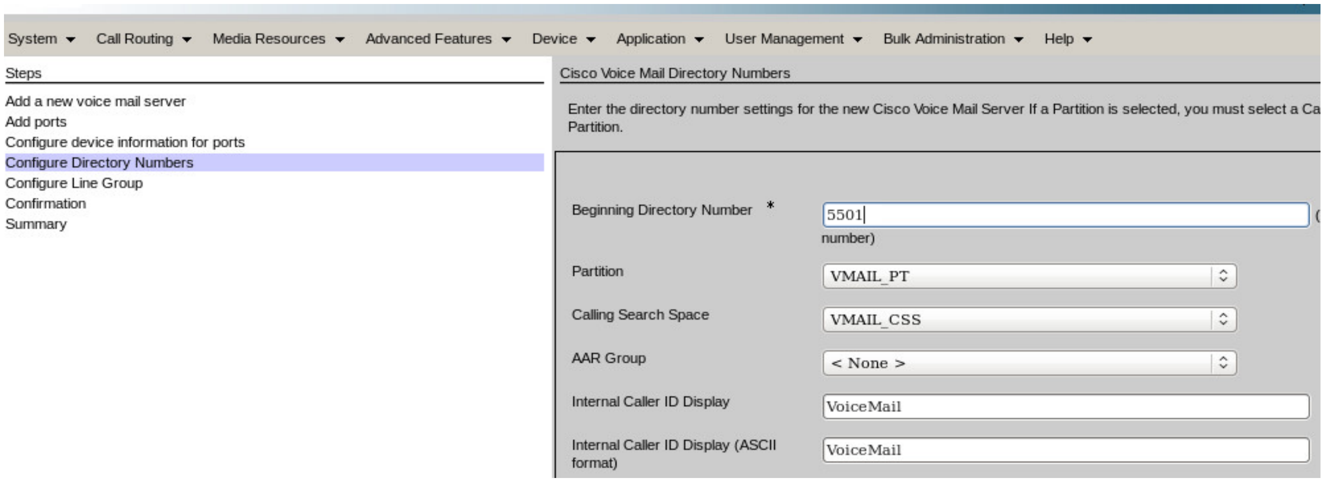
Create Voicemail Profile, ports, MWI

In CUCM administration console, use the Advanced Features ---- >Voicemail ---- > Cisco Voicemail Port Wizard. Select Create a new cisco voicemail server and add ports to it. Name it CCVP-G1 (below is a sample screenshot)

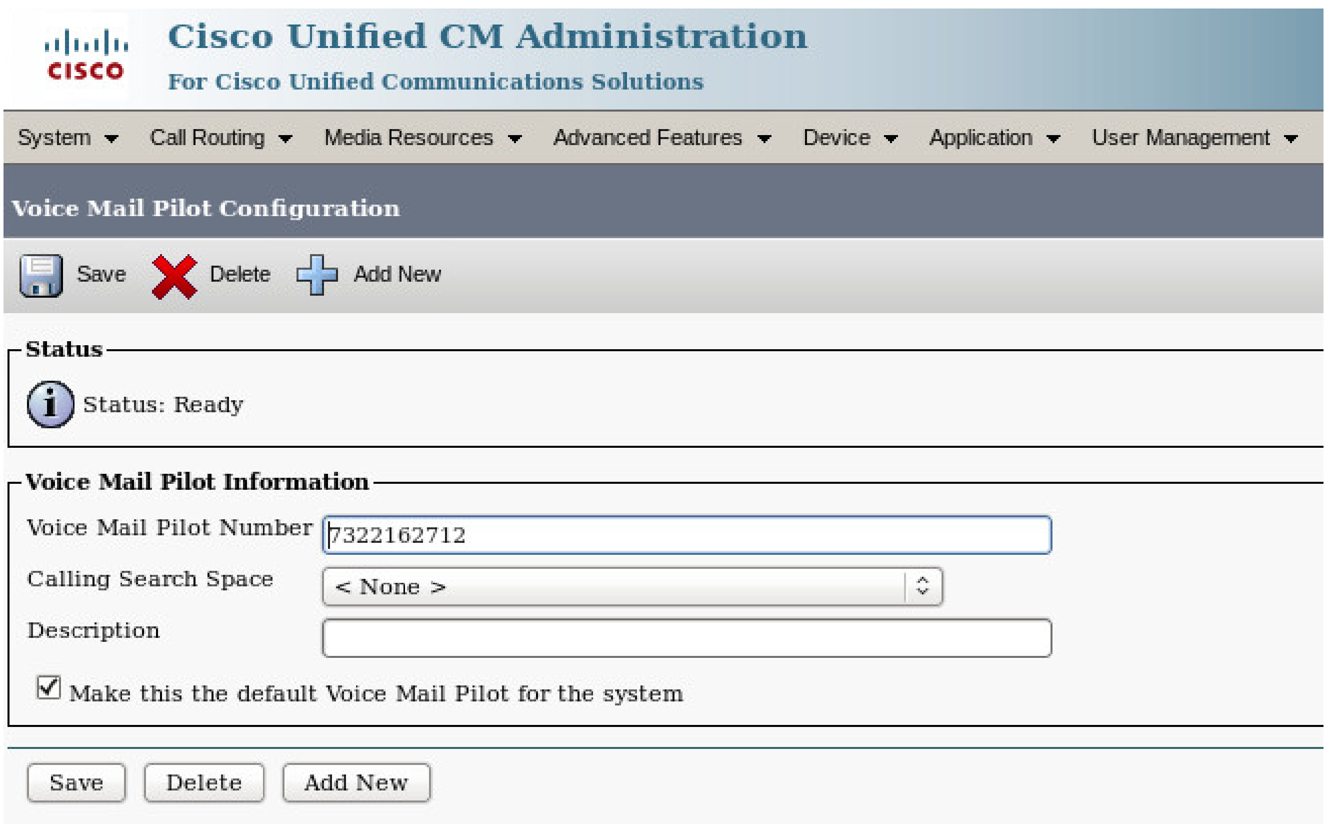


Next, select 2 ports to add. Next screen, select the defined VMAIL_CSS calling search space from drop down, location, non-secure voicemail port.

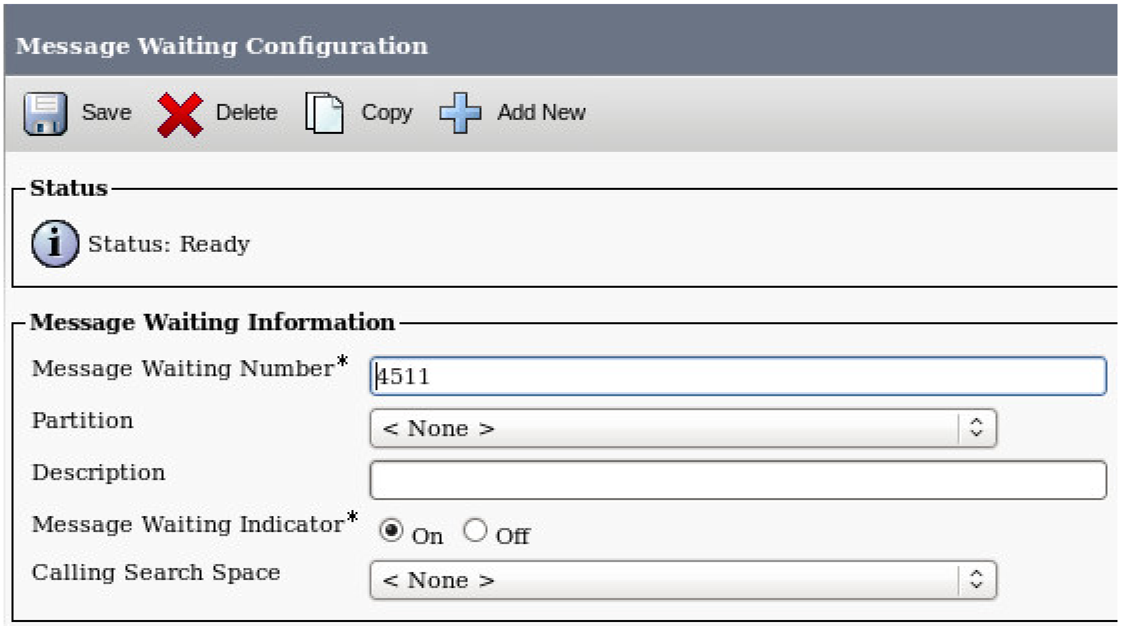
Next screen enter the beginning directory number and assign the correct partition

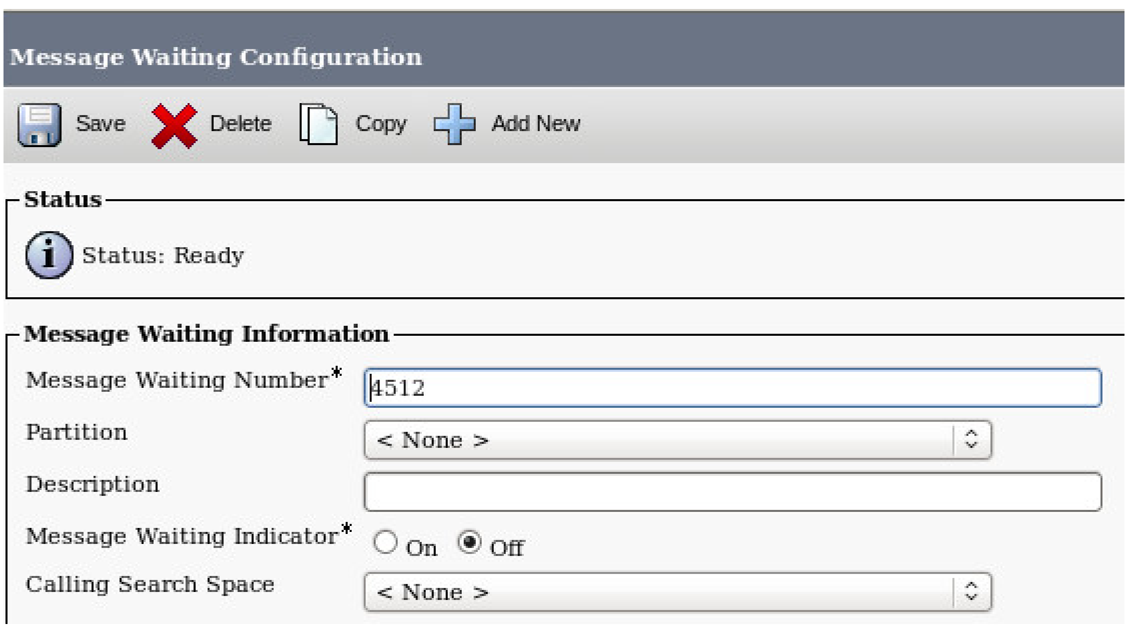


Next screen select adding the directory numbers to a new line group and the line group will be named the same as CCVP-G1. Click on finish to proceed to creating a Voicemail profile and voicemail pilot

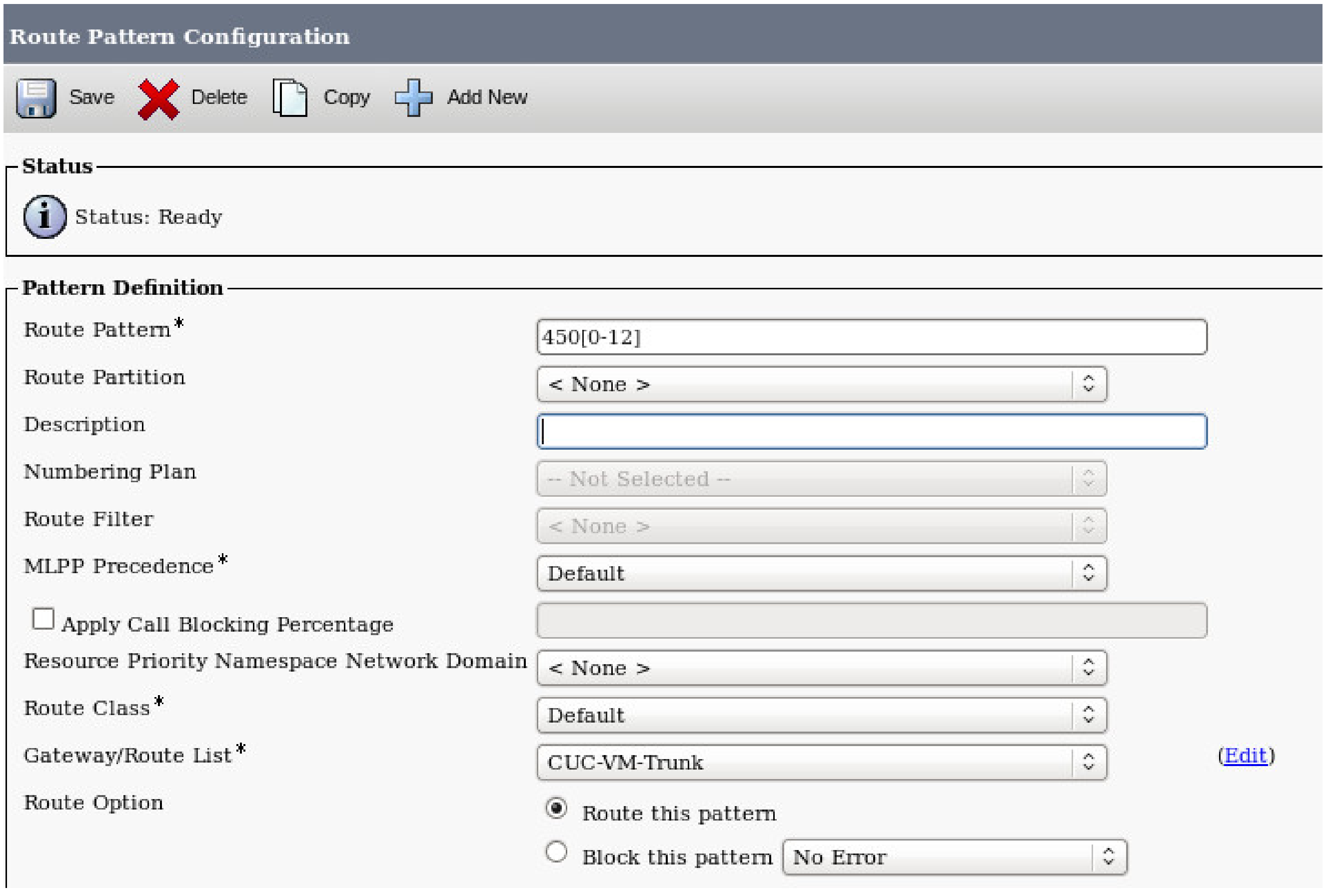


Once this is completed, one can create Message waiting indicators from the Advanced features --- > Voicemail --- > Message Waiting menu path in CUCM



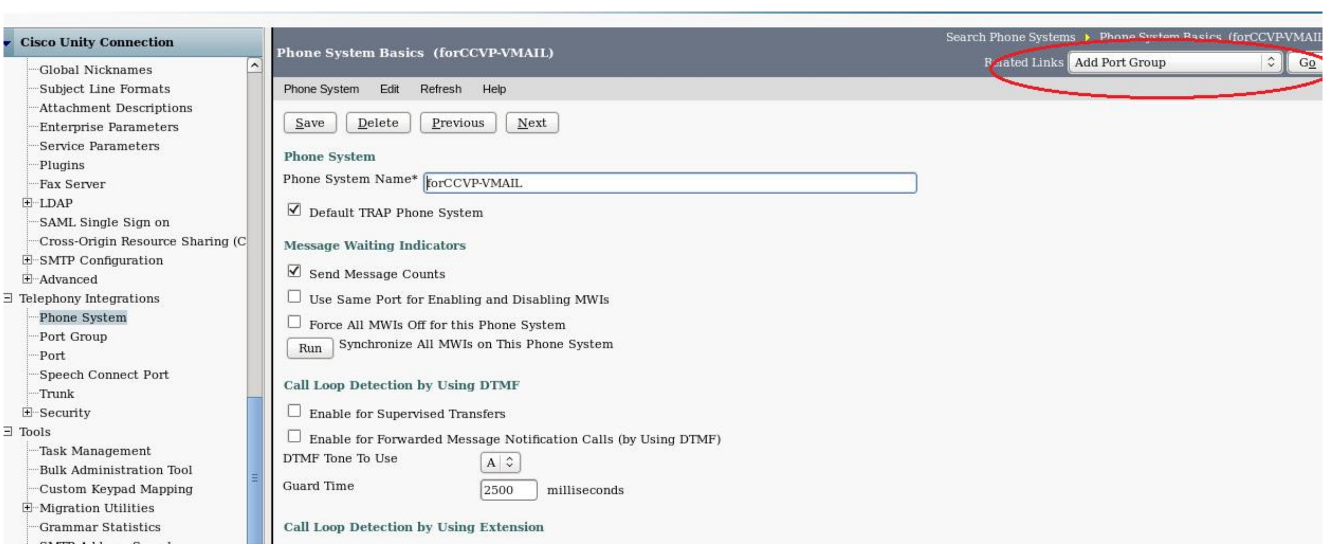


Next, we create route pattern to route locally for CUCM users to access their Voicemail box by dialing 4500 or to dial 4511/4512 to switch on/off MWI. This can be done using the Call Routing --- >Route/Hunt --- >Route Pattern menu path. Select the CUCM-VM-Trunk which was created between CUCM and CUC

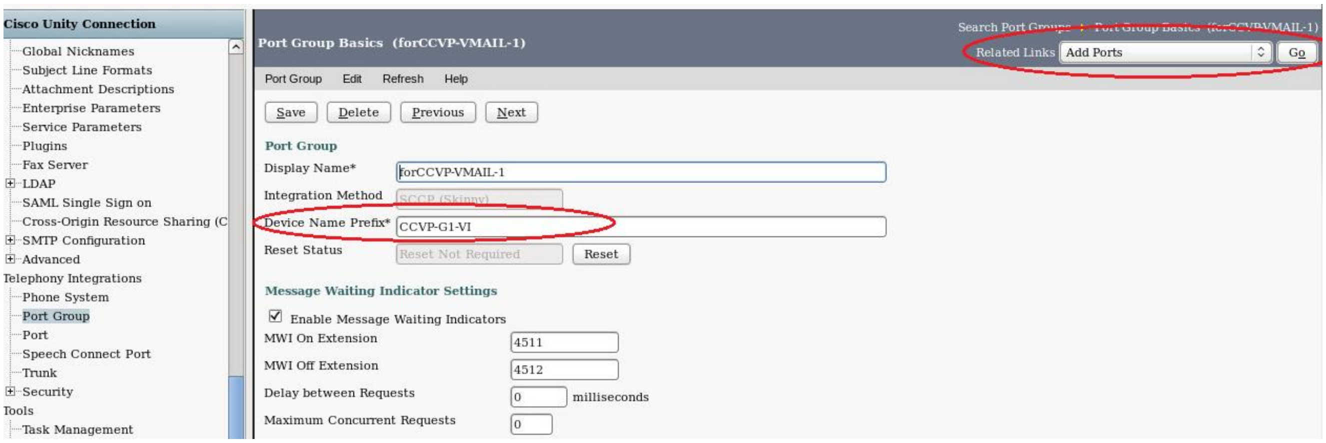


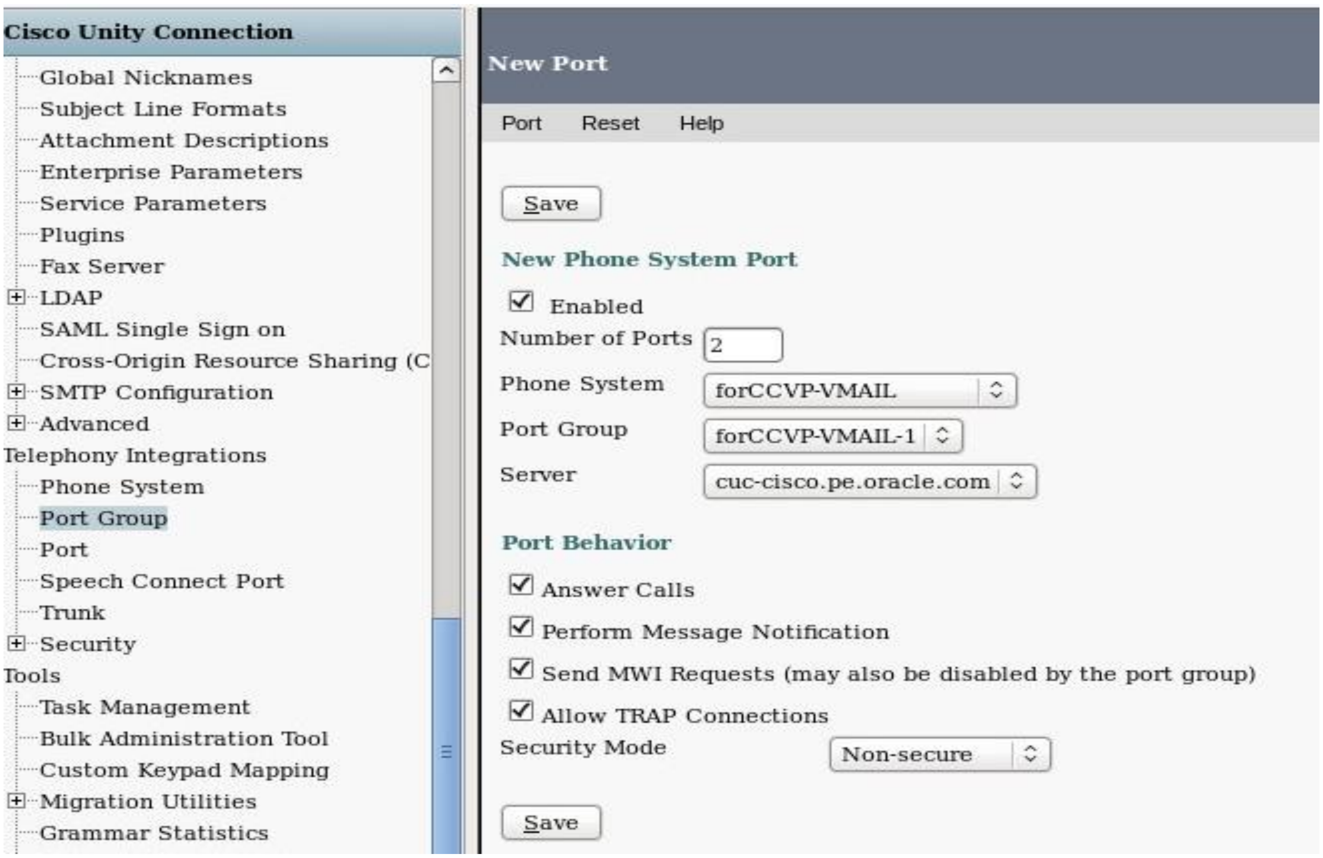
Create Phone system in CUC, add ports and reference Voicemail port

In CUC, use the Telephony Integrations --- > Phone Systems menu path to create a phone system. Add ports to it as well

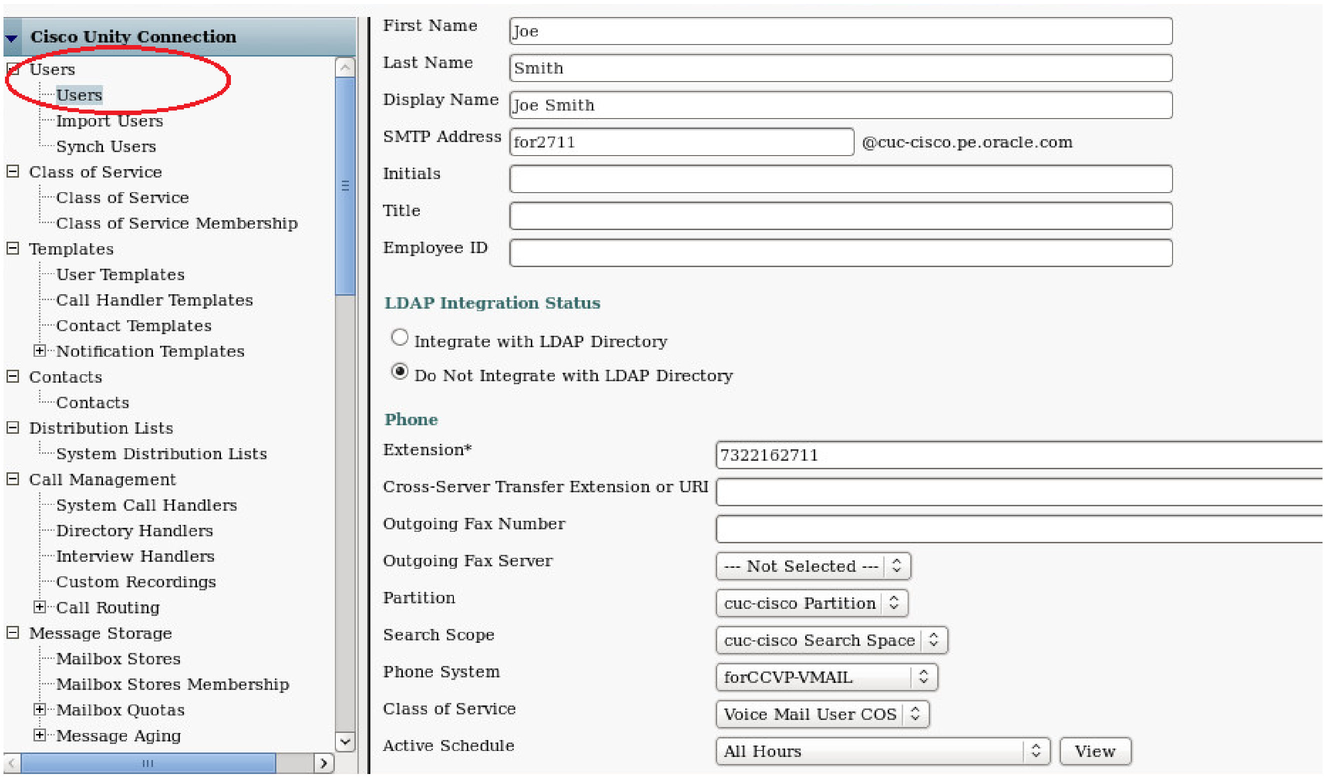


In add port group, use the same CCVP-G1 and add VI to name it CCVP-G1-VI. This will register the Voicemail ports to the CUC.


Cisco Unity Connection
Global Nicknames
Subject Line Formats
Attachment Descriptions
Enterprise Parameters
Service Parameters
Plugins
Fax Server
LDAP
SAML Single Sign on
Cross-Origin Resource Sharing (C
SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security
Tools
Task Management
Port Group Basics (forCCVP-VMAIL-1)
Related Links Add Ports Go
Port Group Edit Refresh Help
Save Delete Previous Next
Port Group
Display Name* forCCVP-VMAIL-1
Integration Method SCCP (Skinny)
Device Name Prefix* CCVP-G1-VI
Reset Status Reset Not Required Reset
Message Waiting Indicator Settings
Enable Message Waiting Indicators
MWI On Extension 4511
MWI Off Extension 4512
Delay between Requests 0 milliseconds
Maximum Concurrent Requests 0


Cisco Unity Connection
Global Nicknames
Subject Line Formats
Attachment Descriptions
Enterprise Parameters
Service Parameters
Plugins
Fax Server
LDAP
SAML Single Sign on
Cross-Origin Resource Sharing (C
SMTP Configuration
Advanced
Telephony Integrations
Phone System
Port Group
Port
Speech Connect Port
Trunk
Security
Tools
Task Management
Bulk Administration Tool
Custom Keypad Mapping
Migration Utilities
Grammar Statistics
New Port
Port Reset Help
Save
New Phone System Port
Enabled
Number of Ports 2
Phone System forCCVP-VMAIL
Port Group forCCVP-VMAIL-1
Server cuc-cisco.pe.oracle.com
Port Behavior
Answer Calls
Perform Message Notification
Send MWI Requests (may also be disabled by the port group)
Allow TRAP Connections
Security Mode Non-secure
Save

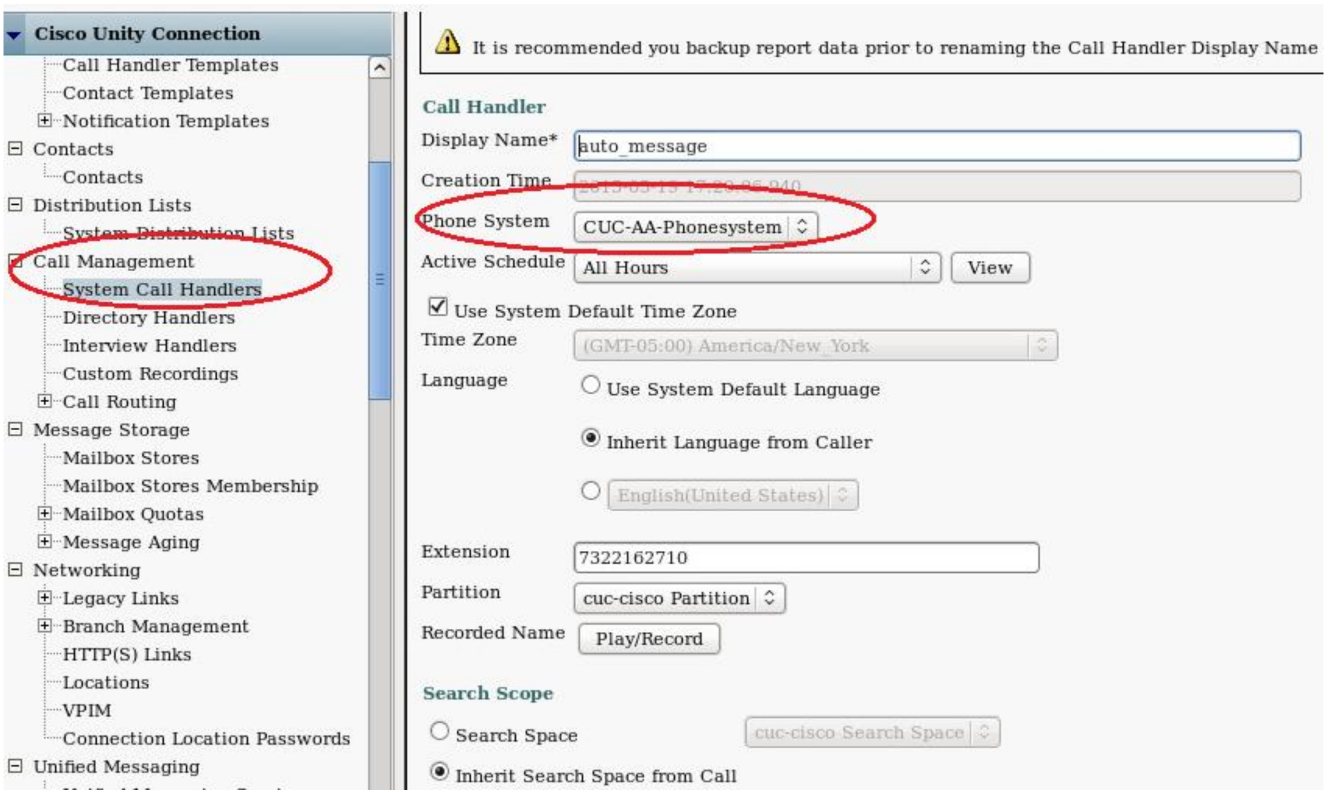
Create Mailbox and users in CUC



For auto Attendant scenario, in addition to the steps described above, one needs to define System call handlers in CUC and define CTI route point in CUCM to complete the integration.

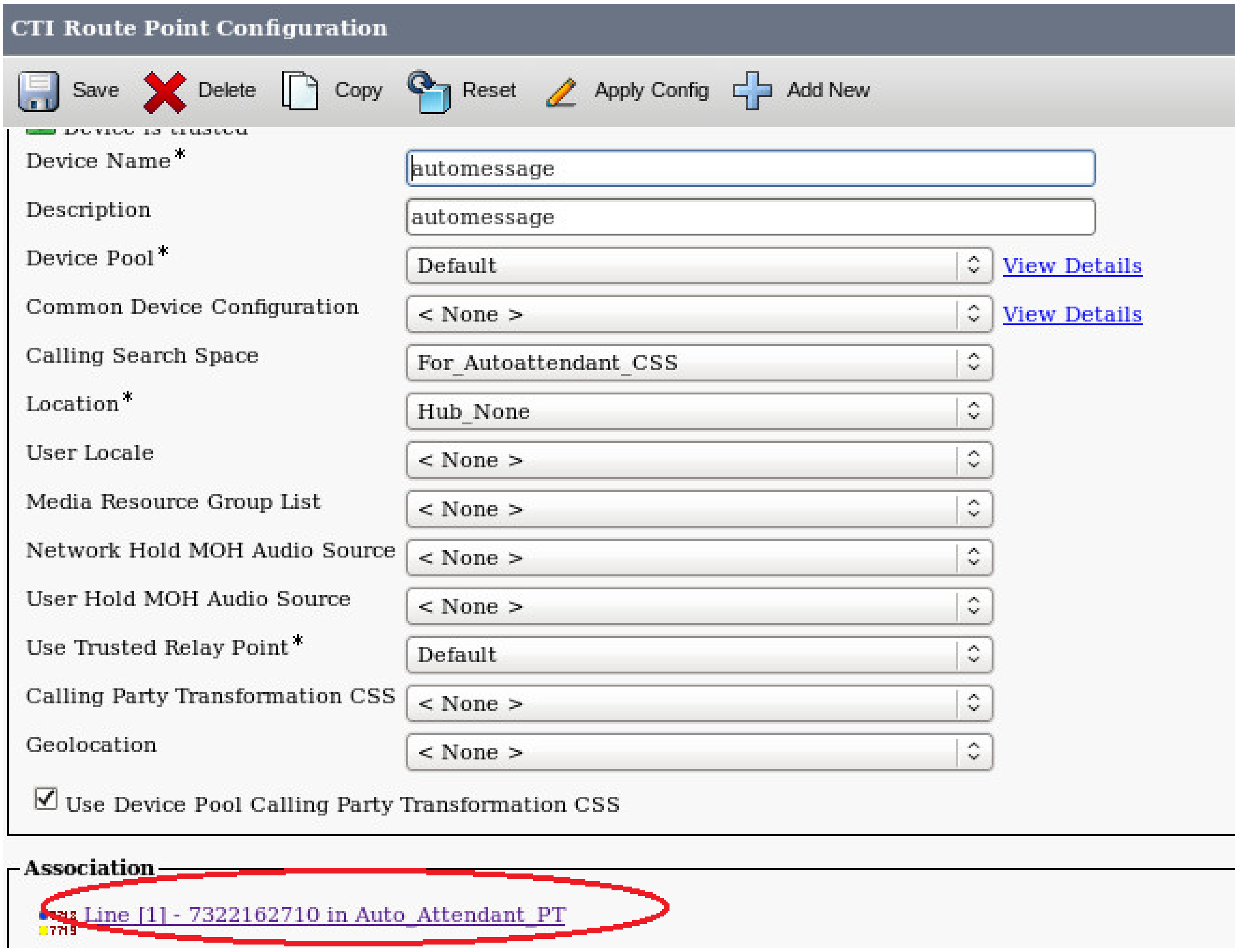
A new phone system, calling search space, partition, Voicemail pilot is defined for Auto attendant scenario to differentiate from Voicemail services.

Creating System Call Handlers in CUC



Define CTI Route Point in CUCM for AA

Use the Devices --- > CTI Route point menu path in CUCM administration console to define a CTI route point and assign it a DN. Sample screenshot provided below



With this, the configuration in CUCM and CUC is complete and one should be able to place calls that bounce to voicemail or interact with the Auto Attendant in CUCM

Test Summary

A comprehensive test plan was executed per ATT test specifications and call flows. For a copy of full test report, please contact your Oracle Sales account team.

Troubleshooting Tools

If you find that you are not able to complete calls or run into issues when going through the test plan, there are a few tools and methodologies available in Cisco Unified serviceability console, Oracle SBC logging and tracing which may be of assistance. In this section we will provide a list of tools which you can use to aid in troubleshooting any issues you may encounter.

A good area to start troubleshooting when calls are not working or having issues is to look at signaling traces for SIP messages during call establishment through traces from CUCM and SBC.

Cisco Unified Serviceability in CUCM

This console gives ability to start trace, monitor alarms and access to Cisco Unified Realm time Monitoring Tool (RTMT). Pavket capturing is also configurable in CUCM.

Configuration checklist when outbound calls are failing:

- Check for Dial plan/route issues
- Check for codec mismatch or signaling complete as CUCM does not offer SDP in outbound INVITE
- Check Gateway and trunk configuration for TCP connections
- SIP OPTIONS message connectivity between SBC and CUCM

Wireshark

Wireshark is also a network protocol analyzer which is freely downloadable from www.wireshark.org. Wireshark could be installed on the server hosting CUCM and have the SBC send packet trace to this remote location.

Oracle E-SBC 3820

The Oracle SBC provides a rich set of statistical counters available from the ACLI, as well as log file output with configurable detail. The follow sections detail enabling, adjusting and accessing those interfaces.

Resetting the statistical counters, enabling logging and restarting the log files.

At the SBC Console:

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

Examining the log files

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

```
C:\Documents and Settings\user>ftp 192.168.5.24
Connected to 192.168.85.55.
220 CUCM-ATTFlexreachFTP server (VxWorks 6.4) ready.
User (192.168.85.55:(none)): user
331 Password required for user.
```

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

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

Through the Web GUI

You can also check the display results of filtered SIP session data from the Oracle Enterprise Session Border Controller, and provides traces in a common log format for local viewing or for exporting to your PC. Please check the “Monitor and Trace” section (page 145) of the Web GUI User Guide available at http://docs.oracle.com/cd/E56581_01/index.htm

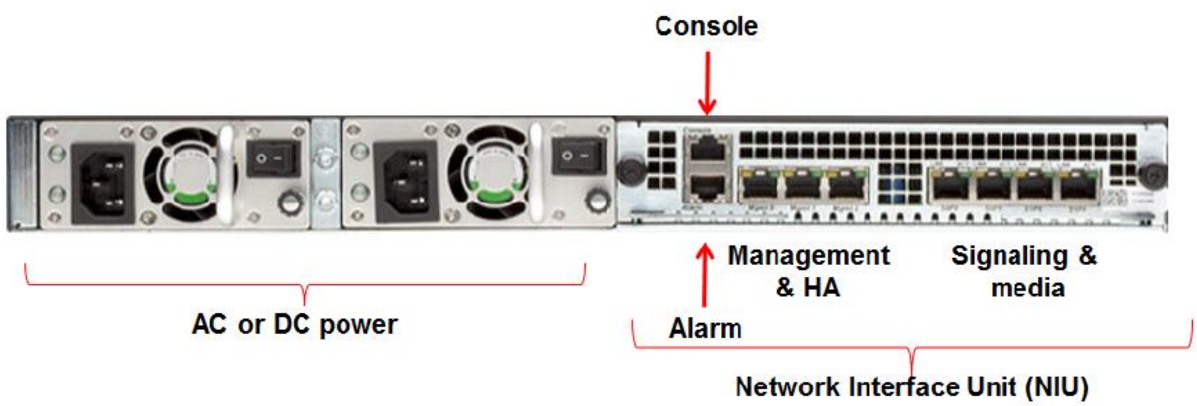
Appendix A

Accessing the ACLI

Access to the ACLI is provided by:

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

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

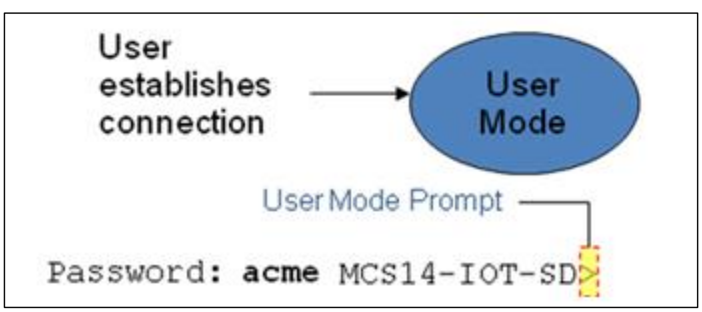


ACLI Basics

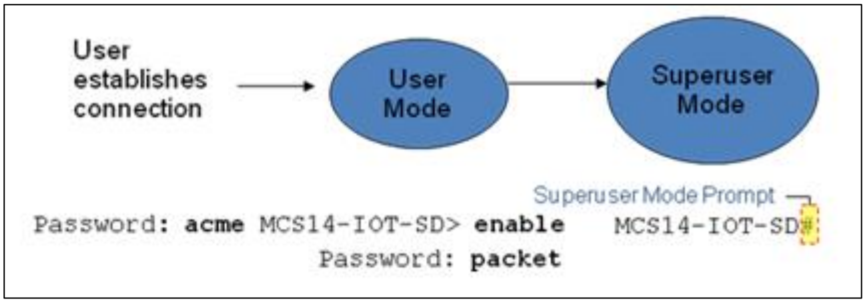
There are two password protected modes of operation within the ACLI, User mode and Superuser mode.

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

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



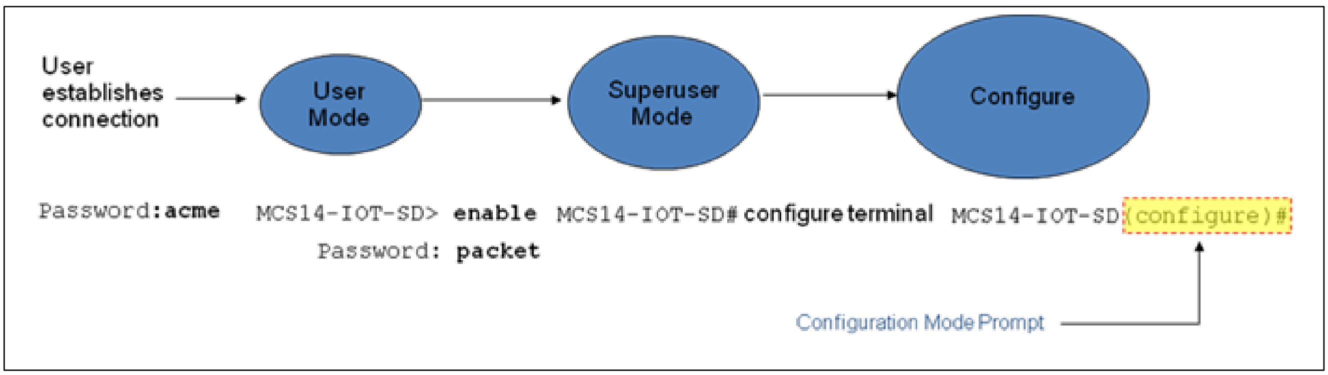
The Superuser mode allows for access to all system commands for operation, maintenance, and administration. This mode is identified by the pound sign (#) in the prompt after the target name. To enter the Superuser mode, issue the enable command in the User mode.



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

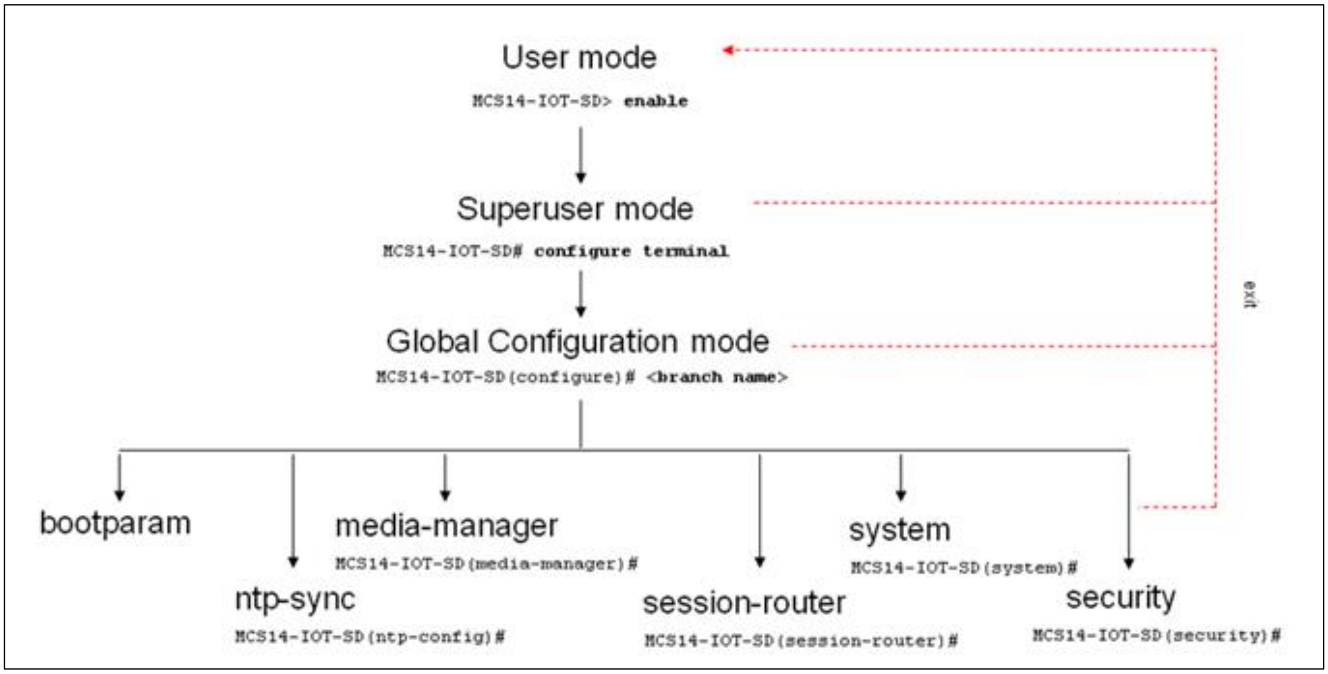
You must enter the Configuration mode to configure elements. For example, you can access the configuration branches and configuration elements for signaling and media configurations. To enter the Configuration mode, issue the **`configure terminal`** command in the Superuser mode.

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



In the configuration mode, there are six configuration branches:

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



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

The bootparam branch provides access to SBC boot parameters. Key boot parameters include:

- boot device – The global management port, usually eth0
- file name – The boot path and the image file.
- inet on ethernet – The IP address and subnet mask (in hex) of the management port of the SD.
- host inet –The IP address of external server where image file resides.
- user and ftp password – Used to boot from the external FTP server.
- gateway inet – The gateway IP address for reaching the external server, if the server is located in a different network.

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

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

The security branch provides access to security configuration.

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

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

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

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

Configuration Elements

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

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

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

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

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

Creating an Element

1. To create a single-instance element, you go to the appropriate level in the ACLI path and enter its parameters. There is no need to specify a unique identifier property because a single-instance element is a global element and there is only one instance of this element.
2. When creating a multiple-instance element, you must specify a unique identifier for each instance of the element.
3. It is important to check the parameters of the element you are configuring before committing the changes. You do this by issuing the **show** command before issuing the **done** command. The parameters that you did not configure are filled with either default values or left empty.

4. On completion, you must issue the **done** command. The done command causes the configuration to be echoed to the screen and commits the changes to the volatile memory. It is a good idea to review this output to ensure that your configurations are correct.
5. Issue the **exit** command to exit the selected element.

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

Editing an Element

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

1. Enter the element that you will edit at the correct level of the ACLI path.
2. Select the element that you will edit, and view it before editing it.
   The **select** command loads the element to the volatile memory for editing. The **show** command allows you to view the element to ensure that it is the right one that you want to edit.
3. Once you are sure that the element you selected is the right one for editing, edit the parameter one by one. The new value you provide will overwrite the old value.
4. It is important to check the properties of the element you are configuring before committing it to the volatile memory. You do this by issuing the **show** command before issuing the **done** command.
5. On completion, you must issue the **done** command.
6. Issue the **exit** command to exit the selected element.

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

Deleting an Element

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

To delete a single-instance element,

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

To delete a multiple-instance element,

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

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

Configuration Versions

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

- The **edited configuration** – this is the version that you are making changes to. This version of the configuration is stored in the SBC's volatile memory and will be lost on a reboot.
  To view the editing configuration, issue the **show configuration** command.

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

Saving the Configuration

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

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

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

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

```
CUCM-ATTFlexreach # 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'.
CUCM-ATTFlexreach #
```

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.

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

Appendix B – SBC configuration

```
CUCM-ATTFlexreach#
capture-receiver
        state                                           enabled
        address                                         10.232.50.81
        network-interface                               s1p0:0
        last-modified-by                                admin@172.18.0.158
        last-modified-date                              2015-04-02 15:45:49
host-route
        dest-network                                    14.1.1.0
        netmask                                         255.255.255.0
        gateway                                         167.167.167.1
        description
        last-modified-by                                admin@172.18.0.150
        last-modified-date                              2015-04-10 21:11:24
local-policy
        from-address                                    *
        to-address                                      *
        source-realm                                    ATT-Trunk
        description
        activate-time
        deactivate-time
        state                                           enabled
        policy-priority                                 none
        policy-attribute
                next-hop                                        10.232.50.89
                realm                                           CUCM
                action                                          none
                terminate-recursion                             disabled
                carrier
                start-time                                      0000
                end-time                                        2400
                days-of-week                                    U-S
                cost                                            0
                state                                           enabled
                app-protocol                                    SIP
                methods
                media-profiles
                lookup                                          single
                next-key
                eloc-str-lkup                                   disabled
                eloc-str-match
        last-modified-by                                admin@172.18.0.158
        last-modified-date                              2015-04-01 18:47:10
local-policy
        from-address                                    *
        to-address                                      *
        source-realm                                    CUCM
        description
        activate-time
        deactivate-time
```

```
        state                                   enabled
        policy-priority                         none
        policy-attribute
                next-hop                                14.1.1.10
                realm                                   ATT-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
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-01 18:46:26
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                                 unique-sdp-id
        red-flow-port                           1985
        red-mgcp-port                           1986
        red-max-trans                           10000
        red-sync-start-time                     5000
        red-sync-comp-time                      1000
        media-policing                          enabled
        max-signaling-bandwidth                 10000000
        max-untrusted-signaling                 100
        min-untrusted-signaling                 30
        app-signaling-bandwidth                 0
        tolerance-window                        30
        trap-on-demote-to-deny                  disabled
        trap-on-demote-to-untrusted             disabled
        syslog-on-demote-to-deny                disabled
        syslog-on-demote-to-untrusted           disabled
        rtcp-rate-limit                         0
        anonymous-sdp                           enabled
```

```
        arp-msg-bandwidth                       32000
        fragment-msg-bandwidth                  0
        rfc2833-timestamp                       disabled
        default-2833-duration                   100
        rfc2833-end-pkts-only-for-non-sig       enabled
        translate-non-rfc2833-event             disabled
        media-supervision-traps                 disabled
        dnsalg-server-failover                  disabled
        syslog-on-call-reject                   disabled
        last-modified-by                        admin@172.18.255.96
        last-modified-date                      2015-04-22 20:37:58
media-profile
        name                                    G729
        subname
        media-type                              audio
        payload-type                            18
        transport                               RTP/AVP
        clock-rate                              0
        req-bandwidth                           0
        frames-per-packet                       0
        parameters                              annexb=no
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
        last-modified-by                        admin@172.18.255.96
        last-modified-date                      2015-04-22 20:35:32
media-profile
        name                                    g729wAnnexB
        subname                                 annexb-yes
        media-type                              audio
        payload-type                            18
        transport                               RTP/AVP
        clock-rate                              0
        req-bandwidth                           0
        frames-per-packet                       0
        parameters                              annexb=yes
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 17:15:31
media-profile
        name                                    PCMA
        subname
        media-type                              audio
```

```
        payload-type                            8
        transport                               RTP/AVP
        clock-rate                              0
        req-bandwidth                           0
        frames-per-packet                       0
        parameters
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 16:34:01
media-profile
        name                                    PCMU
        subname
        media-type                              audio
        payload-type                            0
        transport                               RTP/AVP
        clock-rate                              0
        req-bandwidth                           0
        frames-per-packet                       0
        parameters
        average-rate-limit                      0
        peak-rate-limit                         0
        max-burst-size                          0
        sdp-rate-limit-headroom                 0
        sdp-bandwidth                           disabled
        police-rate                             0
        standard-pkt-rate                       0
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 16:34:16
network-interface
        name                                    s0p0
        sub-port-id                             0
        description
        hostname
        ip-address                              167.167.167.181
        pri-utility-addr
        sec-utility-addr
        netmask                                 255.255.255.0
        gateway                                 167.167.167.1
        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                             167.167.167.181
        ftp-address
        icmp-address                            167.167.167.181
        snmp-address
        telnet-address
        ssh-address
        last-modified-by                        admin@172.18.0.154
        last-modified-date                      2015-04-14 18:52:44
network-interface
        name                                    s1p0
        sub-port-id                             0
        description
        hostname
        ip-address                              10.232.50.50
        pri-utility-addr
        sec-utility-addr
        netmask                                 255.255.255.0
        gateway                                 10.232.50.89
        sec-gateway
        gw-heartbeat
                state                                   disabled
                heartbeat                               0
                retry-count                             0
                retry-timeout                           1
                health-score                            0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                             11
        signaling-mtu                           0
        hip-ip-list                             10.232.50.50
        ftp-address
        icmp-address                            10.232.50.50
        snmp-address
        telnet-address
        ssh-address
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-01 18:43:54
phy-interface
        name                                    s0p0
        operation-type                          Media
        port                                    0
        slot                                    0
        virtual-mac
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode                             FULL
        speed                                   100
```

```
        wancom-health-score                     50
        overload-protection                     disabled
        mac-filtering                           disabled
        last-modified-by                        admin@172.18.0.150
        last-modified-date                      2015-04-09 21:12:41
phy-interface
        name                                    s1p0
        operation-type                          Media
        port                                    0
        slot                                    1
        virtual-mac
        admin-state                             enabled
        auto-negotiation                        enabled
        duplex-mode                             FULL
        speed                                   100
        wancom-health-score                     50
        overload-protection                     disabled
        mac-filtering                           disabled
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-03-31 18:35:15
realm-config
        identifier                              ATT-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
        codec-manip-in-realm                    disabled
        codec-manip-in-network                  enabled
        rtcp-policy
        constraint-name
        call-recording-server-id
        session-recording-server
        session-recording-required              disabled
        manipulation-string
        manipulation-pattern
        stun-enable                             disabled
        stun-server-ip                          0.0.0.0
        stun-server-port                        3478
        stun-changed-ip                         0.0.0.0
        stun-changed-port                       3479
        sip-profile
        sip-isup-profile
```

```
        match-media-profiles
        qos-constraint
        block-rtcp                              disabled
        hide-egress-media-update                disabled
        tcp-media-profile
        monitoring-filters
        node-functionality
        default-location-string
        alt-family-realm
        pref-addr-type                          none
        last-modified-by                        admin@172.18.255.96
        last-modified-date                      2015-04-23 22:25:28
realm-config
        identifier                              CUCM
        description
        addr-prefix                             0.0.0.0
        network-interfaces                      s1p0:0
        mm-in-realm                             disabled
        mm-in-network                           enabled
        mm-same-ip                              enabled
        mm-in-system                            enabled
        bw-cac-non-mm                           disabled
        msm-release                             disabled
        qos-enable                              disabled
        max-bandwidth                           0
        fallback-bandwidth                      0
        max-priority-bandwidth                  0
        max-latency                             0
        max-jitter                              0
        max-packet-loss                         0
        observ-window-size                      0
        parent-realm
        dns-realm
        media-policy
        media-sec-policy
        srtp-msm-passthrough                    disabled
        class-profile
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        average-rate-limit                      0
        access-control-trust-level              none
        invalid-signal-threshold                0
        maximum-signal-threshold                0
        untrusted-signal-threshold              0
        nat-trust-threshold                     0
        max-endpoints-per-nat                   0
        nat-invalid-message-threshold           0
        wait-time-for-invalid-register          0
        deny-period                             30
        cac-failure-threshold                   0
        untrust-cac-failure-threshold           0
```

```
        ext-policy-svr
        diam-e2-address-realm
        subscription-id-type                    END_USER_NONE
        symmetric-latching                      disabled
        pai-strip                               disabled
        trunk-context
        device-id
        early-media-allow
        enforcement-profile
        additional-prefixes
        restricted-latching                     none
        restriction-mask                        32
        user-cac-mode                           none
        user-cac-bandwidth                      0
        user-cac-sessions                       0
        icmp-detect-multiplier                  0
        icmp-advertisement-interval             0
        icmp-target-ip
        monthly-minutes                         0
        options
        spl-options
        accounting-enable                       enabled
        net-management-control                  disabled
        delay-media-update                      disabled
        refer-call-transfer                     disabled
        hold-refer-reinvite                     disabled
        refer-notify-provisional                none
        dyn-refer-term                          disabled
        codec-policy
        codec-manip-in-realm                    disabled
        codec-manip-in-network                  enabled
        rtcp-policy
        constraint-name
        call-recording-server-id
        session-recording-server
        session-recording-required              disabled
        manipulation-string
        manipulation-pattern
        stun-enable                             disabled
        stun-server-ip                          0.0.0.0
        stun-server-port                        3478
        stun-changed-ip                         0.0.0.0
        stun-changed-port                       3479
        sip-profile
        sip-isup-profile
        match-media-profiles
        qos-constraint
        block-rtcp                              disabled
        hide-egress-media-update                disabled
        tcp-media-profile
        monitoring-filters
        node-functionality
        default-location-string
```

```
        alt-family-realm
        pref-addr-type                          none
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-01 18:44:13
response-map
        name                                    change486to603
        entries
                recv-code                               486
                xmit-code                               603
                reason                                  Decline
                method
                register-response-expires
        last-modified-by                        admin@172.18.255.90
        last-modified-date                      2015-04-30 16:10:14
session-agent
        hostname                                10.232.50.89
        ip-address
        port                                    5060
        state                                   enabled
        app-protocol                            SIP
        app-type
        transport-method                        StaticTCP
        realm-id                                CUCM
        egress-realm-id
        description
        carriers
        allow-next-hop-lp                       enabled
        constraints                             disabled
        max-sessions                            0
        max-inbound-sessions                    0
        max-outbound-sessions                   0
        max-burst-rate                          0
        max-inbound-burst-rate                  0
        max-outbound-burst-rate                 0
        max-sustain-rate                        0
        max-inbound-sustain-rate                0
        max-outbound-sustain-rate               0
        min-seizures                            5
        min-asr                                 0
        time-to-resume                          0
        ttr-no-response                         0
        in-service-period                       0
        burst-rate-window                       0
        sustain-rate-window                     0
        req-uri-carrier-mode                    None
        proxy-mode
        redirect-action
        loose-routing                           enabled
        send-media-session                      enabled
        response-map
        ping-method
        ping-interval                           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                       removeplus1
        trust-me                                disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate               0
        early-media-allow
        invalidate-registrations                disabled
        rfc2833-mode                            none
        rfc2833-payload                         0
        codec-policy
        enforcement-profile
        refer-call-transfer                     disabled
        refer-notify-provisional                none
        reuse-connections                       NONE
        tcp-keepalive                           none
        tcp-reconn-interval                     0
        max-register-burst-rate                 0
        register-burst-window                   0
        sip-profile
        sip-isup-profile
        kpml-interworking                       inherit
        monitoring-filters
        session-recording-server
        session-recording-required              disabled
        hold-refer-reinvite                     disabled
        send-tcp-fin                            disabled
        last-modified-by                        admin@172.18.255.90
        last-modified-date                      2015-04-29 21:00:41
session-agent
        hostname                                14.1.1.10
        ip-address
        port                                    5060
        state                                   enabled
        app-protocol                            SIP
        app-type
        transport-method                        UDP
```

```
        realm-id                                ATT-Trunk
        egress-realm-id
        description                             ATT
        carriers
        allow-next-hop-lp                       enabled
        constraints                             disabled
        max-sessions                            0
        max-inbound-sessions                    0
        max-outbound-sessions                   0
        max-burst-rate                          0
        max-inbound-burst-rate                  0
        max-outbound-burst-rate                 0
        max-sustain-rate                        0
        max-inbound-sustain-rate                0
        max-outbound-sustain-rate               0
        min-seizures                            5
        min-asr                                 0
        time-to-resume                          0
        ttr-no-response                         0
        in-service-period                       0
        burst-rate-window                       0
        sustain-rate-window                     0
        req-uri-carrier-mode                    None
        proxy-mode
        redirect-action
        loose-routing                           enabled
        send-media-session                      enabled
        response-map
        ping-method                             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
        trust-me                                disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate               0
```

```
        early-media-allow
        invalidate-registrations               disabled
        rfc2833-mode                           none
        rfc2833-payload                        0
        codec-policy
        enforcement-profile
        refer-call-transfer                    disabled
        refer-notify-provisional               none
        reuse-connections                      NONE
        tcp-keepalive                          none
        tcp-reconn-interval                    0
        max-register-burst-rate                0
        register-burst-window                  0
        sip-profile
        sip-isup-profile
        kpml-interworking                      inherit
        monitoring-filters
        session-recording-server
        session-recording-required             disabled
        hold-refer-reinvite                    disabled
        send-tcp-fin                           disabled
        last-modified-by                       admin@172.18.255.96
        last-modified-date                     2015-04-27 19:36:52
session-translation
        id                                     removeplus1
        rules-calling
        rules-called                           removetheplus1
        last-modified-by                       admin@172.18.255.90
        last-modified-date                     2015-04-29 21:02:49
sip-config
        state                                  enabled
        operation-mode                         dialog
        dialog-transparency                    enabled
        home-realm-id                          CUCM
        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
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 16:38:28
sip-interface
        state                                   enabled
        realm-id                                ATT-Trunk
        description
        sip-port
                address                                 167.167.167.181
                port                                    5060
                transport-protocol                      UDP
                tls-profile
                allow-anonymous                         agents-only
                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                      ChangeforPAIandNAT
        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                            preferred
        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                          invite
        p-early-media-header                    disabled
        p-early-media-direction
        add-sdp-profiles                        G729
                                                g729wAnnexB
                                                PCMU
                                                PCMA
        manipulation-string
        manipulation-pattern
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                          0
        tunnel-name
        register-keep-alive                     none
        kpml-interworking                       disabled
        msrp-delay-egress-bye                   disabled
        send-380-response
        pcscf-restoration
        session-timer-profile
        session-recording-server
        session-recording-required              disabled
        service-tag
        reg-cache-route                         disabled
        last-modified-by                        admin@172.18.255.90
        last-modified-date                      2015-05-01 15:10:54
sip-interface
        state                                   enabled
        realm-id                                CUCM
        description
        sip-port
                address                                 10.232.50.50
                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
        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                          ForPrivacyandcodec_rel
        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                                preferred
        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                        G729
                                                PCMU
                                                PCMA
        manipulation-string
        manipulation-pattern
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                          0
        tunnel-name
        register-keep-alive                     none
        kpml-interworking                       disabled
        msrp-delay-egress-bye                   disabled
        send-380-response
        pcscf-restoration
        session-timer-profile
        session-recording-server
        session-recording-required              disabled
        service-tag
        reg-cache-route                         disabled
        last-modified-by                        admin@172.18.0.141
        last-modified-date                      2015-04-30 16:38:13
sip-manipulation
        name                                    AddDiversion
        description
        split-headers
        join-headers
        header-rule
                name                                    AddDiv
                header-name                             Diversion
                action                                  add
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value                               <sip:7322162712@167.167.167.181>
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-05-04 20:16:08
sip-manipulation
        name                                    ChangeFrom
        description
        split-headers
        join-headers
        header-rule
                name                                    ChangeFrom
                header-name                             From
                action                                  manipulate
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    Newfrom
```

```
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         case-sensitive
                        match-value
                        new-value                               7322162709
        header-rule
                name                                    forPAI
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               ChangePAI
        header-rule
                name                                    forprivacy
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               ACME_NAT_TO_FROM_IP
        last-modified-by                        admin@172.18.0.154
        last-modified-date                      2015-04-06 22:42:59
sip-manipulation
        name                                    ChangePAI
        description
        split-headers
        join-headers
        header-rule
                name                                    Storecontacthost
                header-name                             Contact
                action                                  store
                comparison-type                         pattern-rule
                msg-type                                any
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    storehost
                        parameter-name
                        type                                    uri-host
                        action                                  store
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value
                        new-value
        header-rule
                name                                    ModPAI
                header-name                             P-Asserted-Identity
```

```
                action                                        manipulate
                comparison-type                               boolean
                msg-type                                      any
                methods                                       INVITE
                match-value                                   $Storecontacthost.$storehost.$0
                new-value
                element-rule
                        name                                          modhost
                        parameter-name
                        type                                          uri-host
                        action                                        replace
                        match-val-type                                any
                        comparison-type                               pattern-rule
                        match-value
                        new-value
$Storecontacthost.$storehost.$0
        header-rule
                name                                          StoreFromuser
                header-name                                   From
                action                                        store
                comparison-type                               pattern-rule
                msg-type                                      request
                methods                                       INVITE
                match-value
                new-value
                element-rule
                        name                                          Storeuser
                        parameter-name
                        type                                          uri-user
                        action                                        store
                        match-val-type                                any
                        comparison-type                               pattern-rule
                        match-value
                        new-value
        header-rule
                name                                          ChangePAIuser
                header-name                                   P-Asserted-Identity
                action                                        manipulate
                comparison-type                               case-sensitive
                msg-type                                      request
                methods                                       INVITE
                match-value
                new-value
                element-rule
                        name                                          moduser
                        parameter-name
                        type                                          uri-user
                        action                                        replace
                        match-val-type                                any
                        comparison-type                               case-sensitive
                        match-value
                        new-value
$StoreFromuser.$Storeuser.$0
        header-rule
```

```
                name                                    RPI_Header
                header-name                             Remote-Party-ID
                action                                  manipulate
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value
                element-rule
                        name                                    changehostRPI
                        parameter-name
                        type                                    uri-host
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         case-sensitive
                        match-value
                        new-value                               $LOCAL_IP
        last-modified-by                        admin@172.18.0.148
        last-modified-date                      2015-05-22 21:30:45
sip-manipulation
        name                                    ChangeforPAIandNAT
        description
        split-headers
        join-headers
        header-rule
                name                                    changePAI
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               ChangePAI
        header-rule
                name                                    forprivacy
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               ACME_NAT_TO_FROM_IP
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-05-04 21:30:53
sip-manipulation
        name                                    ForPrivacyandcodec_rel
        description
        split-headers
        join-headers
        header-rule
                name                                    doNATforCUCM
                header-name                             From
                action                                  sip-manip
```

```
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               ACME_NAT_TO_FROM_IP
        header-rule
                name                                    Forrel_711codecstrip
                header-name                             From
                action                                  sip-manip
                comparison-type                         case-sensitive
                msg-type                                any
                methods
                match-value
                new-value                               NAT_IP_rel
    sip-manipulation
        name                                    NAT_IP_rel
        description
        split-headers
        join-headers
        header-rule
                name                                    checkG729Sdp
                header-name                             Content-Type
                action                                  store
                comparison-type                         pattern-rule
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    checksdp
                        parameter-name                          application/sdp
                        type                                    mime
                        action                                  store
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             (\Rm=audio [0-9]{1,5}
RTP/AVP ([0-9]{1,4})*)([0-9 ]*)\b
                        new-value
                element-rule
                        name                                    checkcodec
                        parameter-name                          application/sdp
                        type                                    mime
                        action                                  store
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             a=rtpmap:18.*\R
                        new-value
        header-rule
                name                                    fixSdpRequest
                header-name                             Content-Type
                action                                  manipulate
                comparison-type                         boolean
                msg-type                                request
                methods                                 INVITE
```

```
                match-value                             $checkG729Sdp.$checkcodec
                new-value
                element-rule
                        name                                    removeG711
                        parameter-name                          application/sdp
                        type                                    mime
                        action                                  find-replace-all
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value                             a=rtpmap:0.*\R
                        new-value
                element-rule
                        name                                    mod100mLine
                        parameter-name                          application/sdp
                        type                                    mime
                        action                                  find-replace-all
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value
(m=audio.*RTP/AVP).*( 18).*( 100)\R
                        new-value                               $1+$2+$3+$CRLF
                element-rule
                        name                                    mod101mLine
                        parameter-name                          application/sdp
                        type                                    mime
                        action                                  find-replace-all
                        match-val-type                          any
                        comparison-type                         pattern-rule
                        match-value
(m=audio.*RTP/AVP).*( 18).*( 101)\R
                        new-value                               $1+$2+$3+$CRLF
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-23 18:03:44
sip-manipulation
        name                                    changetoanonymous
        description
        split-headers
        join-headers
        header-rule
                name                                    changetoanonymous
                header-name                             From
                action                                  manipulate
                comparison-type                         case-sensitive
                msg-type                                request
                methods                                 INVITE
                match-value
                new-value
                element-rule
                        name                                    ChngFromuser
                        parameter-name
                        type                                    uri-user
                        action                                  replace
                        match-val-type                          any
                        comparison-type                         case-sensitive
```

```
                        match-value
                        new-value                                      Anonymous
        last-modified-by                        admin@172.18.255.90
        last-modified-date                      2015-05-01 19:21:48
snmp-community
        community-name                          acme
        access-mode                             READ-ONLY
        ip-addresses                            172.18.255.122
        last-modified-by                        admin@172.18.0.145
        last-modified-date                      2015-03-06 15:33:25
steering-pool
        ip-address                              10.232.50.50
        start-port                              16384
        end-port                                32767
        realm-id                                CUCM
        network-interface
        last-modified-by                        admin@172.18.0.154
        last-modified-date                      2015-04-14 19:17:21
steering-pool
        ip-address                              167.167.167.181
        start-port                              16384
        end-port                                32767
        realm-id                                ATT-Trunk
        network-interface
        last-modified-by                        admin@172.18.0.154
        last-modified-date                      2015-04-14 19:17:56
system-config
        hostname
        description
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled                            enabled
        enable-snmp-auth-traps                  disabled
        enable-snmp-syslog-notify               disabled
        enable-snmp-monitor-traps               disabled
        enable-env-monitor-traps                disabled
        snmp-syslog-his-table-length            1
        snmp-syslog-level                       WARNING
        system-log-level                        WARNING
        process-log-level                       DEBUG
        process-log-ip-address                  0.0.0.0
        process-log-port                        0
        collect
                sample-interval                         5
                push-interval                           15
                boot-state                              disabled
                start-time                              now
                end-time                                never
                red-collect-state                       disabled
                red-max-trans                           1000
                red-sync-start-time                     5000
```

```
                red-sync-comp-time                      1000
                push-success-trap-state                 disabled
        comm-monitor
                state                                   disabled
                sbc-grp-id                              0
                tls-profile
                qos-enable                              enabled
        call-trace                              disabled
        internal-trace                          disabled
        log-filter                              all
        default-gateway                         172.18.0.1
        restart                                 enabled
        exceptions
        telnet-timeout                          0
        console-timeout                         0
        remote-control                          enabled
        cli-audit-trail                         enabled
        link-redundancy-state                   disabled
        source-routing                          disabled
        cli-more                                disabled
        terminal-height                         24
        debug-timeout                           0
        trap-event-lifetime                     0
        ids-syslog-facility                     -1
        options
        default-v6-gateway                      ::
        ipv6-signaling-mtu                      1500
        ipv4-signaling-mtu                      1500
        cleanup-time-of-day                     00:00
        snmp-engine-id-suffix
        snmp-agent-mode                         v1v2
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-03-31 19:03:40
translation-rules
        id                                      addplus1
        type                                    add
        add-string                              +1
        add-index                               0
        delete-string
        delete-index                            0
        last-modified-by                        admin@172.18.0.158
        last-modified-date                      2015-04-06 17:53:30
translation-rules
        id                                      removetheplus1
        type                                    delete
        add-string
        add-index                               0
        delete-string                           +1
        delete-index                            0
```

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

Worldwide Inquiries
Phone: +1.650.506.7000
Fax: +1.650.506.7200

ORACLE®

CONNECT WITH US

blogs.oracle.com/oracle

facebook.com/oracle

twitter.com/oracle

oracle.com

Integrated Cloud Applications & Platform Services