Avaya S8300 Release CM 6.0.1 using SIP trunk to Cisco Unified Communications Manager Release 10.0

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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.0 to interoperate with the Avaya S8300 Communication Manager Release 6.0 and Avaya Aura Session Manager Release 6.1 using SIP Early-Offer.

The following items were tested:

- Basic call between the two systems and verification of voice path, using both SIP and digital phones on the Avaya side, and SIP and SCCP IP phones on the Cisco side.
- CLIP/CLIR/CNIP/CNIR features: calling party name and number delivery (allowed and restricted).
- COLP/CONP/COLR/CONR features: connected name and number delivery (allowed and restricted).
- Call transfer: attended, and early attended.
- Alerting Name Identification
- Call forwarding: call forward unconditional (CFU), call forward busy (CFB), and call forward no answer (CFNA).
- Hold and resume with music on hold.
- Three-way conferencing.
- Voice messaging and MWI activation-deactivation.
- Audio Codec Preference List
- Video

Listed below are the highlights of the integration issues:

- Basic calls worked from Cisco UCM to Avaya PBX and vice versa. Avaya's Media Shuffling feature was enabled throughout this testing exercise unless noted.
- CLIR/CNIR—The Avaya SIP trunk does not support calling/connected name and number restriction. Restriction of calling number on Avaya digital and SIP phones is achieved by configuring the Avaya station configuration page and not the SIP trunk page. This restriction is honored by Cisco UCM.

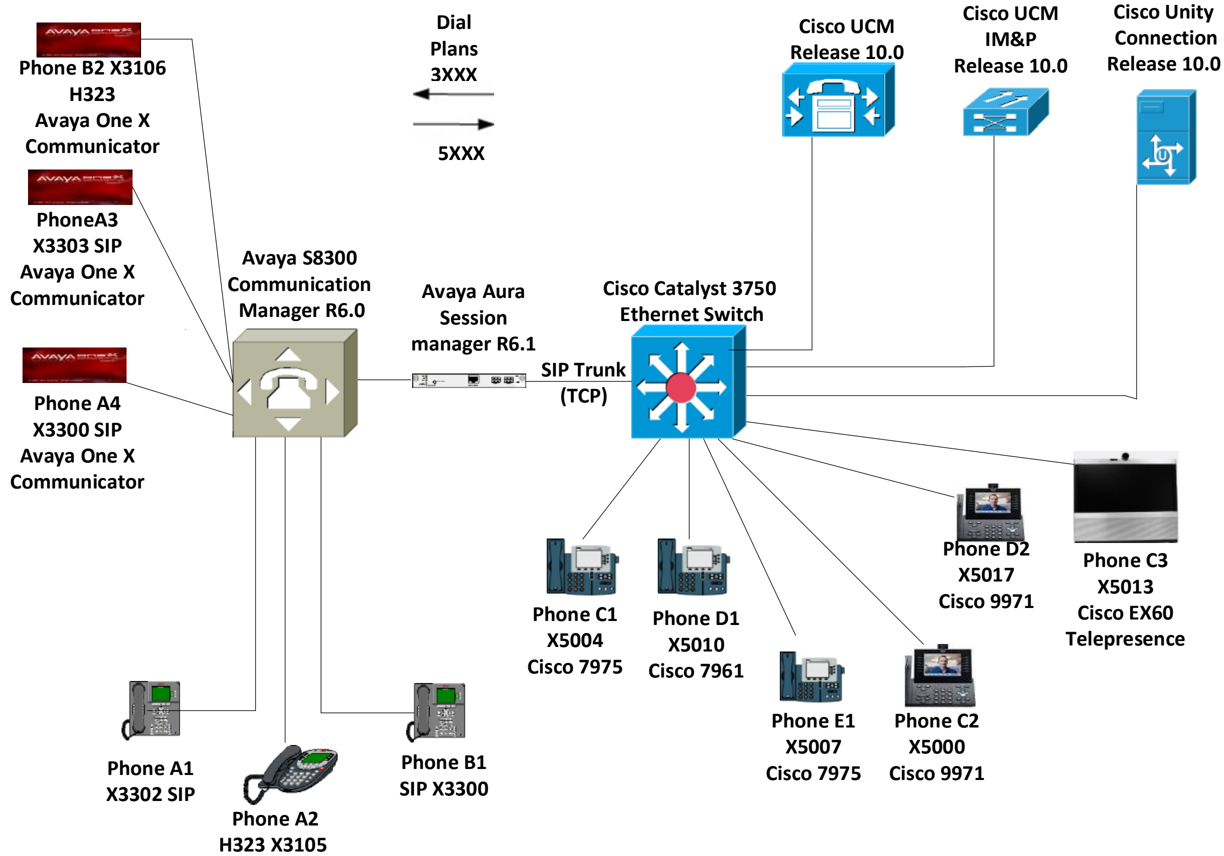
- COLR/CONR—As with calling name and number presentation restrictions, the Avaya PBX does not support connected name and number restriction on SIP trunks. Cisco UCM, on the other hand, restricts the connected name and number information the same way as for calling name and number restriction–by setting the SIP PRIVACY to “id” in the SIP trunk configuration page. Thus, the SIP privacy setting covers all outgoing message presentation restrictions, whether for inbound or outbound calls.
- Both systems support call forwarding (CFU, CFB, and CFNA) features. There are some call forward scenarios where the calling name and number are not updated after the call has been forwarded. This issue is found primarily when an Avaya phone is either the originating or terminating end. The Avaya phones (IP or legacy) do not display the forwarding phone’s name and number information, for a local forwarded call. Cisco phones display the forwarding information only when it is a locally forwarded call.
- Video Call Transfer and Video Conference call failures, appeared to be due to Avaya not responding to a re-INVITE sent from CUCM after the redirection, thus there is no audio after the redirection and the call drops.

Below are the key results:

- Basic call, call transfer, call forwarding, conference call, and hold and resume work successfully.
- Centralized voicemail, using Unity Connection server integrated to Cisco UCM via SIP was used for testing. This voicemail solution can provide centralized voicemail services, supporting both Avaya and Cisco end-users.

Network Topology

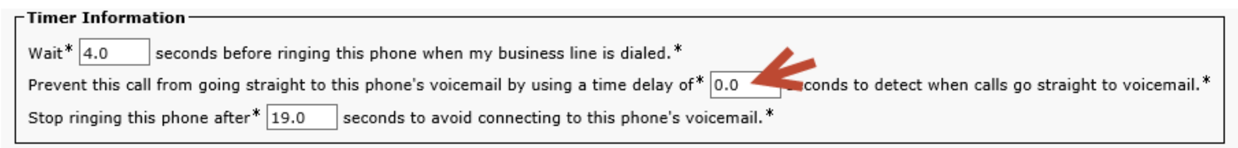
Basic Call Setup



Limitations

These are the known limitations, caveats, or integration issues:

- Avaya doesn't support Alerting Name feature.
- Avaya couldn't block caller id when calls were local (internal).
- Although the Codec Preference List was used and the INVITE message displayed the right codec, Avaya would respond with to the INVITE with their preferred Codec Preference for the call.
- Avaya experienced Music on Hold, DTMF to Voice Mail, and one way audio on conference calls where the distant end (CUCM) couldn't hear. Avaya Media Server had to be powered off by unplugging the AC cord and back on. Once this was performed above issues were resolved. This occurred twice during testing.
- For Extend & Connect Remote Destination to receive Voicemail when they are busy or call forward unconditional, Their Remote Destination Timer Information had to be set to 0.0.



- Video Call Transfer and Conference call failures, appeared to be due to Avaya not responding to a re-INVITE sent from CUCM after the redirection, thus there is no audio after the redirection and the call drops.

System Components

Hardware Requirements

The following hardware was used

- Cisco UCS-C240-M3S VMWare Host
- Catalyst switch 3750 WS-C3750X-48
- Cisco 7961, 7975, and 9971 IP phones
- Cisco EX60 Telepresence
- Avaya S8300D PBX with G6430 Media Gateway
- Avaya Common Server HP DL360 G7 with Session Manager
- Avaya Common Server HP DL360 G7 with System Manager
- Avaya Common Server HP DL360 G7 with Session border Controller

Software Requirements

The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.0
- Cisco Unified Communications Manager IM & P release 10.0
- Catalyst 3750 Cisco IOS Software, C3750E Software (C3750E-UNIVERSALK9-M), Version 12.2(55)SE5
- Cisco Unity Connection release 10.0
- Avaya Communication Manager release 6.01 Service Pack 11(patch 20685) (System Platform 6.0.3.10.3)
- Avaya G430 Media Gateway firmware release 30.12.1
- Avaya Aura® Session Manager R6.1 (6.1.2.0.612004) Service Pack 2
- Avaya Aura® System Manager R6.1 (System Platform 6.0.3.0.3, Template 6.1.5.0) Service Pack 2
- Avaya Aura® Session Border Controller 6.1 (System Platform 6.0.3.0.3, Template E362P4)
- Avaya One-X Communicator Release 6.1

Features

This section lists supported and unsupported features. Please see the Limitations section on page 7 for more information.

Features Supported

- CLIP—calling line (number) identification presentation.
- CLIR—calling line (number) identification restriction.
- CNIP—calling name identification presentation.
- CNIR—calling name identification restriction.
- Alerting name.
- Attended call transfer.
- Early attended call transfer.
- CFU—call forwarding unconditional.
- CFB—call forwarding busy.
- CFNA—call forwarding no answer.
- COLP—connected line (number) identification presentation.
- COLR—connected line (number) identification restriction.
- CONP—connected name identification presentation.
- CONR—connected name identification restriction.
- Hold and resume.
- Conference call.
- MWI—Message Waiting Indicator (lamp ON, lamp OFF).
- Audio Codec Preference List
- Video

Features Not Supported or Not Tested

- Call completion (callback, automatic callback).
- Inter-working Test Cases with Various Calling/Connected Name and Number.
- Shared Line - Hold & Resume with MOH
- Call Park/Pickup
- Interworking Test Cases for Call Transfer

Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Avaya (CM, SM) PBX's. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:

Avaya S8300 PBX:

1. Configure the IP-Codec-Set, and IP-Network-Region.
2. Configure the IP interface for C-LAN and IP Media Processor cards.
3. Configure Cisco UCM as an IP node-name.
4. Configure the signaling group for the SIP trunk to Cisco UCM.
5. Configure the trunk group for the SIP trunk to Cisco UCM.
6. Configure the SIP and digital station phone extension.
7. Configure the uniform dialing plan to the Cisco UCM extensions.
8. Configure the route pattern to the Cisco UCM extensions.

Cisco Unified Communications Manager:

1. SIP trunk security profile.
2. Device setting SIP profile.
3. Media resource group and media resource group list.
4. Partitions and calling search space.
5. Assign media resource group list (MRGL) in the default device pool.
6. SIP trunk to Avaya S8300 PBX.
7. SIP Trunk Normalization Script
8. SIP Trunk to Cisco Unity
9. Assign User in Cisco Unity

10. SIP and SCCP phones device configuration.
11. Route pattern to the Avaya S8300 PBX.
12. CallManager Service Parameter “Duplex Streaming Enabled” set to “True”.
13. Audio Codec Preference Configuration
14. Region Configuration

Configuring the Avaya S8300

Avaya S8300D Software Version and Hardware Configuration List

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10.70.2.14 - PuTTY
list configuration all

                              SYSTEM CONFIGURATION

Board                                                     Assigned Ports
Number   Board Type                 Code     Vintage    u=unassigned t=tti p=psa

001V1    ICC MM                      S8300D  HW01 FW001
001V2    DCP MM                      MM712AP HW05 FW009 01 02 u  u  u  u  u  u
001V9    MG-ANNOUNCEMENT            VMM-ANN             01 02 03 04 05 06 07 08
                                                        09 10 11 12 13 14 15 16

Command successfully completed
Command:
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

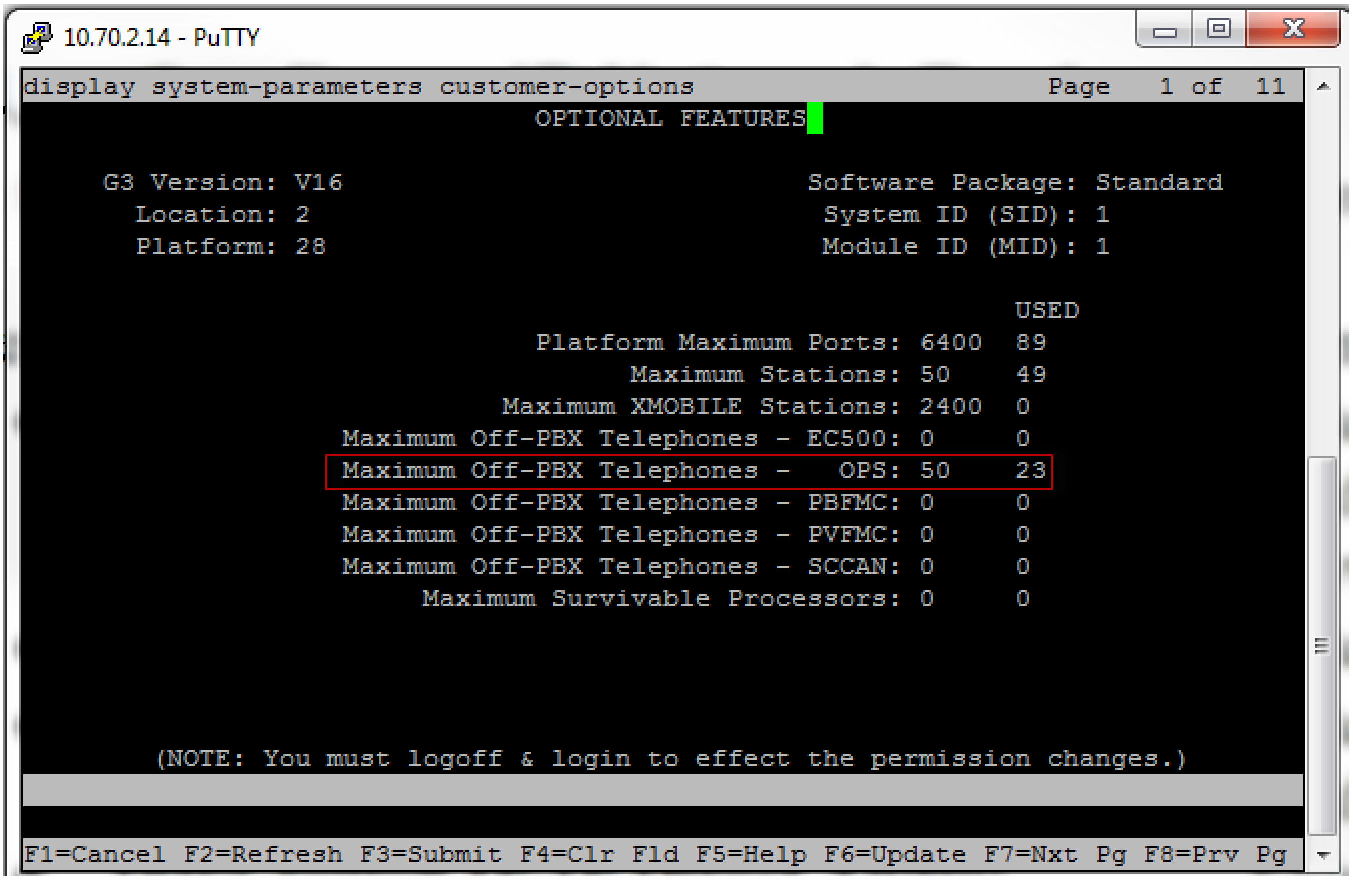
Verify system capacities and licensing:

Make sure system have enough license for SIP trunk and Video. Also make sure on page 10, the following features are enabled:

ARS? Verify “y” is displayed.

ARS/AAR Partitioning? Verify “y” is displayed

ARS/AAR Dialing without FAC? Verify “y” is displayed



10.70.2.14 - PuTTY

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 4000   0
           Maximum Concurrently Registered IP Stations: 2400   1
             Maximum Administered Remote Office Trunks: 4000   0
Maximum Concurrently Registered Remote Office Stations: 2400   0
              Maximum Concurrently Registered IP eCons: 50     0
  Max Concur Registered Unauthenticated H.323 Stations: 100    0
                        Maximum Video Capable Stations: 2400   3
                   Maximum Video Capable IP Softphones: 4      3
                       Maximum Administered SIP Trunks: 4000   40
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000   0
   Maximum Number of DS1 Boards with Echo Cancellation: 80     0
                             Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 50     1
           Maximum TN2602 Boards with 80 VoIP Channels: 128    0
          Maximum TN2602 Boards with 320 VoIP Channels: 128    0
   Maximum Number of Expanded Meet-me Conference Ports: 0      0

        (NOTE: You must logoff & login to effect the permission changes.)

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

10.70.2.14 - PuTTY

```
display system-parameters customer-options                      Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y  Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? y
          ASAI Link Core Capabilities? n                DCS Call Coverage? y
          ASAI Link Plus Capabilities? n               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n  Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                        DS1 MSP? y
                                ATMS? y           DS1 Echo Cancellation? y
                  Attendant Vectoring? y


        (NOTE: You must logoff & login to effect the permission changes.)

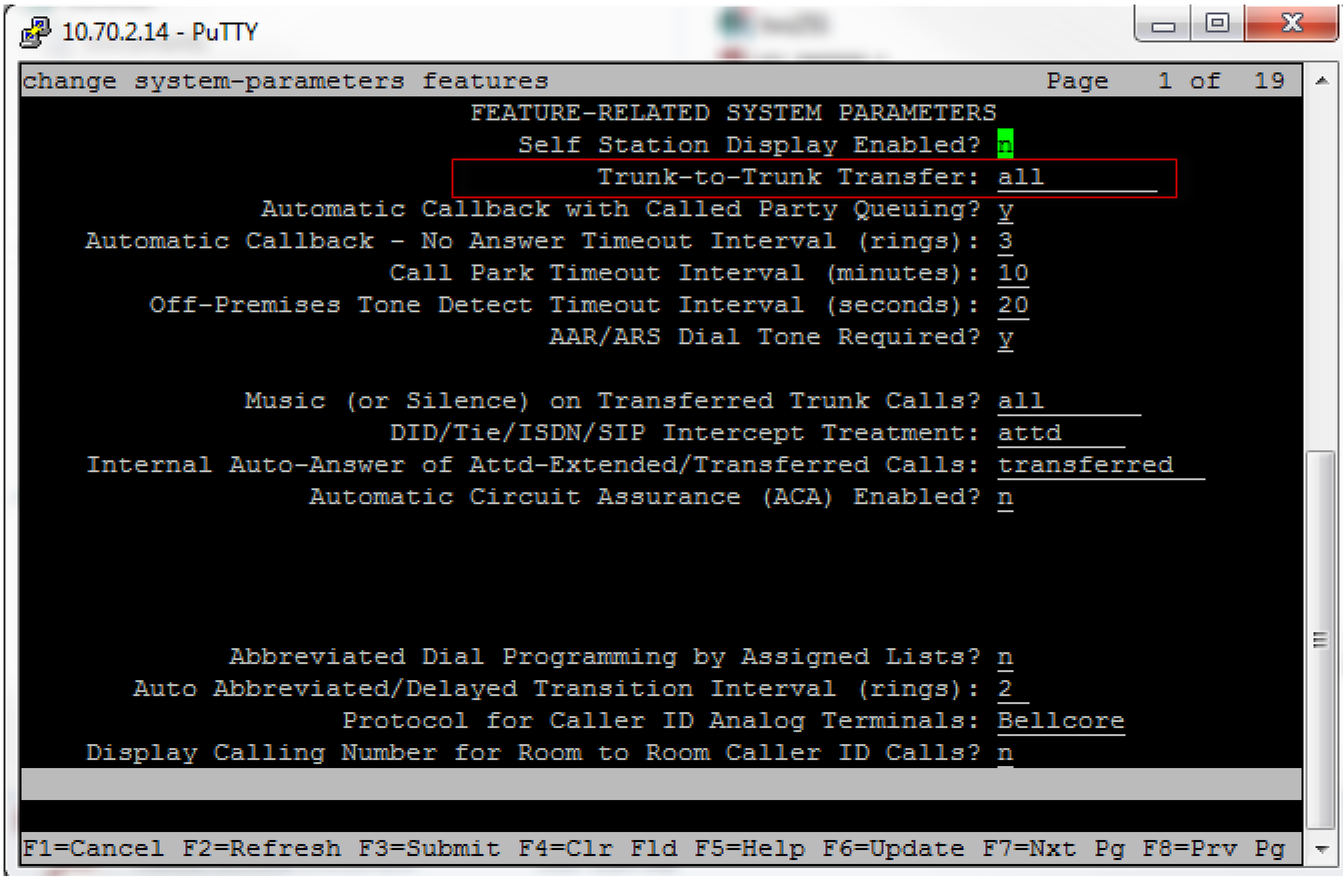
F1=Cancel  F2=Refresh  F3=Submit  F4=Clr Fld  F5=Help  F6=Update  F7=Nxt Pg  F8=Prv Pg
```

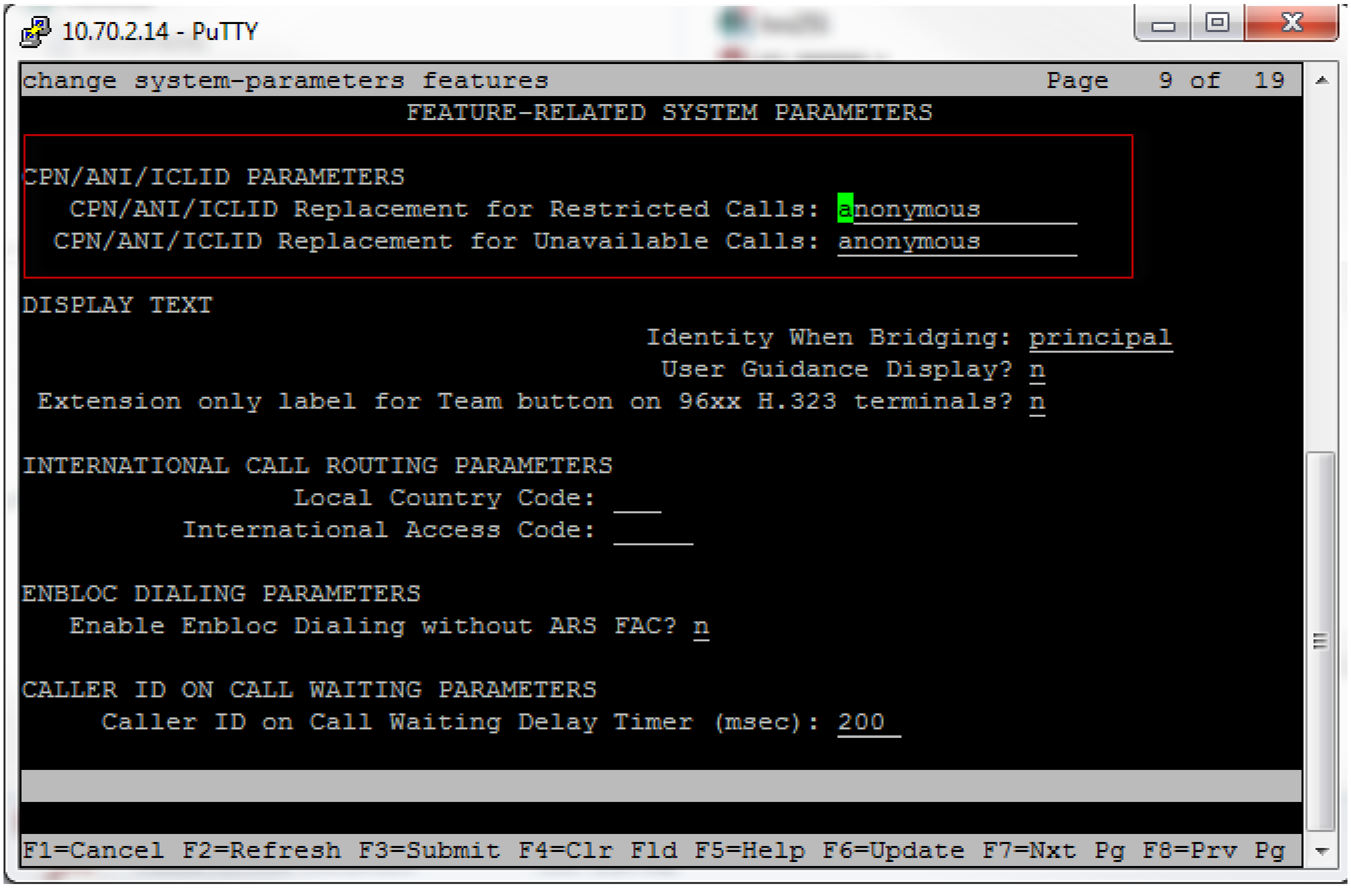
Configure System Feature:

On page 1,

Set **Trunk-to-Trunk Transfer** to All

Set **CPN/ANI/ICLID Replacement for Restricted/Unavailable calls** to anonymous




10.70.2.14 - PuTTY
change system-parameters features Page 9 of 19
FEATURE-RELATED SYSTEM PARAMETERS
CPN/ANI/ICLID PARAMETERS
CPN/ANI/ICLID Replacement for Restricted Calls: anonymous
CPN/ANI/ICLID Replacement for Unavailable Calls: anonymous
DISPLAY TEXT
Identity When Bridging: principal
User Guidance Display? n
Extension only label for Team button on 96xx H.323 terminals? n
INTERNATIONAL CALL ROUTING PARAMETERS
Local Country Code:
International Access Code:
ENBLOC DIALING PARAMETERS
Enable Enbloc Dialing without ARS FAC? n
CALLER ID ON CALL WAITING PARAMETERS
Caller ID on Call Waiting Delay Timer (msec): 200
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Config IP Codec Set and IP Network Region:

Codec set 1 is configured fior this test.

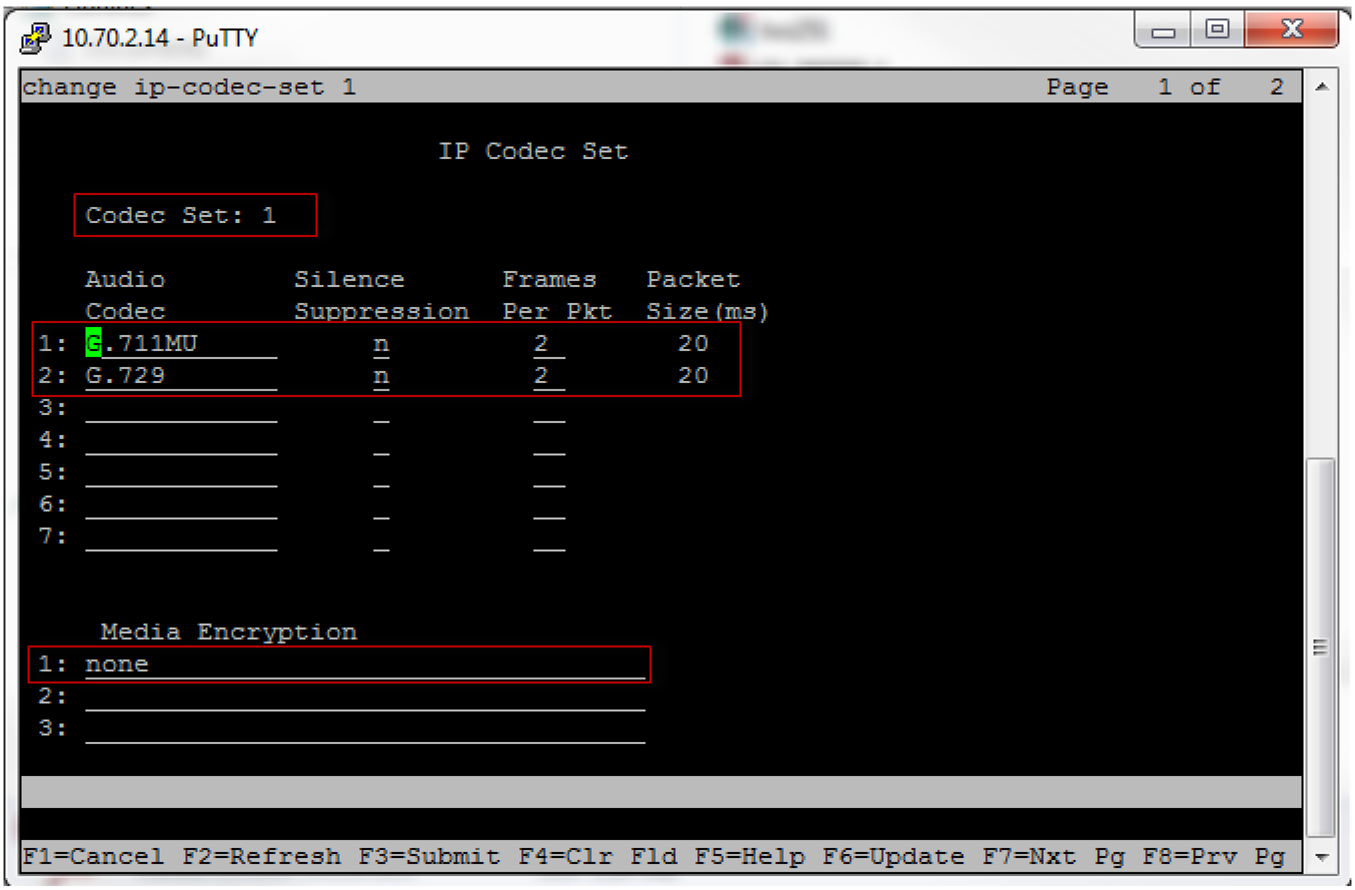
Audio Codec G711MU and G.729 are select codec

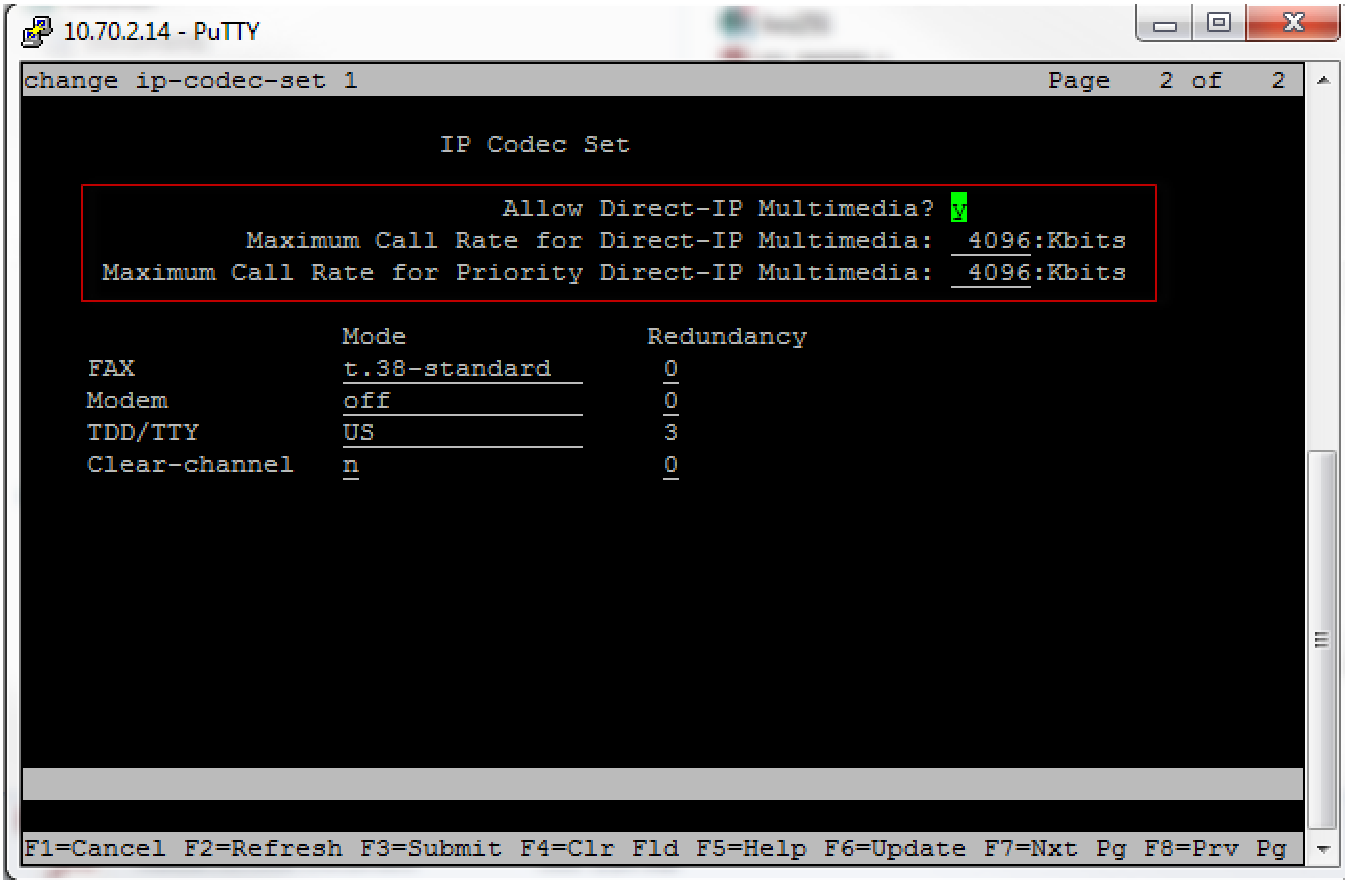
Media Encryption is set to none

Allow Direct-IP Multimedia set to ‘y”

Set Maximum Call Rate for Direct-IP Multimedia:4096:Kbits

Set Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits




10.70.2.14 - PuTTY
change ip-codec-set 1
Page 2 of 2
IP Codec Set
Allow Direct-IP Multimedia? y
Maximum Call Rate for Direct-IP Multimedia: 4096:Kbits
Maximum Call Rate for Priority Direct-IP Multimedia: 4096:Kbits
Mode
Redundancy
FAX t.38-standard 0
Modem off 0
TDD/TTY US 3
Clear-channel n 0
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Configure IP-Network-region 1:

Location:1

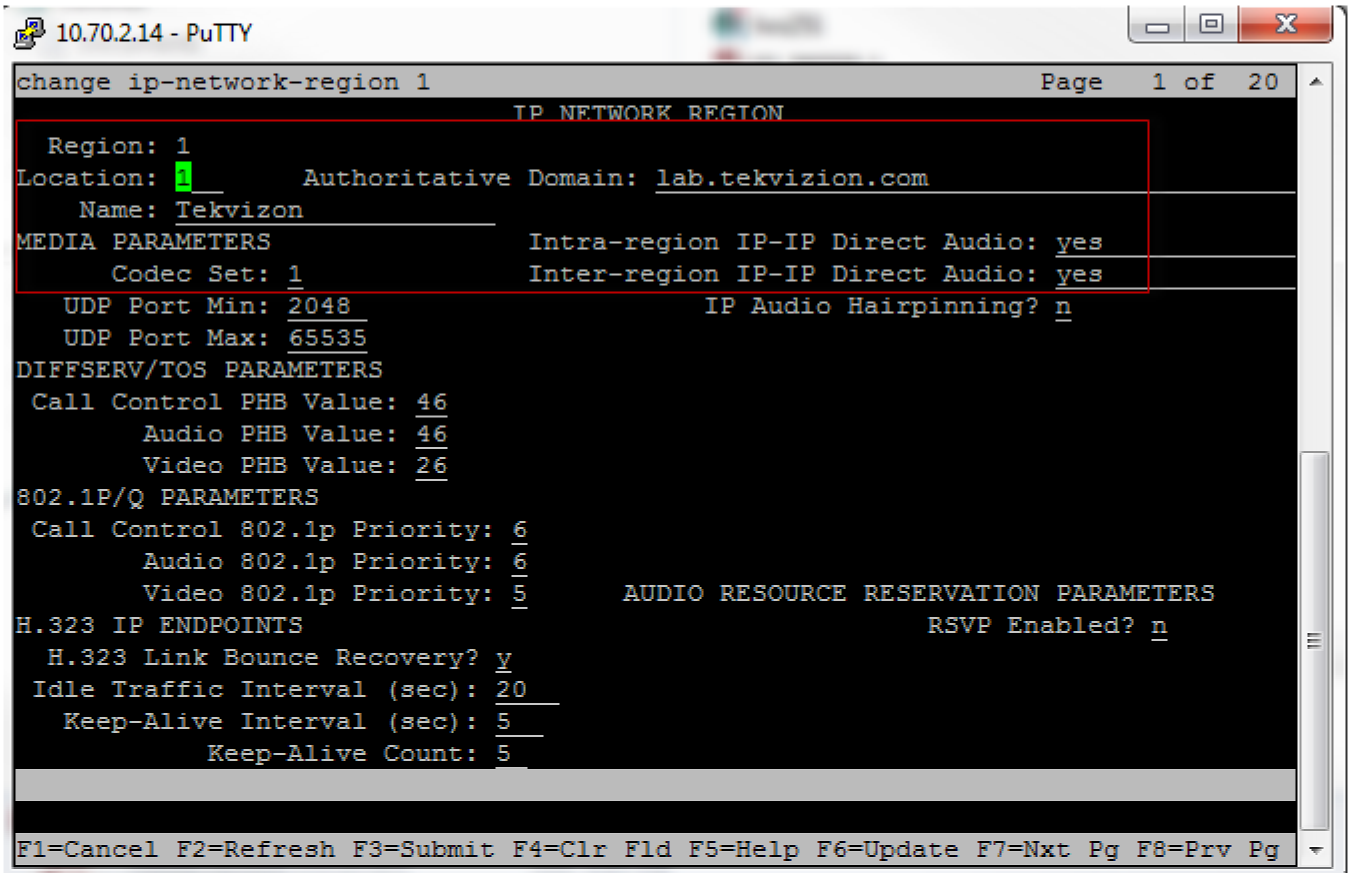
Authoritative Dimain:lab.tekvizion.com

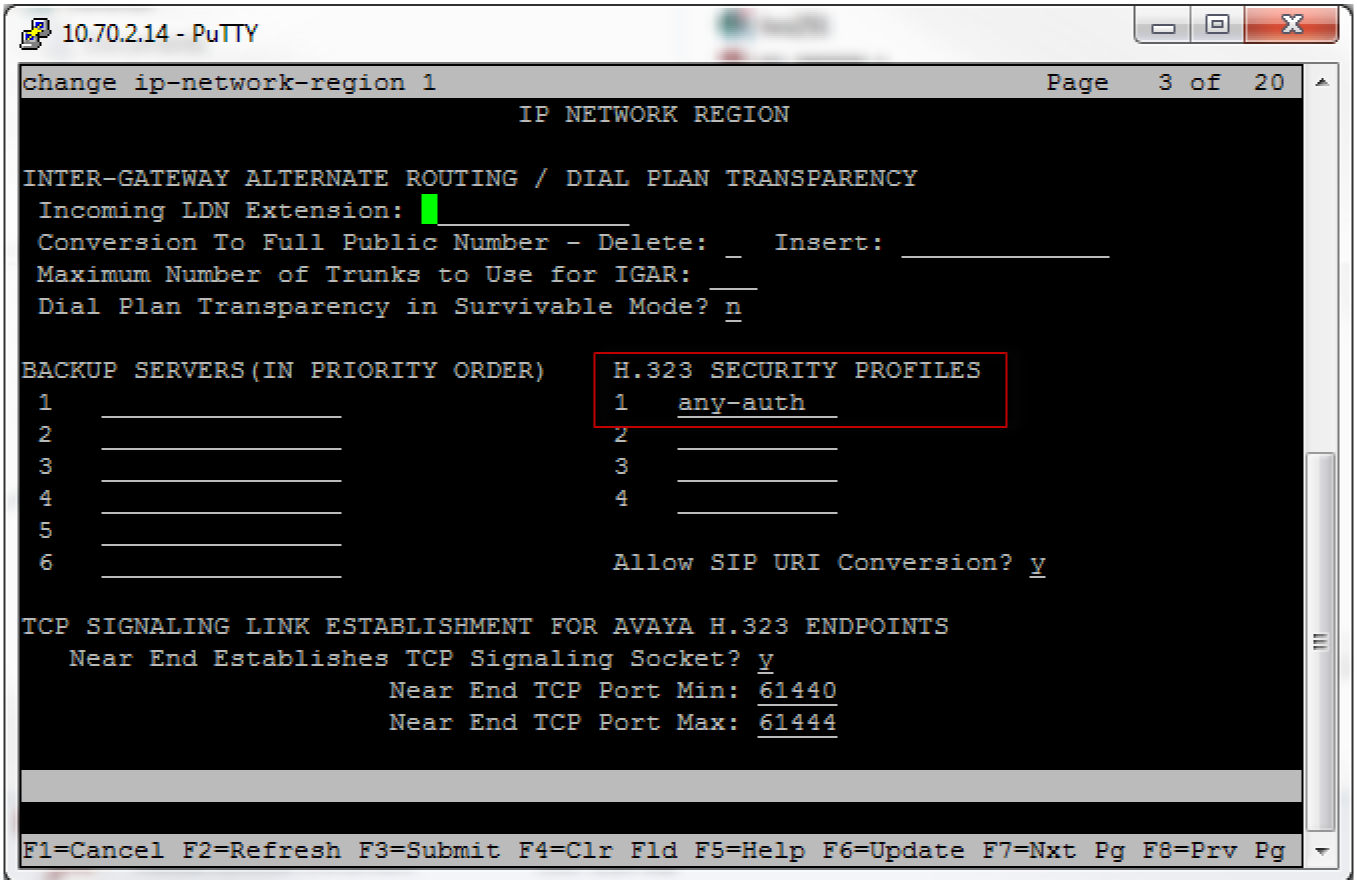
Name:tekvizion

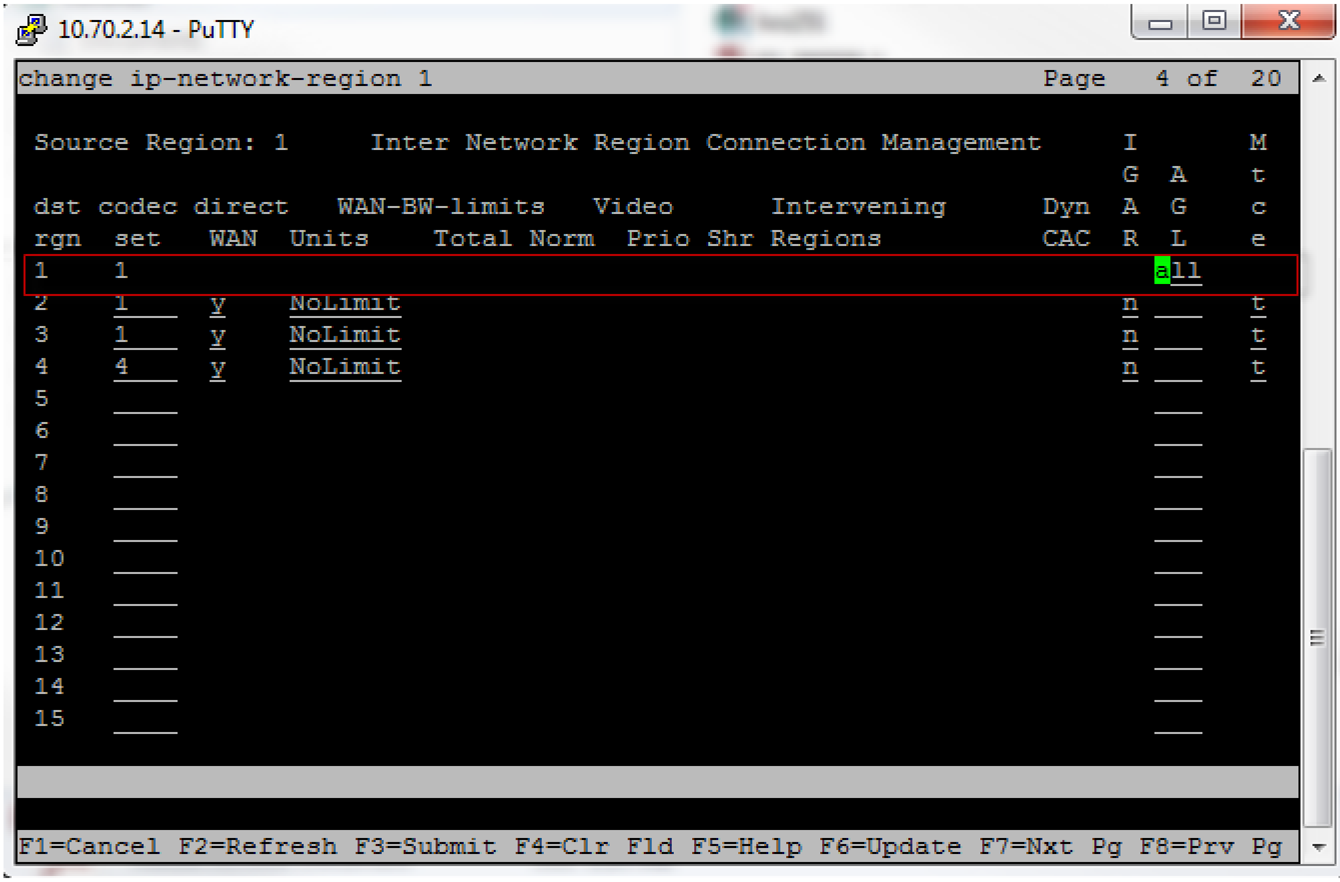
Codec Set: 1 which programmed in previous step

Inter/Intra-region IP-IP Direct Audio:YES

H.323 SECURITY PROFILES: any-auth




10.70.2.14 - PuTTY
change ip-network-region 1 Page 3 of 20
IP NETWORK REGION
INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
Incoming LDN Extension:
Conversion To Full Public Number - Delete: Insert:
Maximum Number of Trunks to Use for IGAR:
Dial Plan Transparency in Survivable Mode? n
BACKUP SERVERS(IN PRIORITY ORDER)
1
2
3
4
5
6
H.323 SECURITY PROFILES
1 any-auth
2
3
4
Allow SIP URI Conversion? y
TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
Near End Establishes TCP Signaling Socket? y
Near End TCP Port Min: 61440
Near End TCP Port Max: 61444
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg



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10.70.2.14 - PuTTY
change ip-network-region 1                                      Page   4 of  20

 Source Region: 1      Inter Network Region Connection Management      I       M
                                                                       G  A    t
 dst codec direct   WAN-BW-limits   Video       Intervening      Dyn   A  G    c
 rgn  set   WAN  Units     Total Norm  Prio Shr Regions          CAC   R  L    e
 1    1                                                                   all
 2    1     y    NoLimit                                               n       t
 3    1     y    NoLimit                                               n       t
 4    4     y    NoLimit                                               n       t
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```
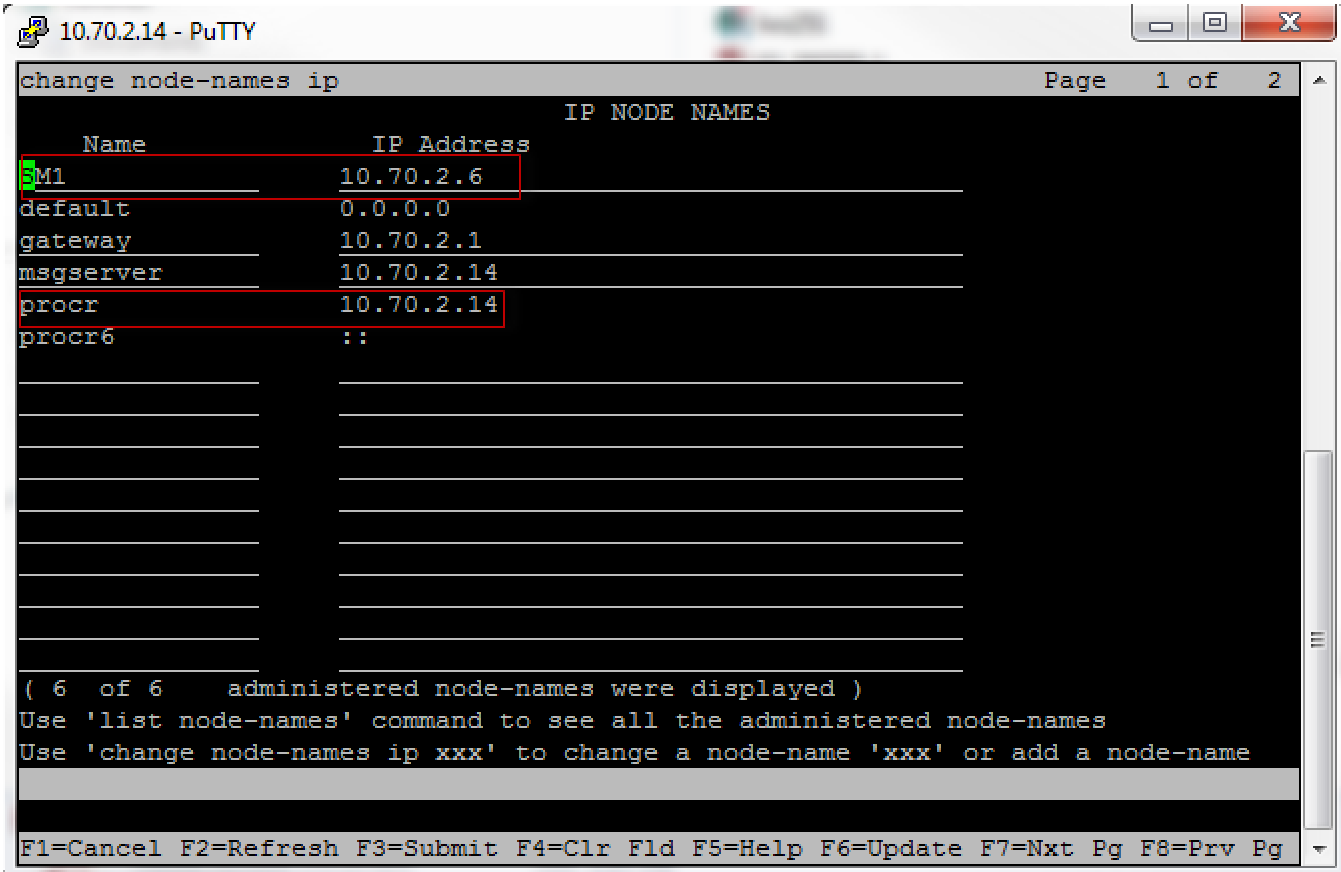

Configure the Signaling group and trunk Group

Configure the Node IP for Avaya Session manager and CM



Configure the Signaing Group 4:

Set Group Type: sip

IMS Enabled? N

Transport Method: tcp

IP Video? Y

Priority Video? Y

Peer Detection Enabled? Y

Near-end Node Name: procr

Far-end Node Name: SM1

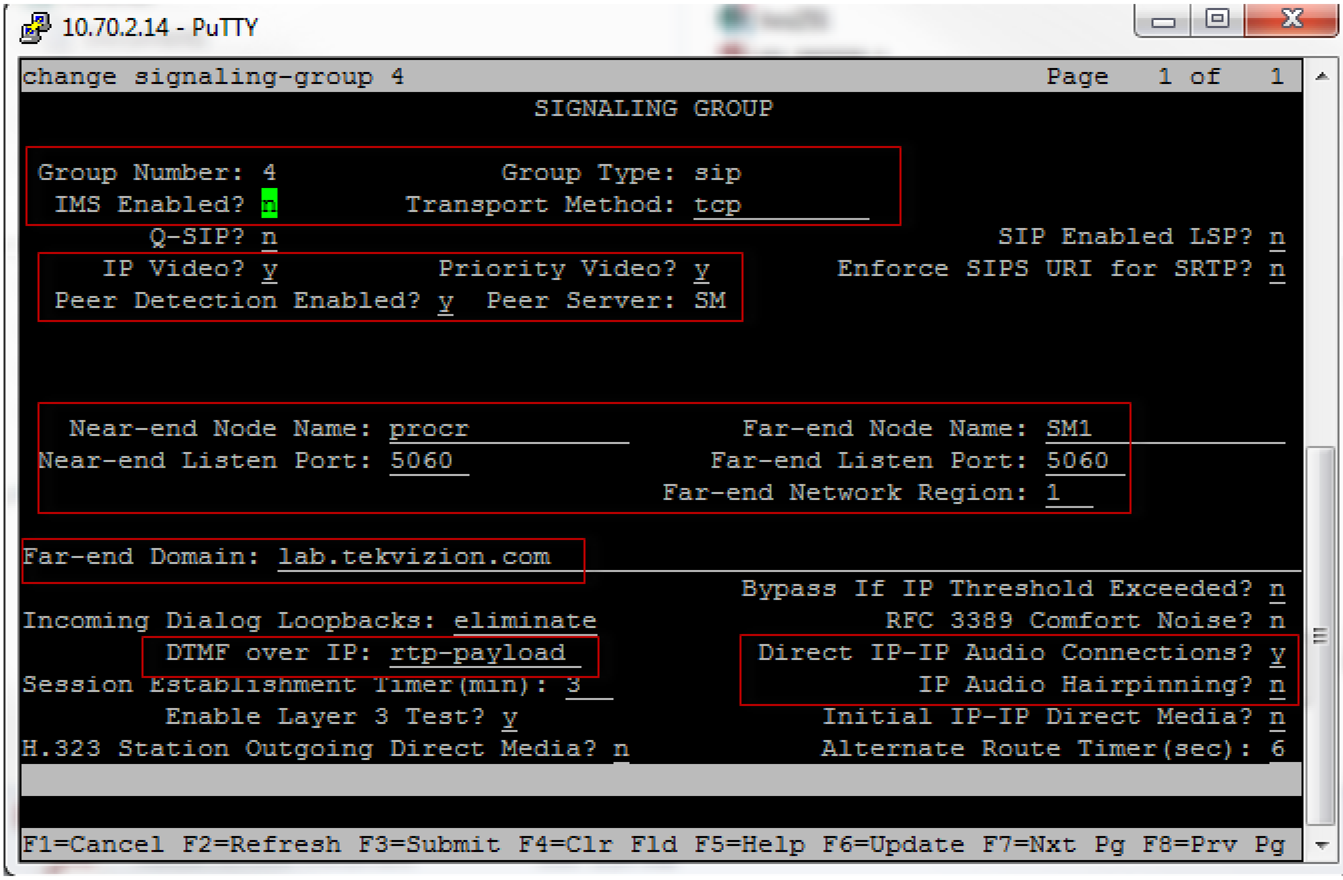
Near-end Listen Port: 5060

Far-end Listen Port: 506

Far-end Network Region: 1

DTMF over IP: rtp-payload

Direct IP-IP Audio Connections? Y



Configure trunk group 4:

Group number: 4

Group Type:sip

Group Name:SIP to Cisco

TAC:*104

Member Assignment Method:auto

Service Type:tieSignaling Group:4

Number of Members:10

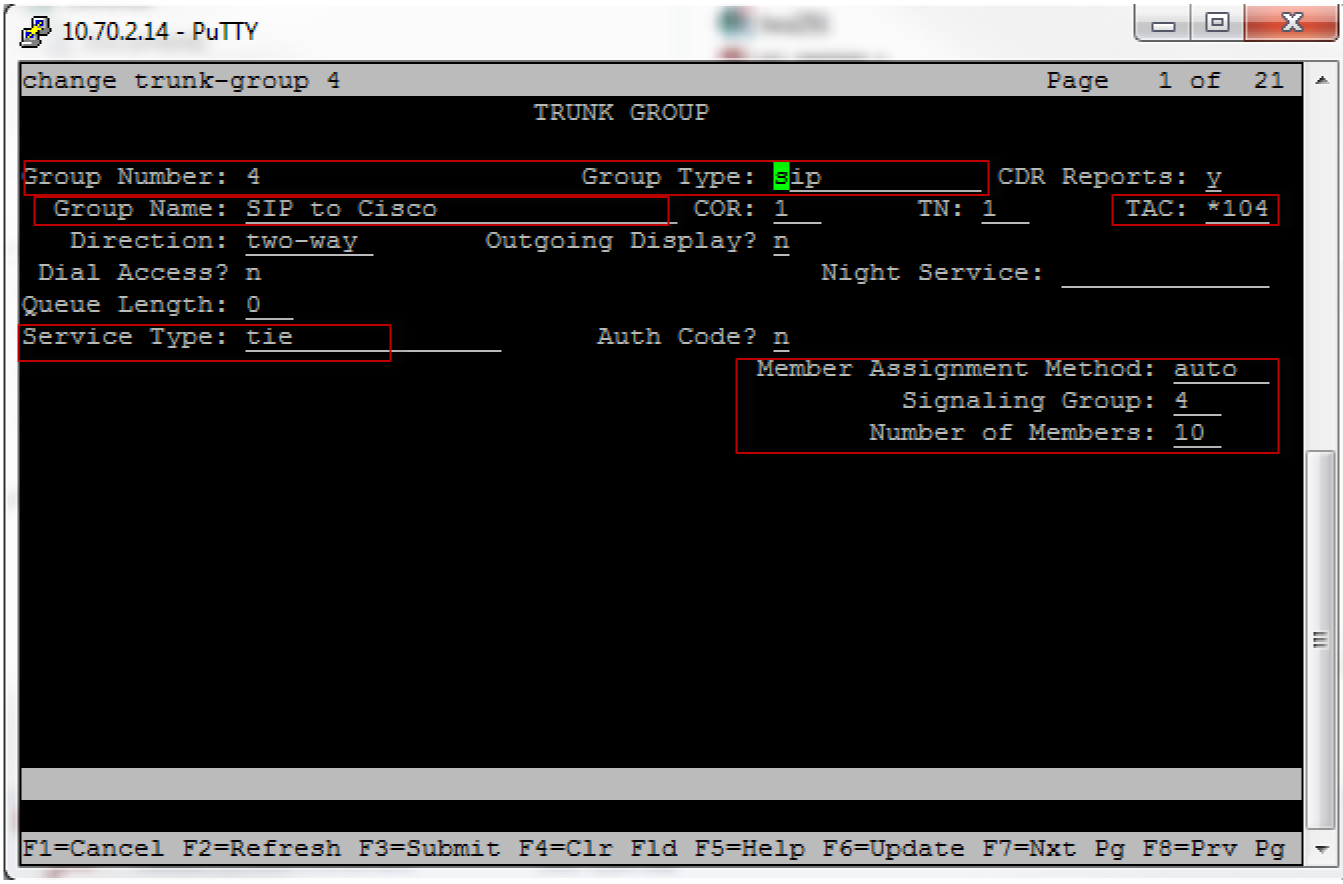
Preferred Minimum Session Refresh Interval(sec): 900

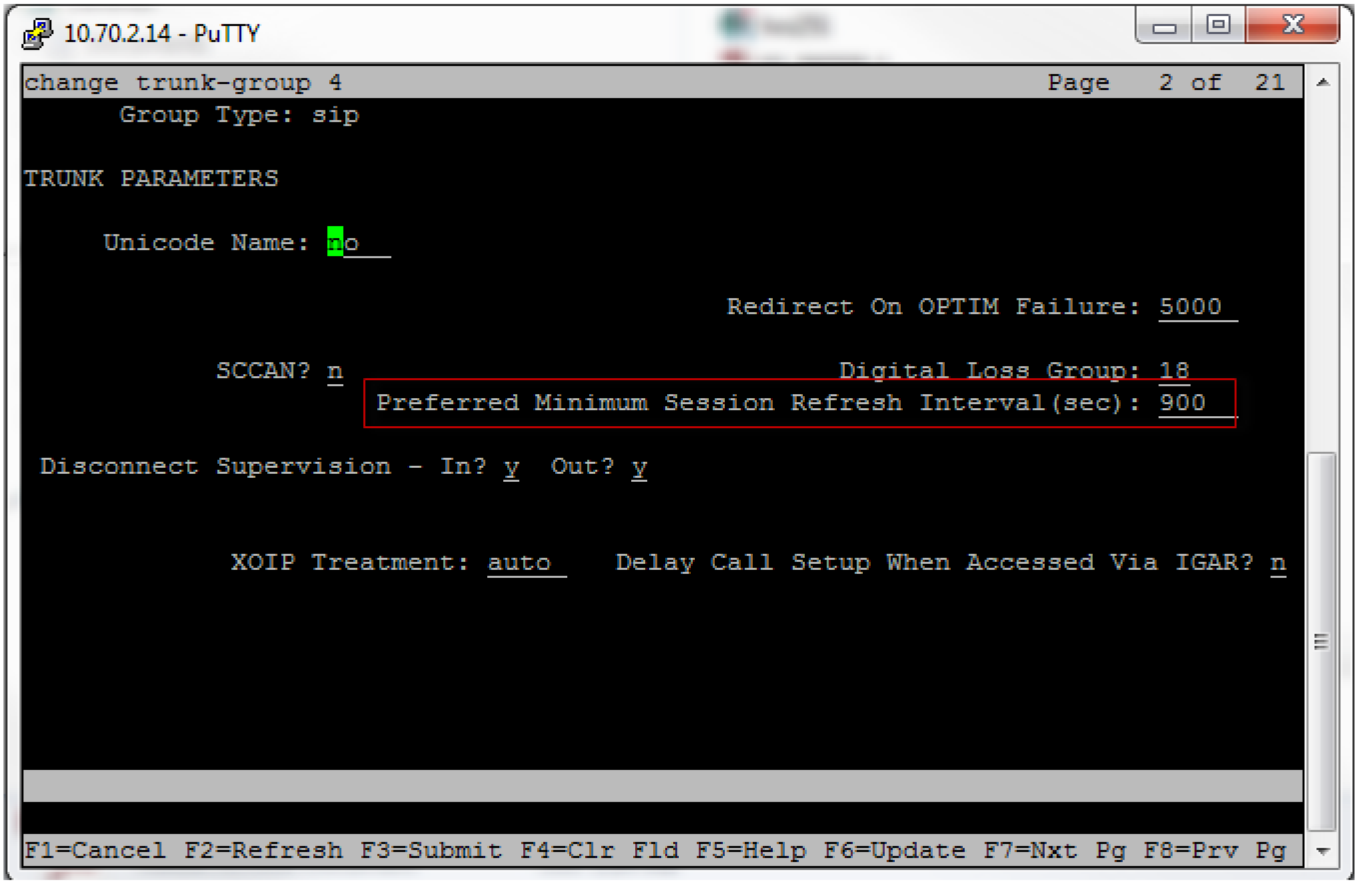
Numbering Format: private

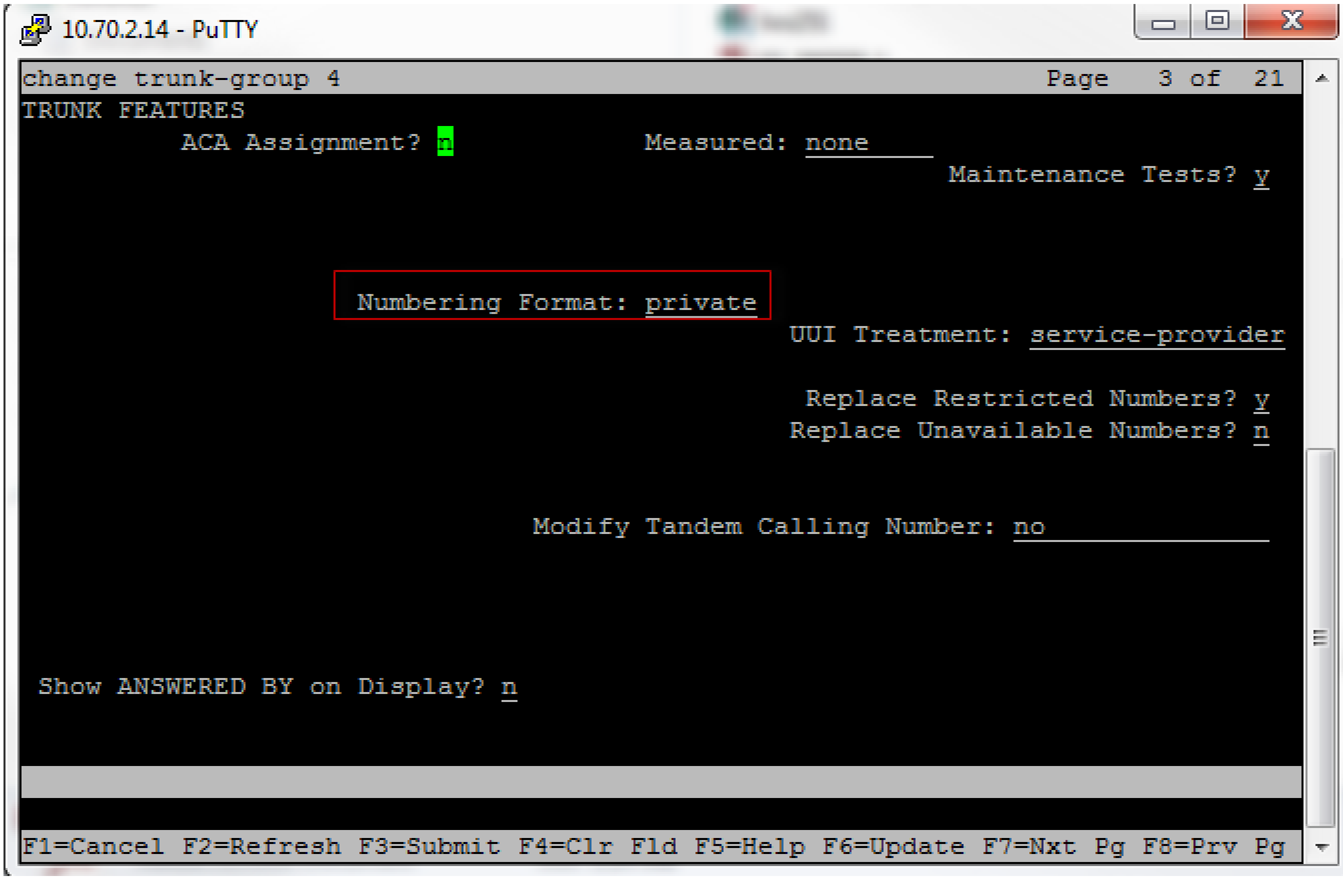
Mark Users as Phone? Y

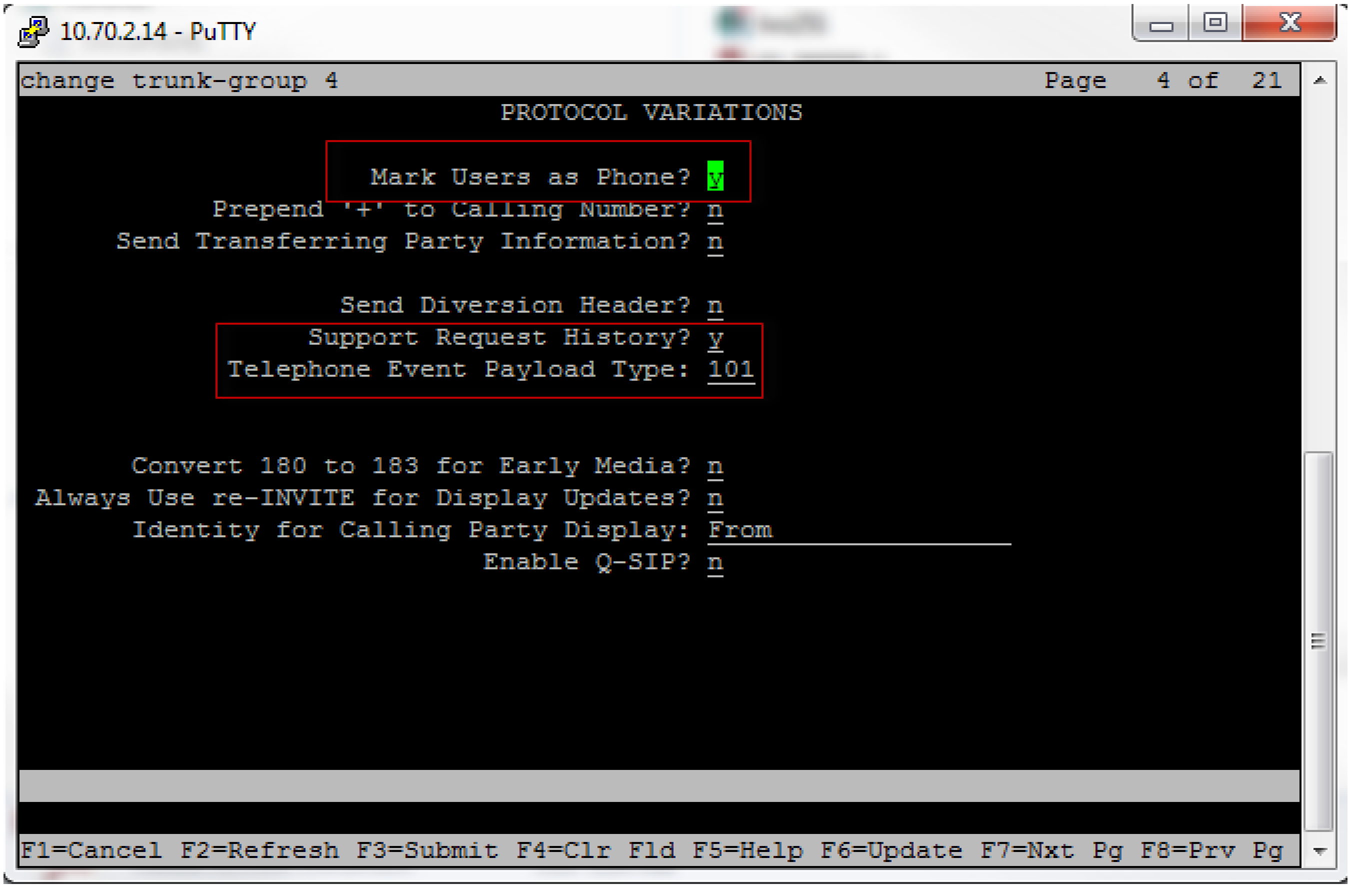
Support Request History? Y

Telephone Event Payload Type: 101


10.70.2.14 - PuTTY
change trunk-group 4 Page 1 of 21
TRUNK GROUP
Group Number: 4 Group Type: sip CDR Reports: y
Group Name: SIP to Cisco COR: 1 TN: 1 TAC: *104
Direction: two-way Outgoing Display? n
Dial Access? n Night Service:
Queue Length: 0
Service Type: tie Auth Code? n
Member Assignment Method: auto
Signaling Group: 4
Number of Members: 10
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg


10.70.2.14 - PuTTY
change trunk-group 4 Page 2 of 21
Group Type: sip
TRUNK PARAMETERS
Unicode Name: no
Redirect On OPTIM Failure: 5000
SCCAN? n Digital Loss Group: 18
Preferred Minimum Session Refresh Interval(sec): 900
Disconnect Supervision - In? y Out? y
XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg


10.70.2.14 - PuTTY
change trunk-group 4
Page 3 of 21
TRUNK FEATURES
ACA Assignment? n
Measured: none
Maintenance Tests? y
Numbering Format: private
UUI Treatment: service-provider
Replace Restricted Numbers? y
Replace Unavailable Numbers? n
Modify Tandem Calling Number: no
Show ANSWERED BY on Display? n
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg


10.70.2.14 - PuTTY
change trunk-group 4
Page 4 of 21
PROTOCOL VARIATIONS
Mark Users as Phone? y
Prepend '+' to Calling Number? n
Send Transferring Party Information? n
Send Diversion Header? n
Support Request History? y
Telephone Event Payload Type: 101
Convert 180 to 183 for Early Media? n
Always Use re-INVITE for Display Updates? n
Identity for Calling Party Display: From
Enable Q-SIP? n
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Configure Route pattern:

Pattern Number: 4

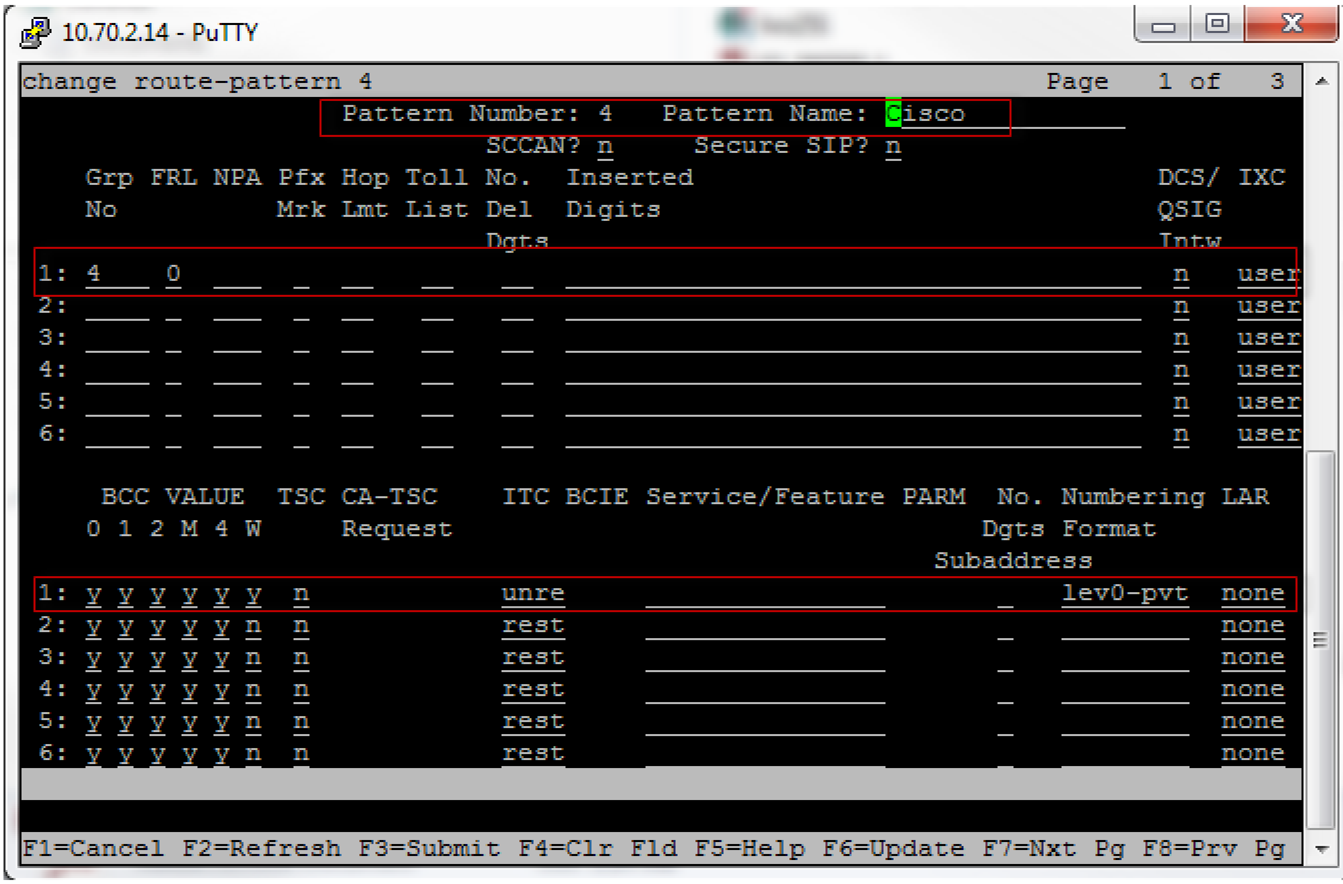
Pattern name: Cisco

Grp No: 4

FRL: 0

ITC:unre

Numbering Format:lev0-pvt

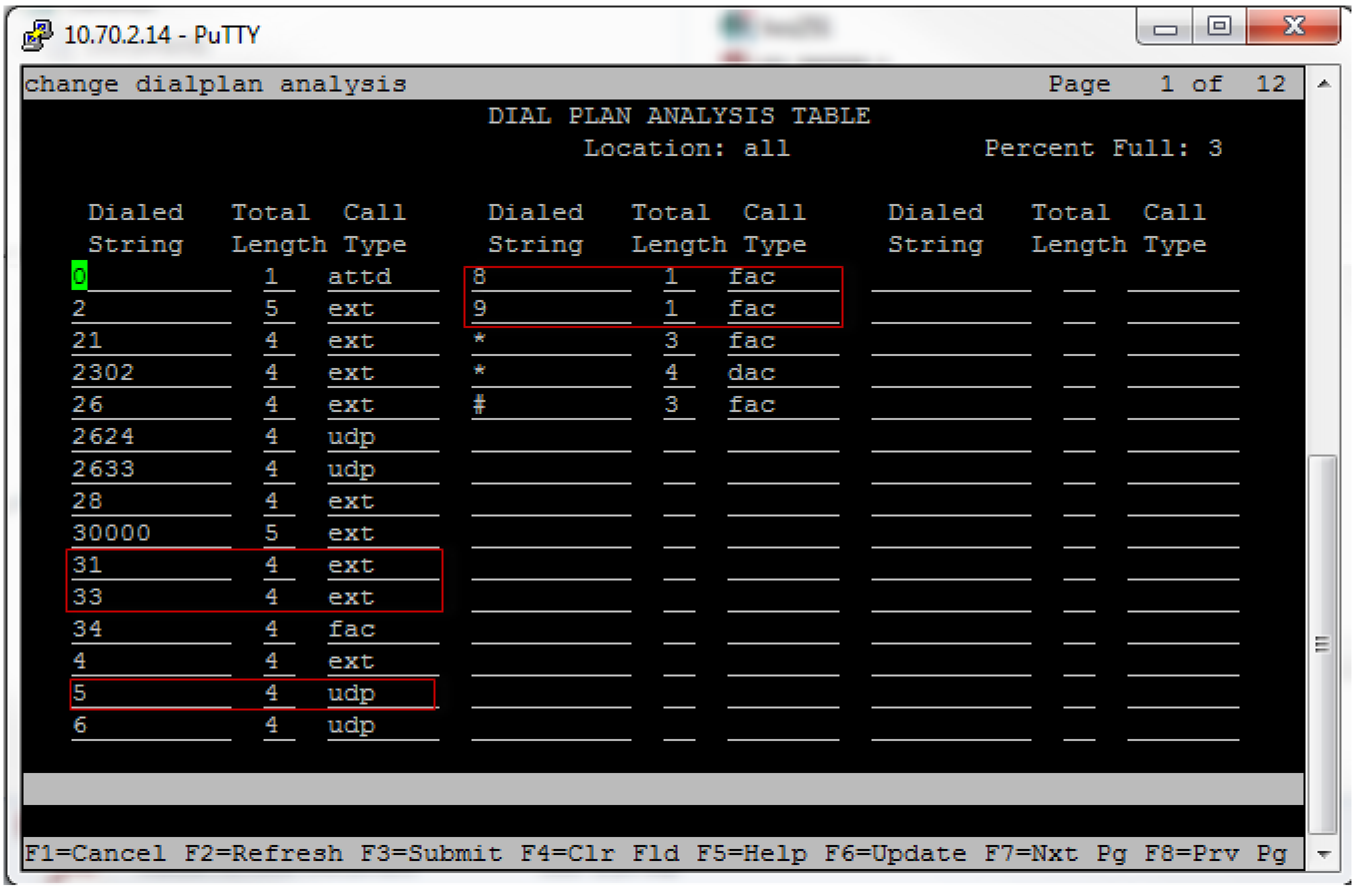


Dialing plan:

Configure 4 digits number start with 31 and 33 as ext

Configure 4 digit number start with 5 as udp

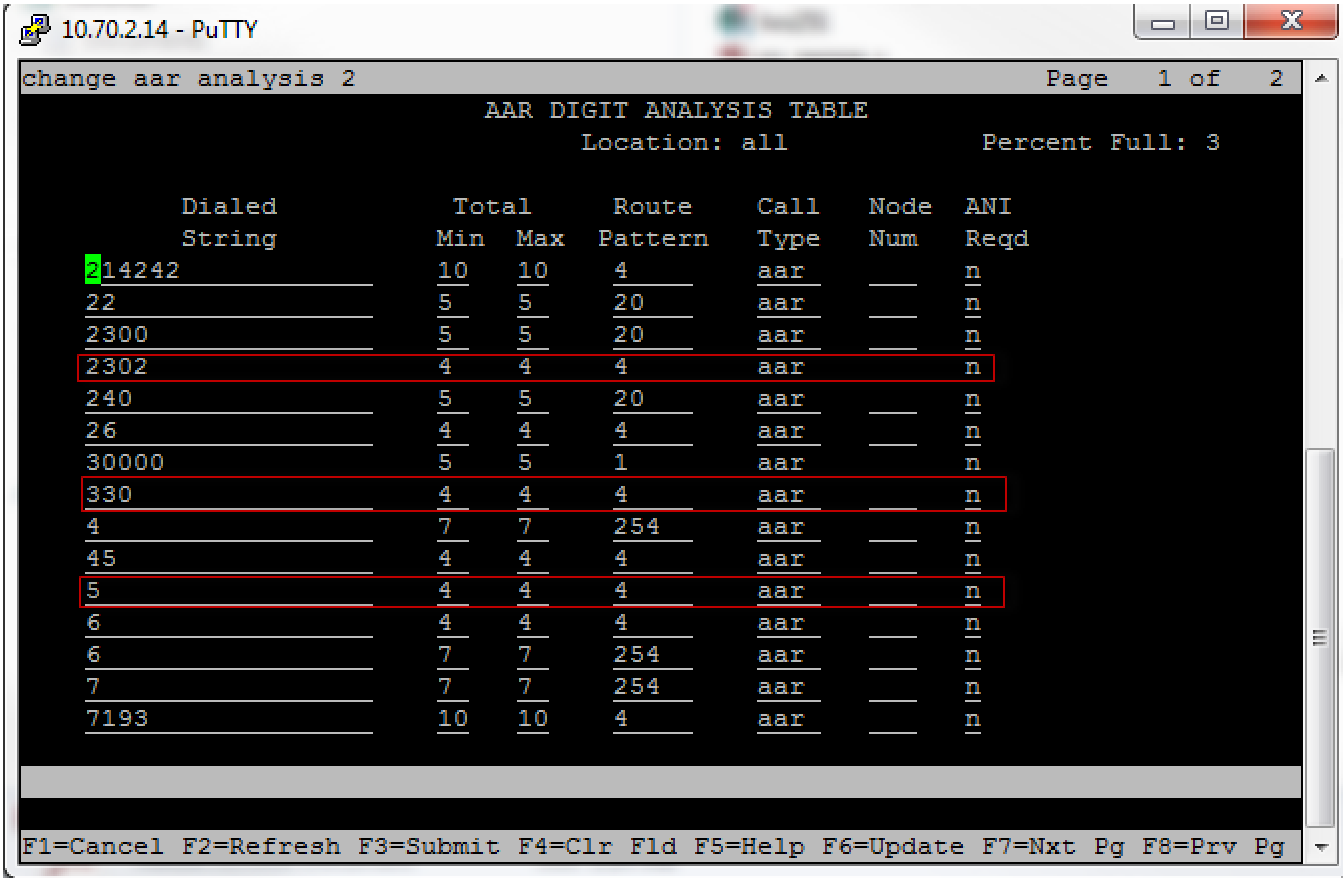
8 and 9 are set as 1 digit fac code.




```
change dialplan analysis                                      Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all            Percent Full: 3

    Dialed   Total  Call       Dialed   Total  Call       Dialed   Total  Call
    String   Length Type       String   Length Type       String   Length Type
   0           1    attd      8           1   fac
   2           5    ext       9           1   fac
   21          4    ext       *           3   fac
   2302        4    ext       *           4   dac
   26          4    ext       #           3   fac
   2624        4    udp
   2633        4    udp
   28          4    ext
   30000       5    ext
   31          4    ext
   33          4    ext
   34          4    fac
   4           4    ext
   5           4    udp
   6           4    udp

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

Configure the AAR dialplan:

Set 4 digits dial string start with 2302(Unity mail), 330(Avaya SIP phone) and 5(Cisco phone) to use Route pattern 4 with Call Type aar.


10.70.2.14 - PuTTY

change aar analysis 2   Page   1 of   2

AAR DIGIT ANALYSIS TABLE

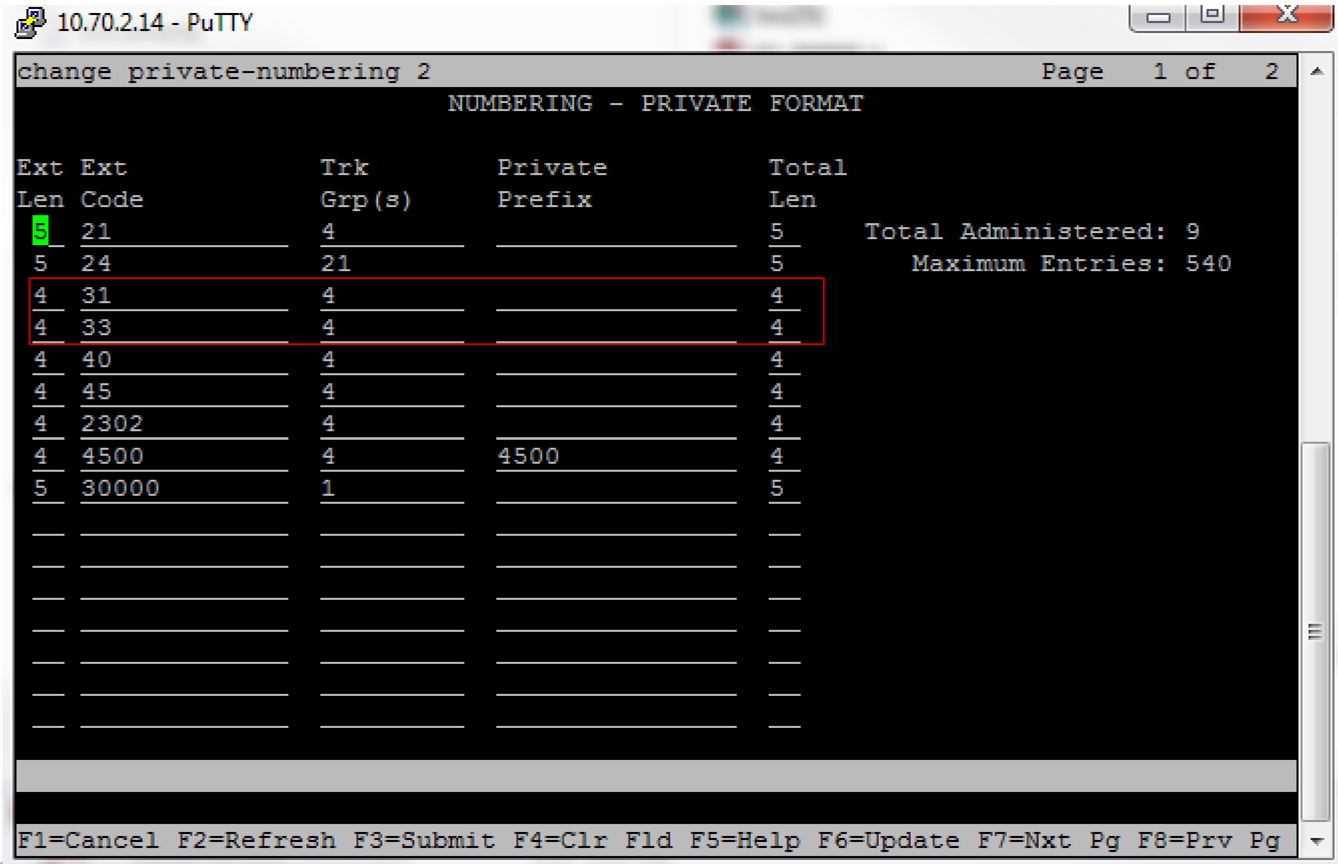
Location: all   Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
214242	10	10	4	aar		n
22	5	5	20	aar		n
2300	5	5	20	aar		n
2302	4	4	4	aar		n
240	5	5	20	aar		n
26	4	4	4	aar		n
30000	5	5	1	aar		n
330	4	4	4	aar		n
4	7	7	254	aar		n
45	4	4	4	aar		n
5	4	4	4	aar		n
6	4	4	4	aar		n
6	7	7	254	aar		n
7	7	7	254	aar		n
7193	10	10	4	aar		n

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Configure Private numbering plan:

Use the **change private-numbering** command to define the calling party number to be sent out through SIP trunk.in our case, 4 digits extension with leading digits 31 and 33 via trunk group 4 will result in a 4-digit calling number.



Fill in the indicated fields as shown below and use default values for remaining fields.

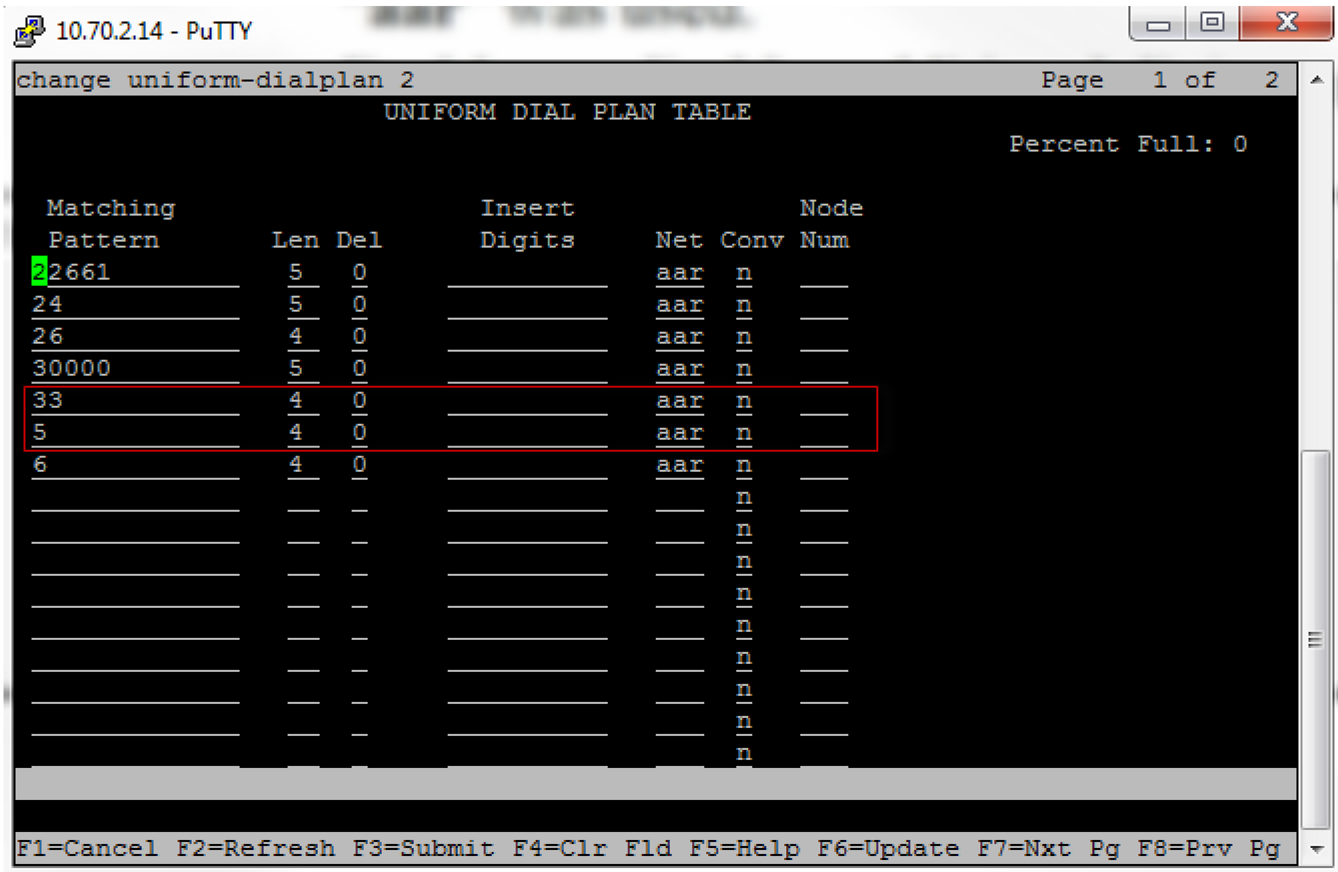
Configure Uniform dialplan:

Matching Pattern Enter the number Communication Manager matches to dialed numbers. Accepts up to seven digits. 33 and 5 are used in the example

Len Enter the number of user-dialed digits the system collects to match to this Matching Pattern value. 4 is used in the example

Del Enter number of digits to delete before routing the call. 0 is selected

Net The server or switch network used to analyze the converted, aar is used here



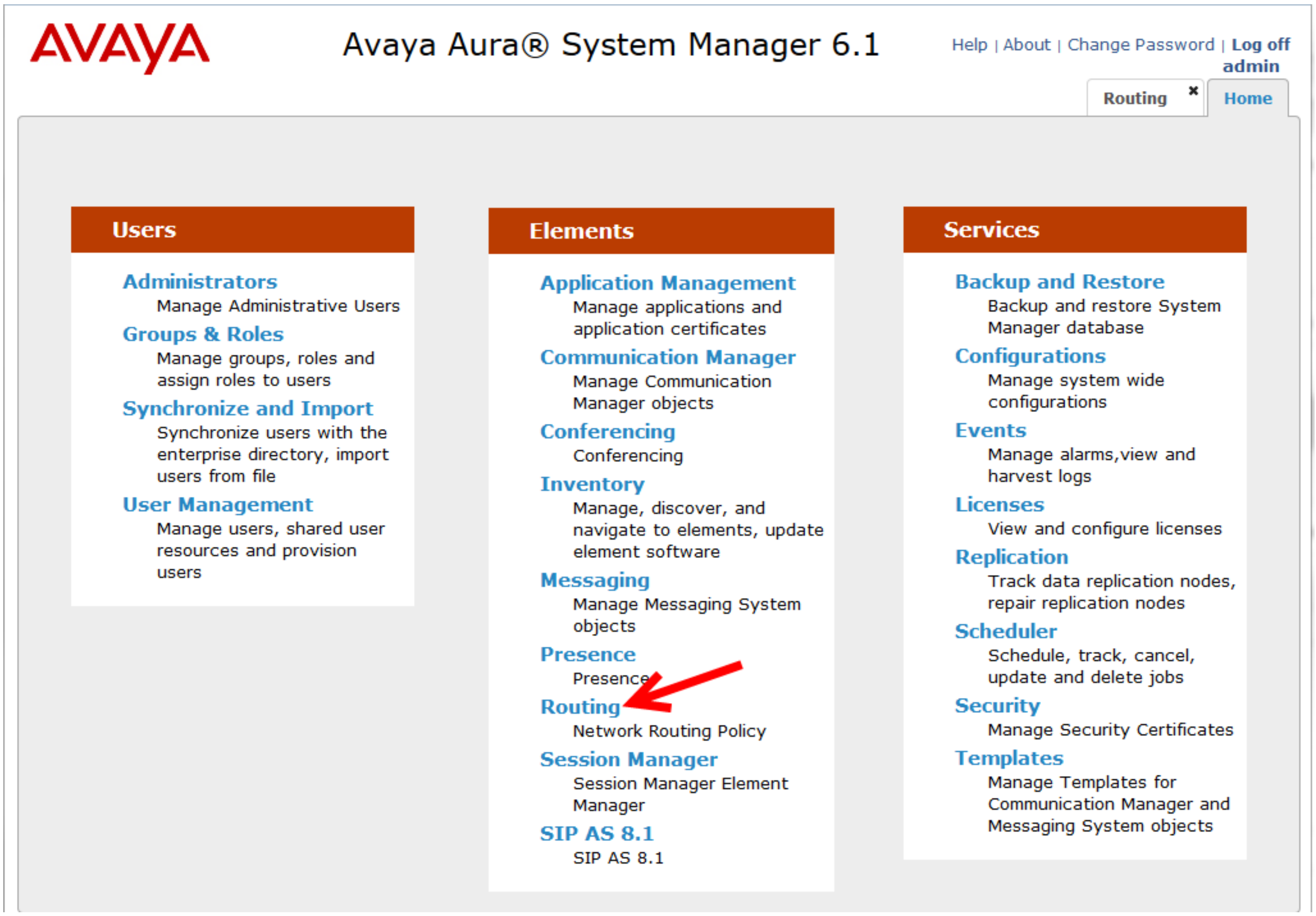
Save Translation

After finished above configuration, use the “save translation” command to save these changes.

Configure Avaya Aura Session Manager

Access Avaya Aura System Manager web login screen via https://<IP Address/FQDN>, For this test, IP address used is 10.70.2.4. Use admin as User ID and associated password, and then "Log on"

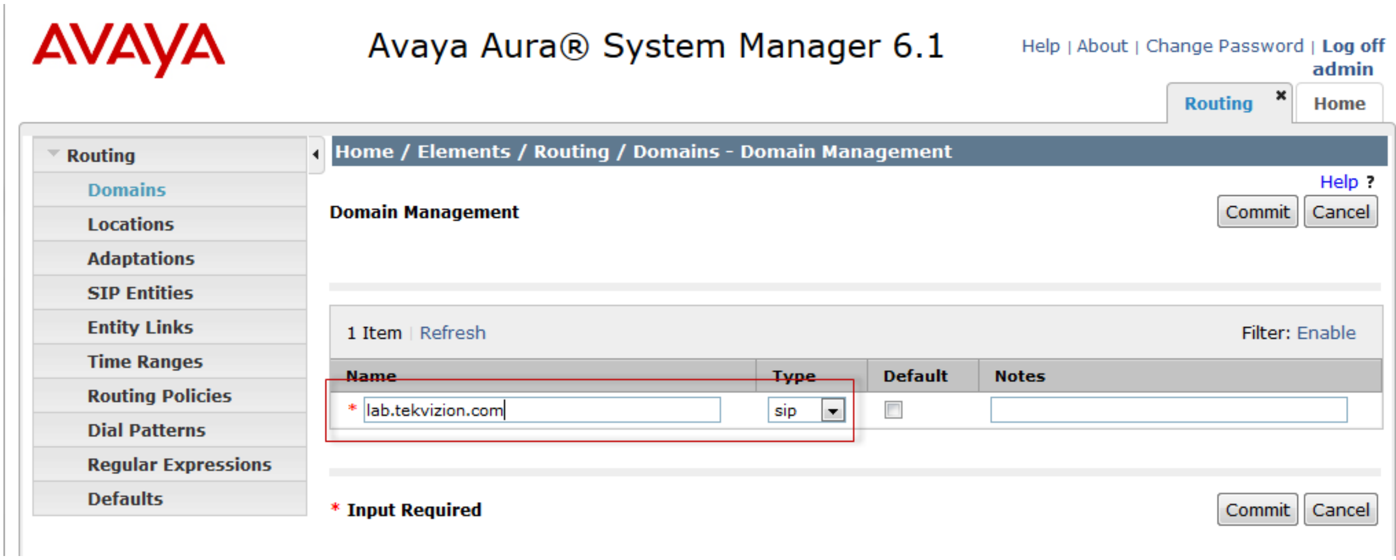
Navigation: Home→Elements→Routing



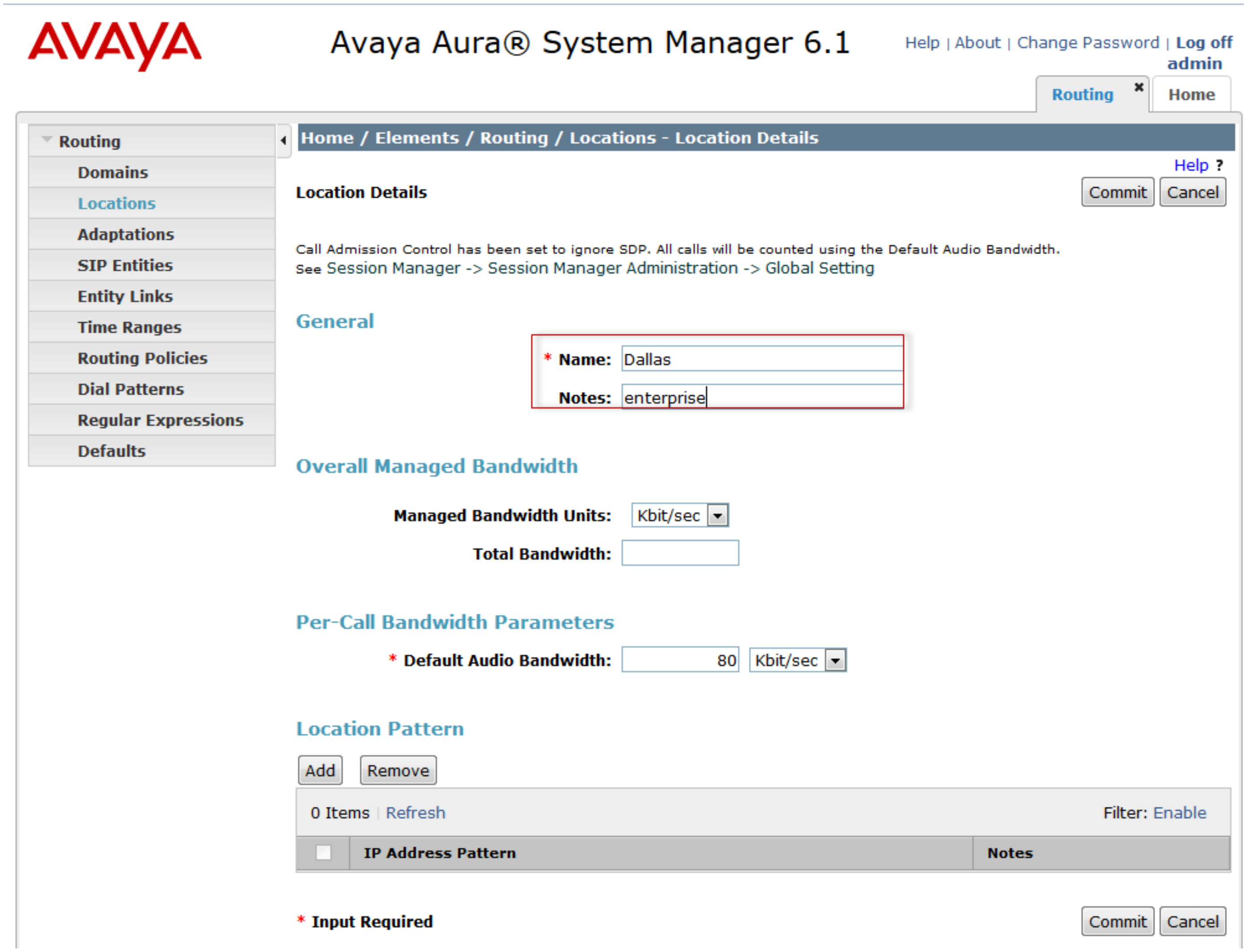
Add Domains
Under page Domain Management:

Name: lab.tekvizion.com

Type:sip



Add Location
Name: Dallas



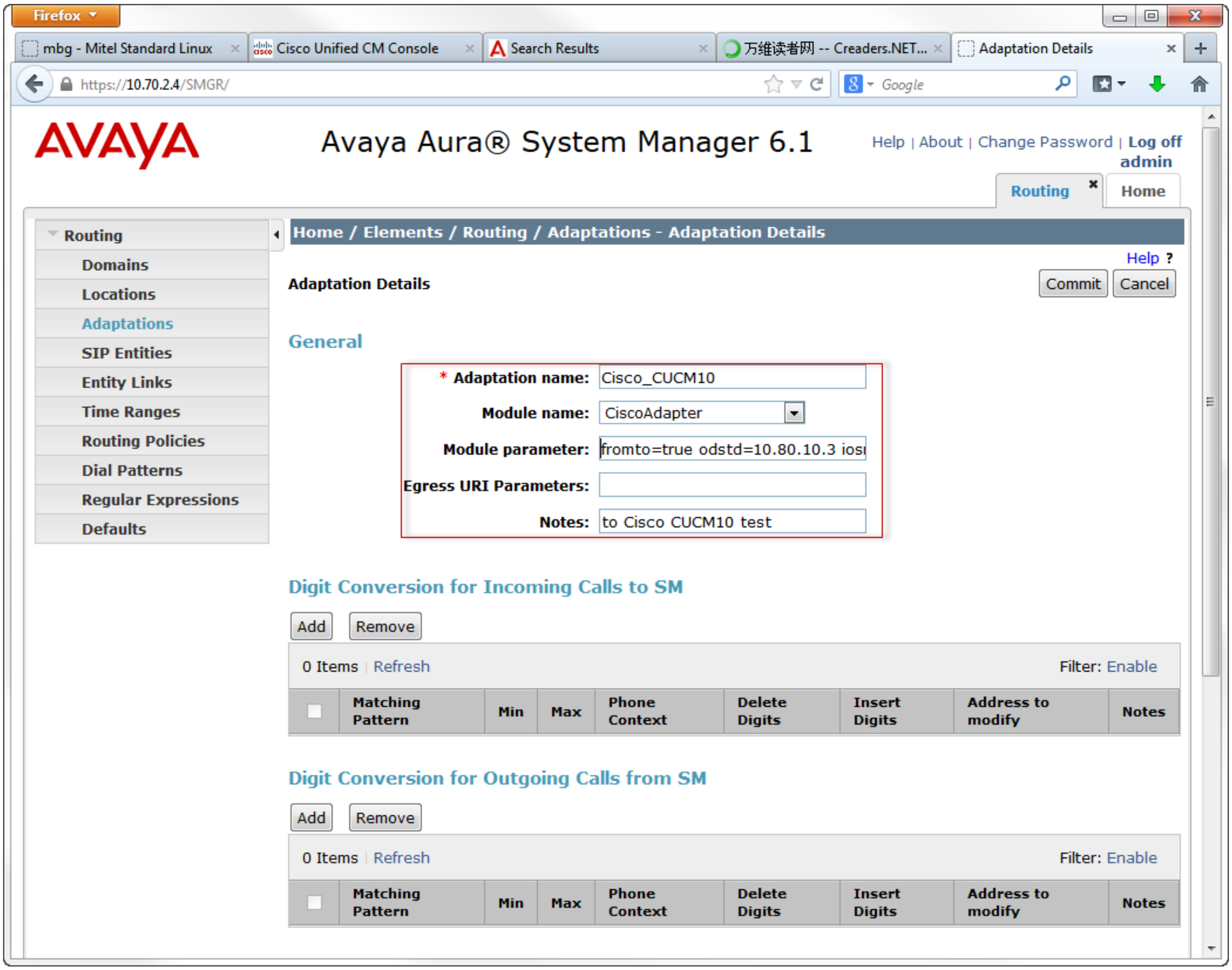
Add Adaptations

Adaptation for Cisco CUCM

Adaptation name: Cisco_CUCM10

Module name: CiscoAdapter

Module Parameter: fromto=true odstd=10.80.10.3 iosrcd=lab.tekvizion.com

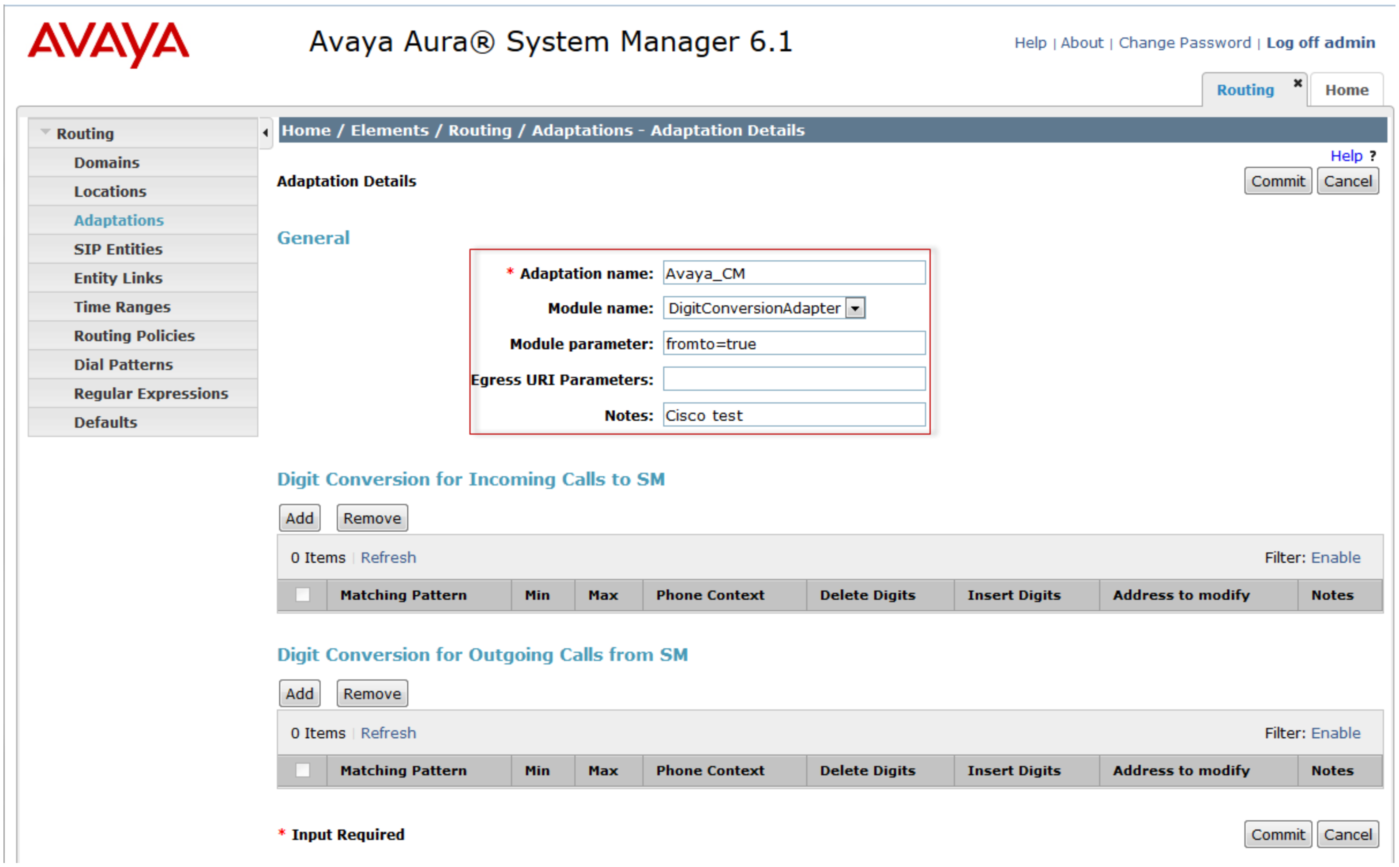


Adaptation for Avaya Aura CM

Adaptation name:Avaya_CM

Module name: DigitConversionAdapter

Module Parameter: fromto=true



Add SIP Entities and Entity Link

SIP Entity for Session Manager

Name: teksm

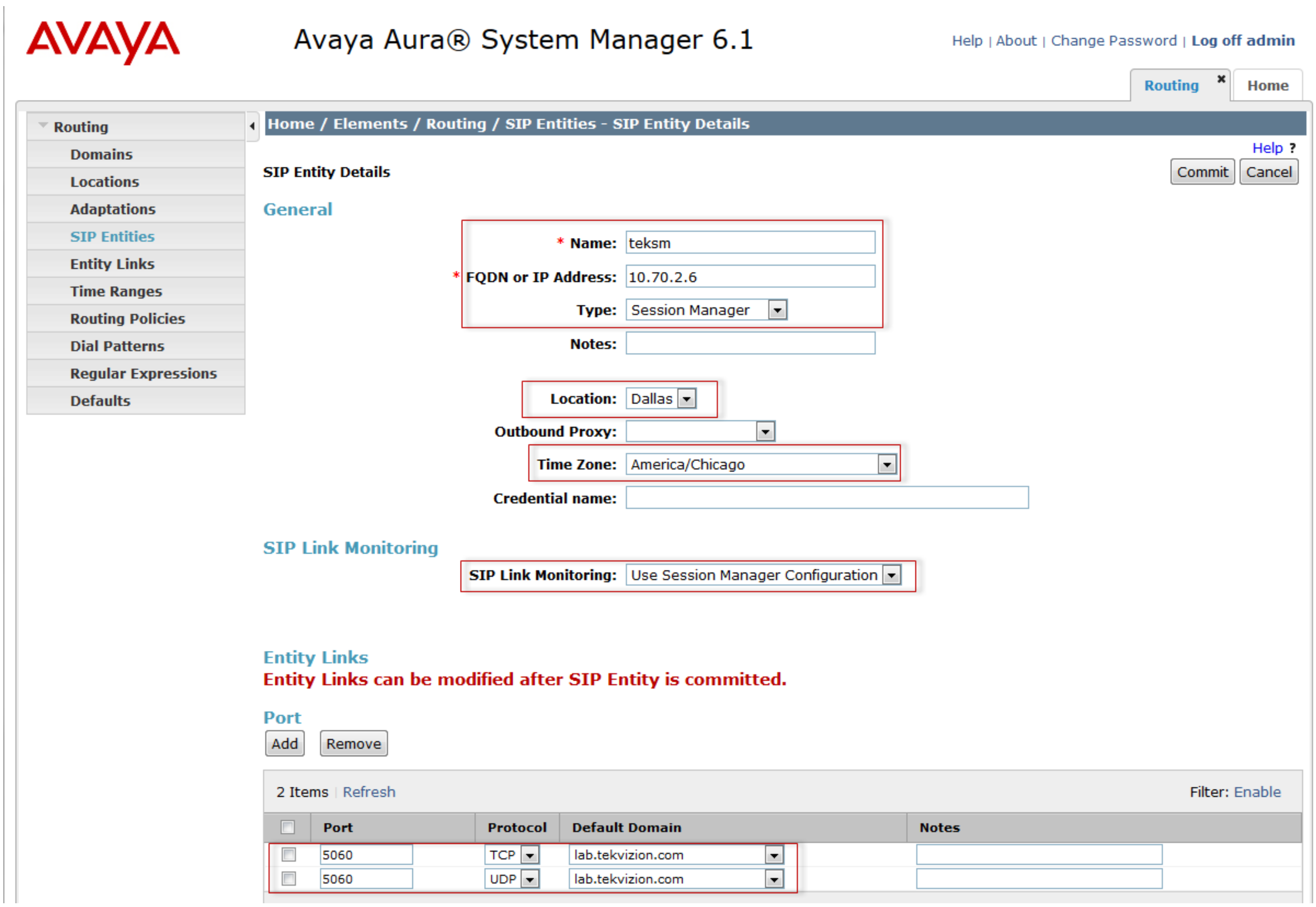
FQDN or IP Address: 10.70.2.6

Type: Session Manager

Location: Dallas

Time Zone: America/Chicago

SIP Link Monitoring: Use Session manager Configuration



SIP Entity and entity Link for CUCM

Name: Cisco_CUCM10

FQDN or IP Address: 10.80.10.3

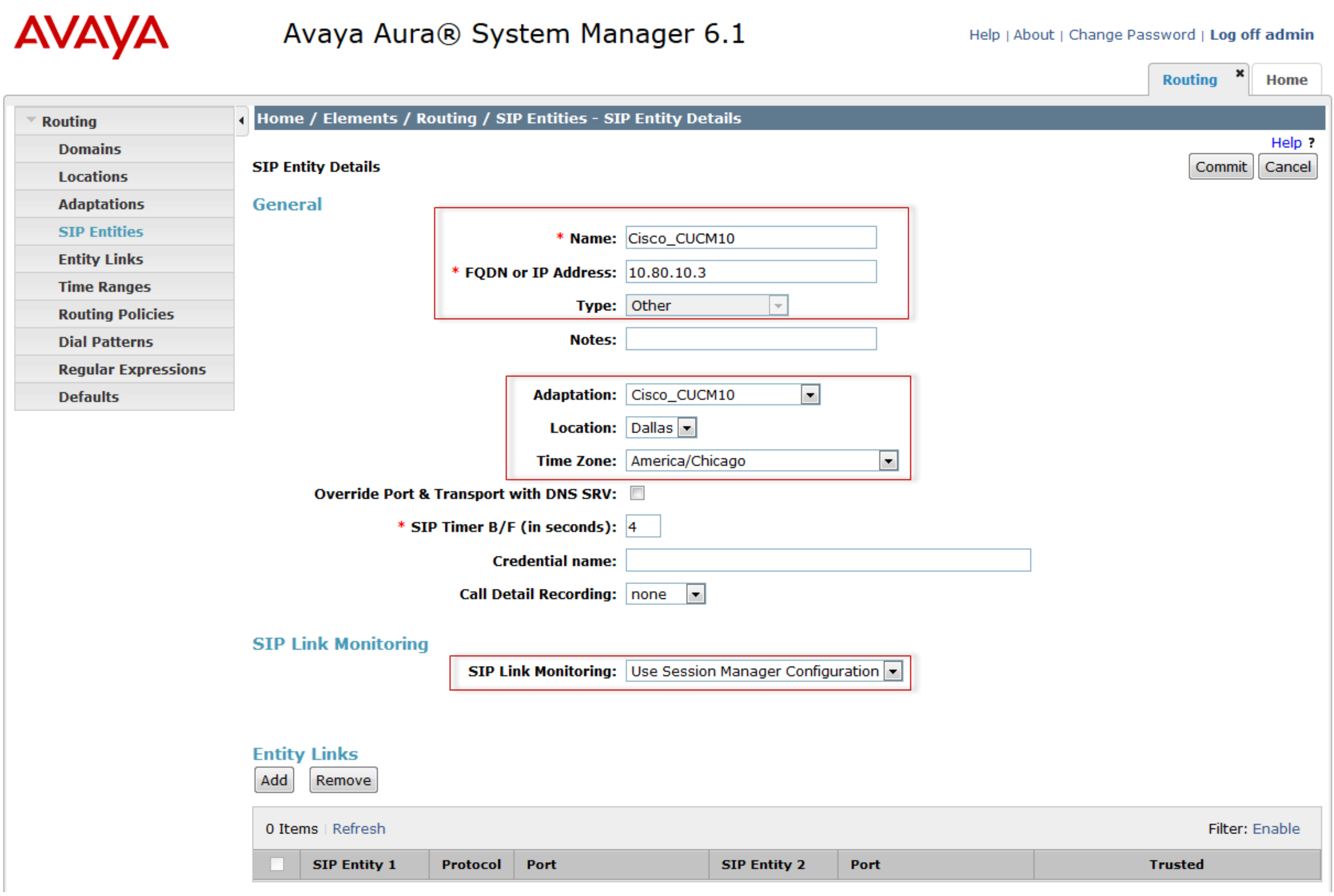
Type: Other

Adaptation: Cisco_CUCM10

Location: Dallas

Time Zone: America/Chicago

SIP Link Monitoring:Use Session Manager Configuration



SIP Entity and Entity Link for Avaya Aura Communication manager

Name: tekcm

FQDN or IP Address: 10.70.2.14

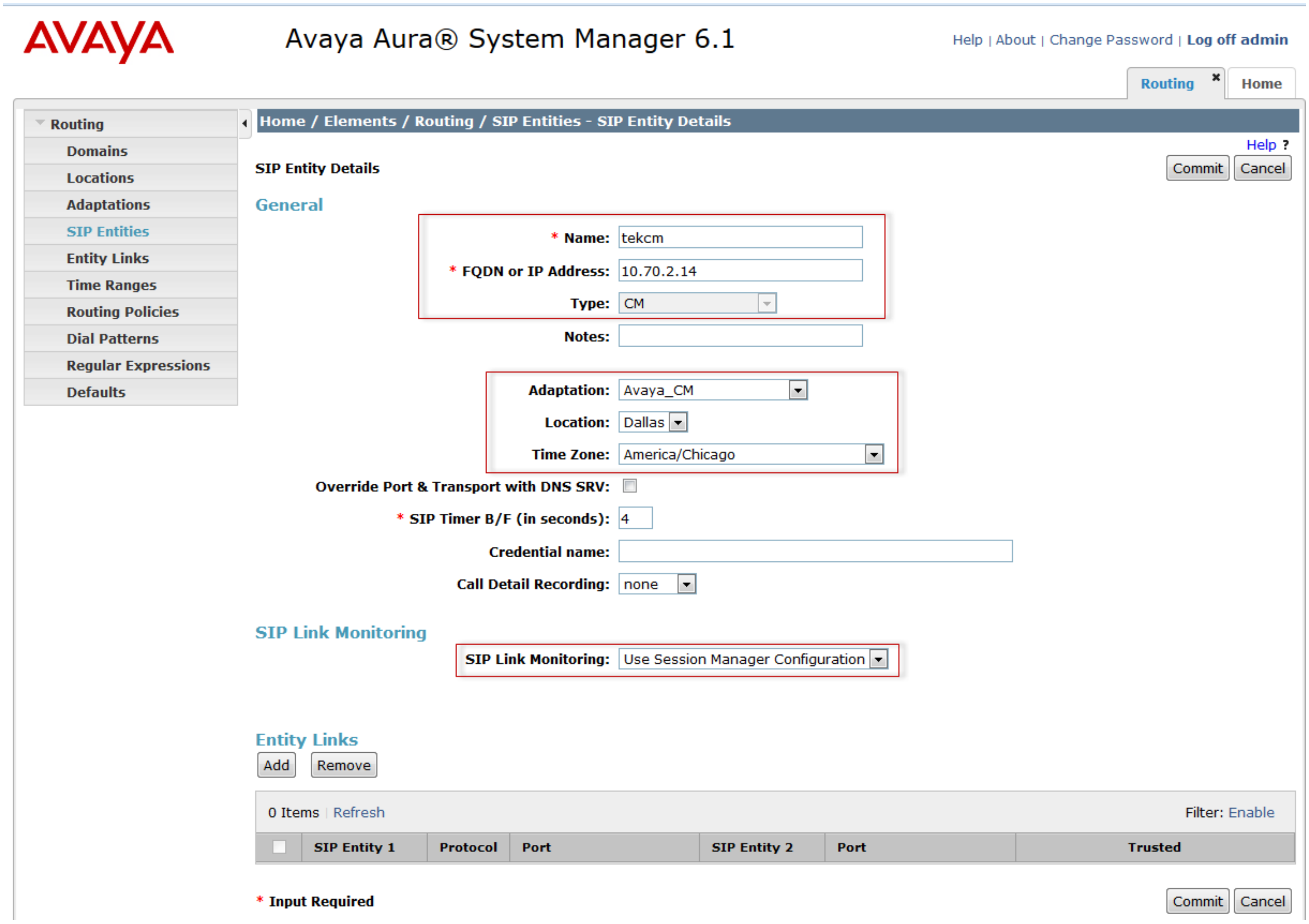
Type: CM

Adaptation: Avaya_CM

Location: Dallas

Time Zone: Chicago

Sip Link Monitoring: Use Session Manager Configuration



Add Entity Links
Add entity link between Avaya Session manager and Cisco CUCM:

Name:ASM to CUCM10

SIP Entity 1:teksm

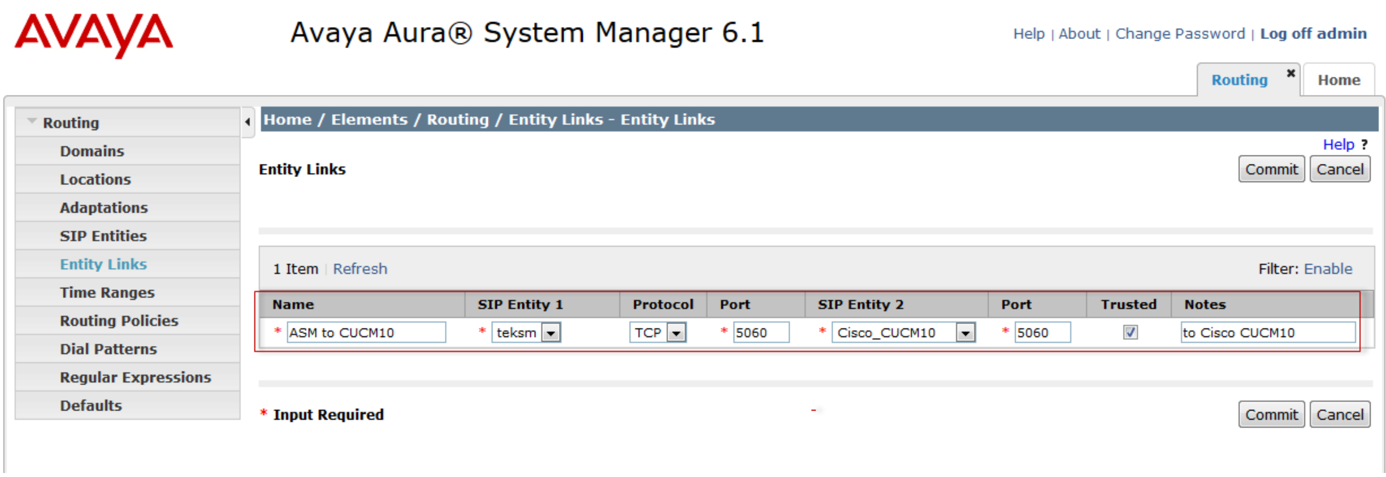
Protocol:tcp

Port 5060

SIP Entity 2:Cisco_CUCM10

Port 5060

Trusted:checked



Add entity link between Avaya Session manager and Avaya Aura Communication Manager:

Name:teksm_tekcm_5060_TCP

SIP Entity 1:teksm

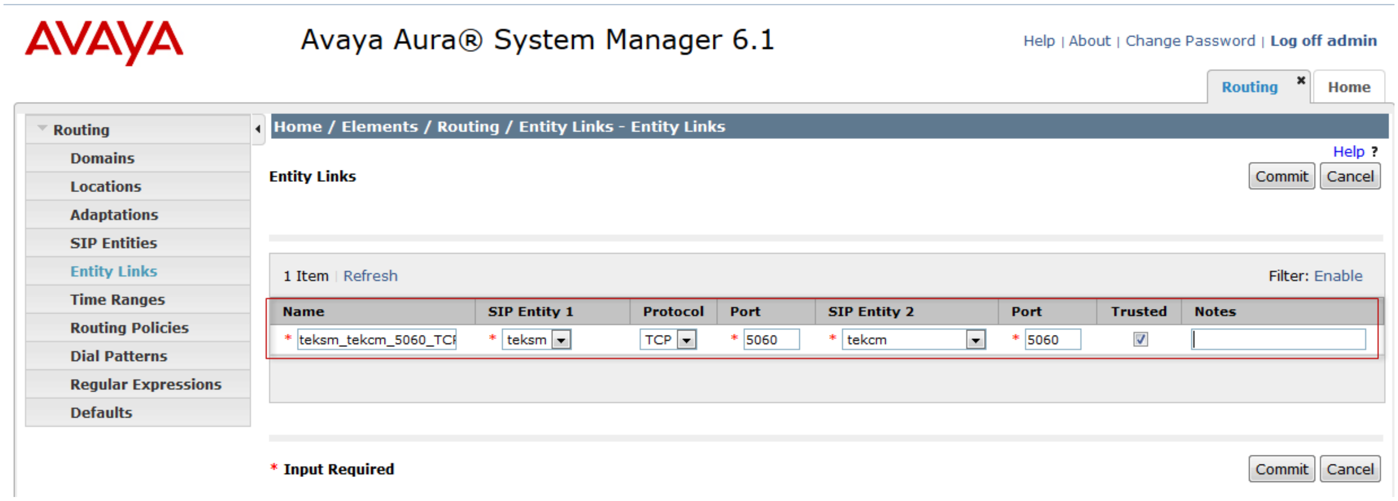
Protocol:tcp

Port 5060

SIP Entity 2:tekcm

Port 5060

Trusted:checked

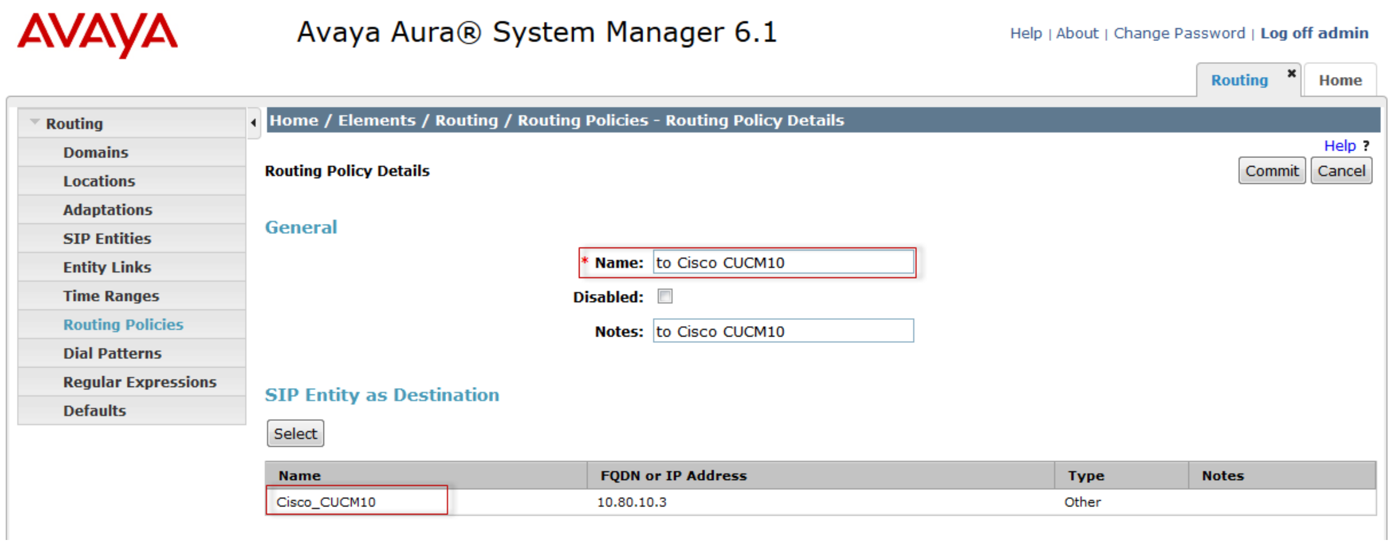


Add Routing Polices

Routing policy for call to go to Cisco CUCM

Name: to Cisco CUCM10

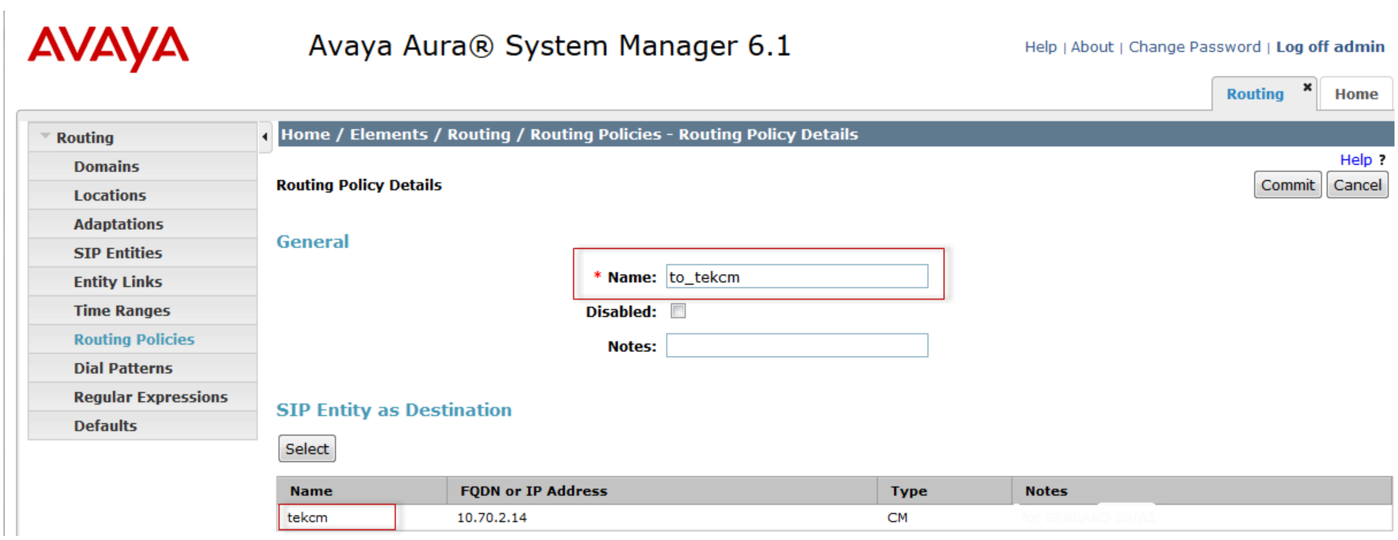
Select SIP Entity “Cisco_CUCM10” for SIP Entity as Destination



Routing Policy for calls to go to Avaya Aura Communication Manager

Name: To_tekcm

Select SIP Entity “tekcm” for SIP Entity as Destination



Add Dial Pattern

Dial pattern to Cisco CUCM

Pattern: 5

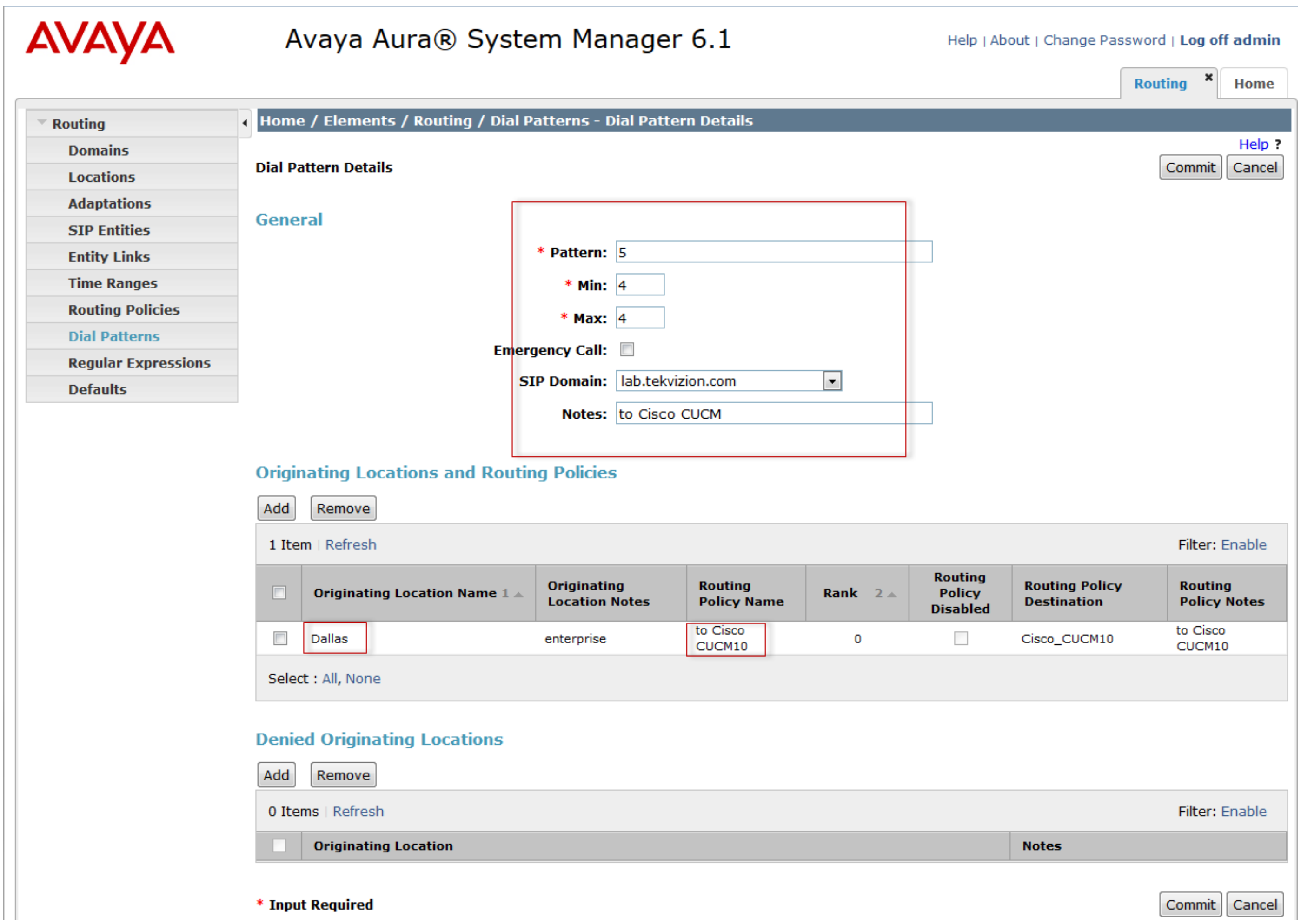
Min: 4

Max: 4

SIP Domain: lab.tekvizion.com

Original Location Name: Dallas

Routing Policy Name: to Cisco CUCM10



Dial Pattern to Avaya Aura Communication Manager

Pattern: 310

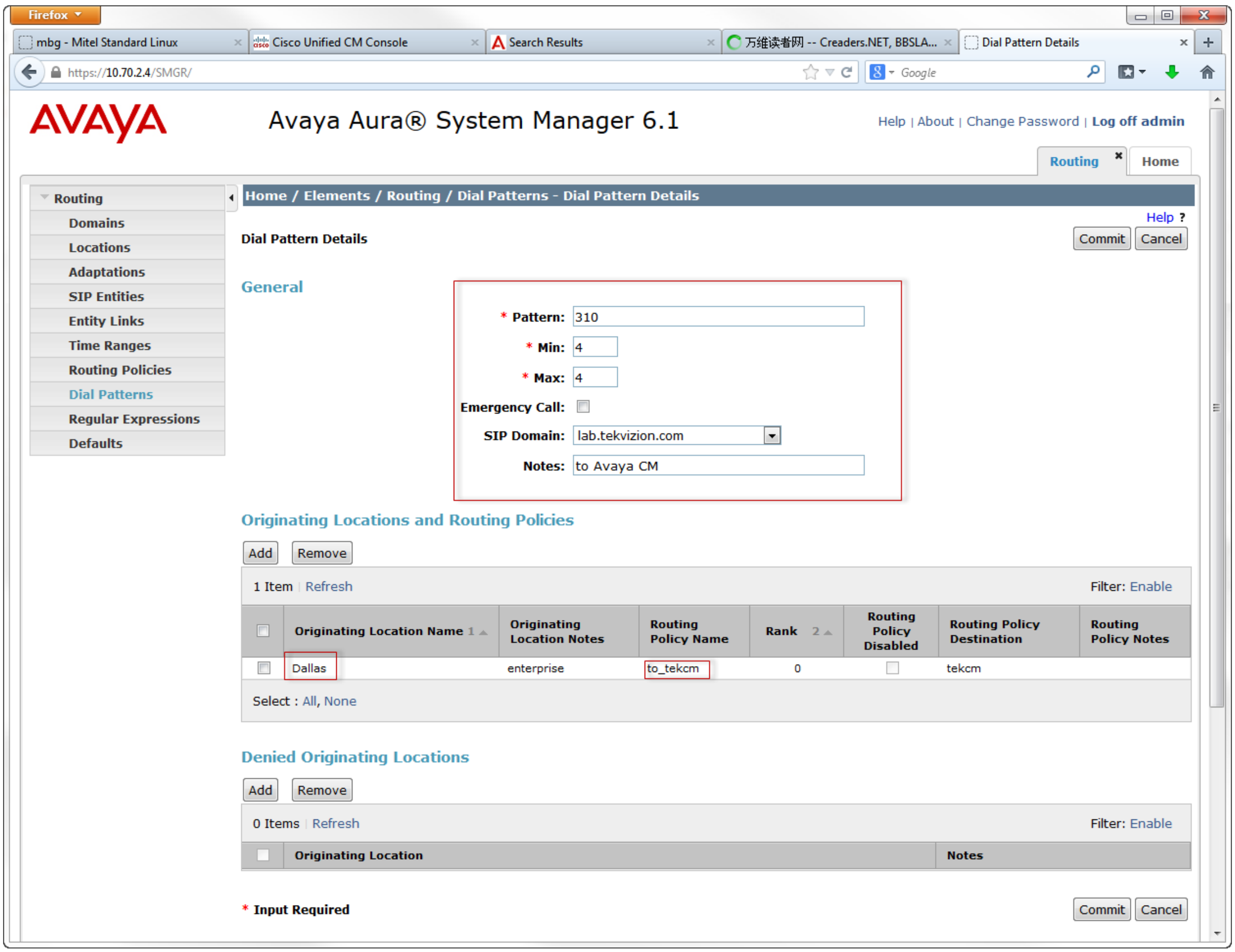
Min: 4

Max: 4

SIP Domain: lab.tekvizion.com

Original Location Name: Dallas

Routing Policy Name: to_tekcm



Pattern: 330

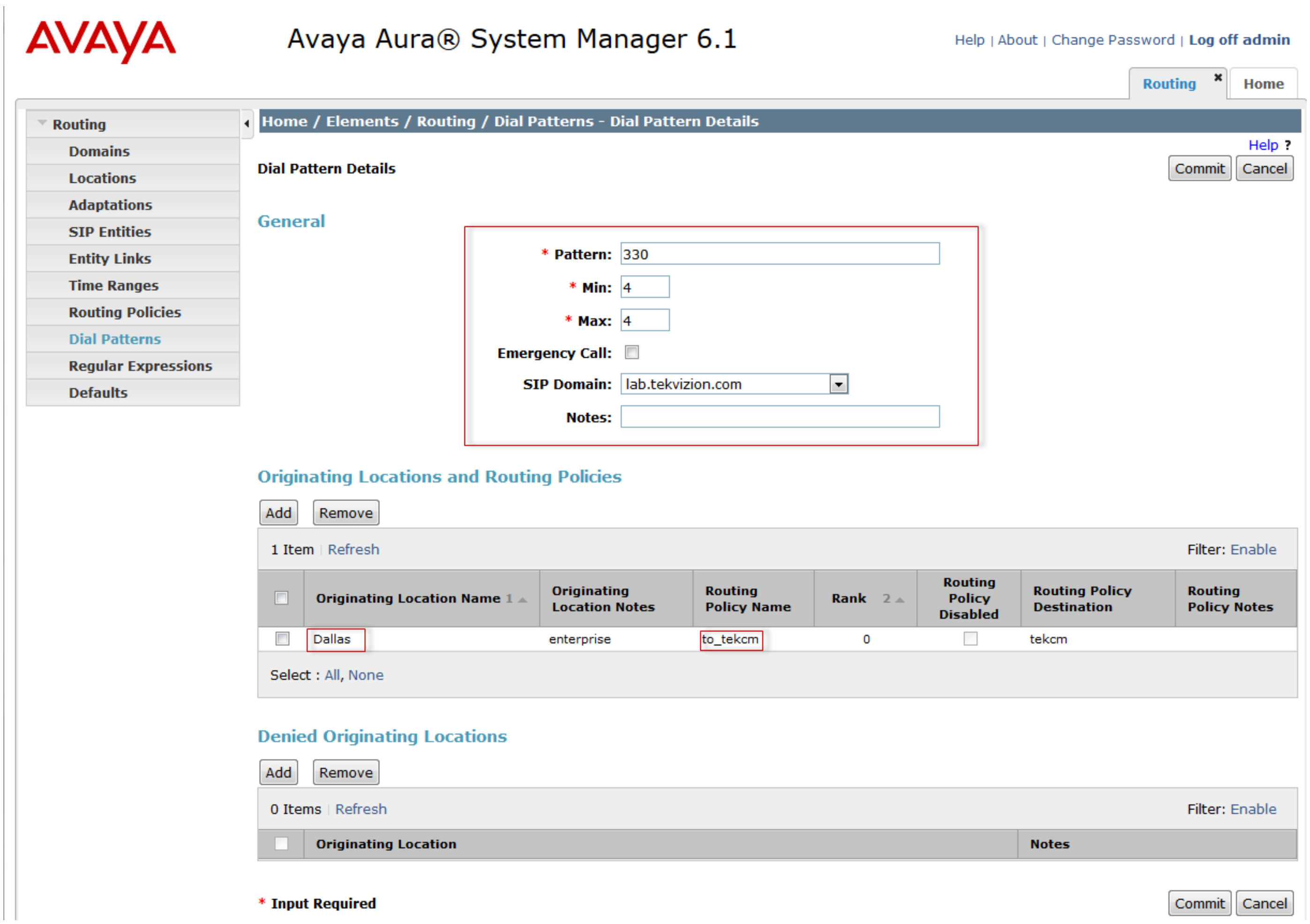
Min: 4

Max: 4

SIP Domain: lab.tekvizion.com

Original Location Name: Dallas

Routing Policy Name: to_tekcm



Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

Cisco Unified Communications Manager SIP Trunk Security Profile

Set Name*= Non Secure SIP Trunk Profile. This is used for this example.

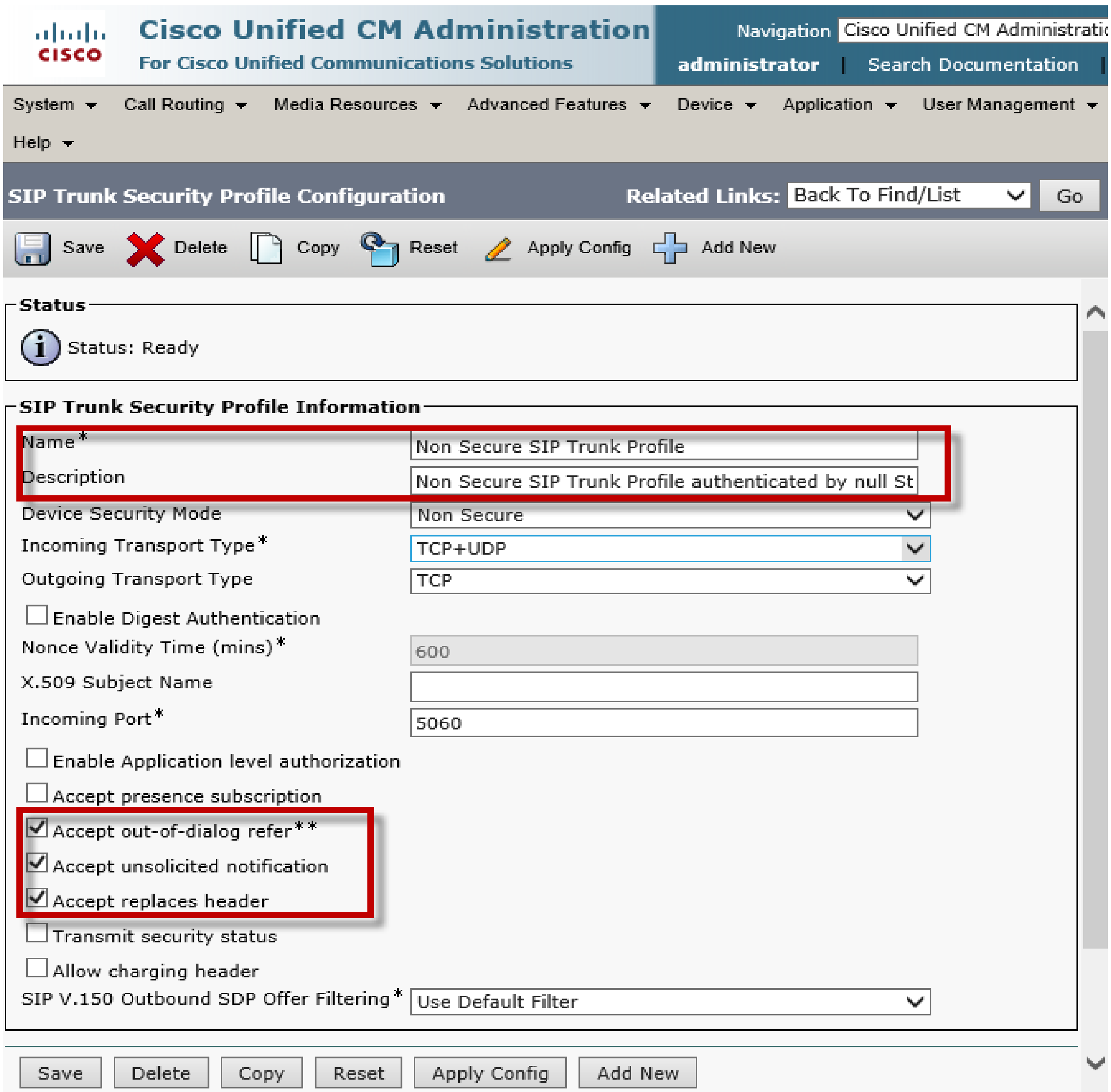
Set Description = This text is used to identify this SIP Trunk Security Profile.

Check Accept out of dialog refer

Check Accept unsolicited notification

Check Accept replaces header

All other values are default.



Cisco Unified Communications Manager SIP Trunk Security Profile for Unity Connection

Set Name*= Non Secure SIP Trunk to VM Profile. This is used for this example.

Set Description = This text is used to identify this SIP Trunk Security Profile.

Check Accept presence subscription

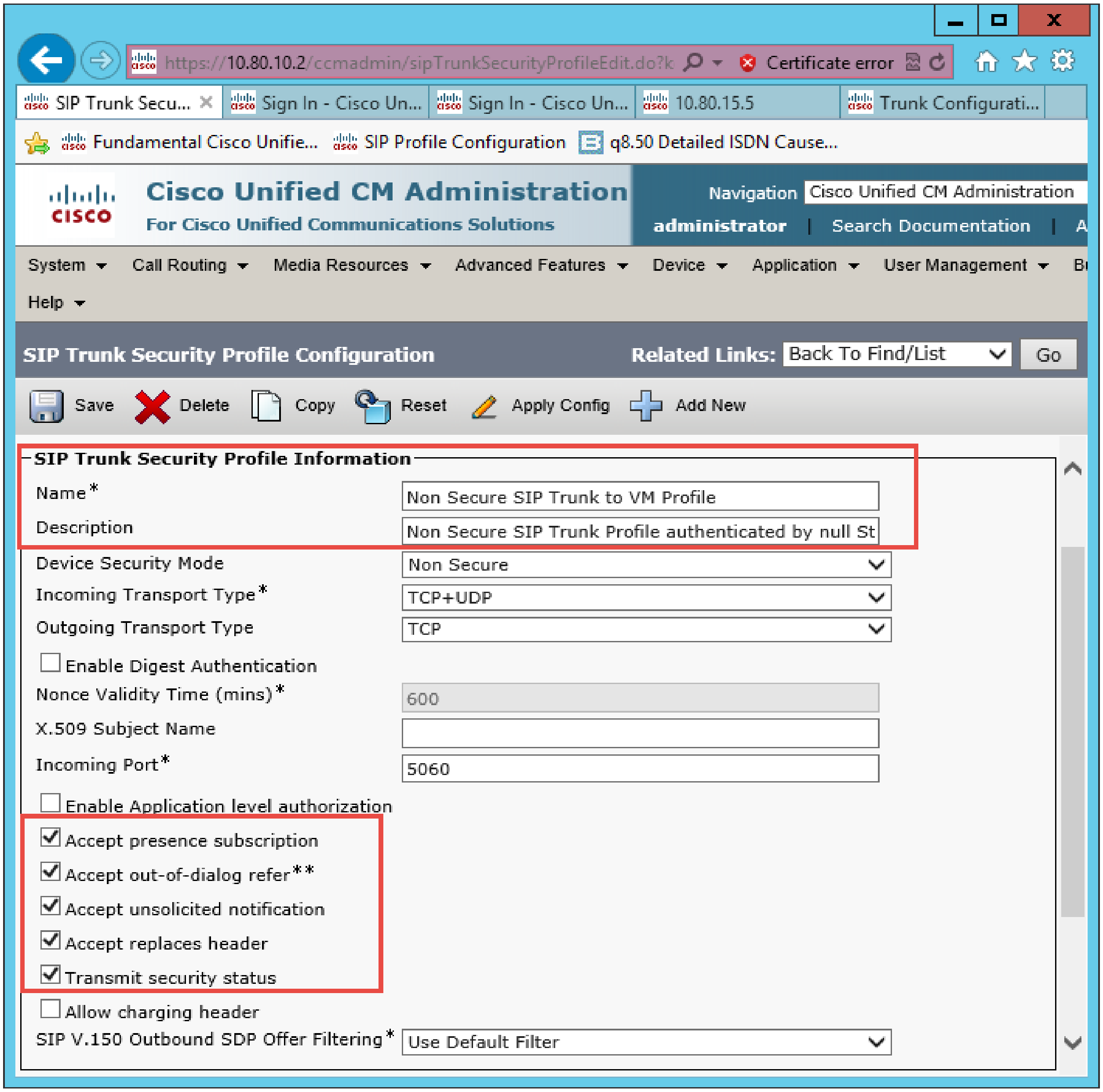
Check Accept out of dialog refer**

Check Accept unsolicited notification

Check Accept replaces header

Check Transmit security status

All other values are default.



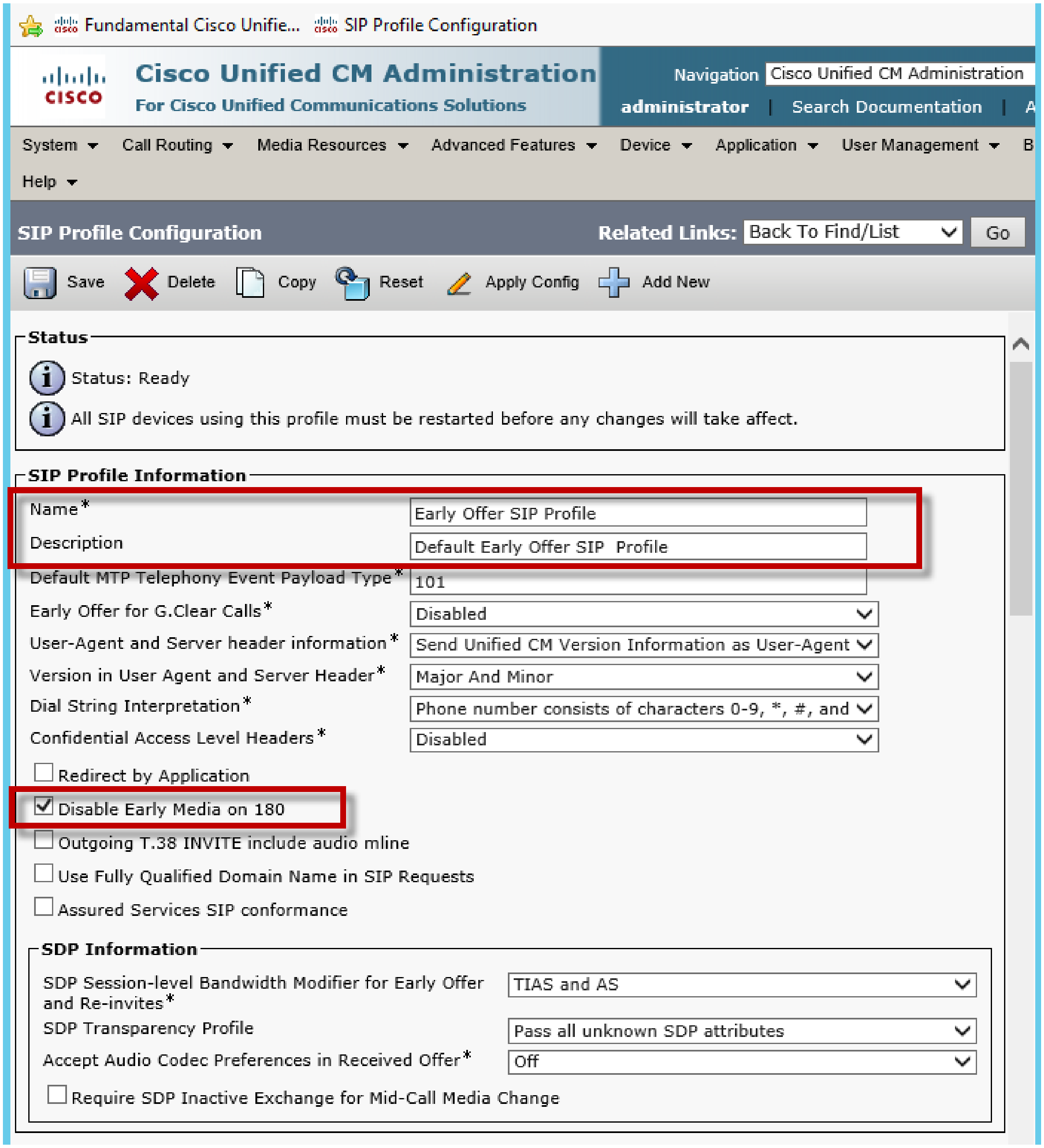
Cisco Unified Communications Manager SIP Profile

Set Name*= Early Offer SIP Profile. This is used for this example.

Set Description = This text is used to identify this SIP Profile.

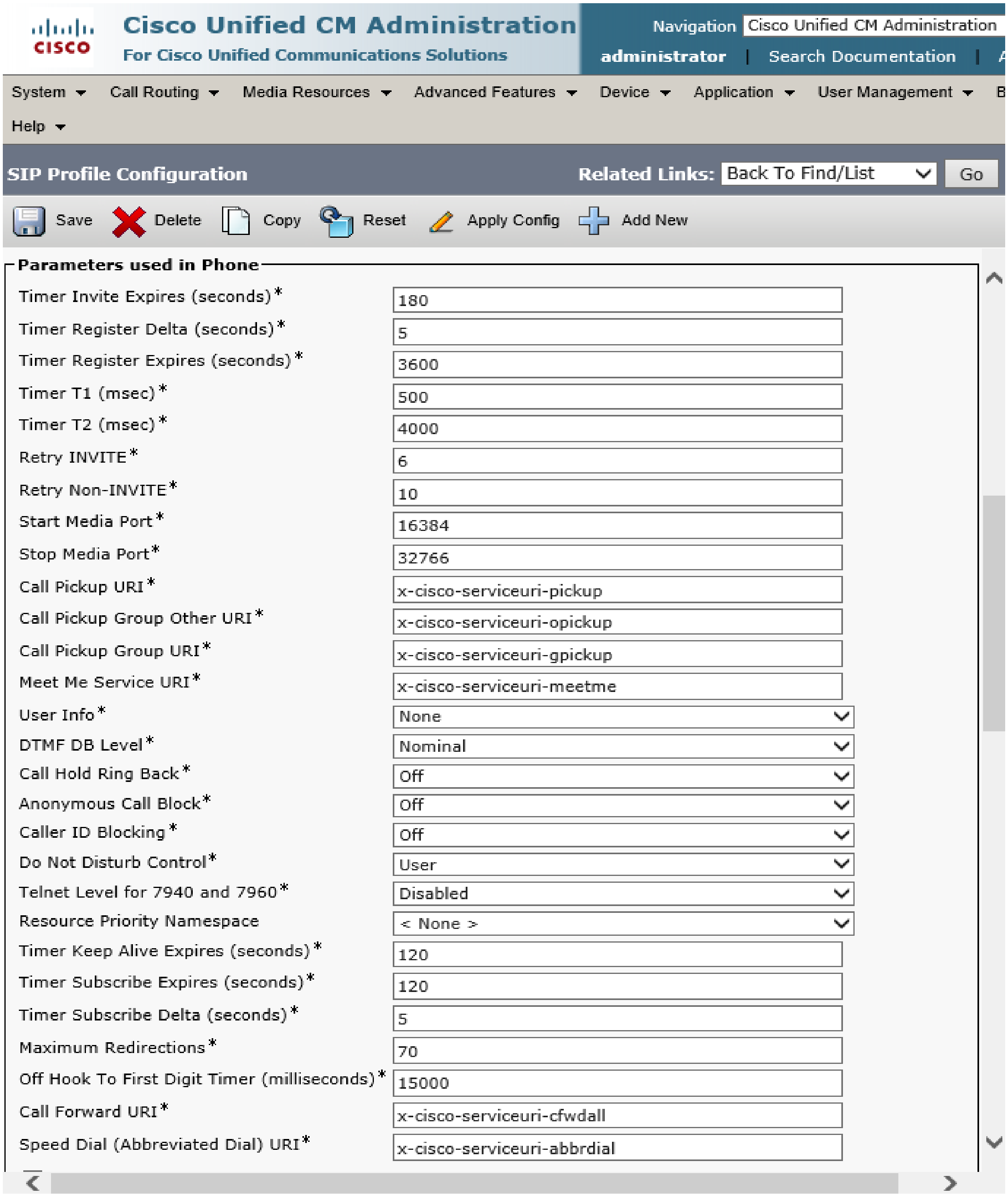
Check Disable Early Media on 180

All other values are default.


Fundamental Cisco Unifie... SIP Profile Configuration
Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
administrator | Search Documentation
System Call Routing Media Resources Advanced Features Device Application User Management
Help
SIP Profile Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
Status
Status: Ready
All SIP devices using this profile must be restarted before any changes will take affect.
SIP Profile Information
Name* Early Offer SIP Profile
Description Default Early Offer SIP Profile
Default MTP Telephony Event Payload Type* 101
Early Offer for G.Clear Calls* Disabled
User-Agent and Server header information* Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header* Major And Minor
Dial String Interpretation* Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers* Disabled
Redirect by Application
Disable Early Media on 180
Outgoing T.38 INVITE include audio mline
Use Fully Qualified Domain Name in SIP Requests
Assured Services SIP conformance
SDP Information
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* TIAS and AS
SDP Transparency Profile Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer* Off
Require SDP Inactive Exchange for Mid-Call Media Change

Cisco Unified Communications Manager SIP Profile (Continued)

These values are default.



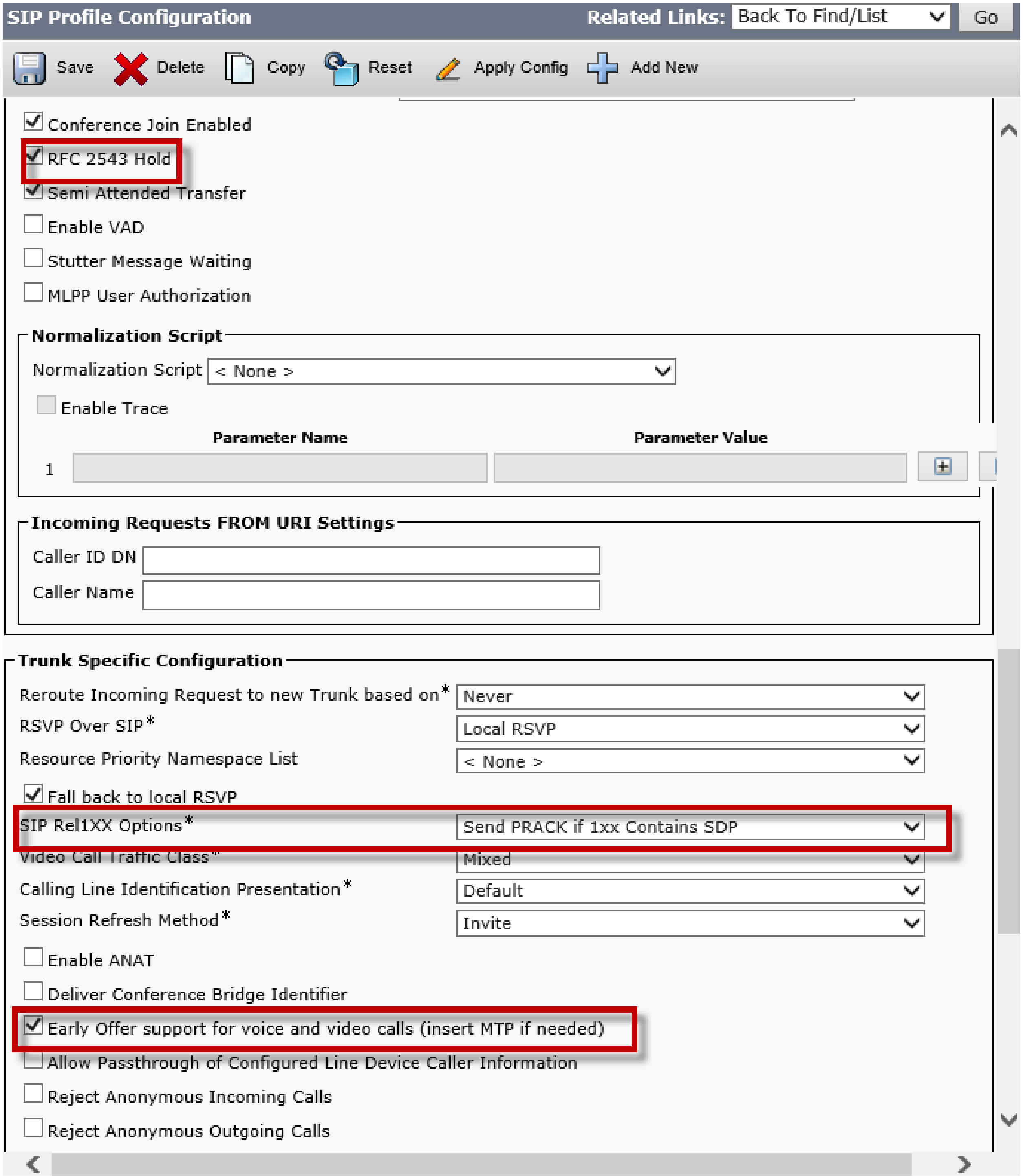
Cisco Unified Communications Manager SIP Profile (Continued)

Check RFC 2543 Hold

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP

Check Early Offer support for voice and video calls (insert MTP if needed)

All other values are default.

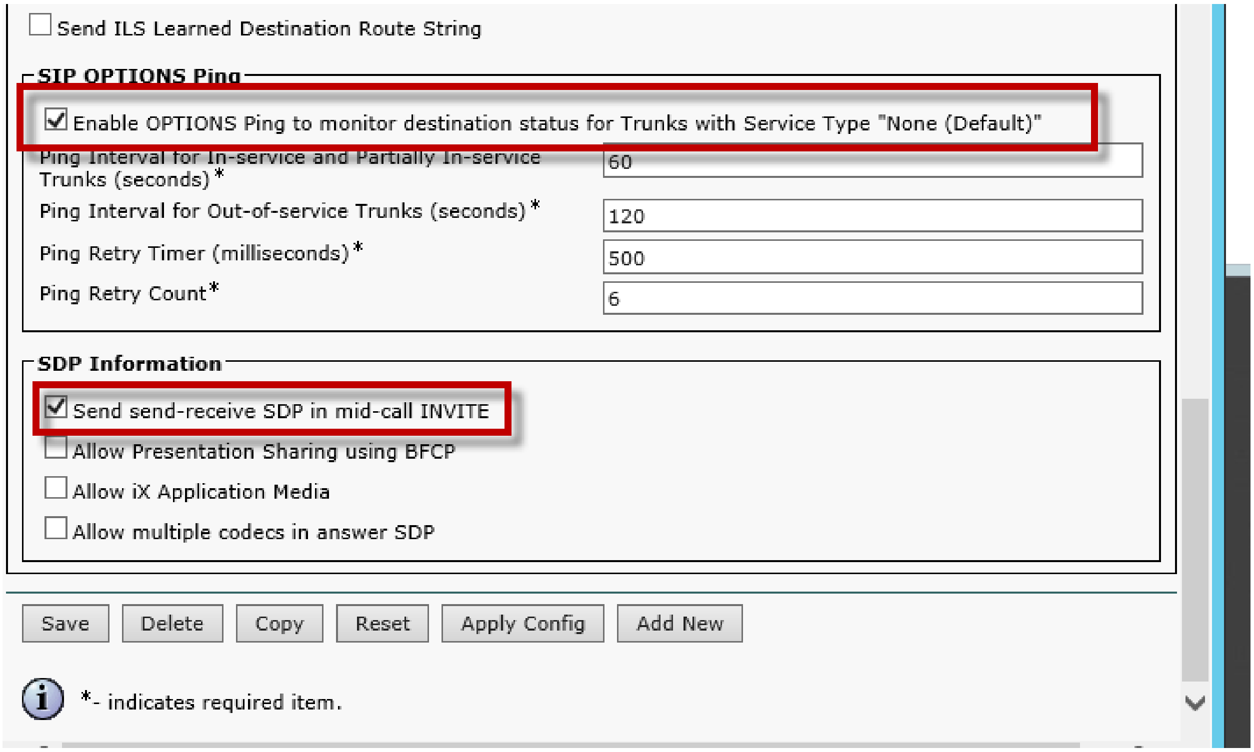


Cisco Unified Communications Manager SIP Profile (Continued)

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

Check Send send-receive SDP in mid-call INVITE

All other values are default.



Cisco Unified Communications Manager SIP Trunk to Avaya Configuration

Set Device Name*= Trunk_to_Avaya_SM. This is used for this example.

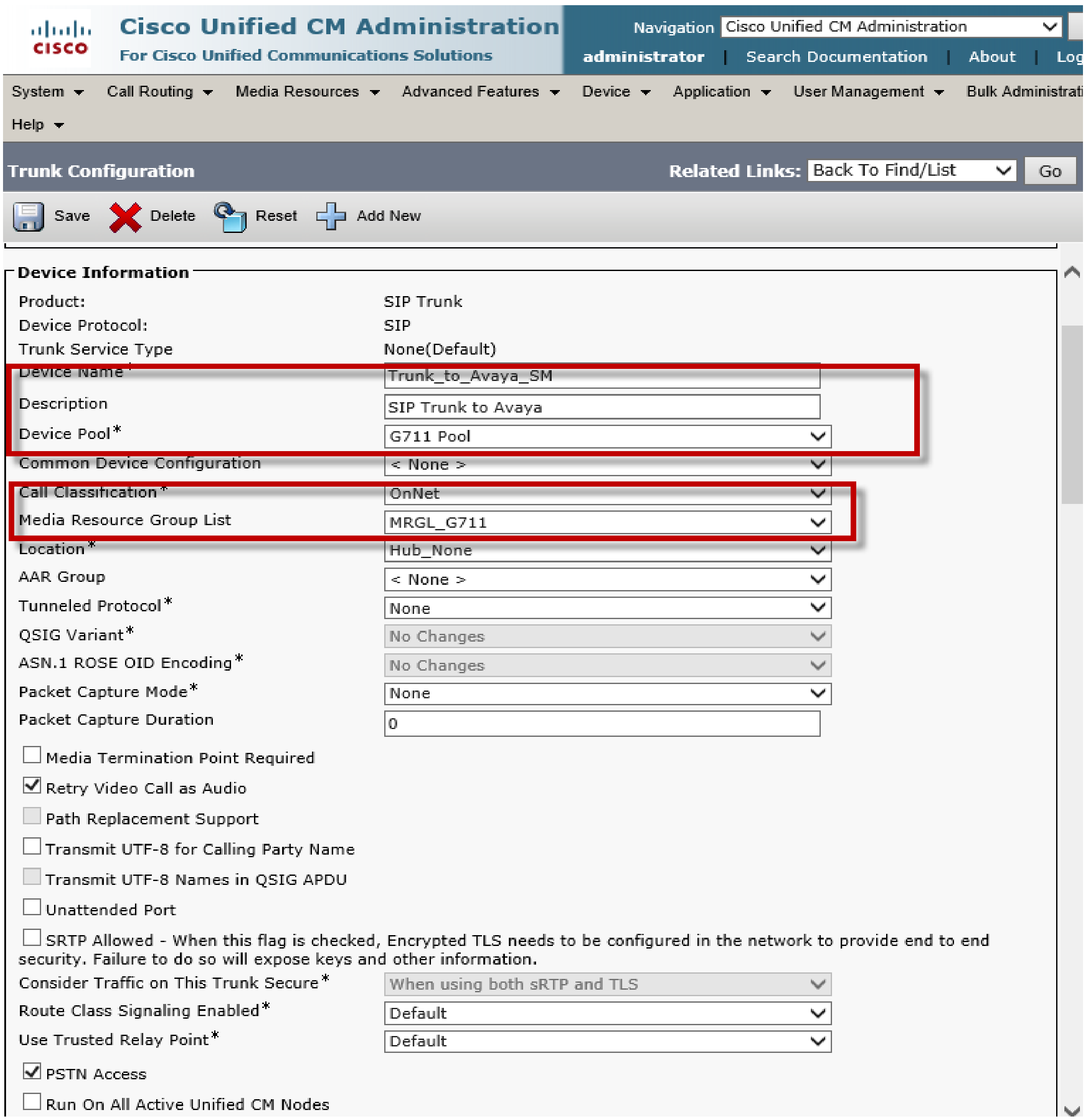
Set Description = This text is used to identify this Trunk Group.

Set Device Pool* = G711 Pool This is used for this example

Set Call Classification*= OnNet. This is used for this example

Set Media Resource Group List = MRGL_G711. This is used for this example

All other values are default.



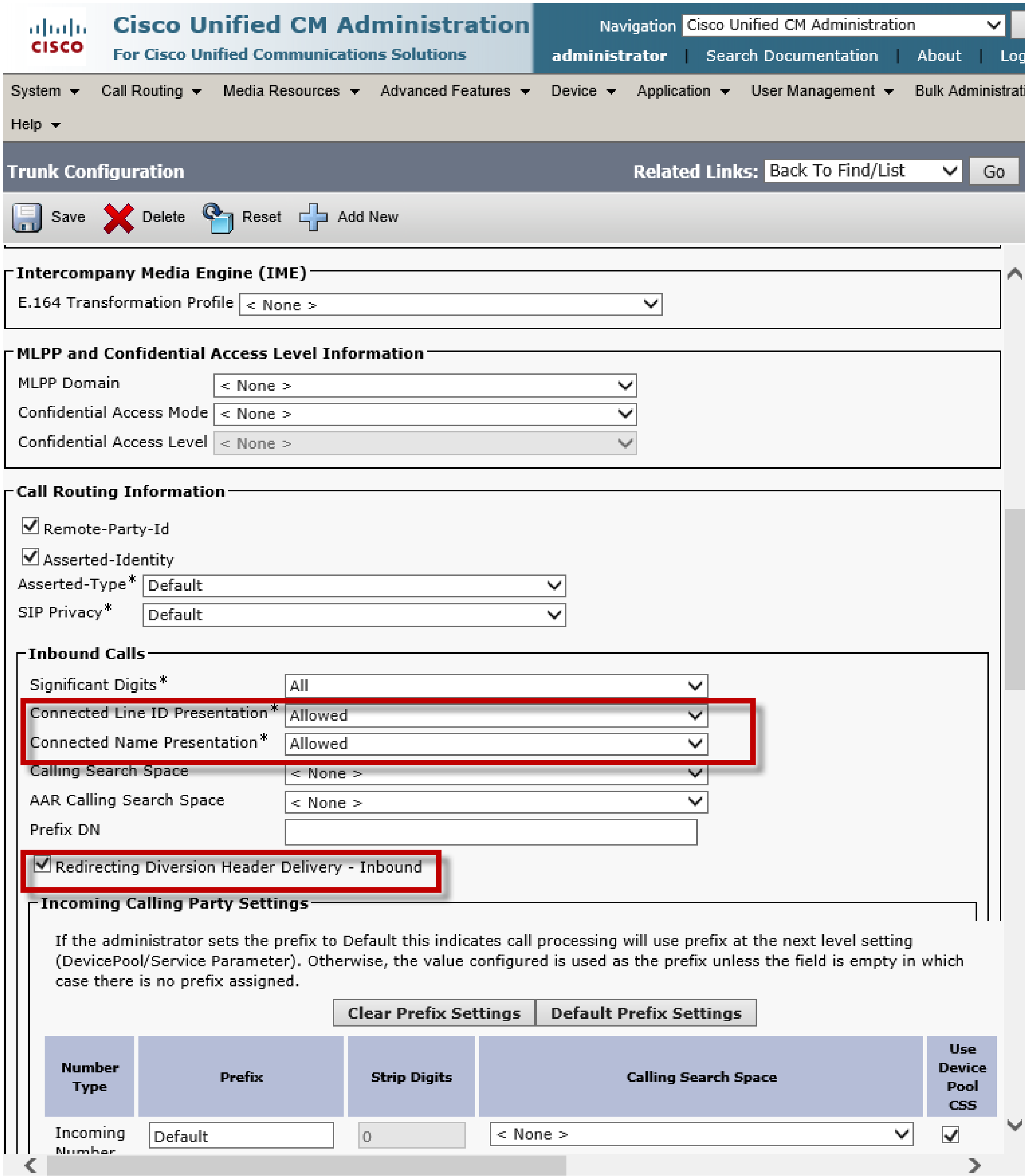
Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Connected Line ID Presentation*= Allowed

Set Connected Name Presentation* = Allowed

Check Redirecting Diversion Header Delivery - Inbound

All other values are default.



Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

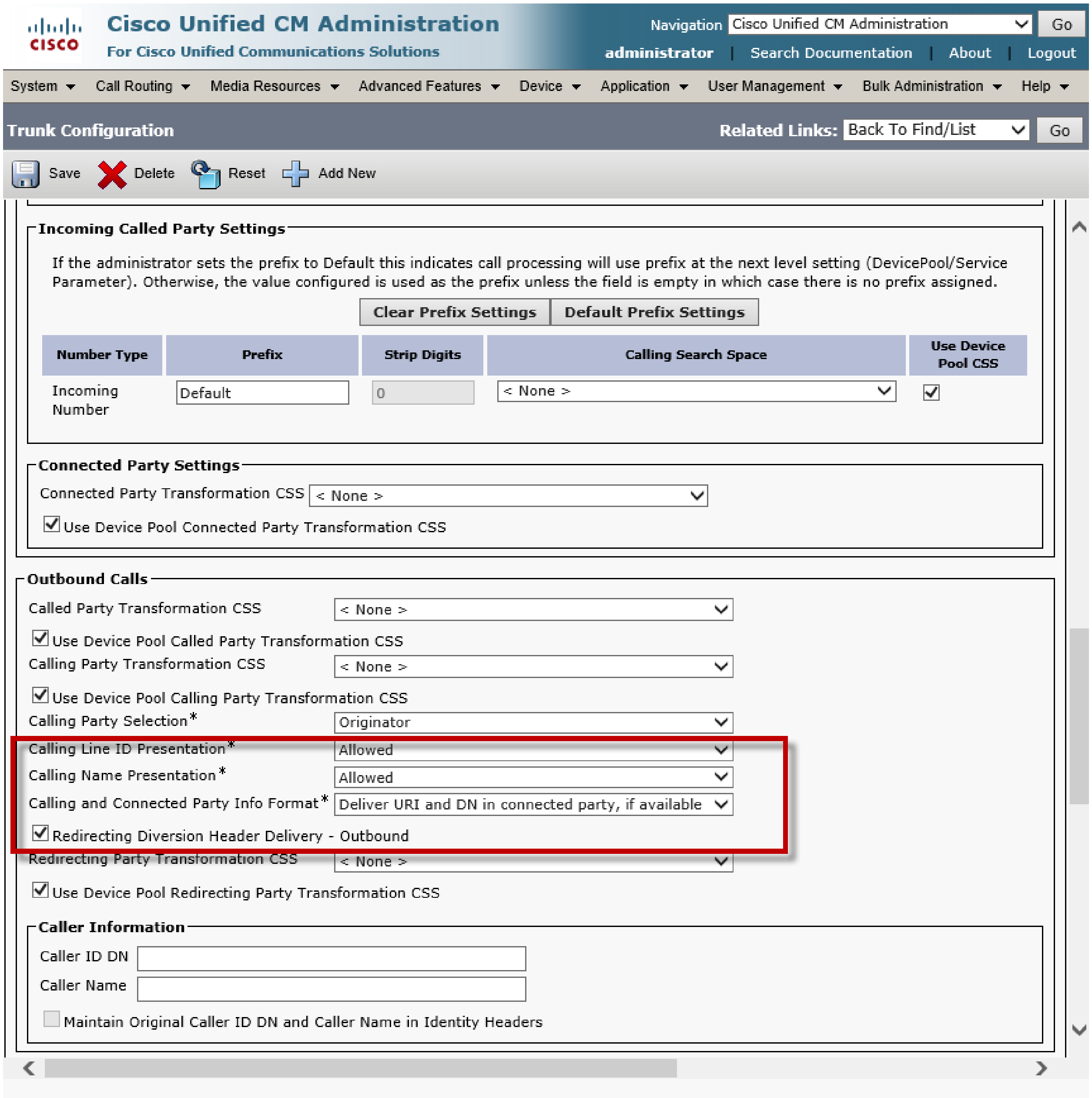
Set Calling Line ID Presentation*= Allowed

Set Calling Name Presentation*= Allowed

Set Calling and Connected Party Info Format* = Deliver URI and DN in connected party, if available

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.



Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Destination Address = 10.70.2.6. This is used in this example.

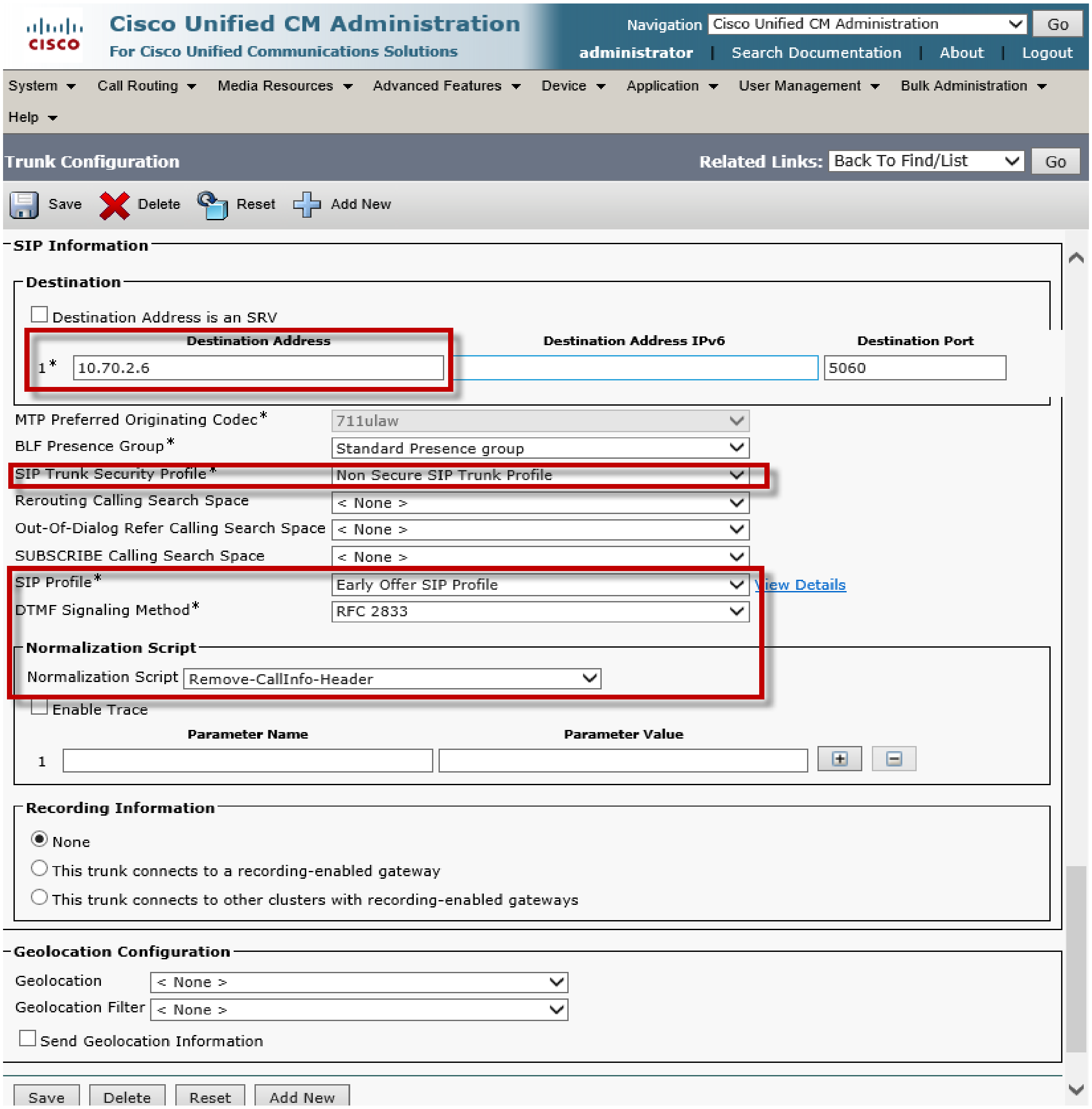
Set SIP Trunk Security Profile*= Non Secure SIP Trunk Profile

Set SIP Profile*= EarlyOffer SIP Profile

Set DTMF Signaling Method*= RFC 2833

Set Normalization Script = Remove-Call-Info-Header. This example script name was used to remove Call-Info Header to Avaya

All other values are default.



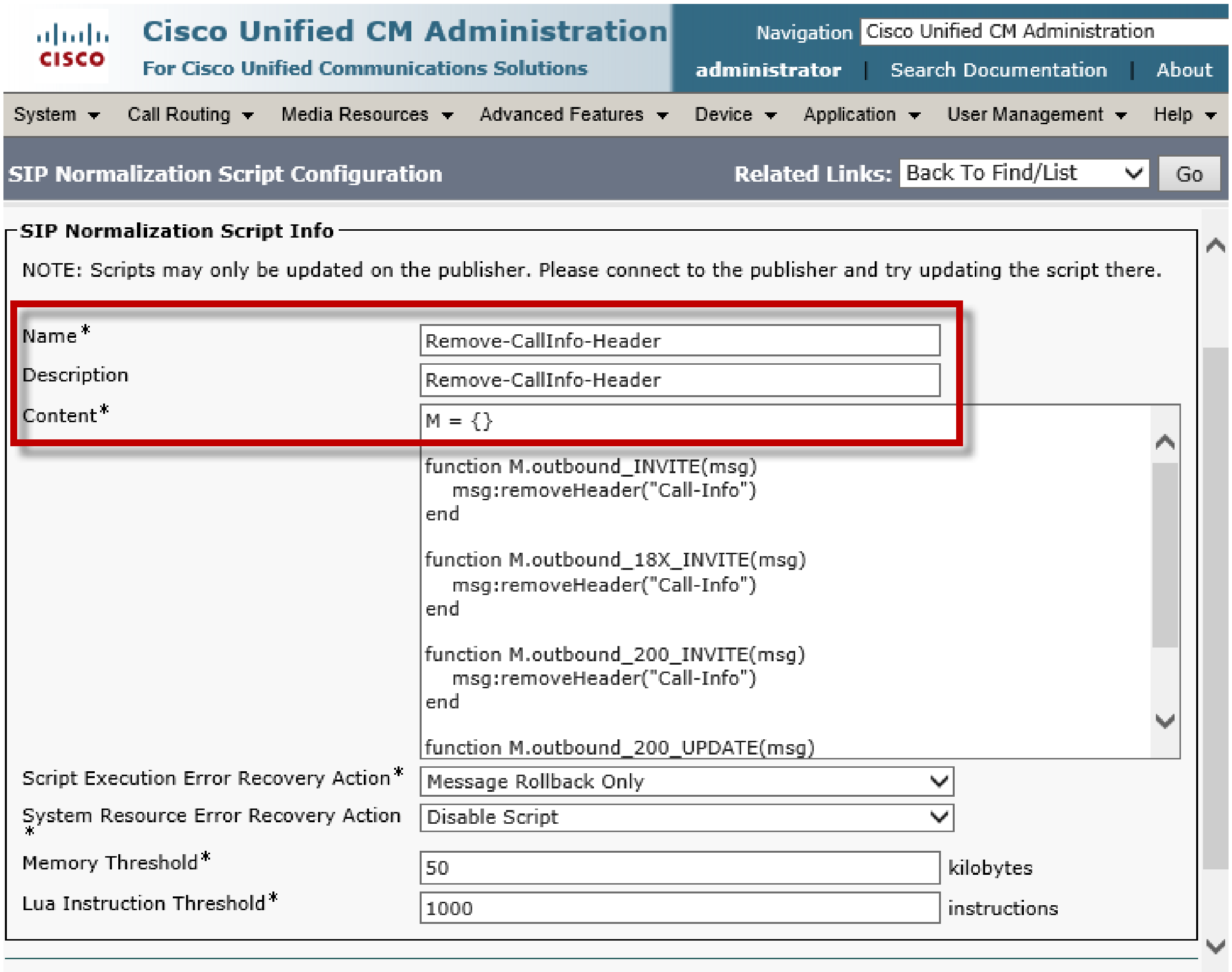
Cisco Unified Communications Manager SIP Trunk Normalization Script

Set Name*= Remove-CallInfo-Header. This is used for this example.

Set Description = This text is used to identify this SIP Normalization Script.

Set Content*= Please see full contents on next page.

All Other values are default



Note: SIP Normalization script was used to remove the Call-Info Header from Cisco to Avaya.

The full content of the SIP Normalization Script is captured below:

```
M = {}

function M.outbound_INVITE(msg)

   msg:removeHeader("Call-Info")

end

function M.outbound_18X_INVITE(msg)

   msg:removeHeader("Call-Info")

end

function M.outbound_200_INVITE(msg)

   msg:removeHeader("Call-Info")

end

function M.outbound_200_UPDATE(msg)

   msg:removeHeader("Call-Info")

end

return M
```

Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration

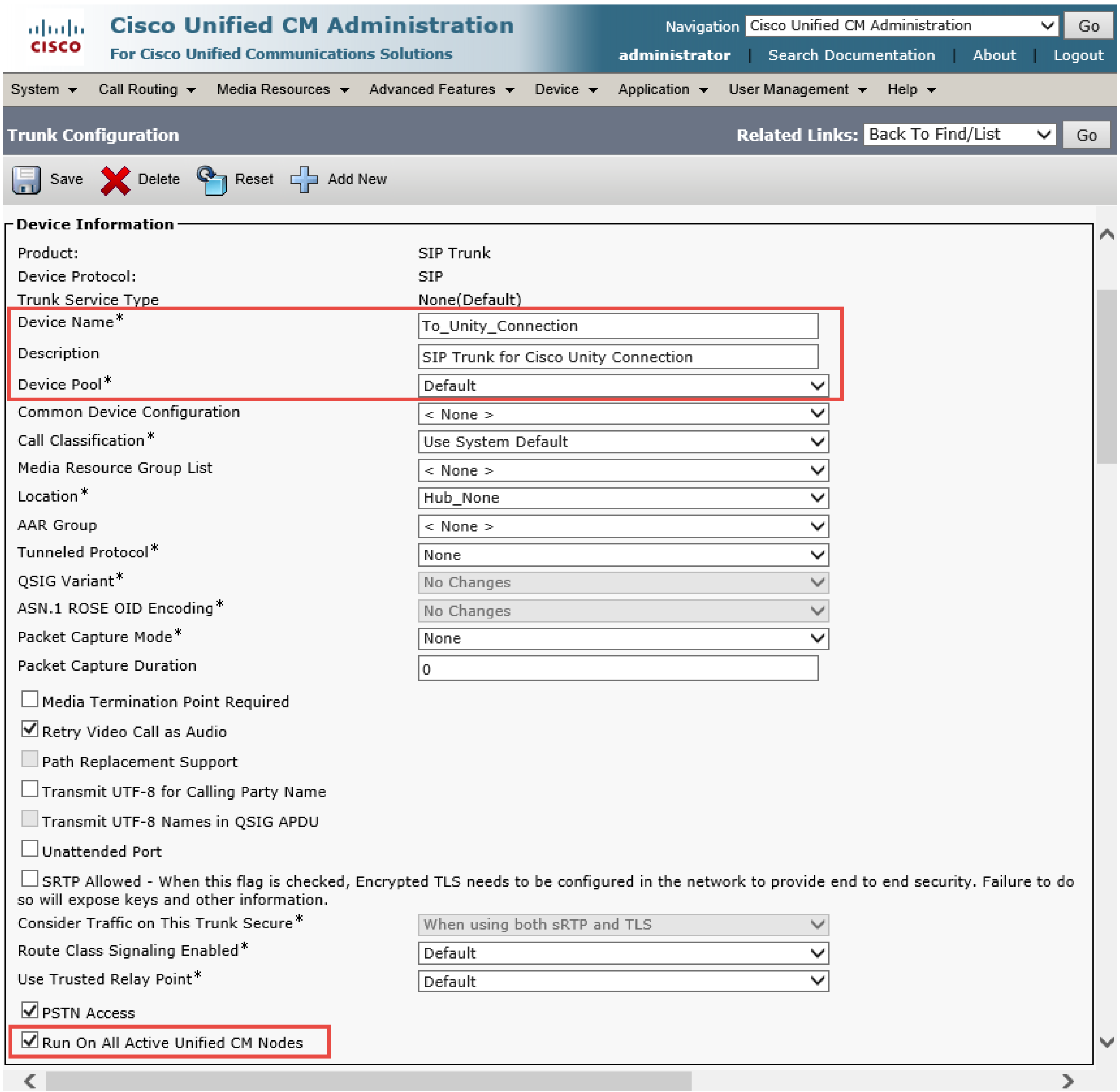
Set Device Name*= To_Unity_Connection. This is used for this example.

Set Description = This text is used to identify this Trunk Group.

Set Device Pool* = Default This is used for this example

Check Run On All Active Unified CM Nodes

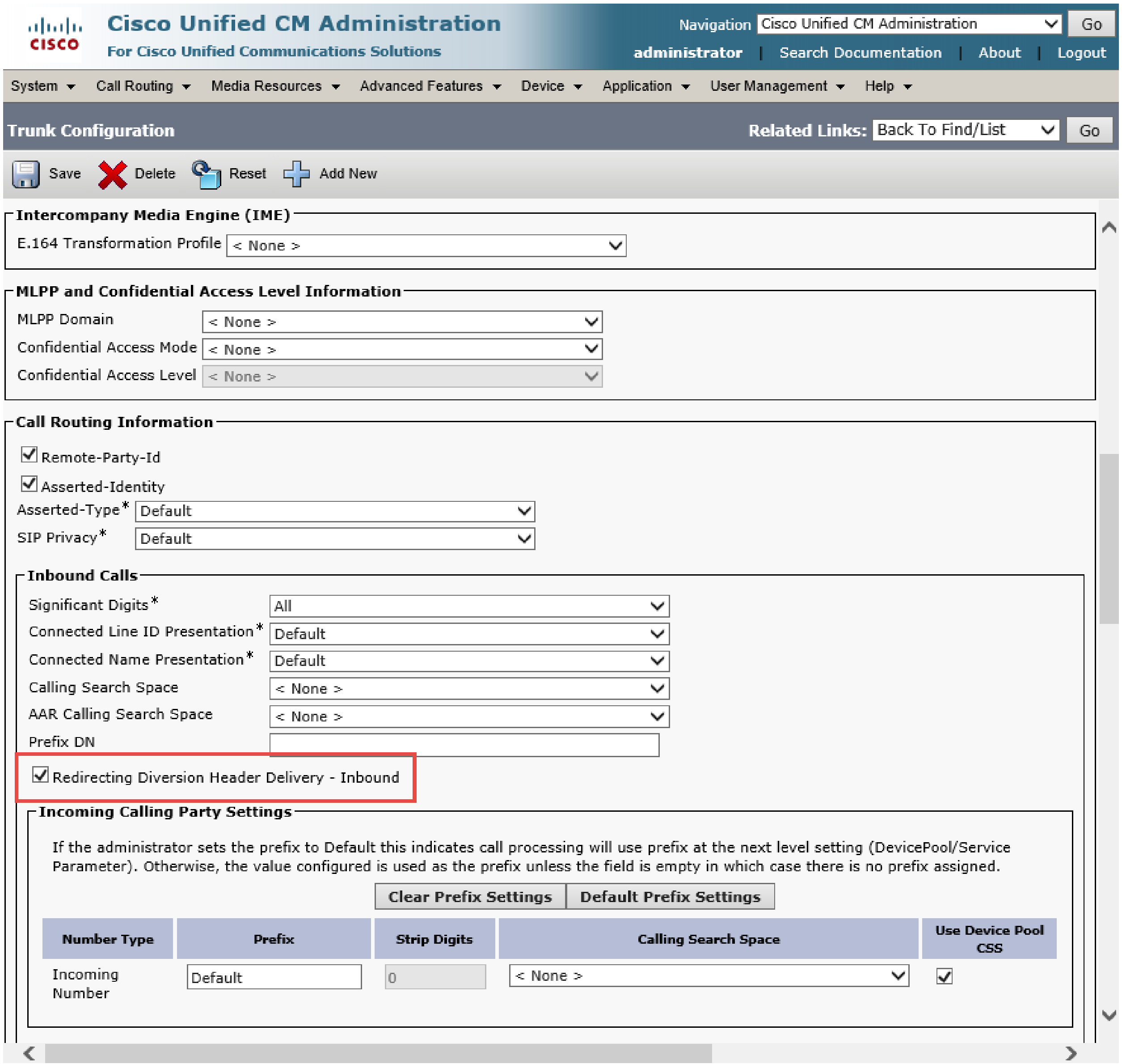
All other values are default.



Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration (Continued)

Check Redirecting Diversion Header Delivery - Inbound

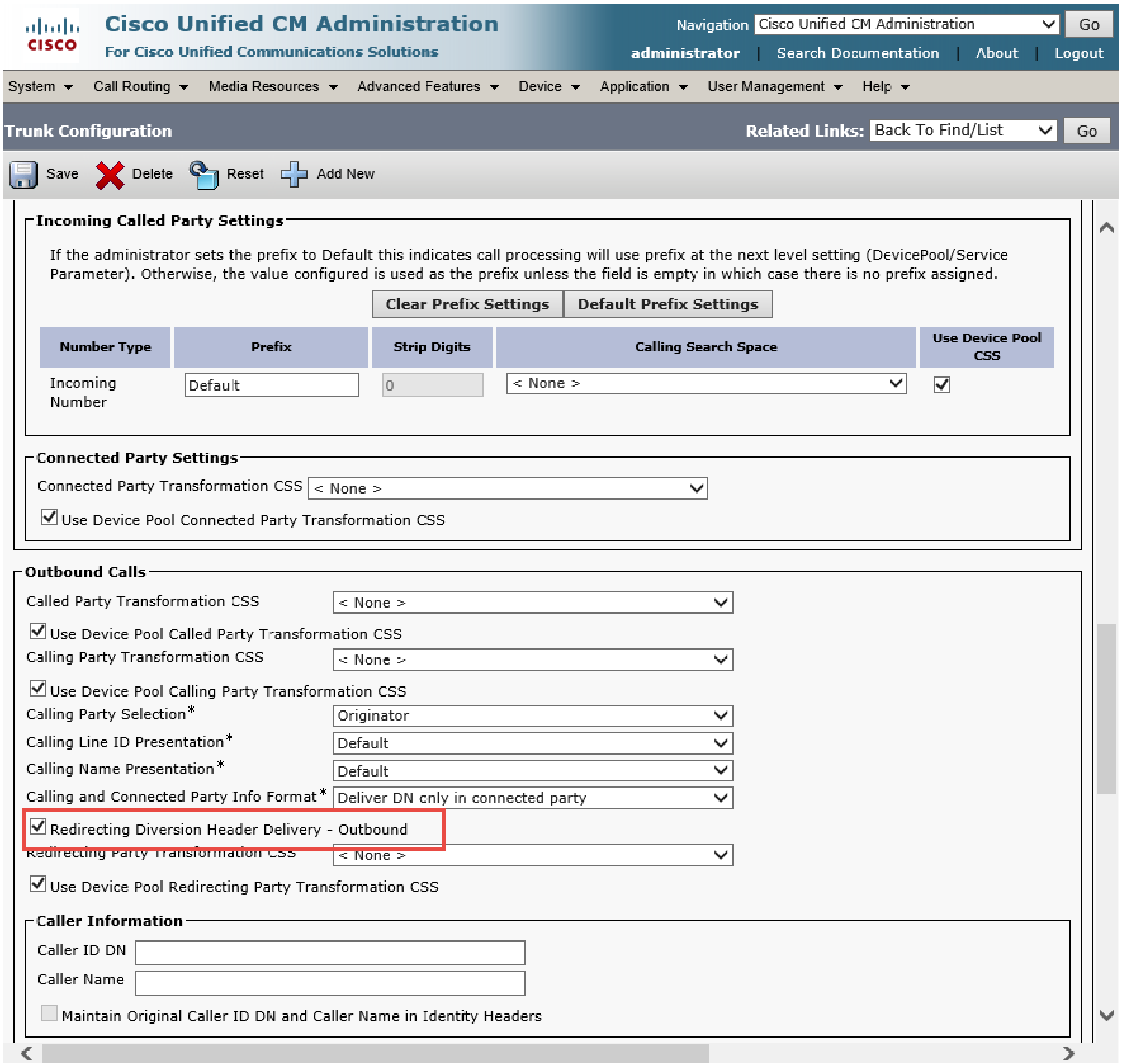
All other values are default.



Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration (Continued)

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.



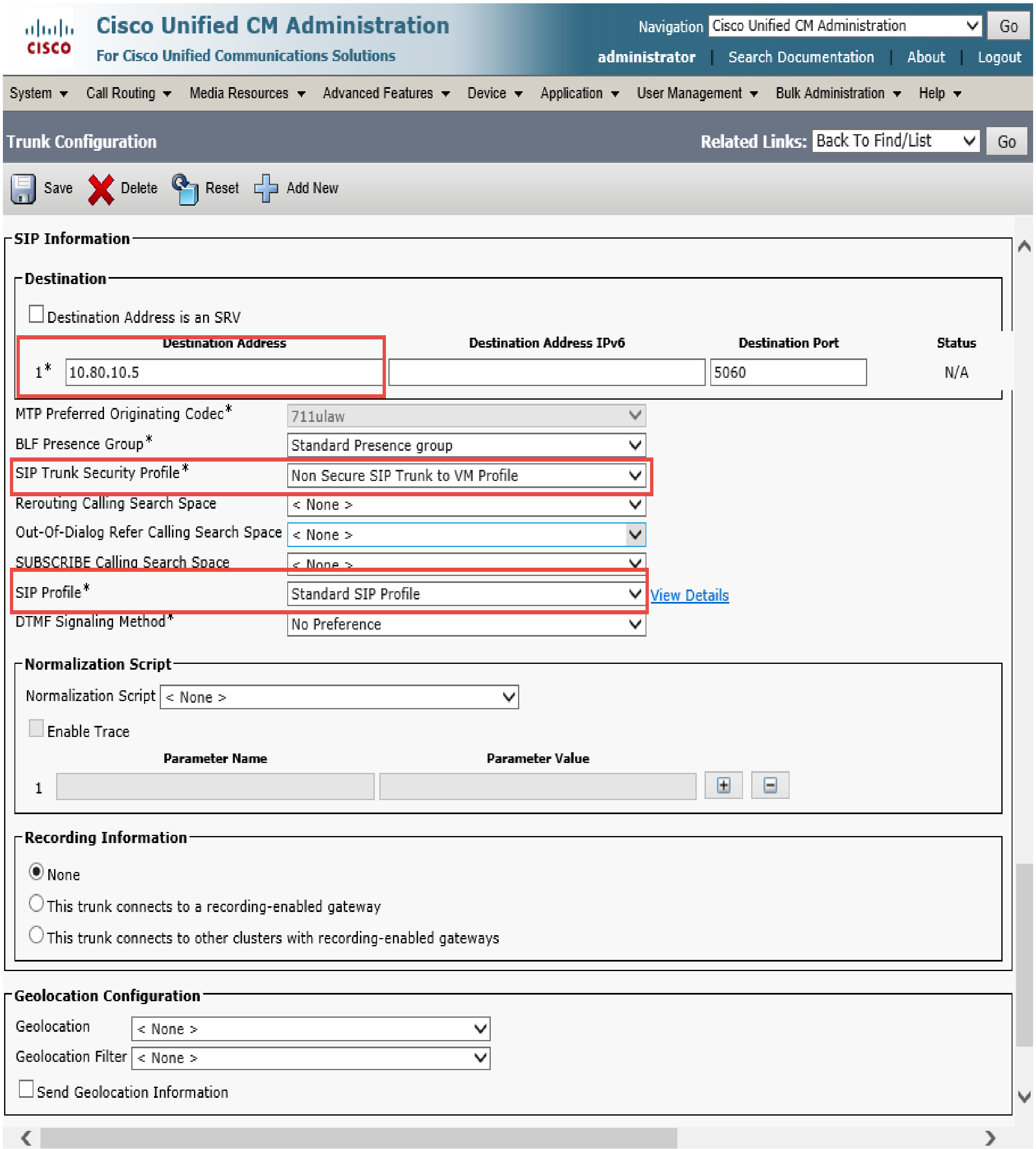
Cisco Unified Communications Manager SIP Trunk to Cisco Unity Configuration (Continued)

Set Destination Address = 10.80.10.5. This is used in this example.

Set SIP Trunk Security Profile*= Non Secure SIP Trunk to VM Profile

Set SIP Profile*= Standard SIP Profile

All other values are default.



Cisco Unity Connection User 5017 Configuration

Set Alias*= 5017. This is used for this example.

Set First Name = This text is used to identify this User.

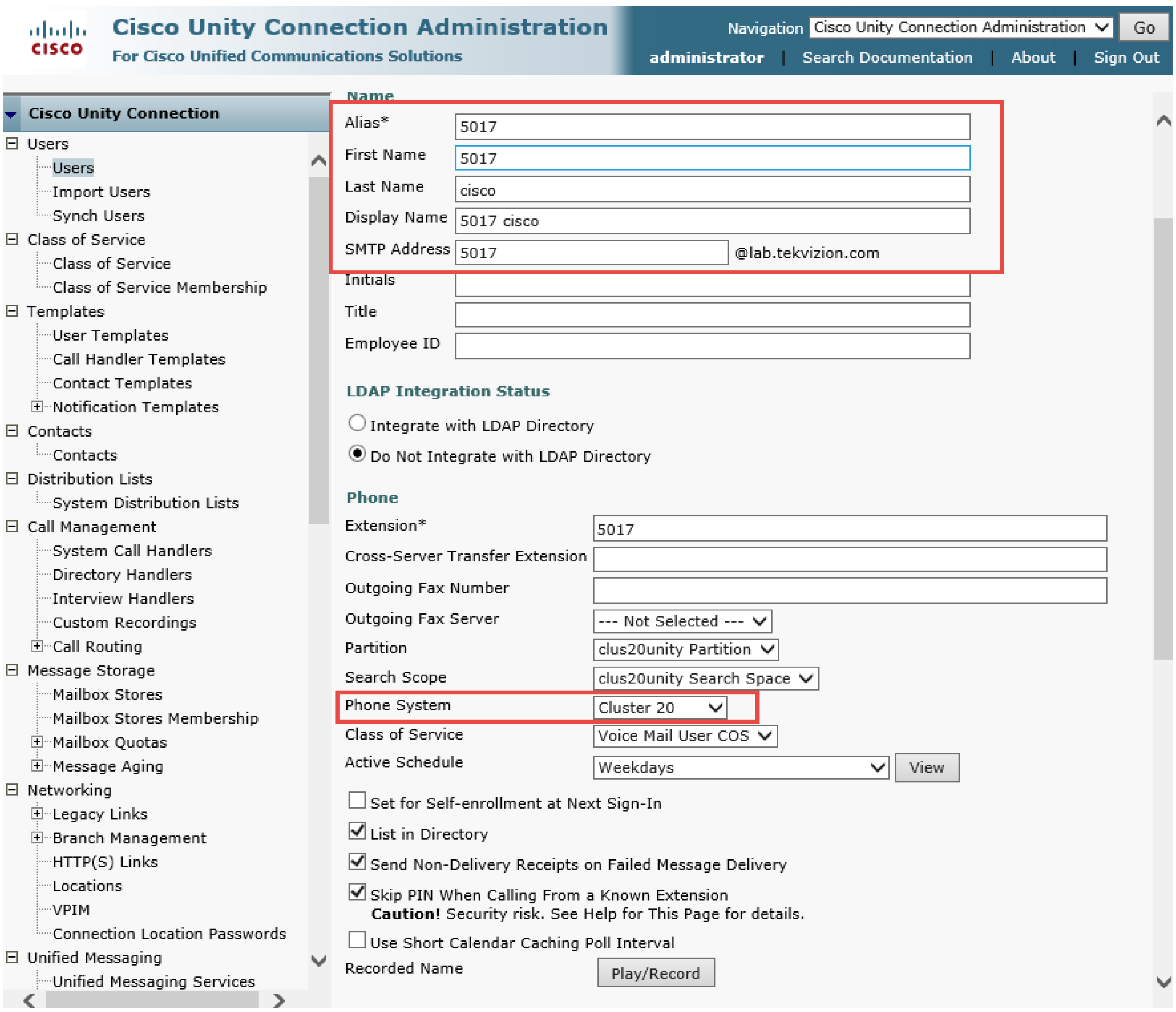
Set Last Name* = cisco This is used for this example

Set Display Name= 5017 cisco. This is used in this example.

Set SMTP Address =5017. This is used in this example.

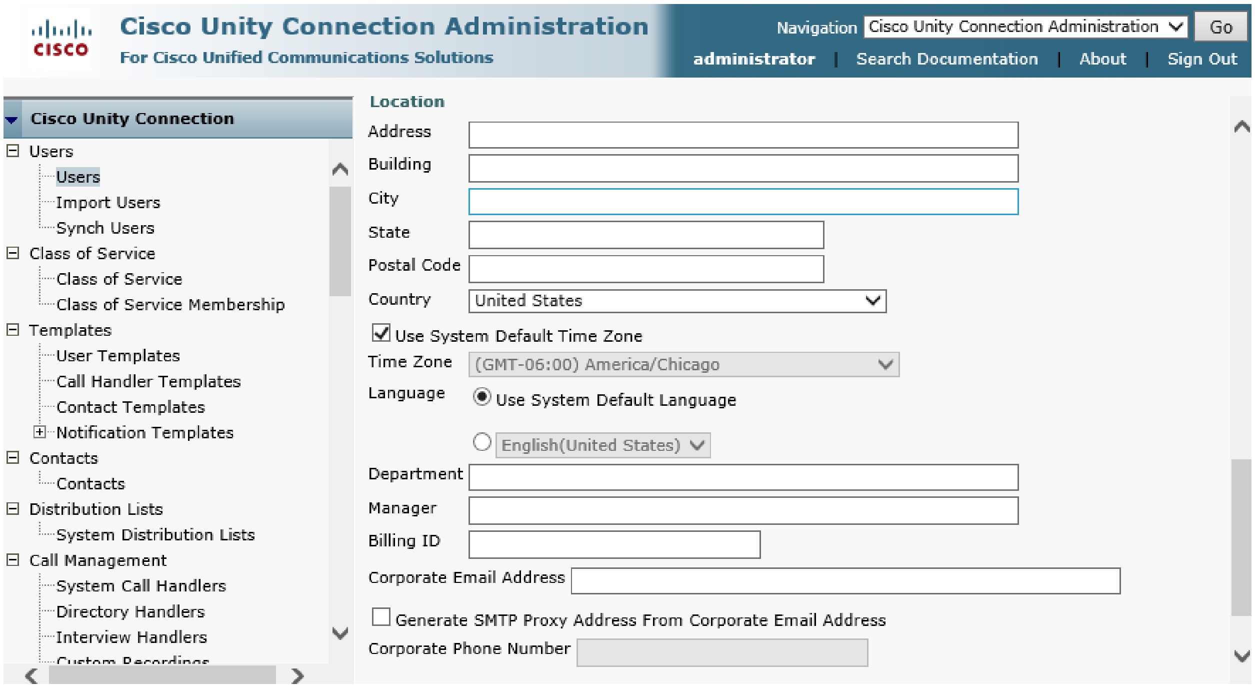
Set Phone System= Cluster 20. This is used in this example.

All other values are default.



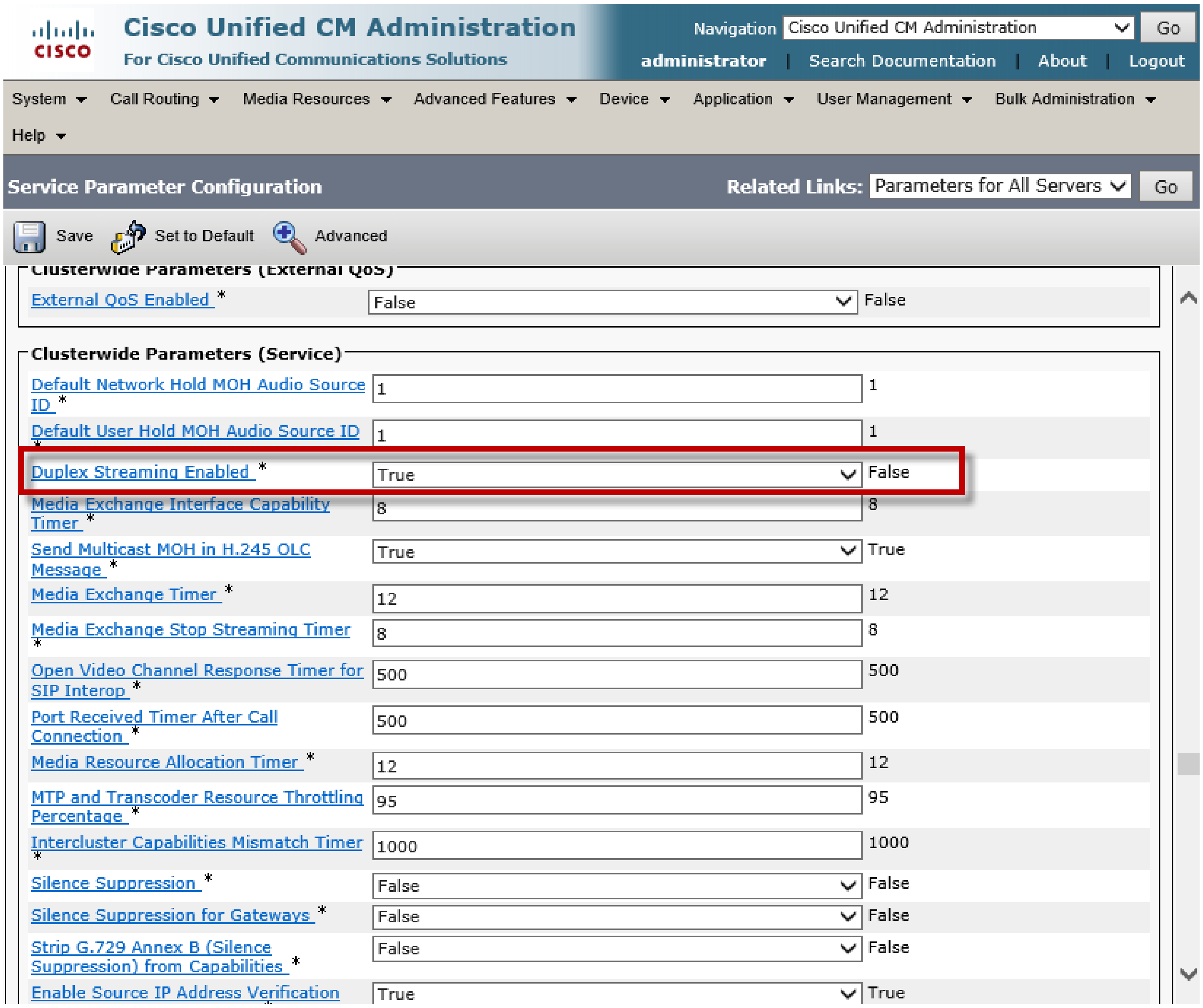
Cisco Unity Connection User 5017 Configuration (Continued)

All values are default.



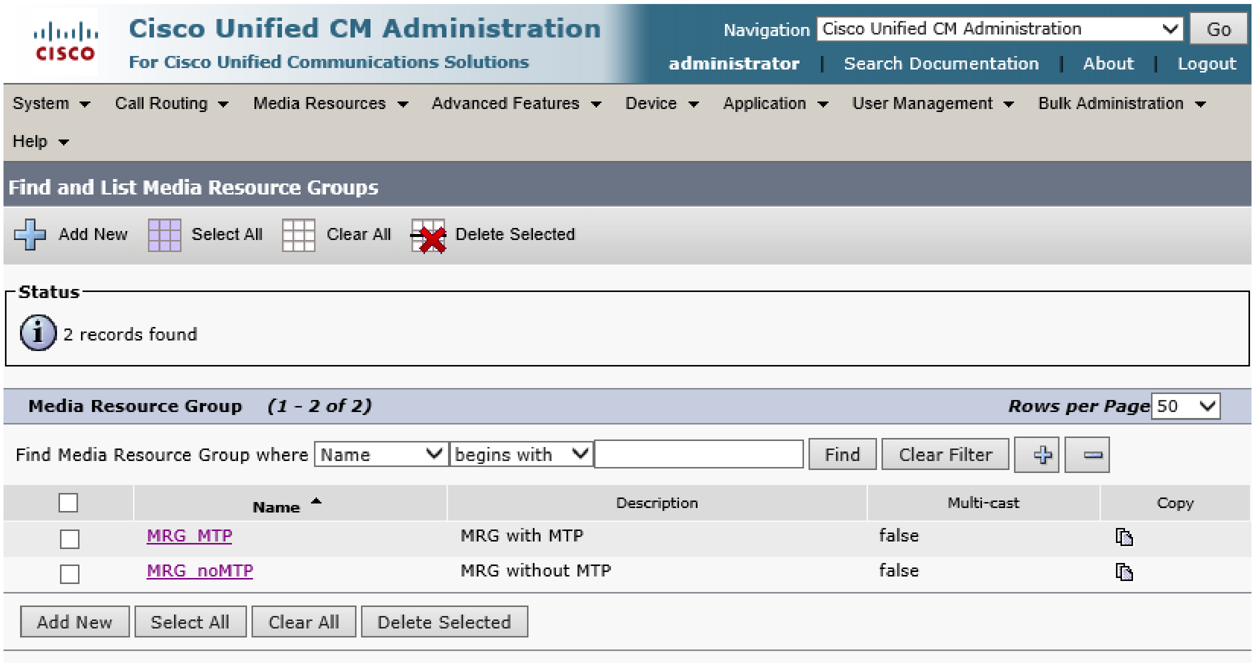
Cisco Unified Communications Manager Service Parameter

Set Duplex Streaming Enabled* = True. See Note under capture for more info.



Note: Cisco Unified Communications Manager Service Parameter "Duplex Streaming Enabled" should be set to "True" in order for MoH and ringback to work properly during call transfers/conferences initiated by Cisco stations to Avaya IP endpoints.

Cisco Unified Communications Manager Media Resource Group



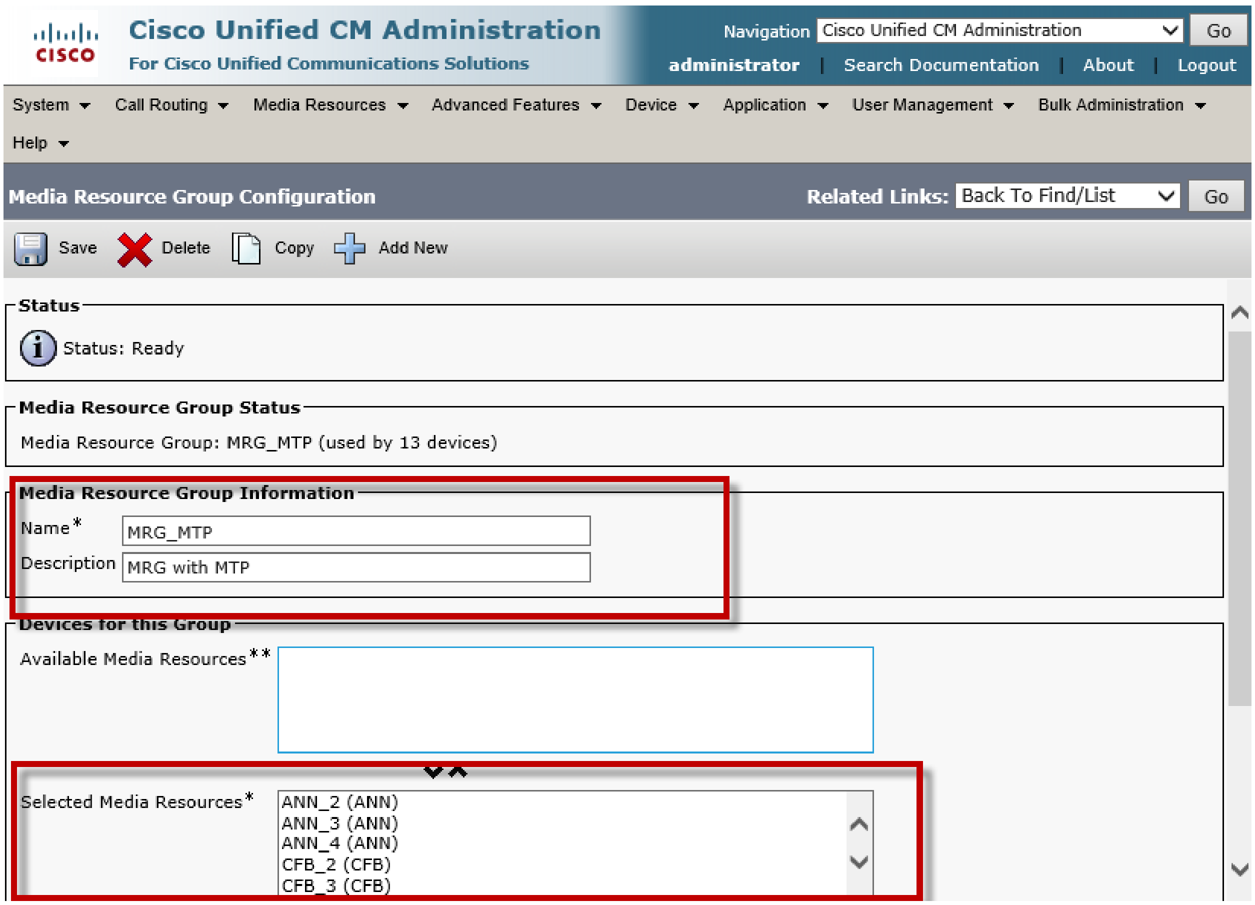
Media Resource Group MRG_MTP

Set Name*= MRG_MTP This is used for this example.

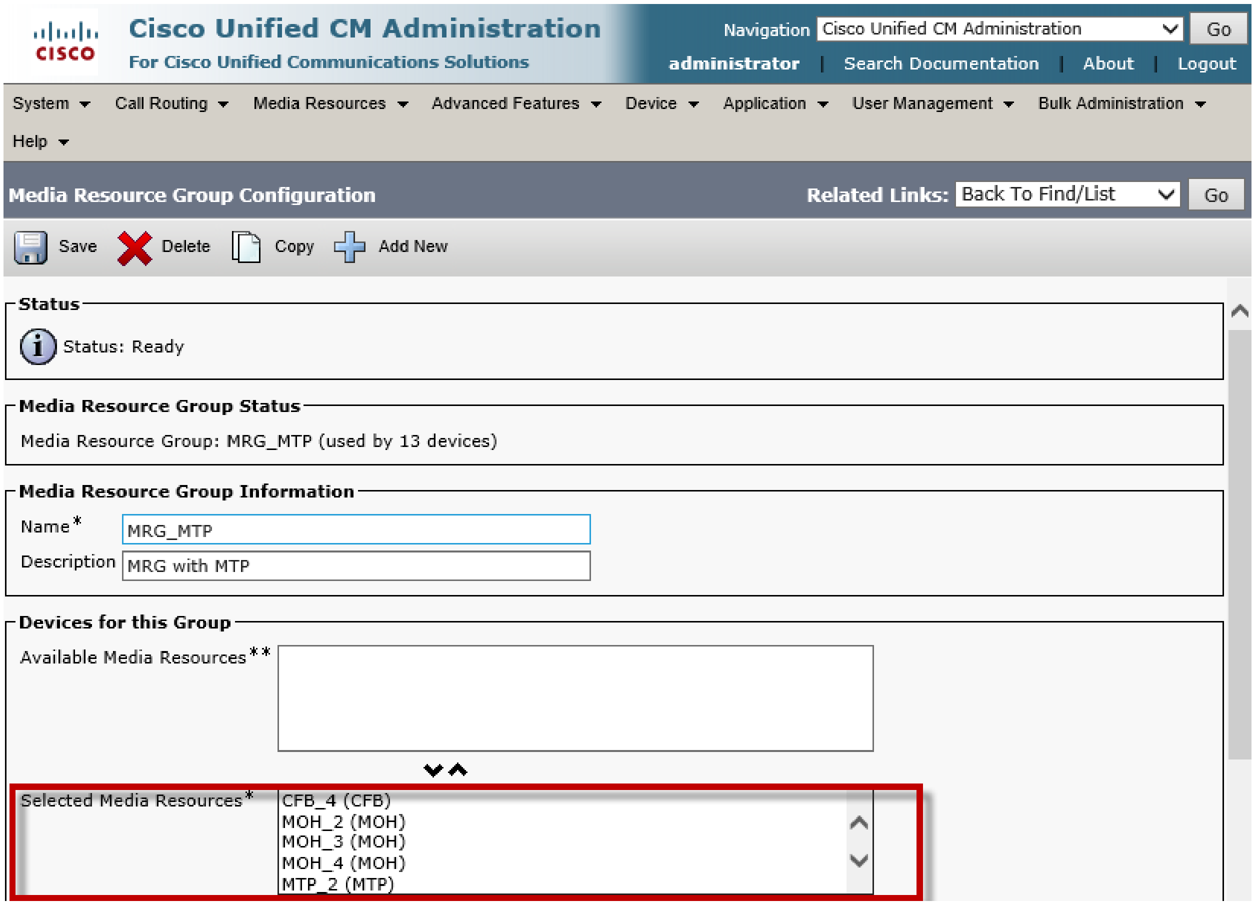
Set Description = This text is used to identify this Media Resource Group.

Set all resources in the Selected Media Resources* Box.

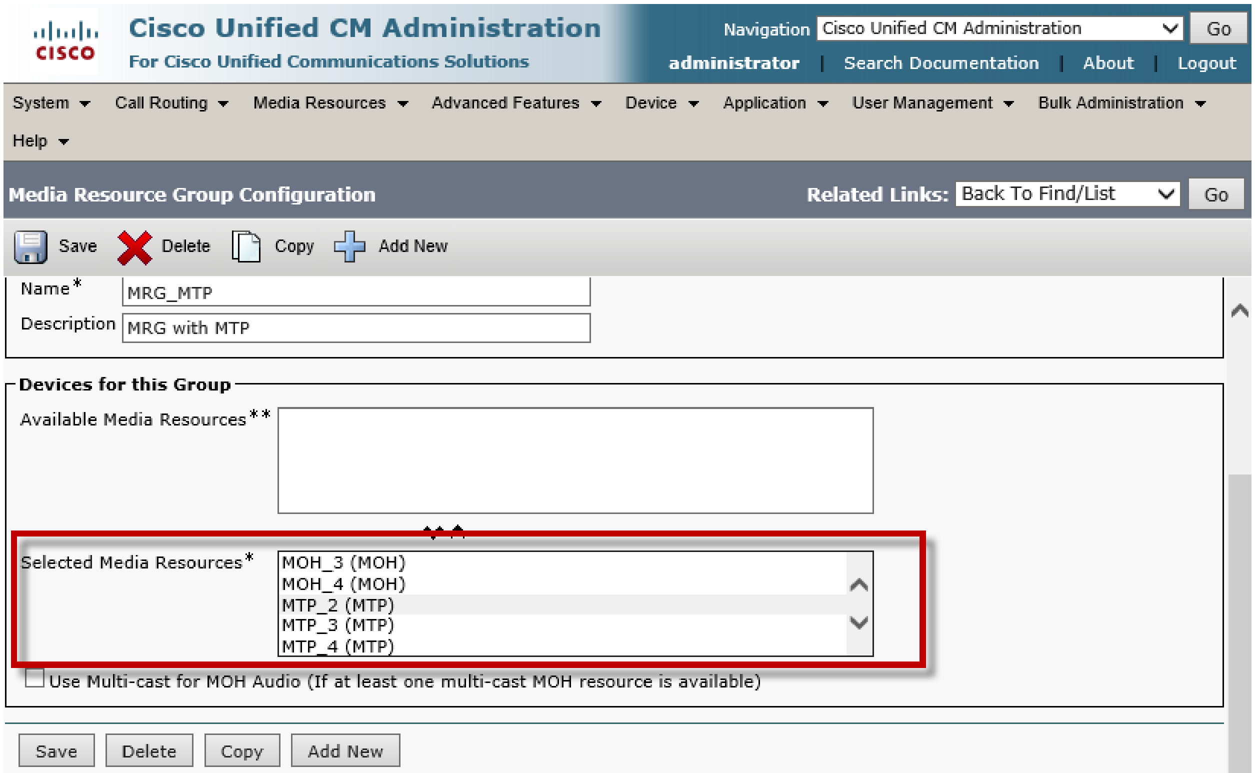
All other values are default.



MRG_MTP Resource Group (Continued)



MRG_MTP Resource Group (Continued)



Resource Group for MRG noMTP

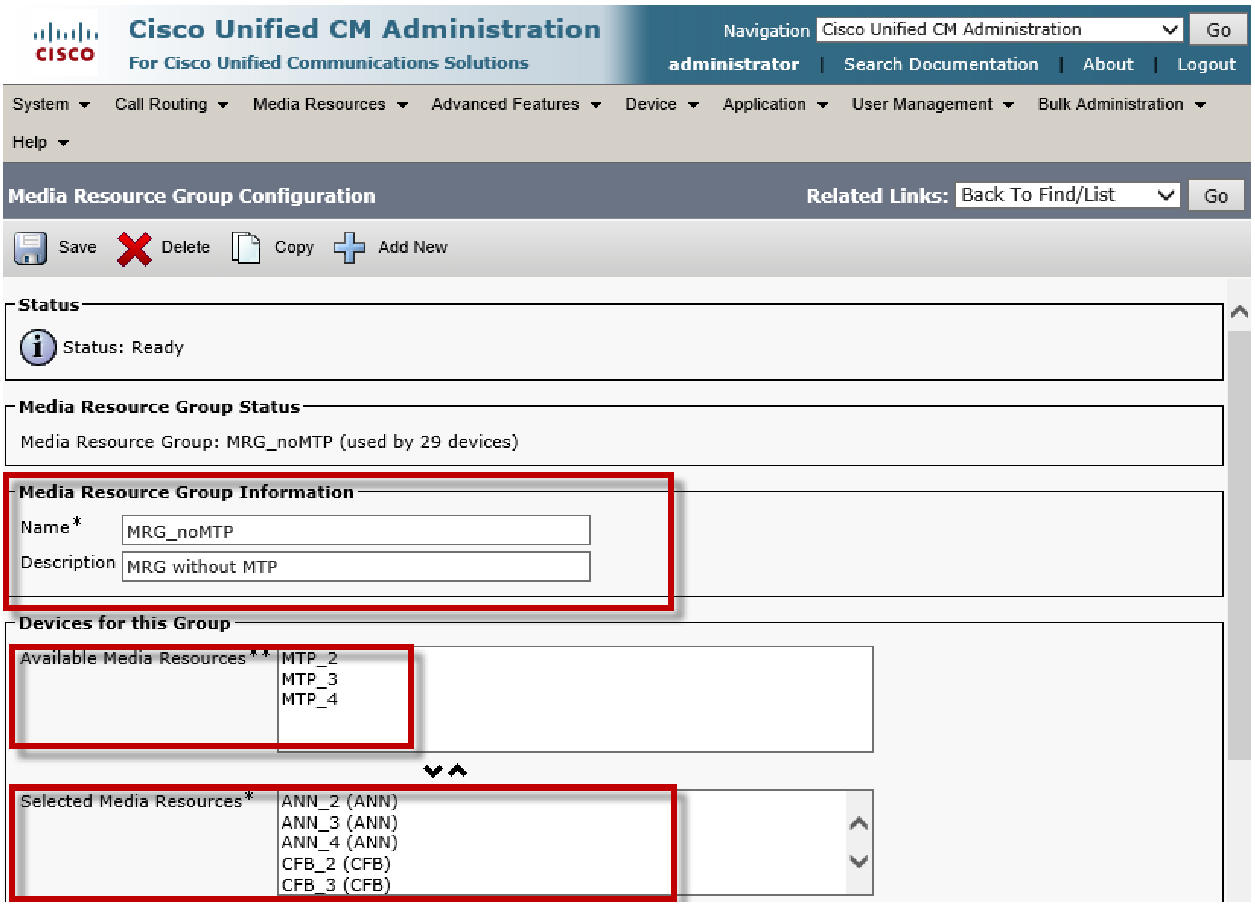
Set Name*= MRG_noMTP This is used for this example.

Set Description = This text is used to identify this Media Resource Group.

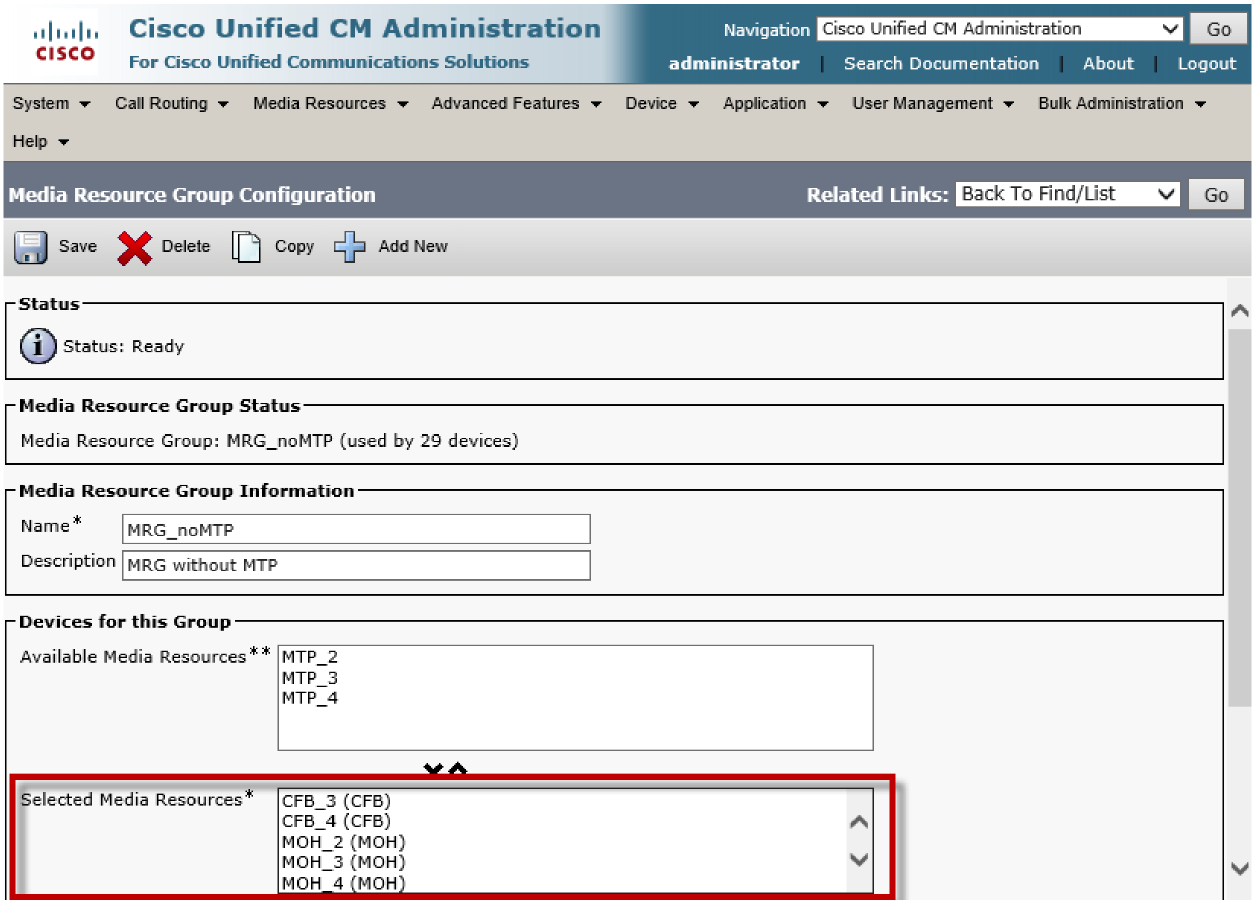
Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set other resources in the Selected Media Resources*

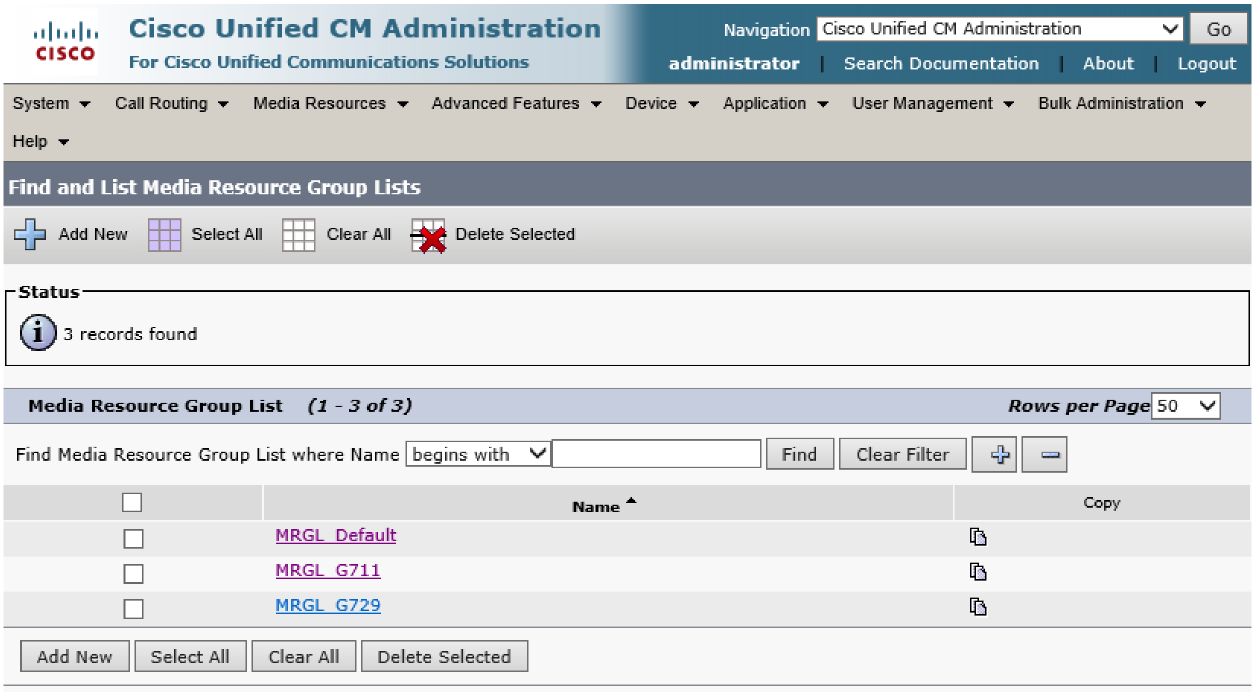
All other values are default.



Resource Group for MRG noMTP (Continued)



Cisco Unified Communications Manager Media Resource Group List

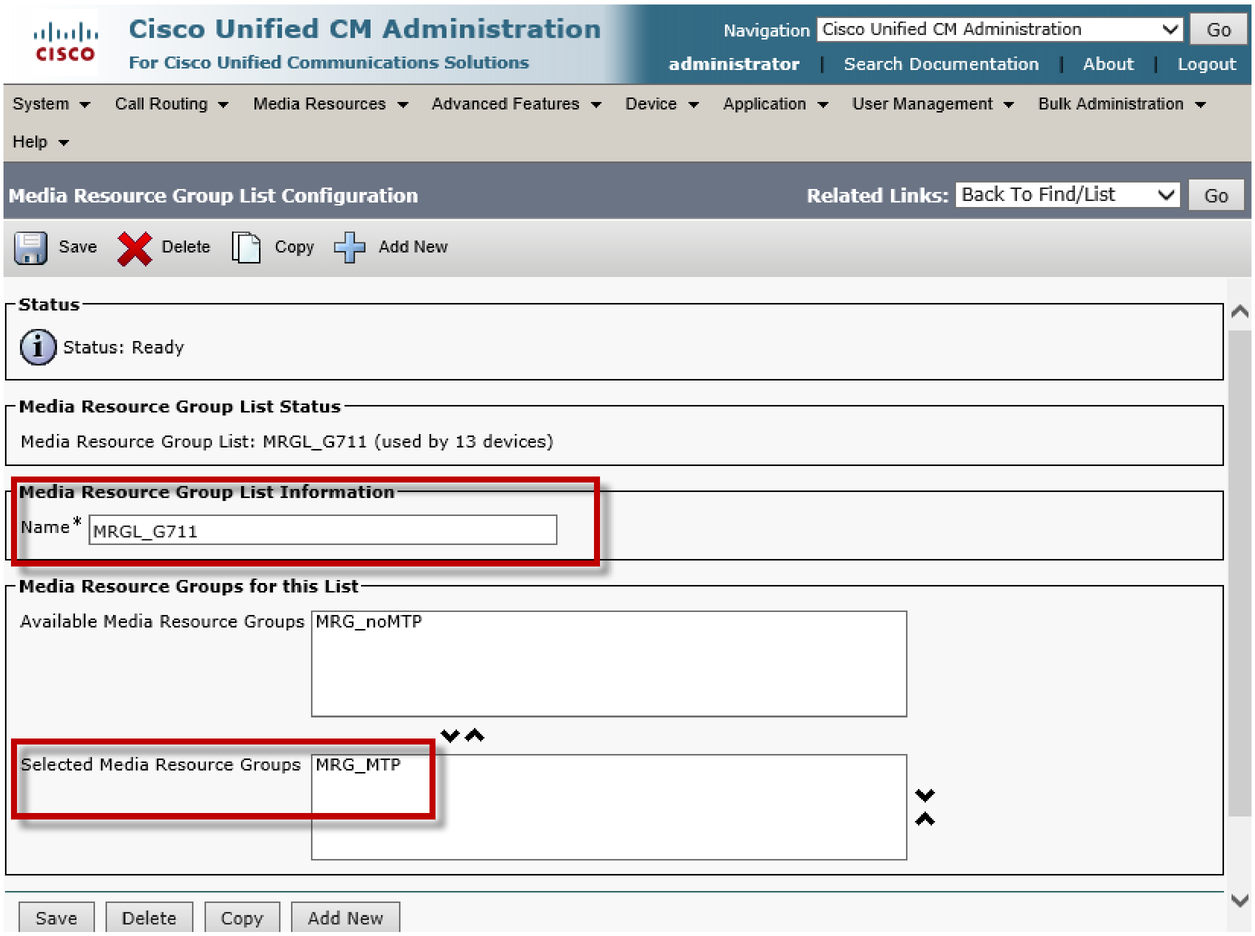


Set Name*= MRGL_G711 This is used for this example.

Set Description = This text is used to identify this Media Resource Group List.

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

Set Selected Media Resource Groups= MRG_MTP



Note: This Media Resource Group List was added to provide early offer on the invite from Cisco to Avaya for SCCP phones.

Cisco Unified Communications Manager Route Pattern to Avaya

Set Route Pattern* =3XXX This is used to route to Avaya in this example.

Set Description = This text is used to identify this Route Pattern.

Set Gateway/Route List* = Trunk_to_Avaya_SM. This is used for this example.

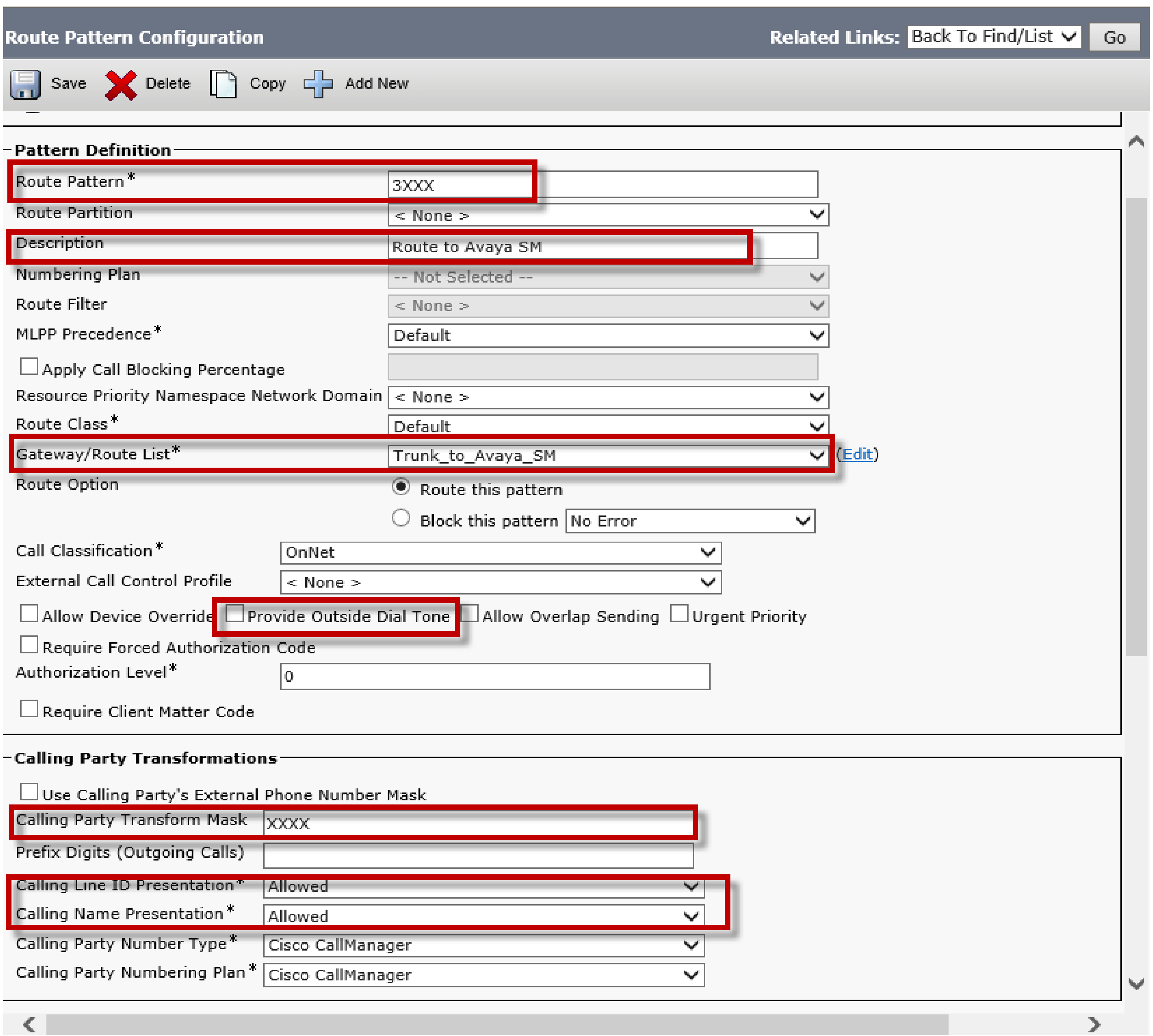
Uncheck Provide Outside Dial Tone

Set Calling Party Transform Mask = XXXX

Set Calling Line ID Presentation= Allowed

Set Calling Name Presentation= Allowed

All other values are default.

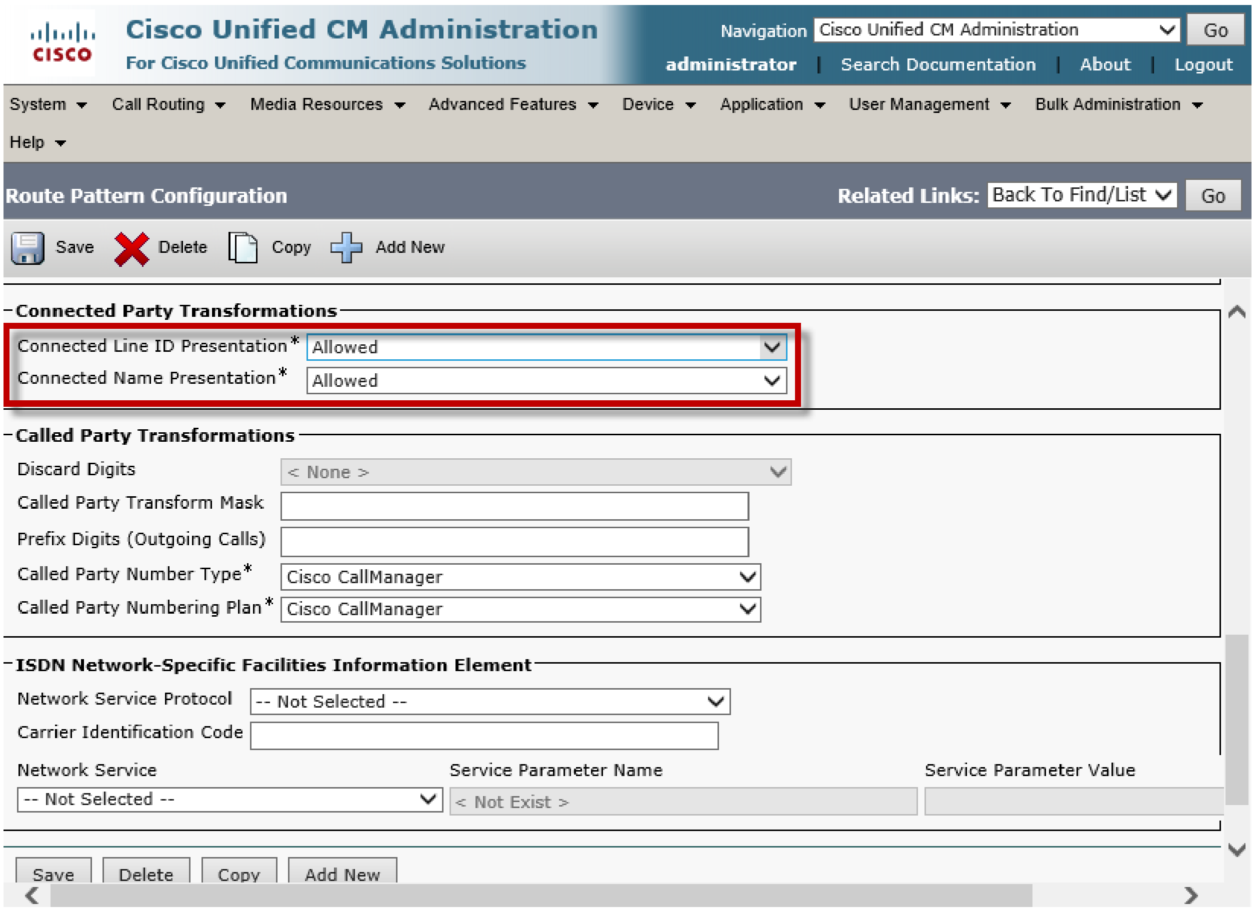


Route Patter Configuration for 3xxx (Continued)

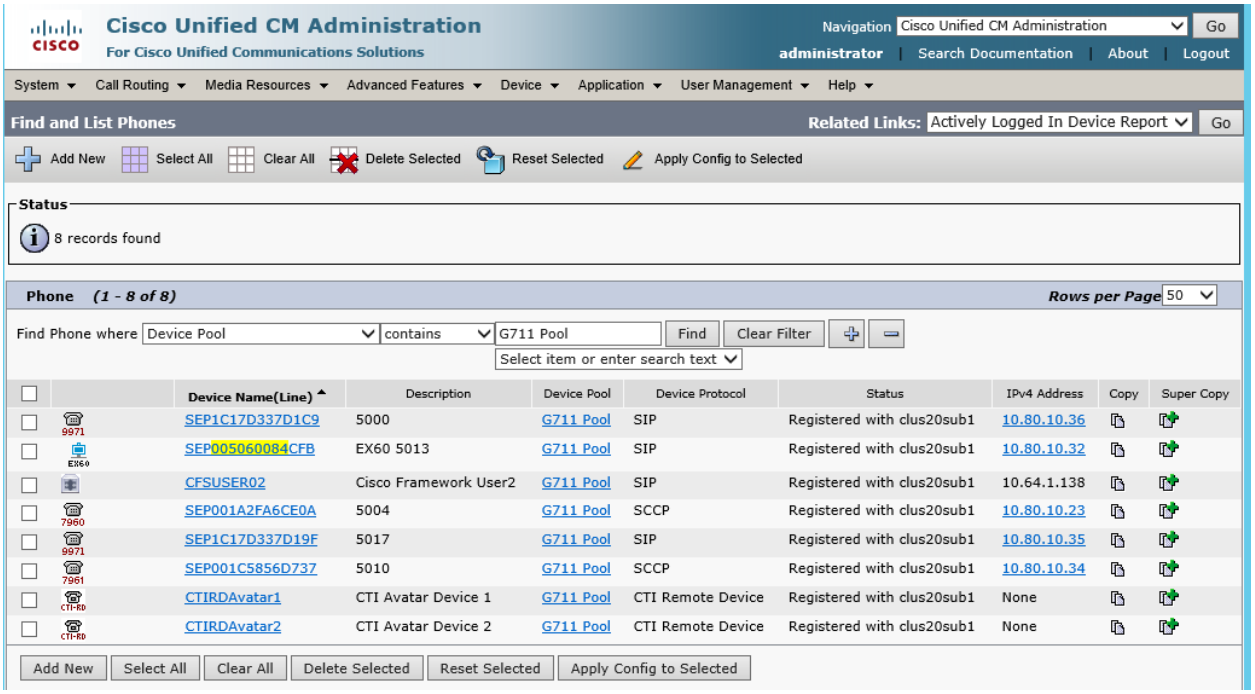
Set Connected Line ID Presentation*= Allowed

Set Calling Name Presentation* = Allowed

All other values are default.



Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration



Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set MAC Address* = 1C17D337D1C9 . This is used in this example.

Set Description = This text is used to identify this Phone

Set Device Pool*= G711 Pool. This is used in this example.

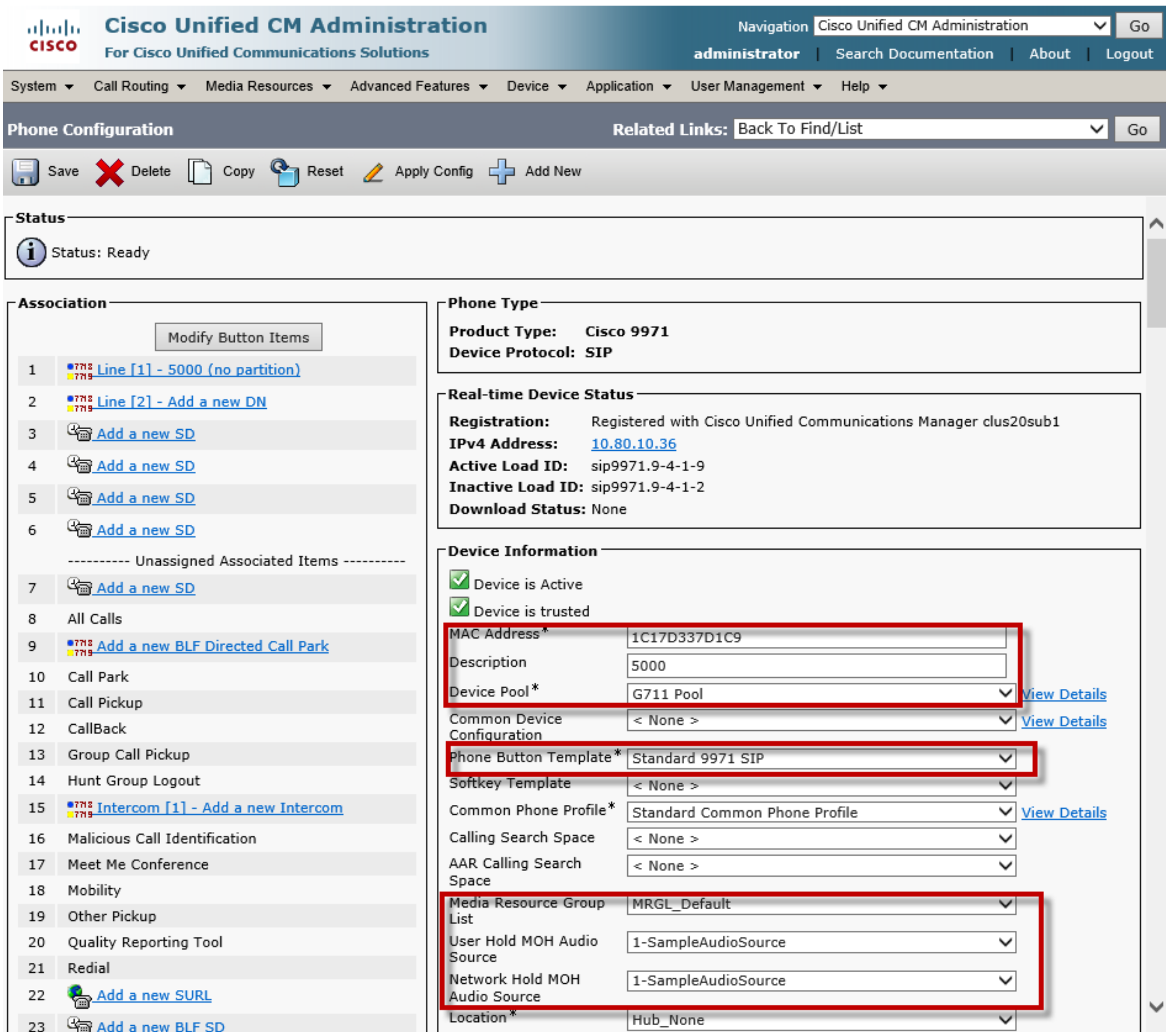
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.

Set Media Resource Group List = MRGL_Default. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

All other values are default.

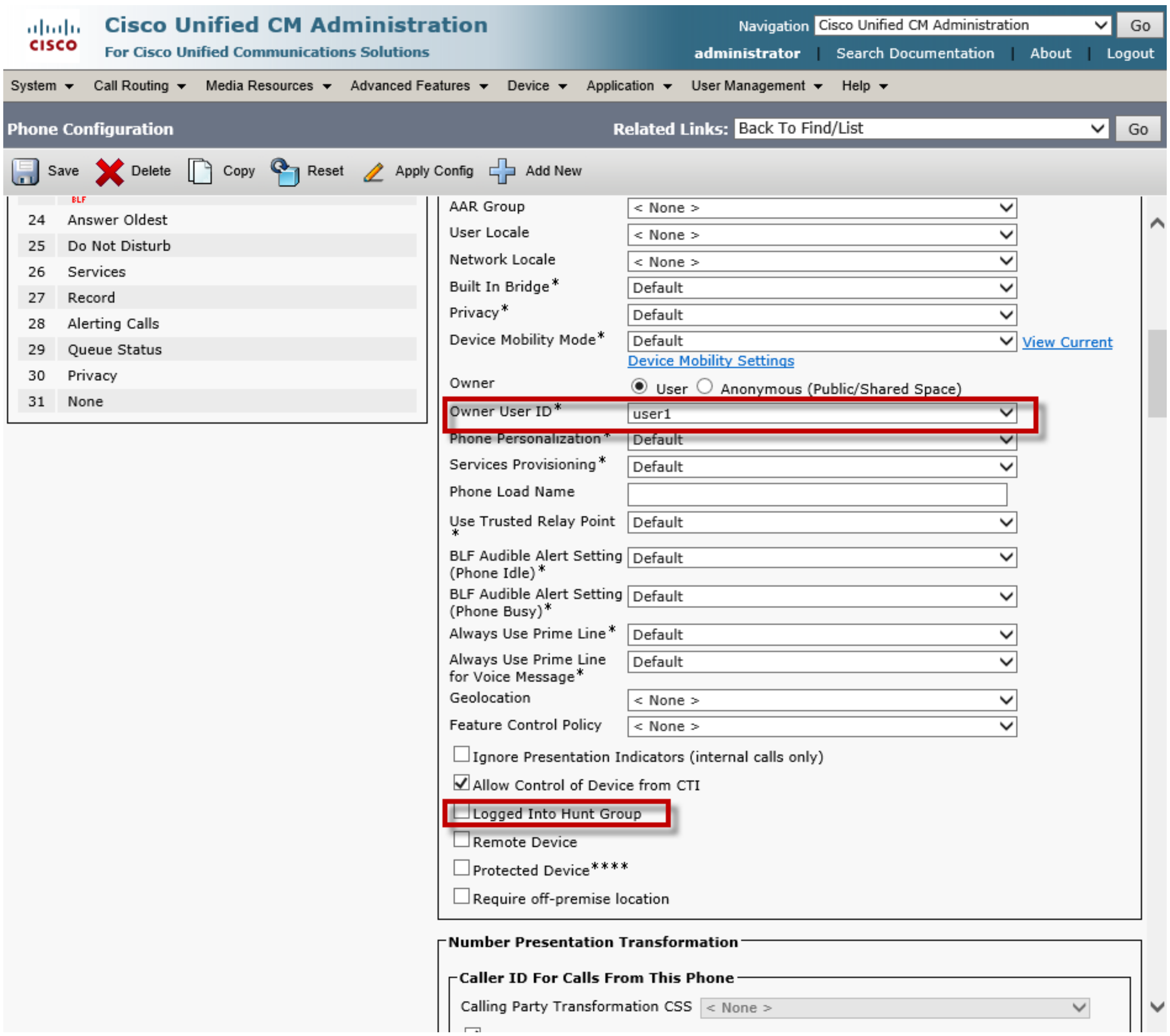


Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set Owner User ID*= User1. Leave Blank if Phone is not provisioned for Jabber Avatar

Uncheck Logged Into Hunt Group

All other values are default.



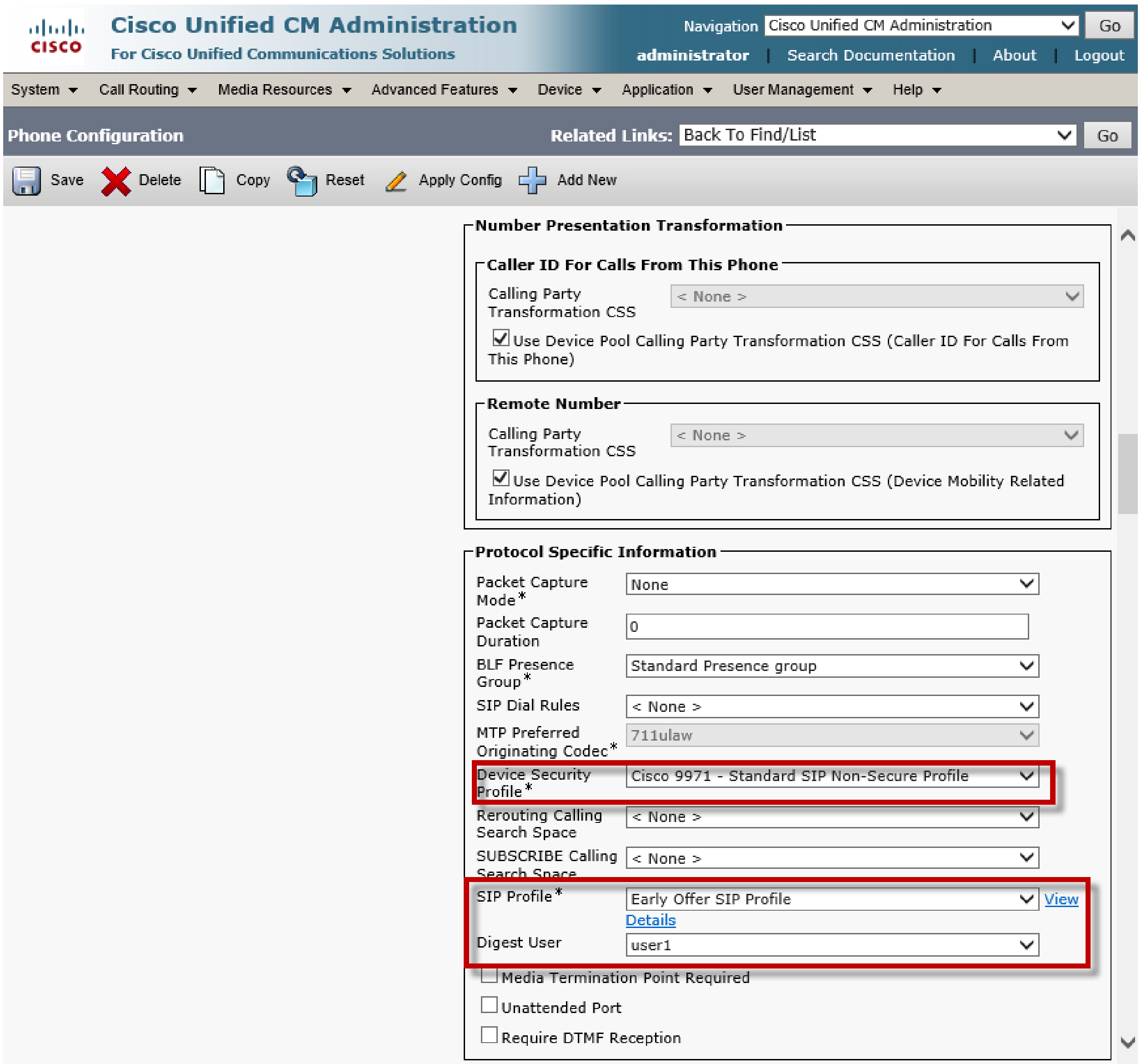
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set Device Security Profile* = Cisco 9971- Standard SIP Non-Secure Profile. This is used in this example.

Set SIP Profile*= Early Offer SIP Profile. This is used in this example.

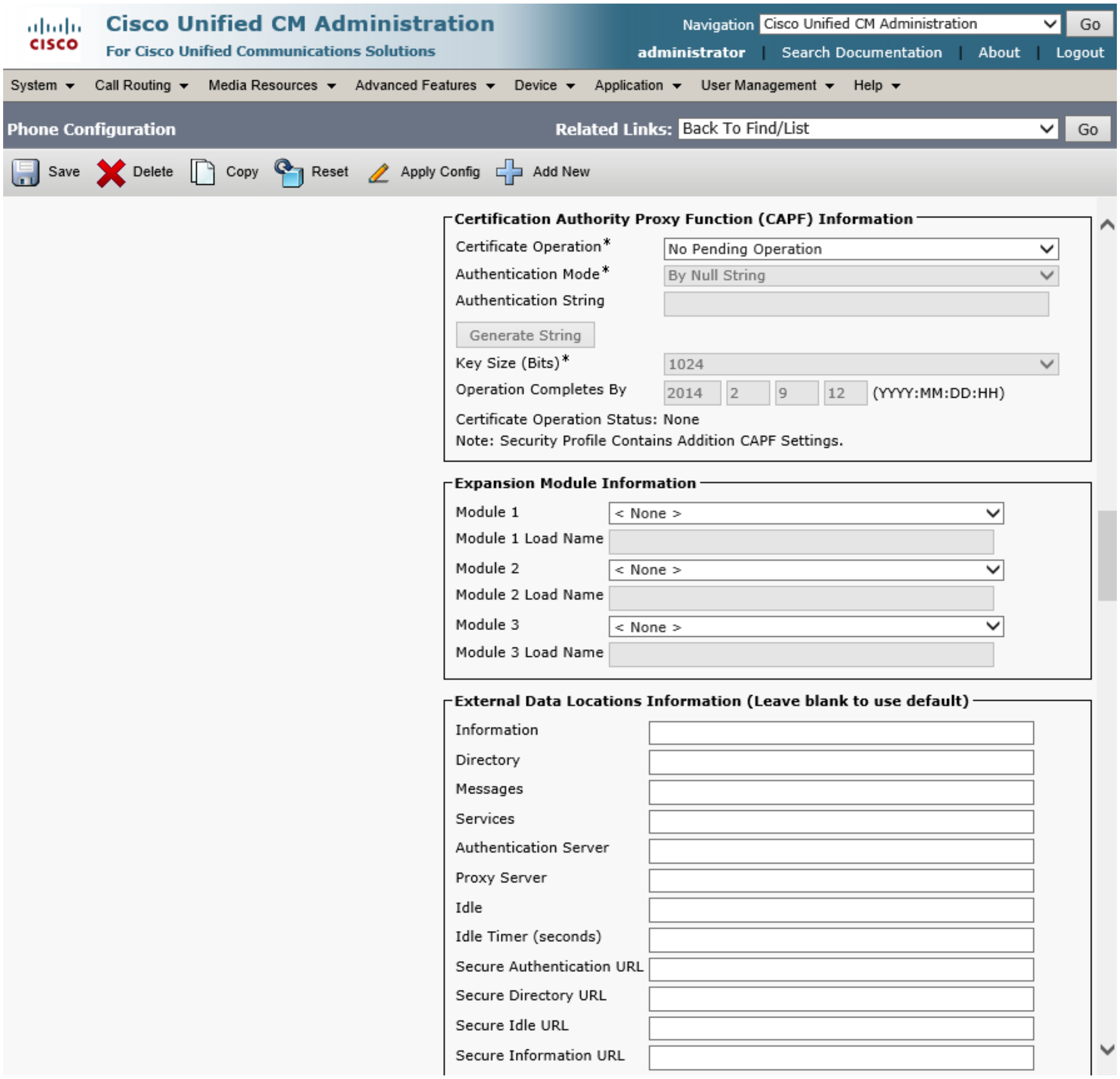
Set Digest User = user1. If this is not a Jabber Avatar Phone. Leave as none.

All other values are default.



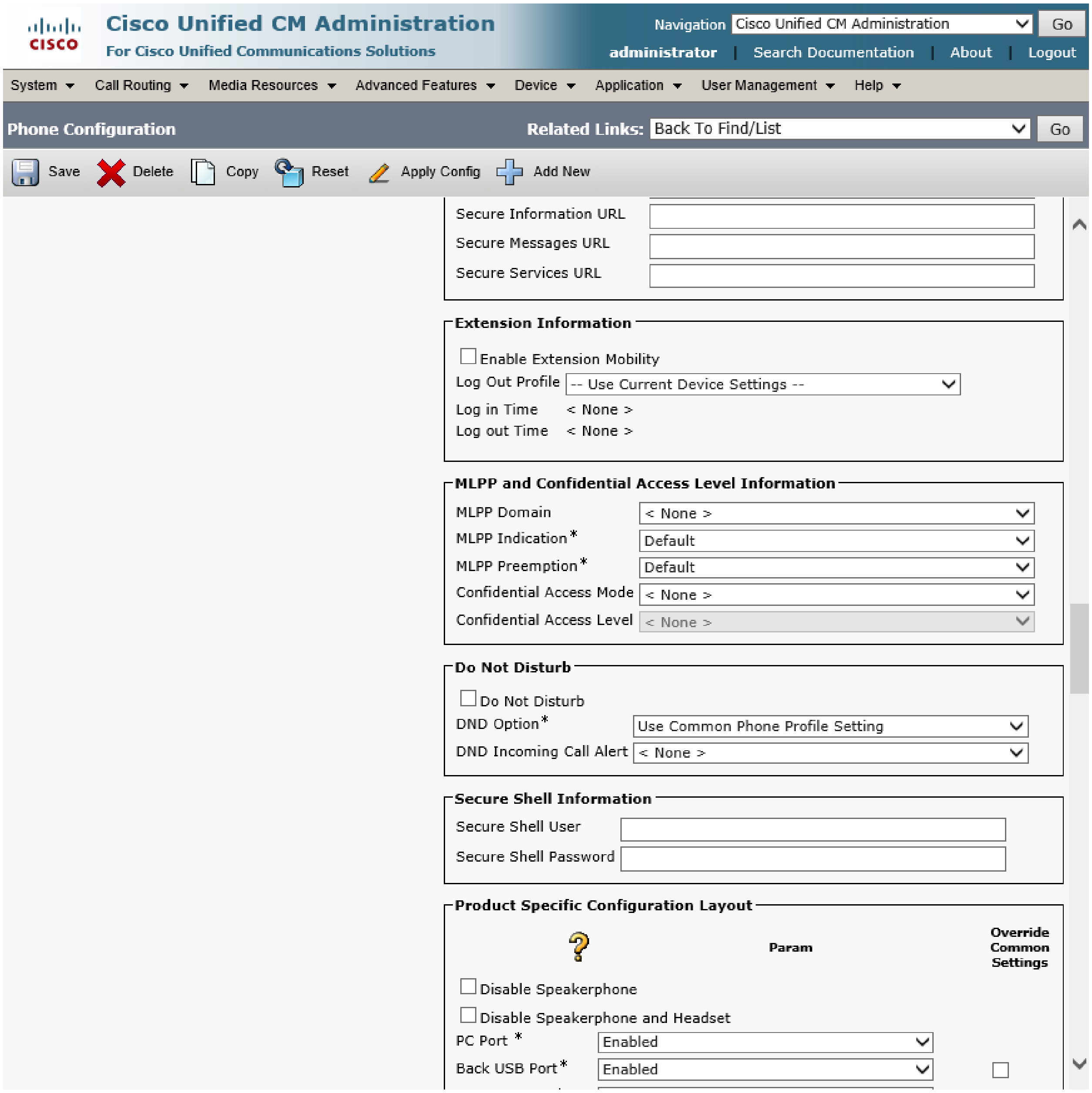
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All other values are default.



Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.

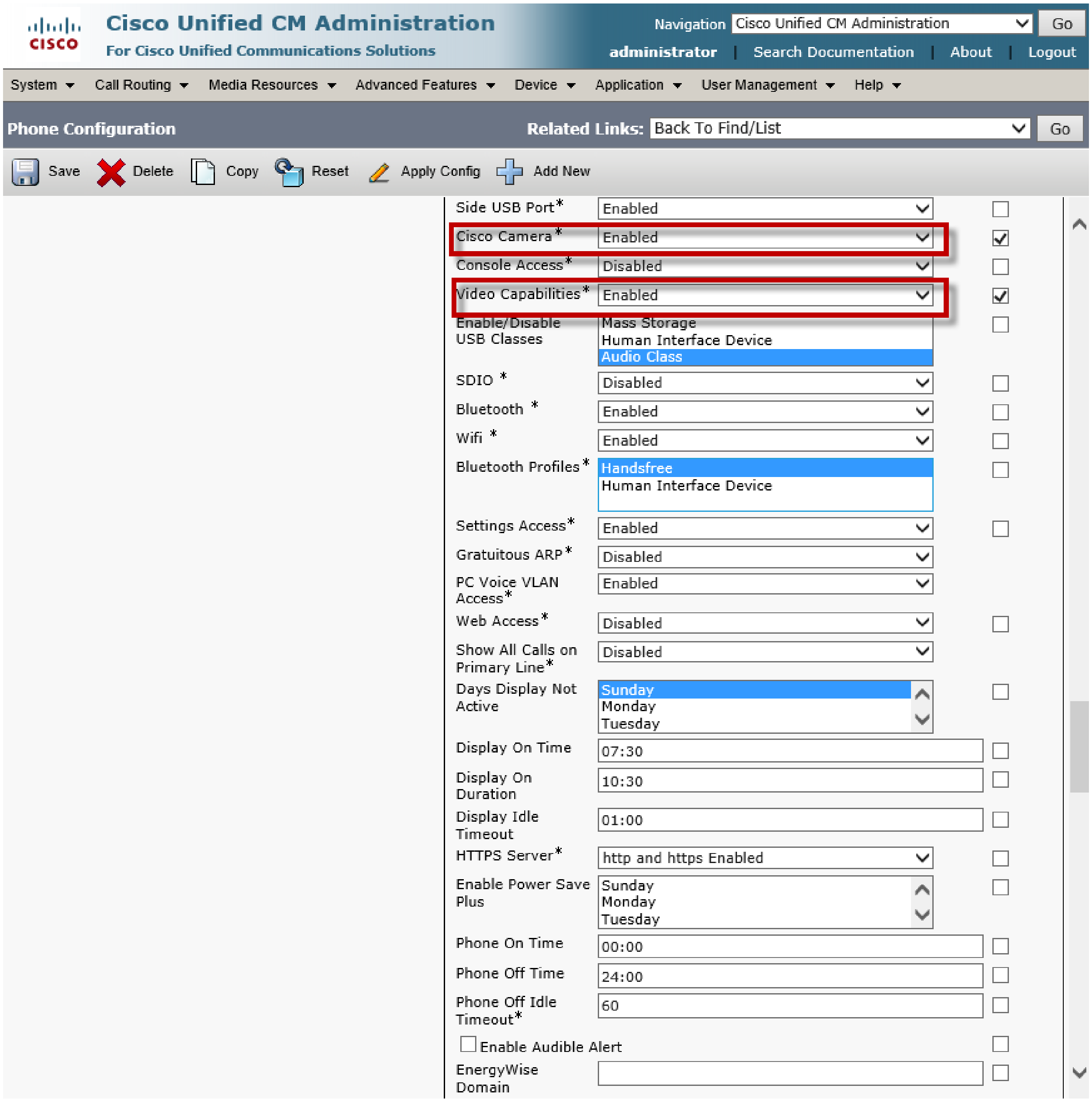


Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

Set Cisco Camera* = Enabled. This is used in this example.

Set Video Capablities* = Enabled. This is used in this example.

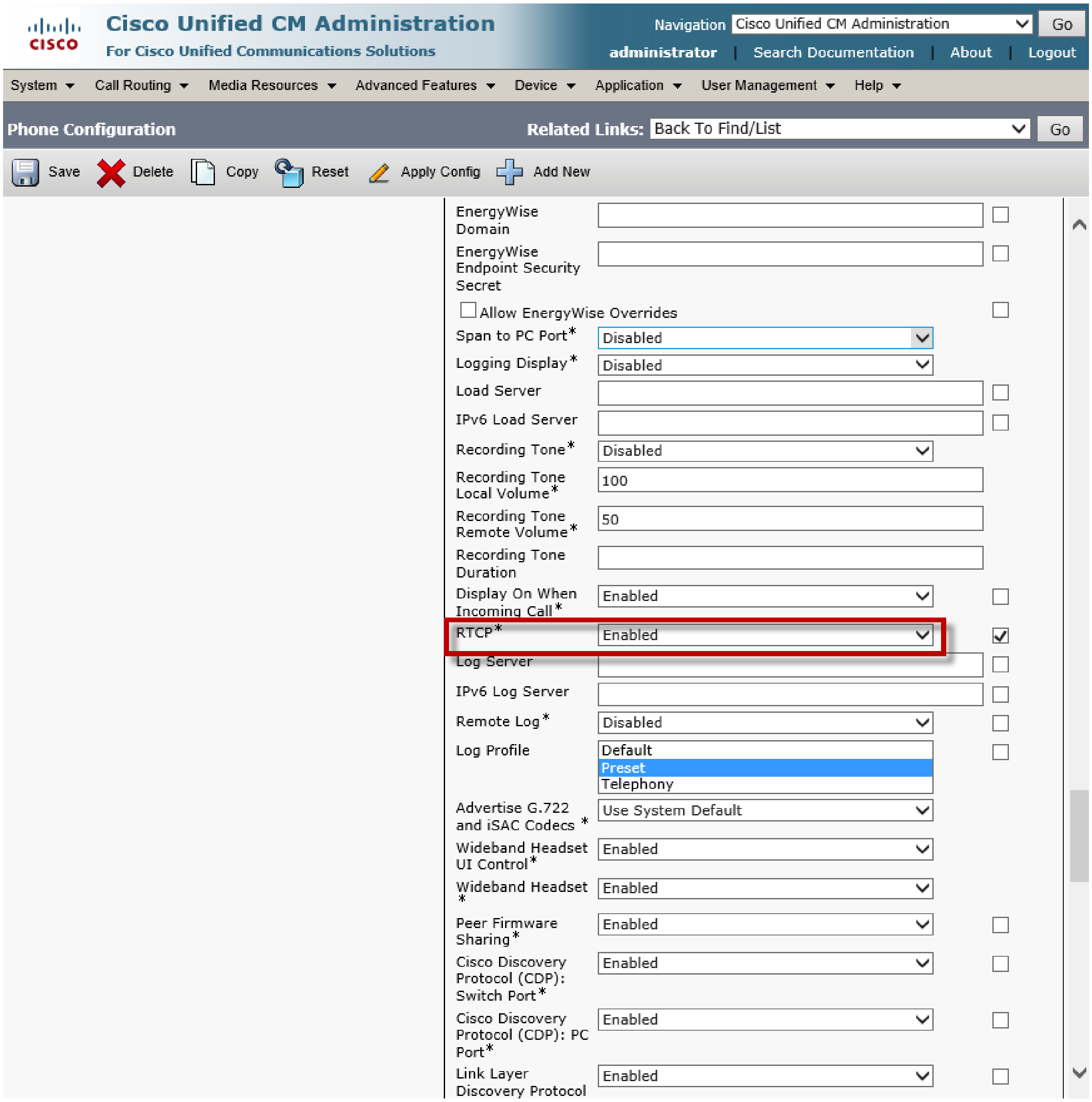
All other values are default.



Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

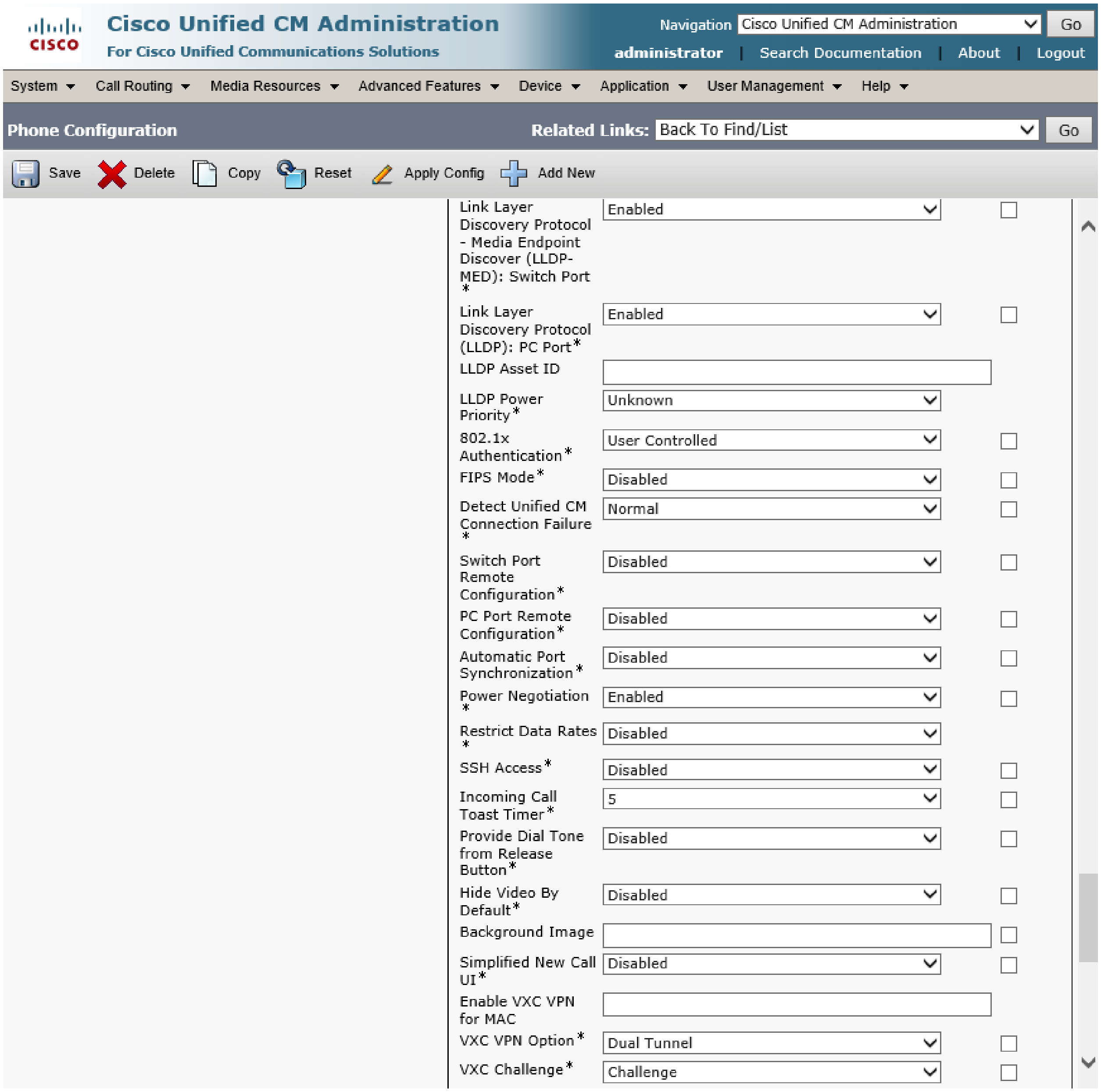
Set RTCP* = Enabled. This is used in this example.

All other values are default.



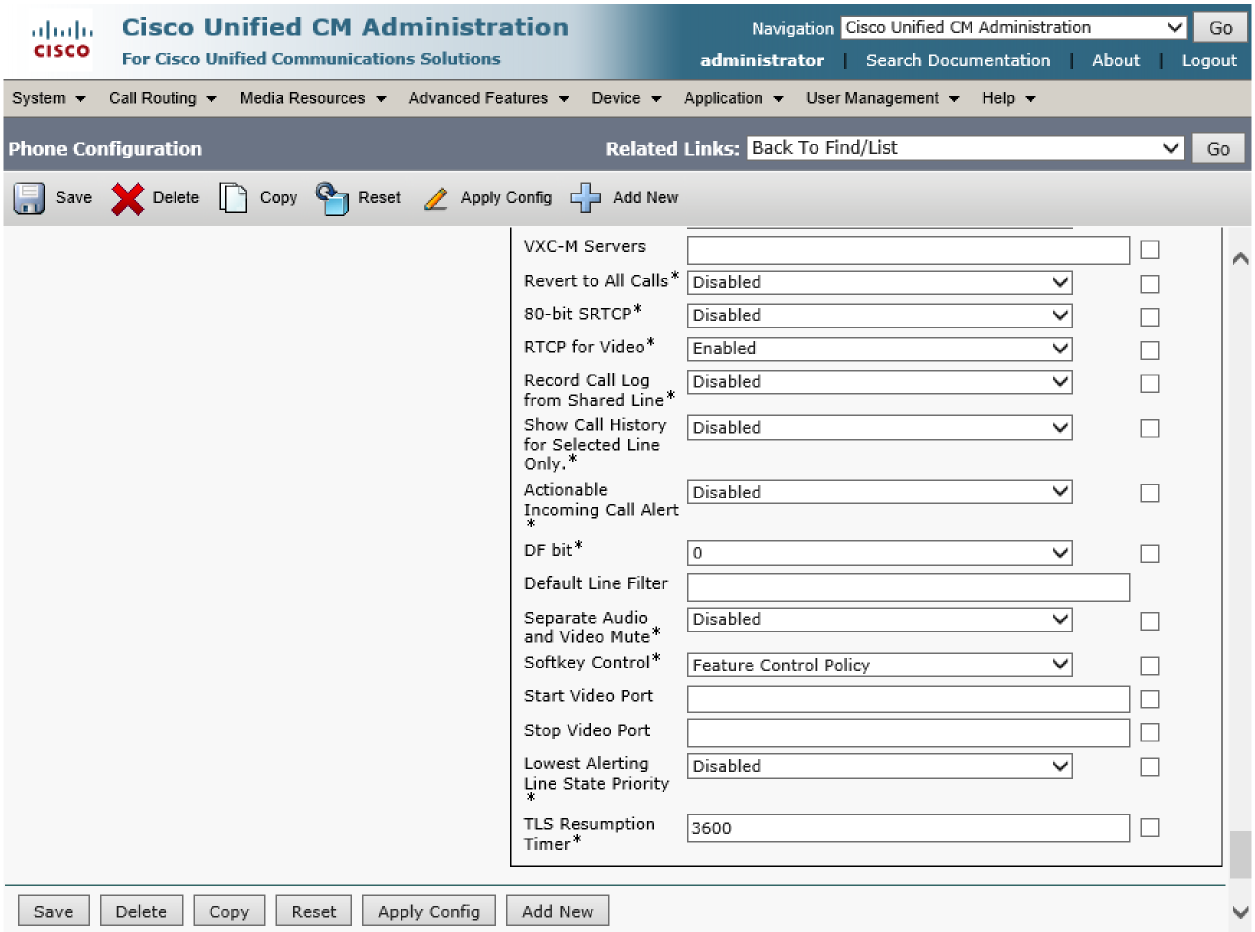
Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.



Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

All values are default.



Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration

Set MAC Address* = 001C5856D737. This is used in this example.

Set Description = This text is used to identify this Phone

Set Device Pool*= G711 Pool . This is used in this example.

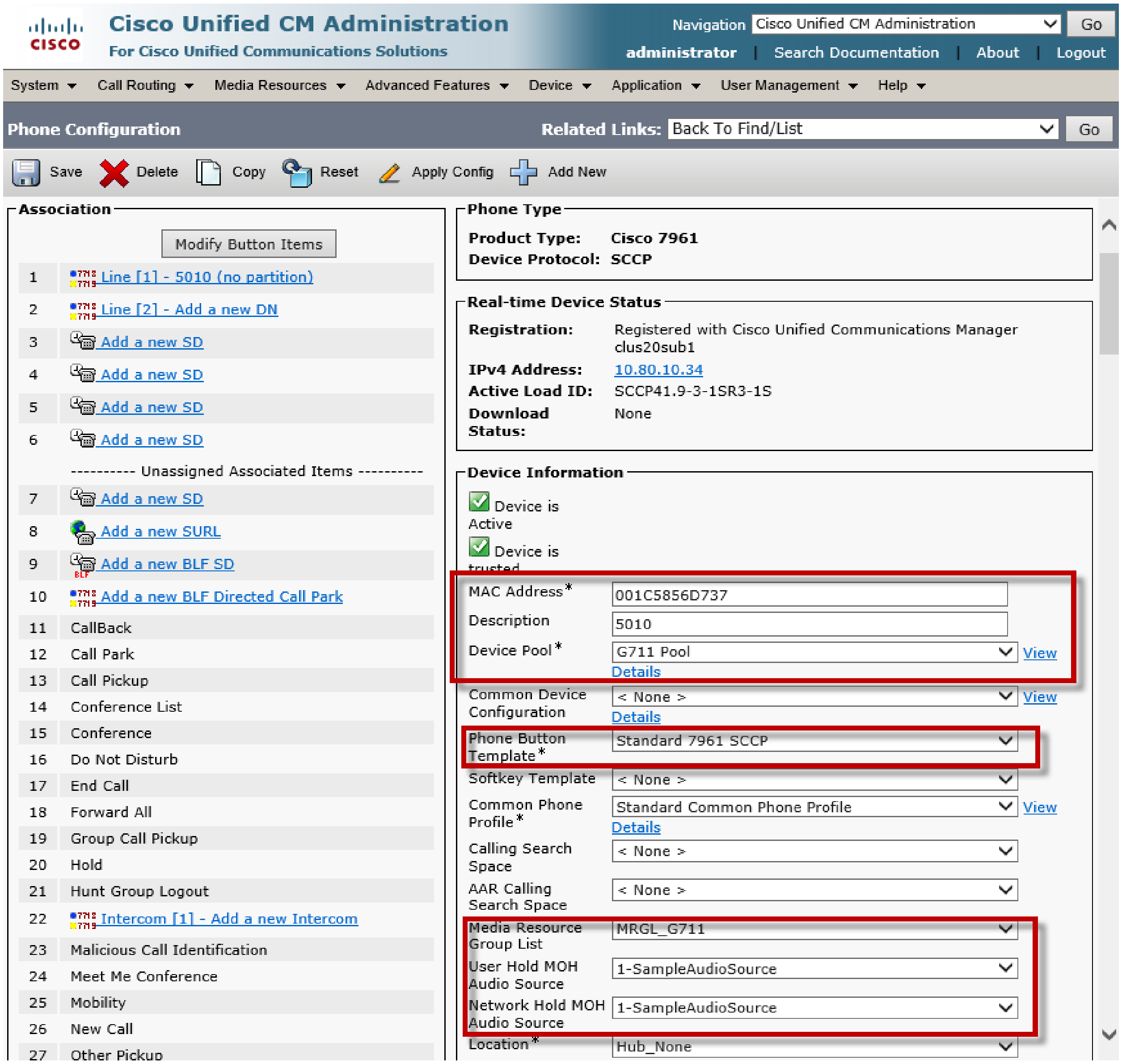
Set Phone Button Template*= Standard 7961 SCCP. This is used in this example

Set Media Resource Group List = MRGL_G711. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

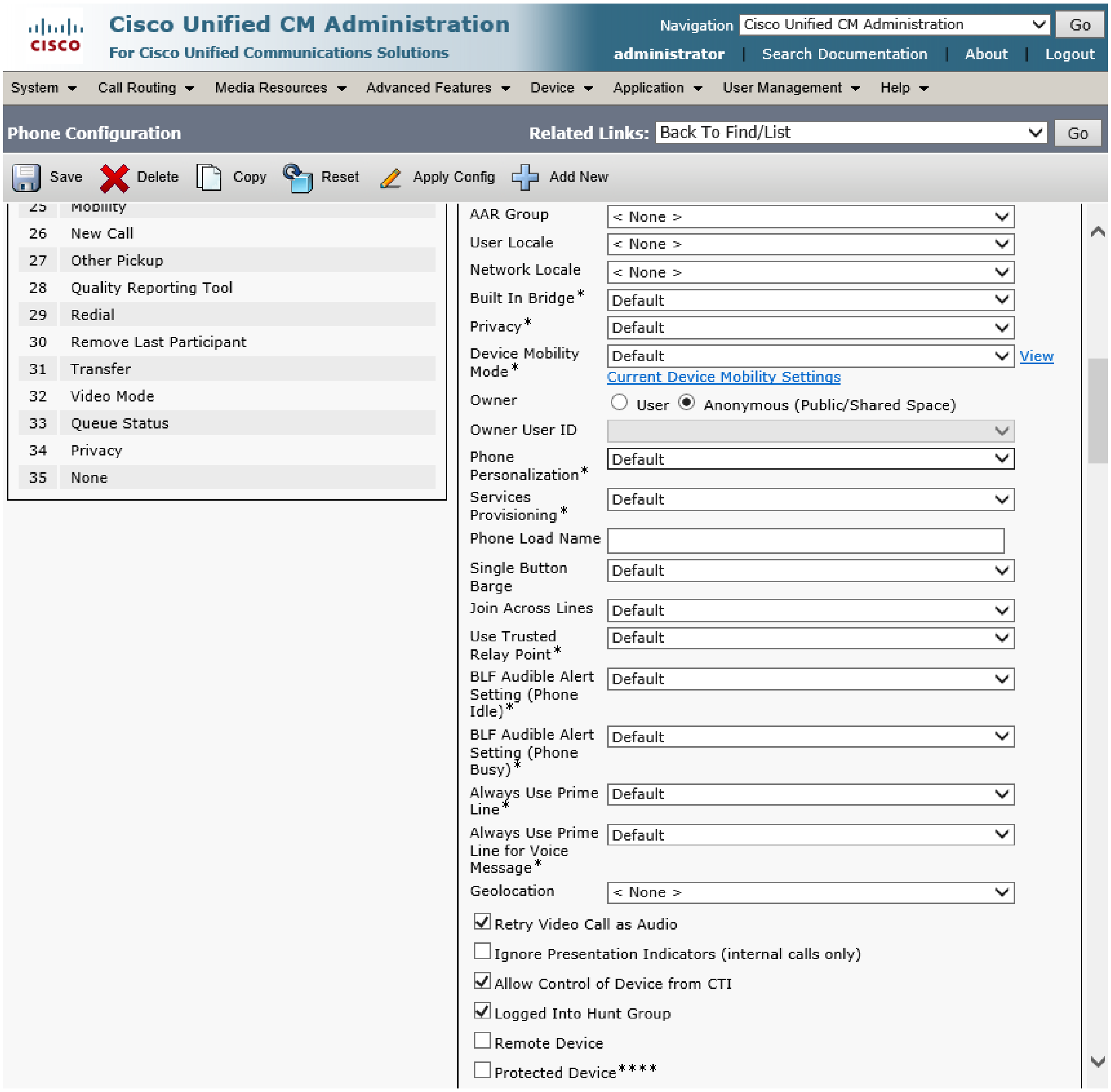
Set Network Hold MOH Audio Source = 1-SampleAudioSource.

All other values are default.



Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

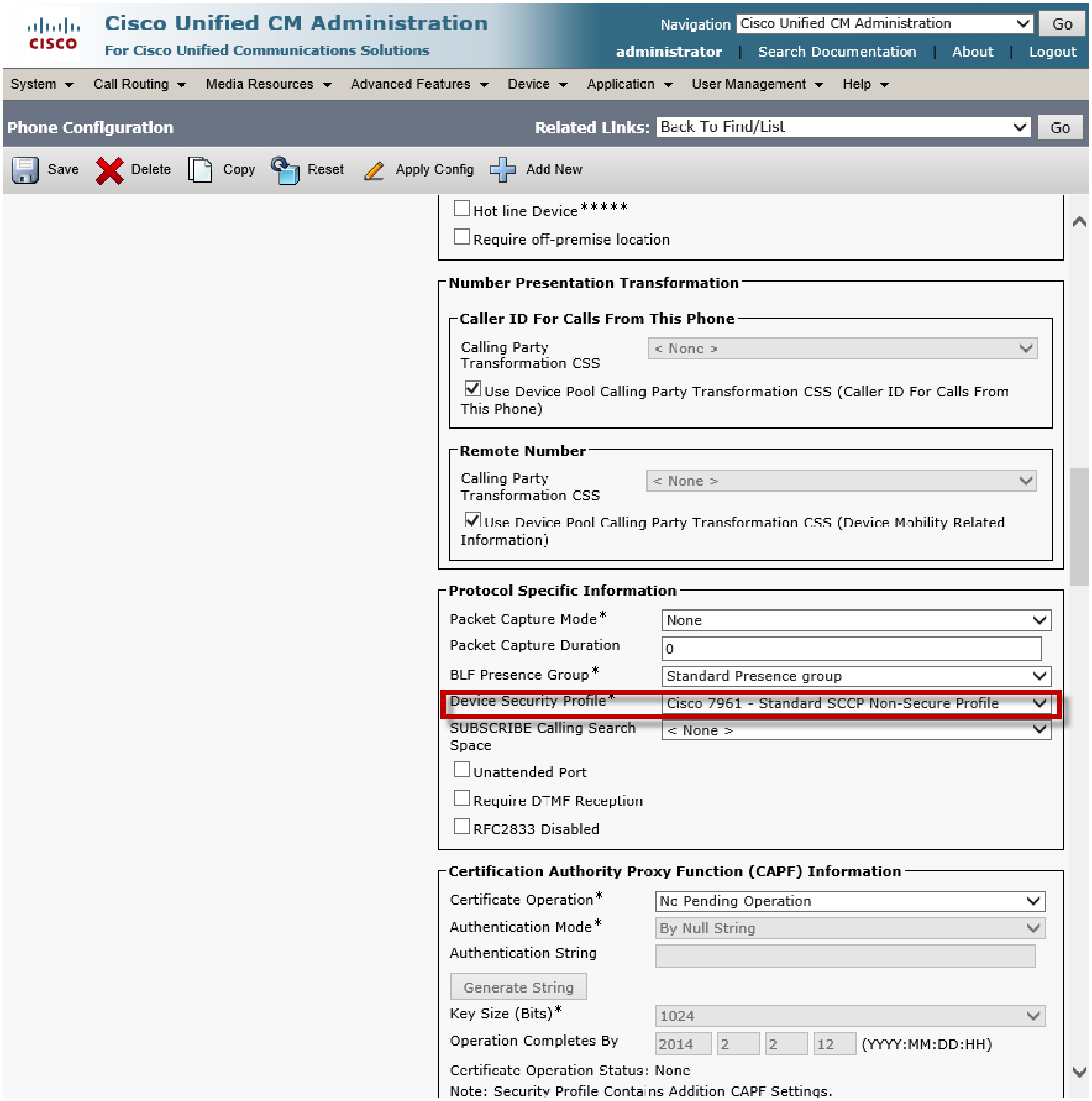
All other values are default.



Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

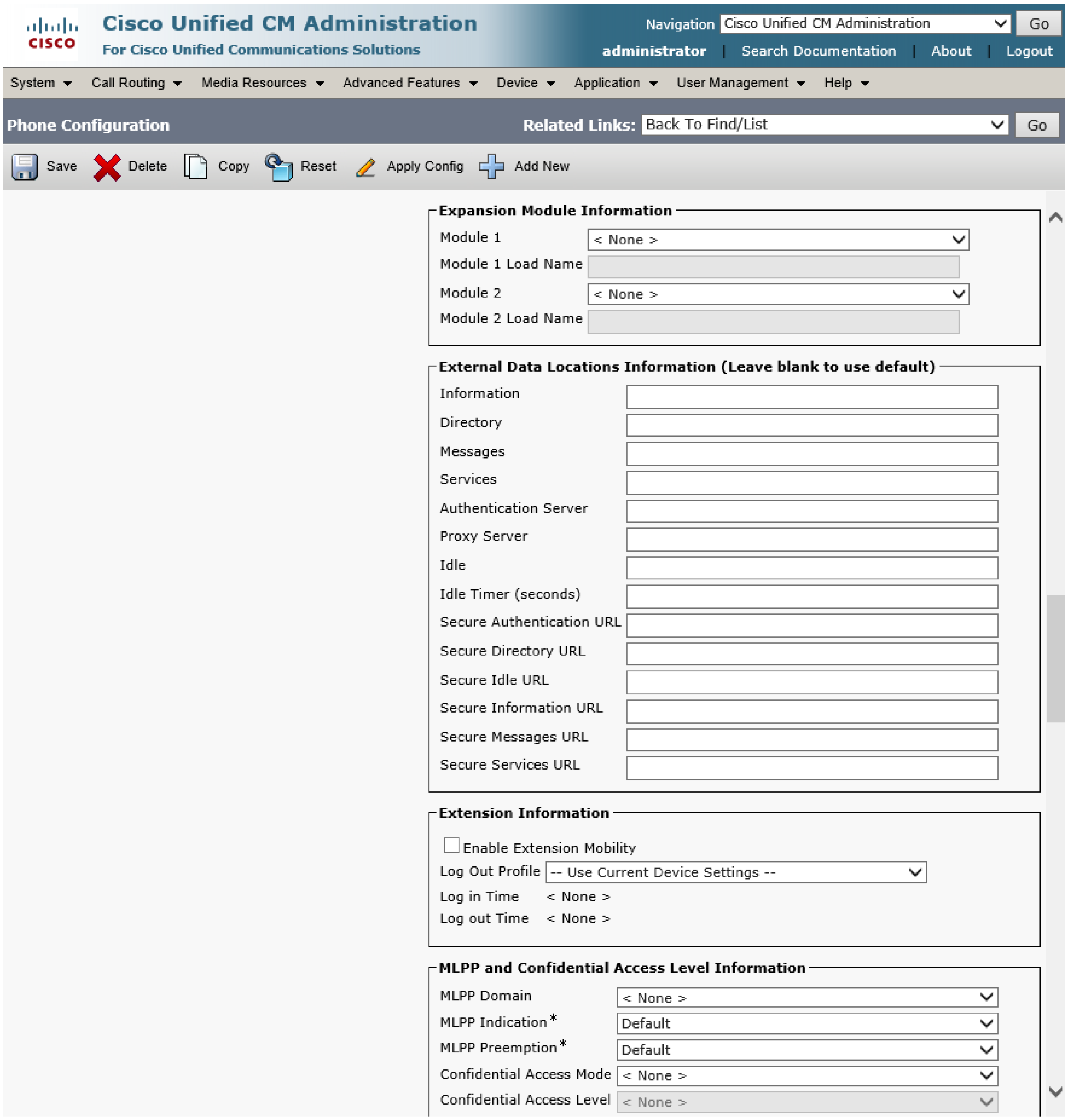
Set Device Security Profile* = Cisco 7961 – Standard SCCP Non-Secure Profile. This is used in this example

All other values are default.



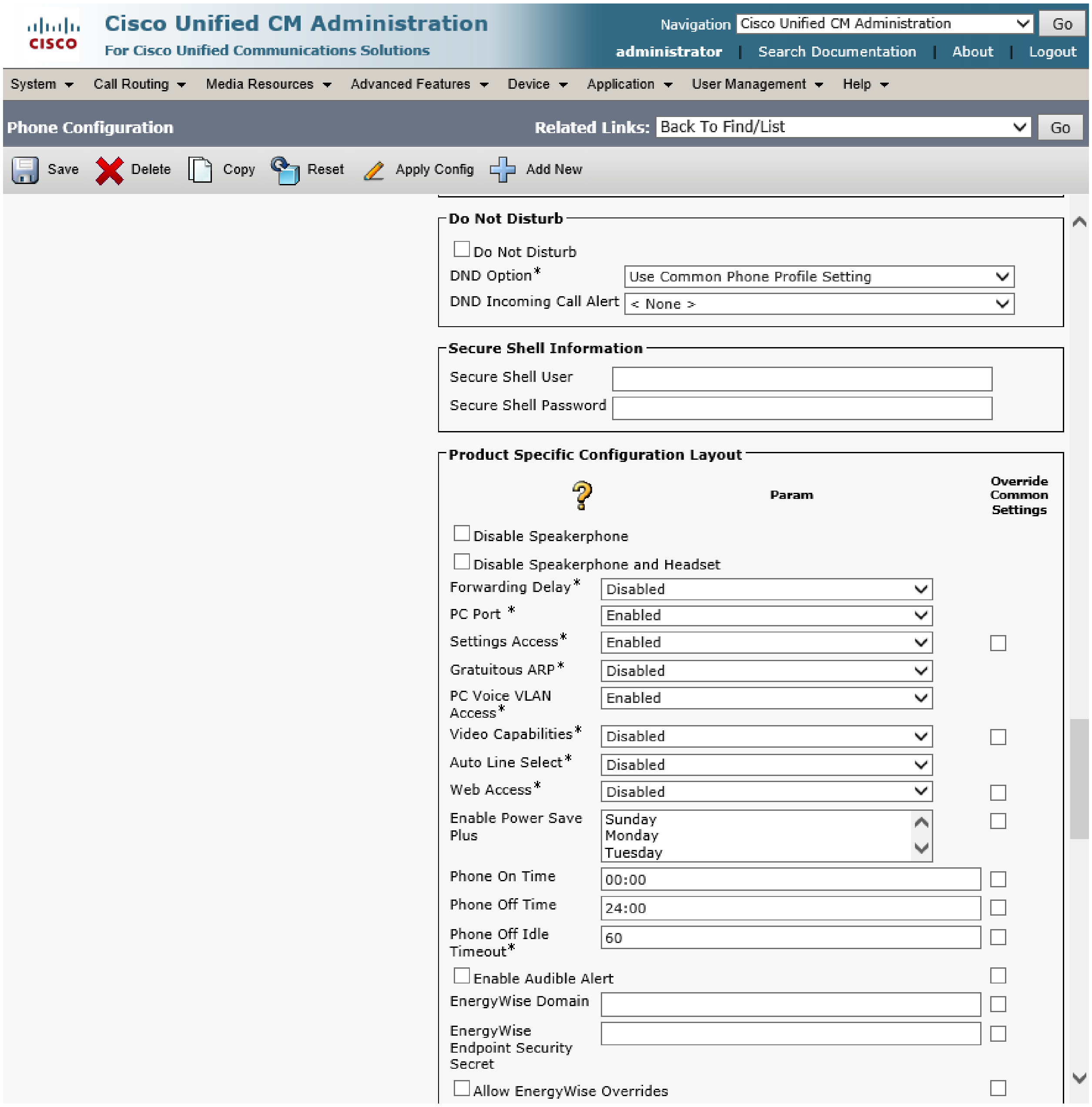
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.



Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.

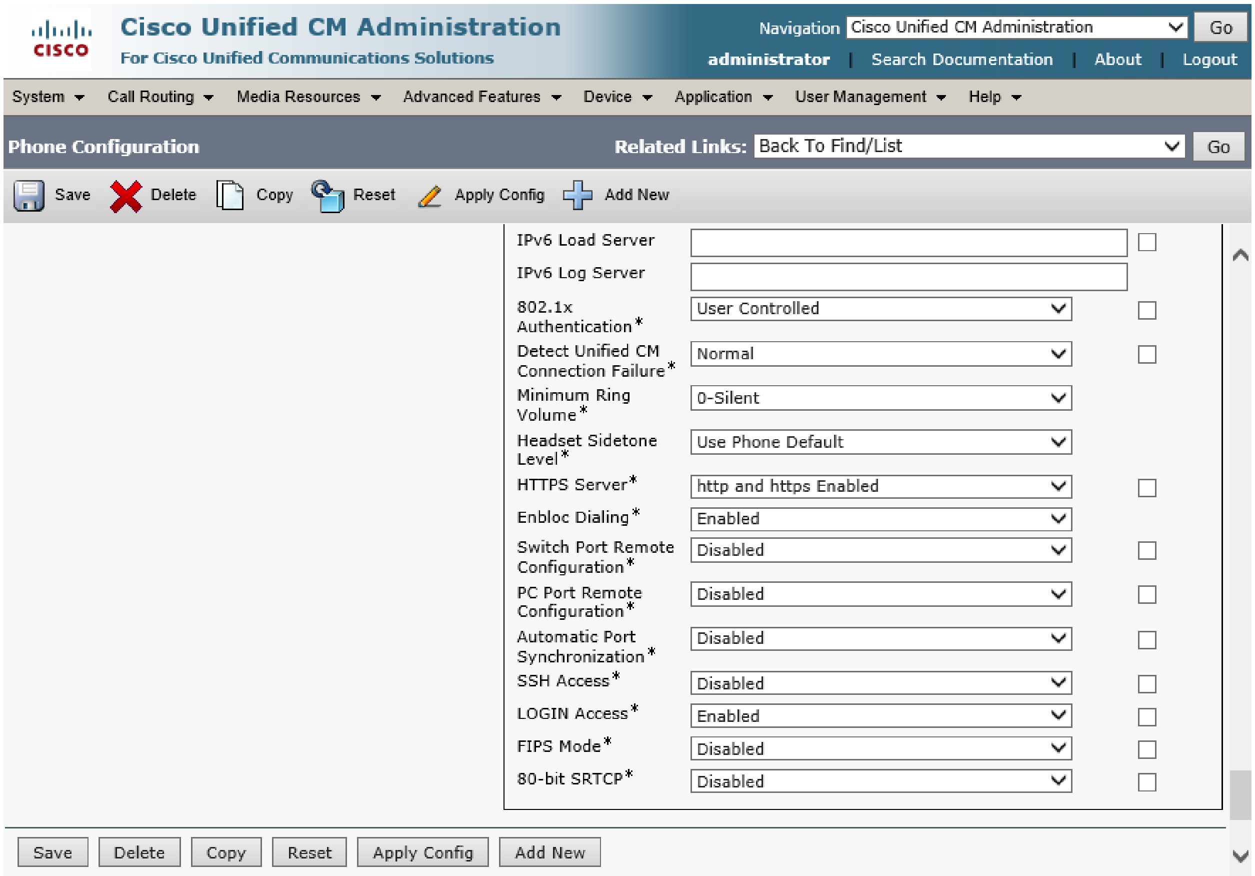


Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.

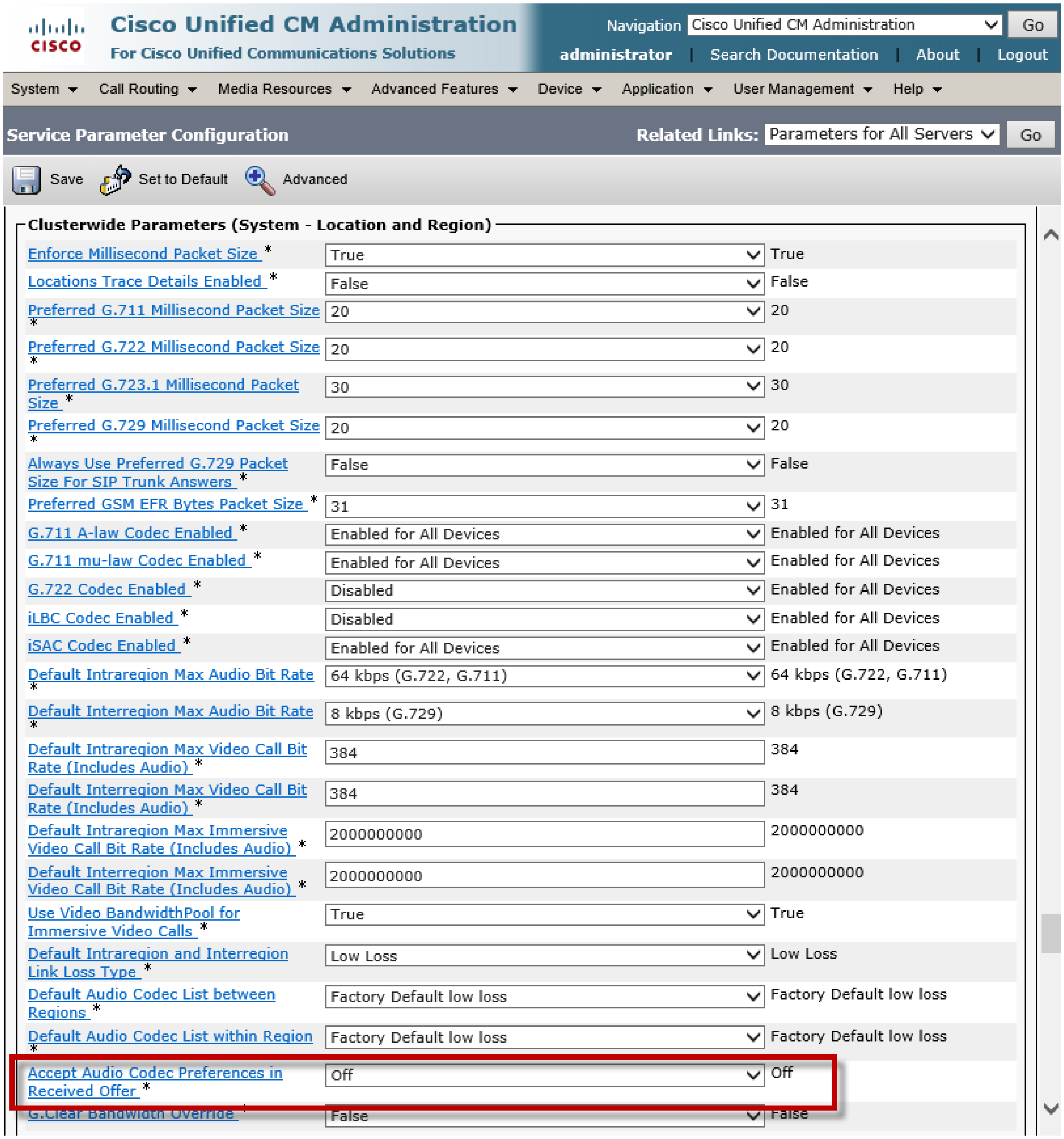
Cisco Unified Communications Manager SCCP Phone Ext. 5010 Device Level Configuration (Continued)

These values are default.



Cisco Unified Communications Manager Audio Codec Preference List Configuration

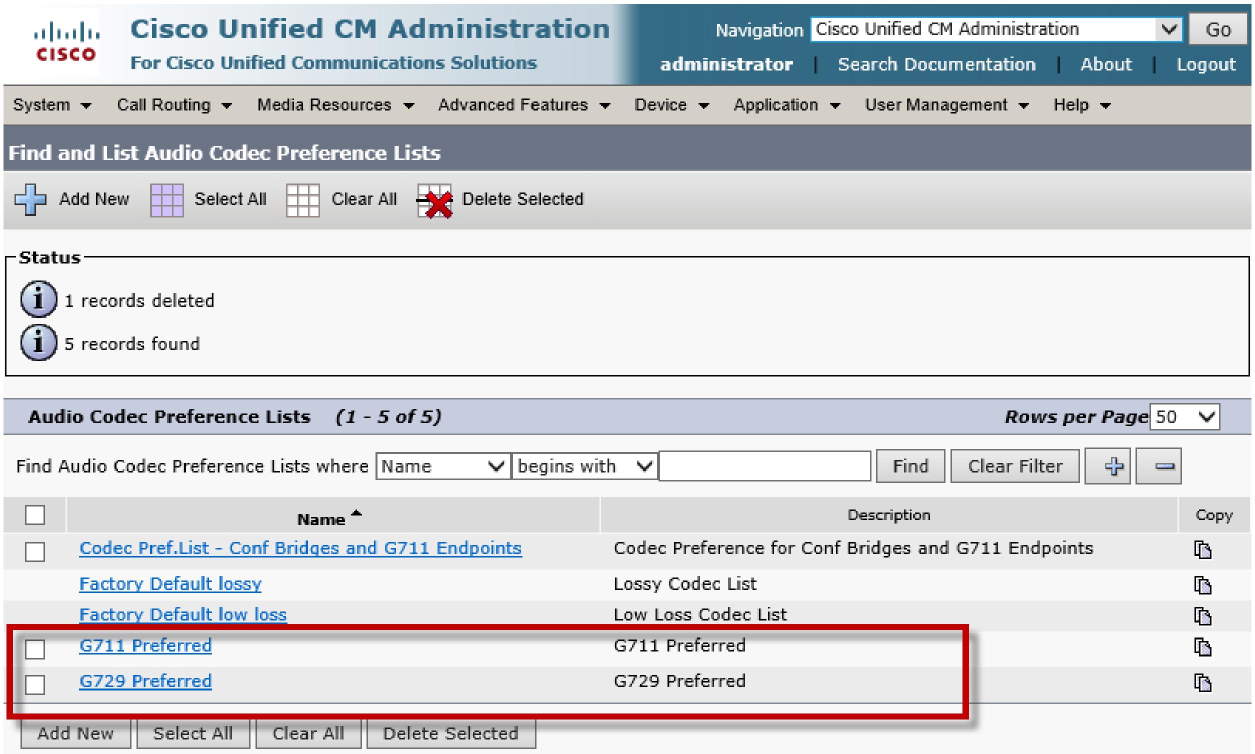
Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.



Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

G711 Preferred and G729 Preferred Audio Codec Preference List created in this example.

All other values are default.



Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

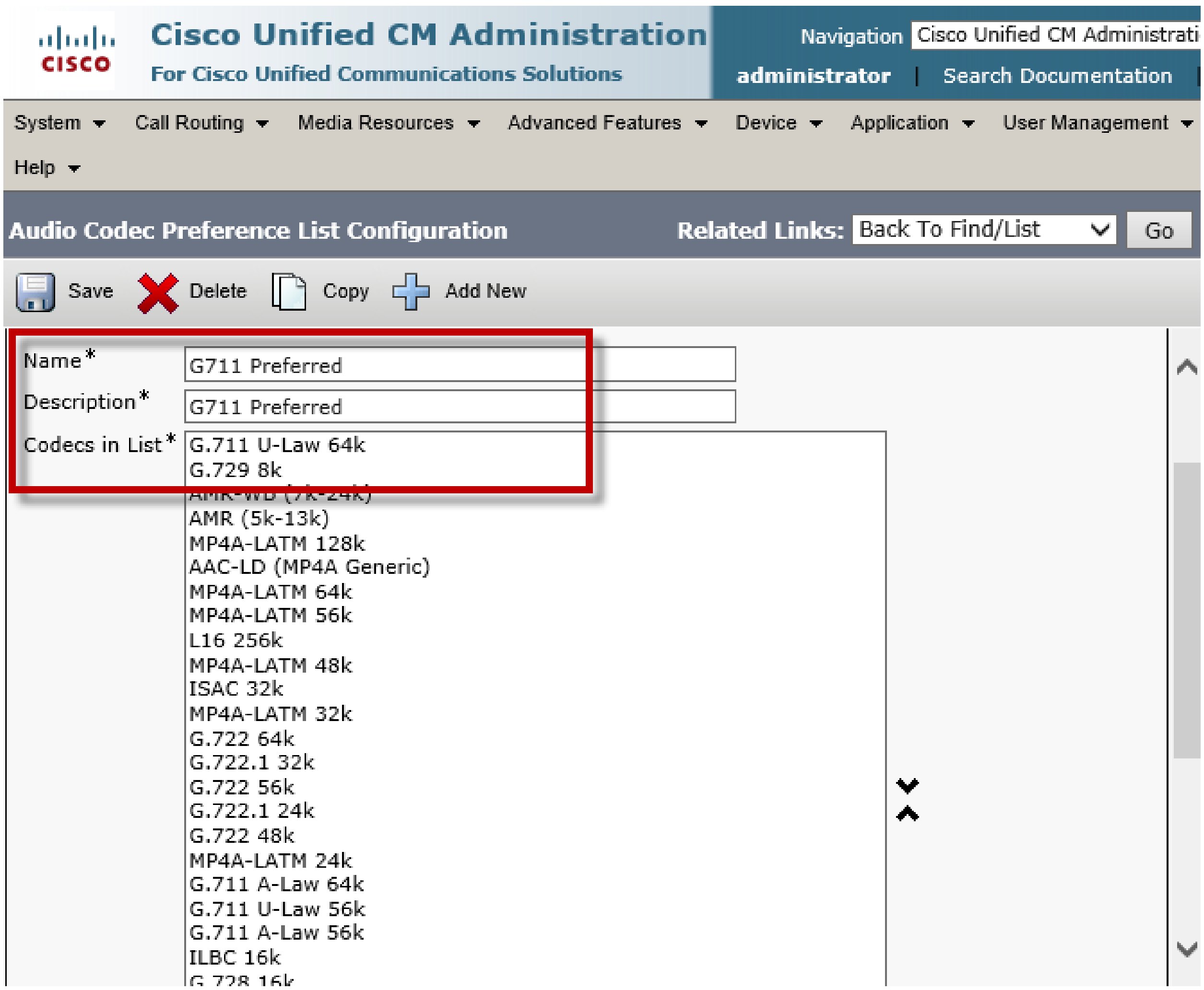
Set Name*= G711 Preferred. This is used for this example.

Set Description*= This text is used to identify this Audio Codec Preference List.

Set Codec in List*= G.711 U-Law 64k . First choice in this example.

Set Codec in List*= G.729 8k. Second choice in this example.

All other values are default.



Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

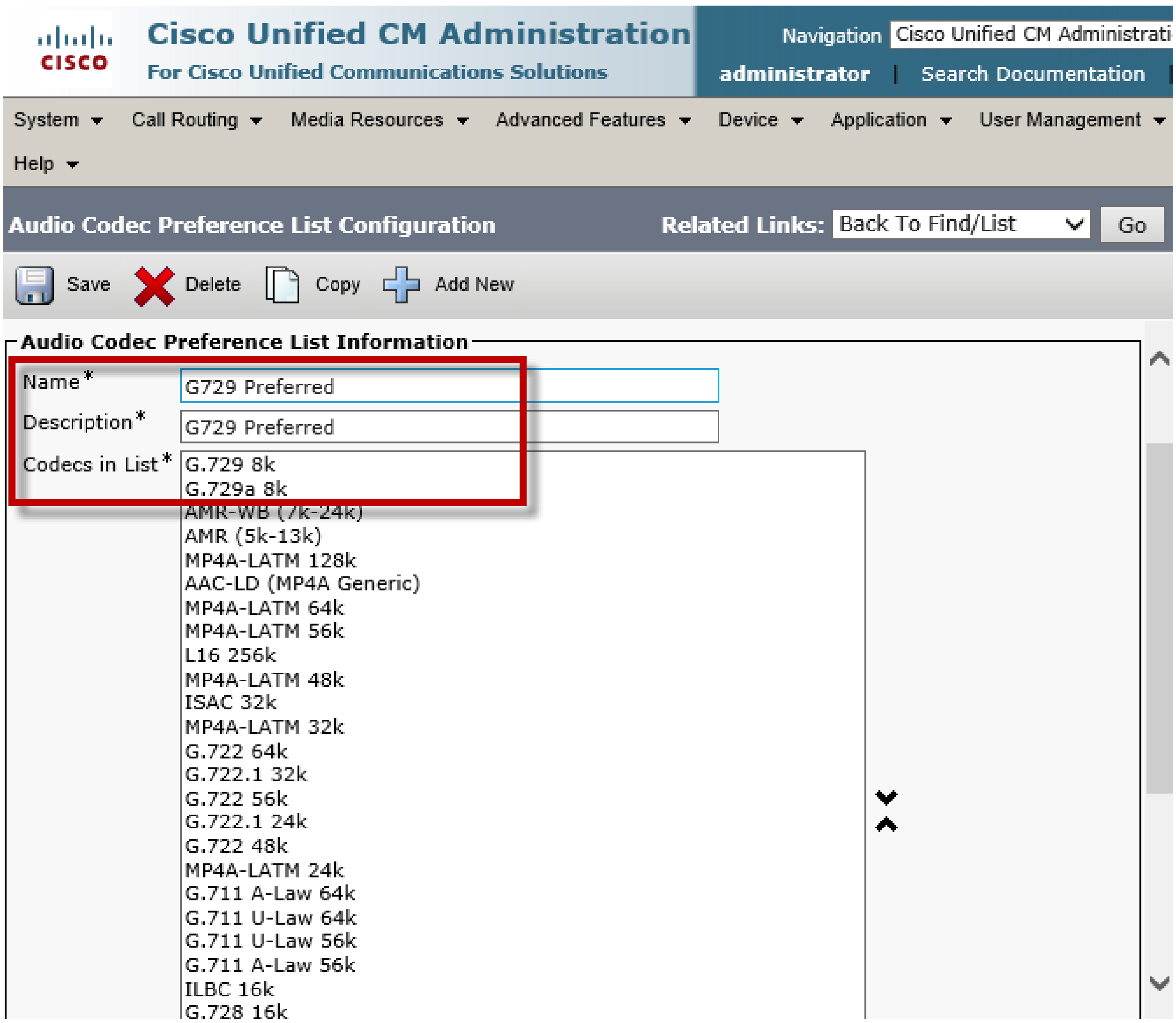
Set Name*= G729 Preferred. This is used for this example.

Set Description* = This text is used to identify this Audio Codec Preference List.

Set Codec in List*= G.729 8k. First choice for this example.

Set Codec in List*= G.729a 8k. Second choice for this example.

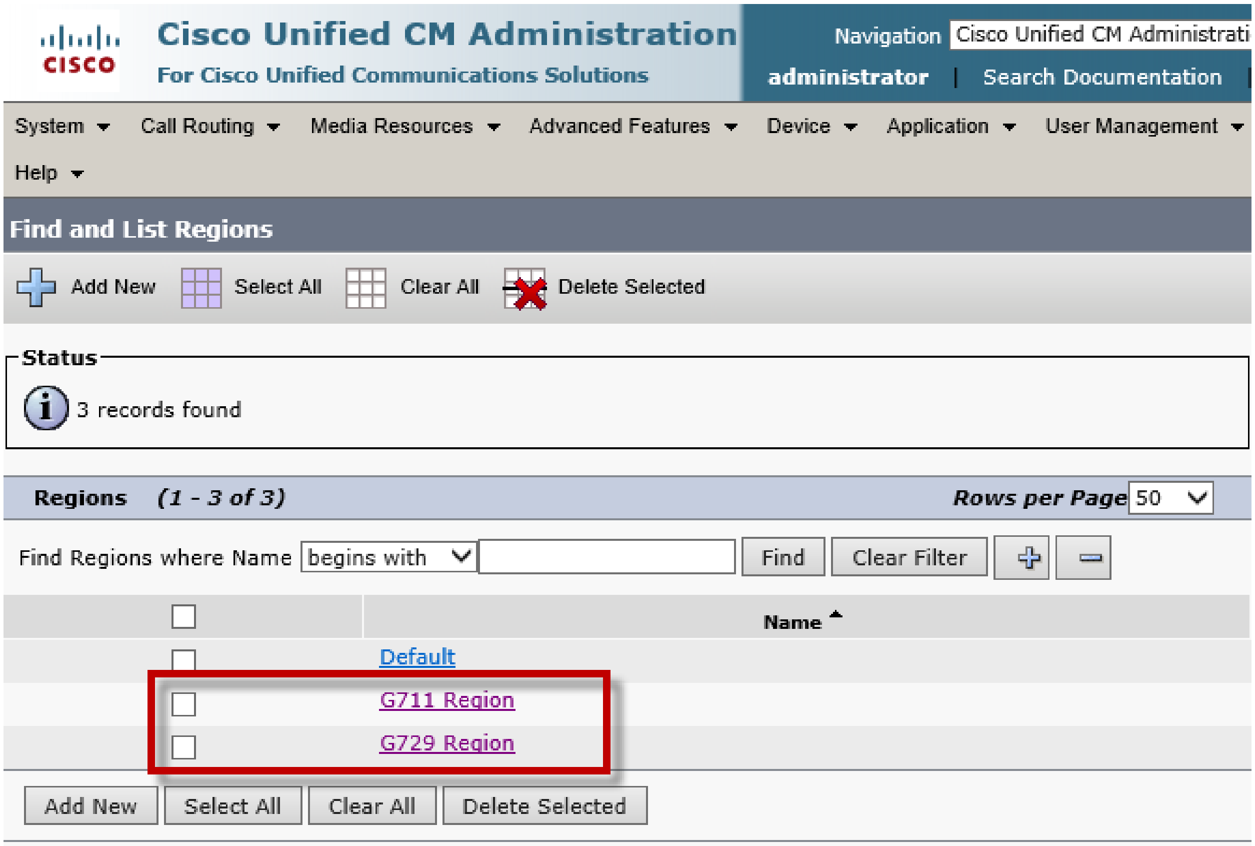
All other values are default.



Cisco Unified Communications Manager Region Configuration

G711 Region and G729 Region created in this example.

All other values are default.



Cisco Unified Communications Manager Region Configuration (Continued)

Set Name*= G711 Region. This is used in this example.

Set Region= G711 Region. This is used in this example

Set Audio Codec Preference List= G711 Preferred.

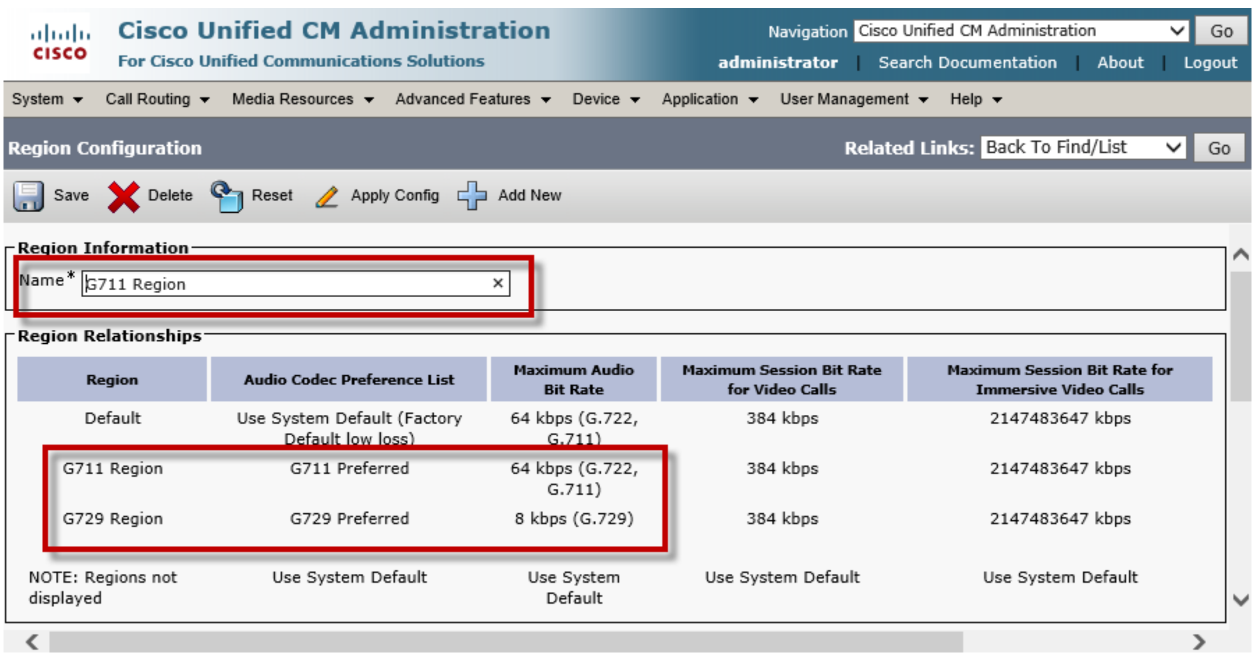
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.

Set Region=G729 Region. This is used in this example.

Set Audio Codec Preference List= G729 Preferred. This is used in this example

Set Maximum Audio Bit Rate= 8 Kbps (G7.29). This is used in this example

All other values are default.



Cisco Unified Communications Manager Region Configuration (Continued)

Set Name*= G729 Region. This is used in this example.

Set Region= G711 Region. This is used in this example

Set Audio Codec Preference List= G729 Preferred. This is used in this example

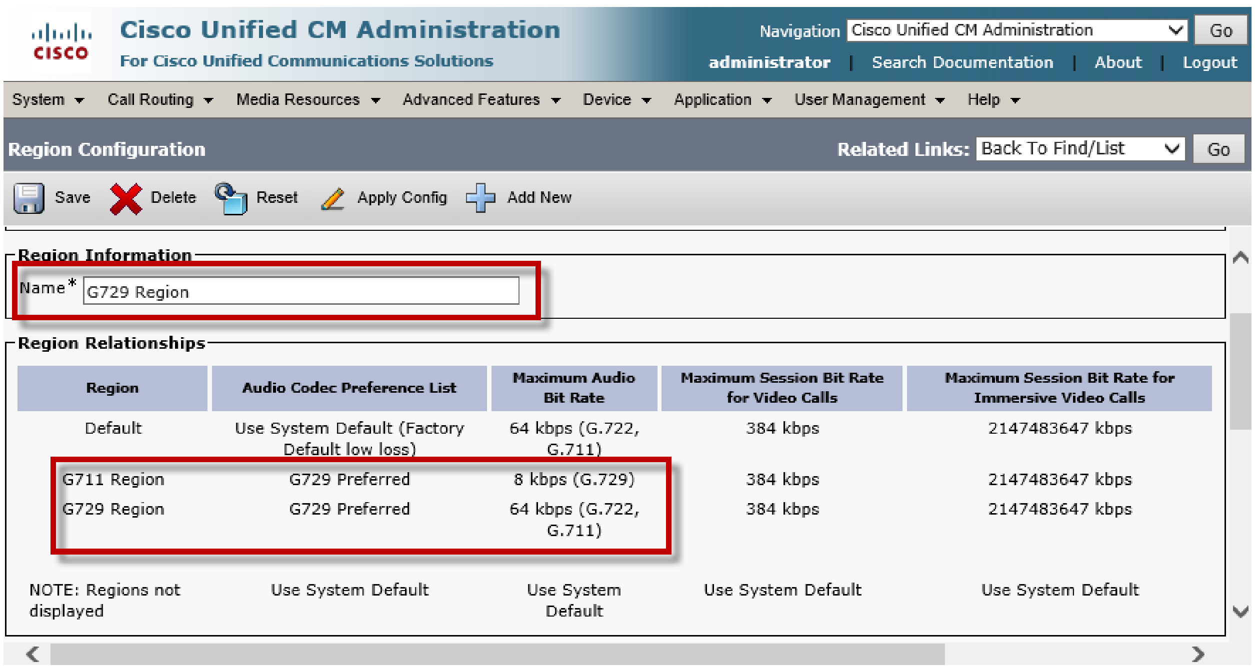
Set Maximum Audio Bit Rate= 8 Kbps (G.729). This is used in this example.

Set Region=G729 Region. This is used in this example.

Set Audio Codec Preference List= G729 Preferred. This is used in this example

Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example

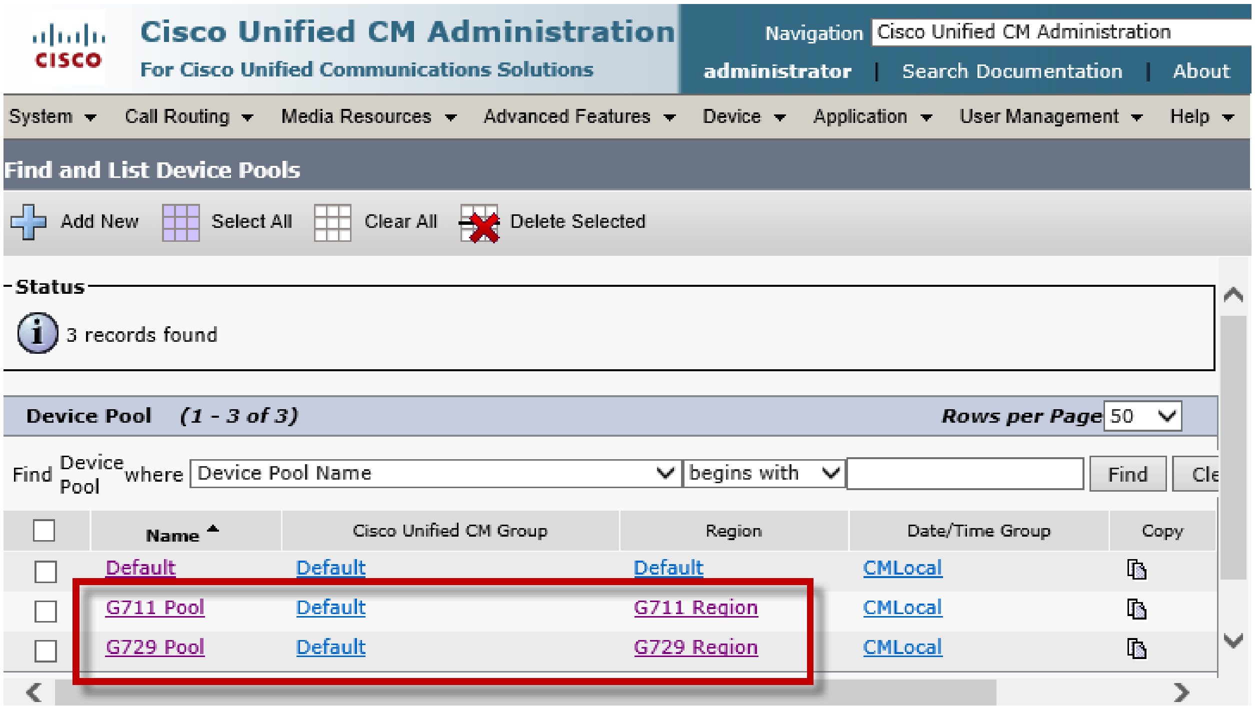
All other values are default.



Cisco Unified Communications Manager Device Pool Configuration

G711 Pool and G729 Pool created in this example.

All other values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G711 Pool. This is used in this example.

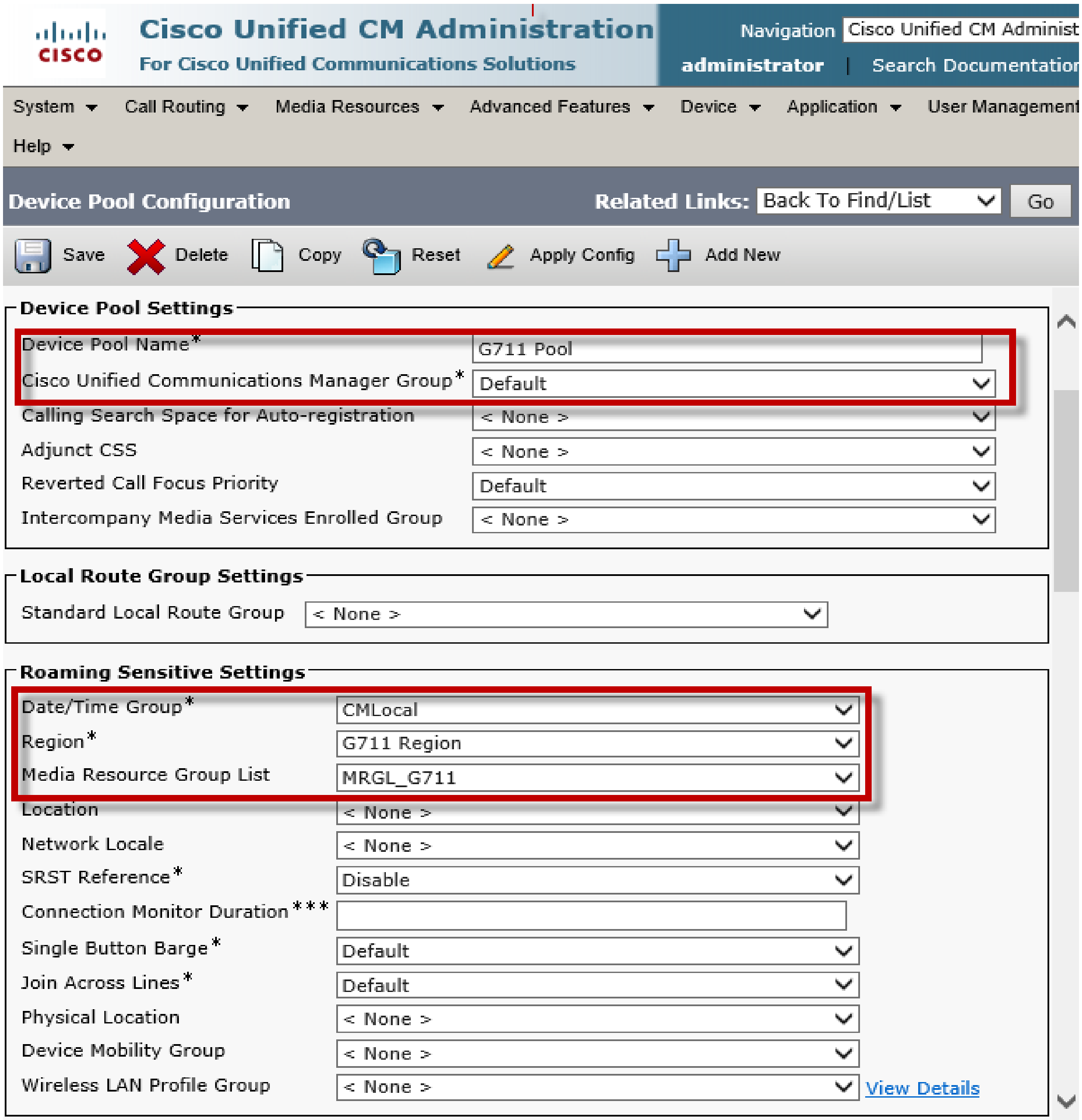
Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group* = CMLocal

Set Region* =G711 Region. This is used in this example

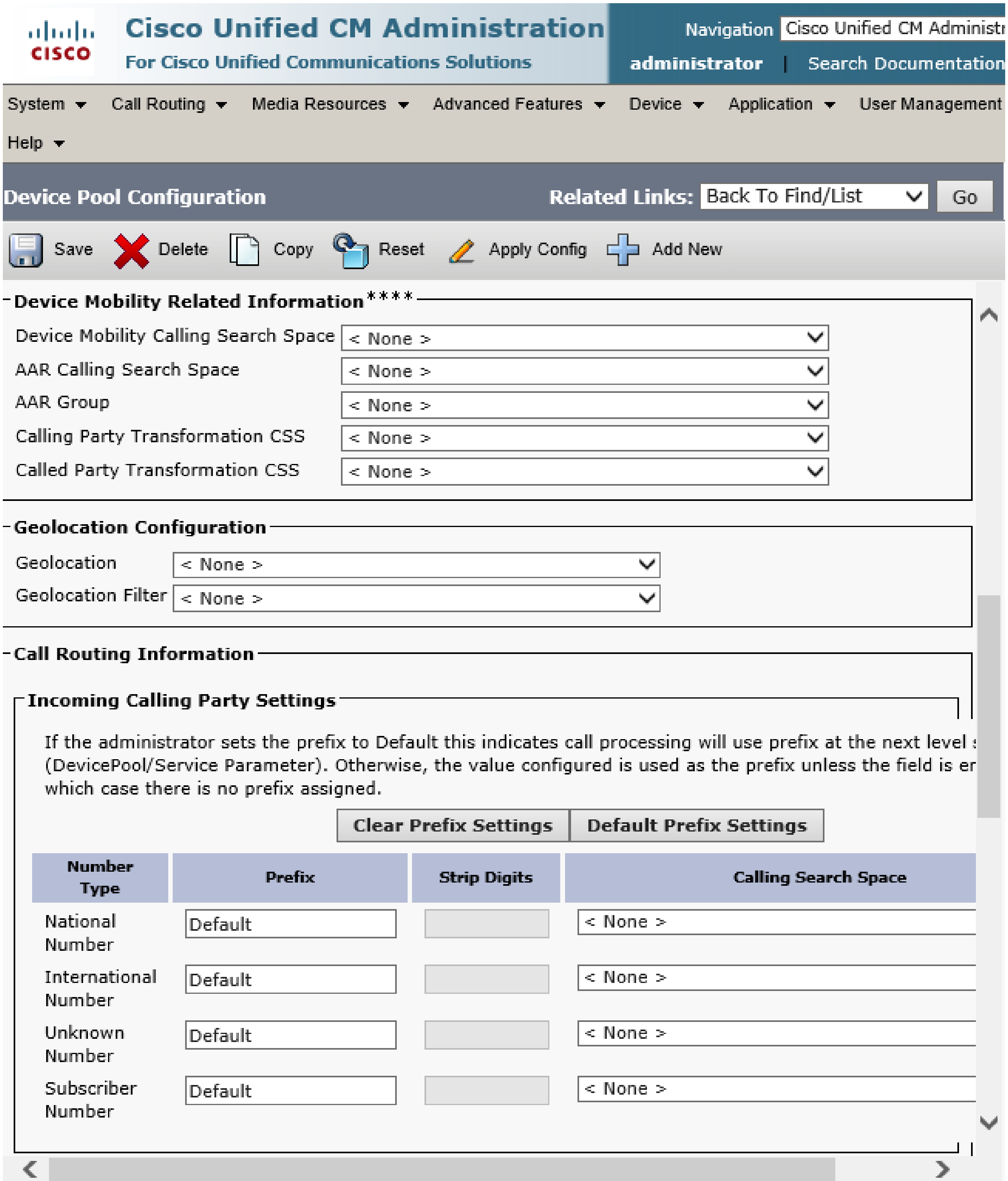
Set Media Resource Group List =MRGL_G711. This is used in this example.

All other values are default.



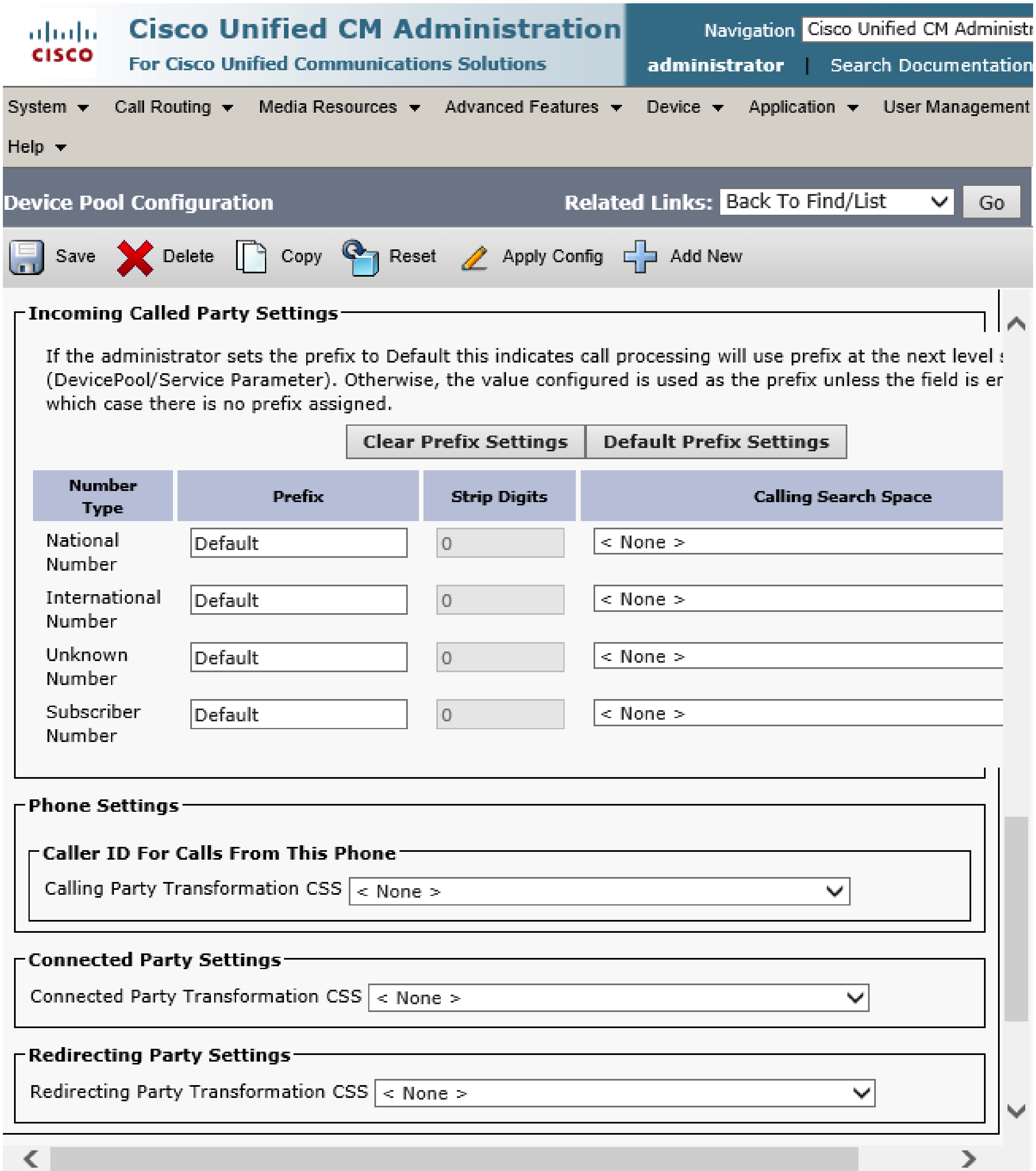
Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.


Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation
administrator
Search Documentation
System
Call Routing
Media Resources
Advanced Features
Device
Application
User Management
Help
Device Pool Configuration
Related Links:
Back To Find/List
Go
Save
Delete
Copy
Reset
Apply Config
Add New
Device Mobility Related Information****
Device Mobility Calling Search Space
< None >
AAR Calling Search Space
< None >
AAR Group
< None >
Calling Party Transformation CSS
< None >
Called Party Transformation CSS
< None >
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Call Routing Information
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level
(DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is
which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
National Number
Default
< None >
International Number
Default
< None >
Unknown Number
Default
< None >
Subscriber Number
Default
< None >

Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name*= G729 Pool. This is used in this example.

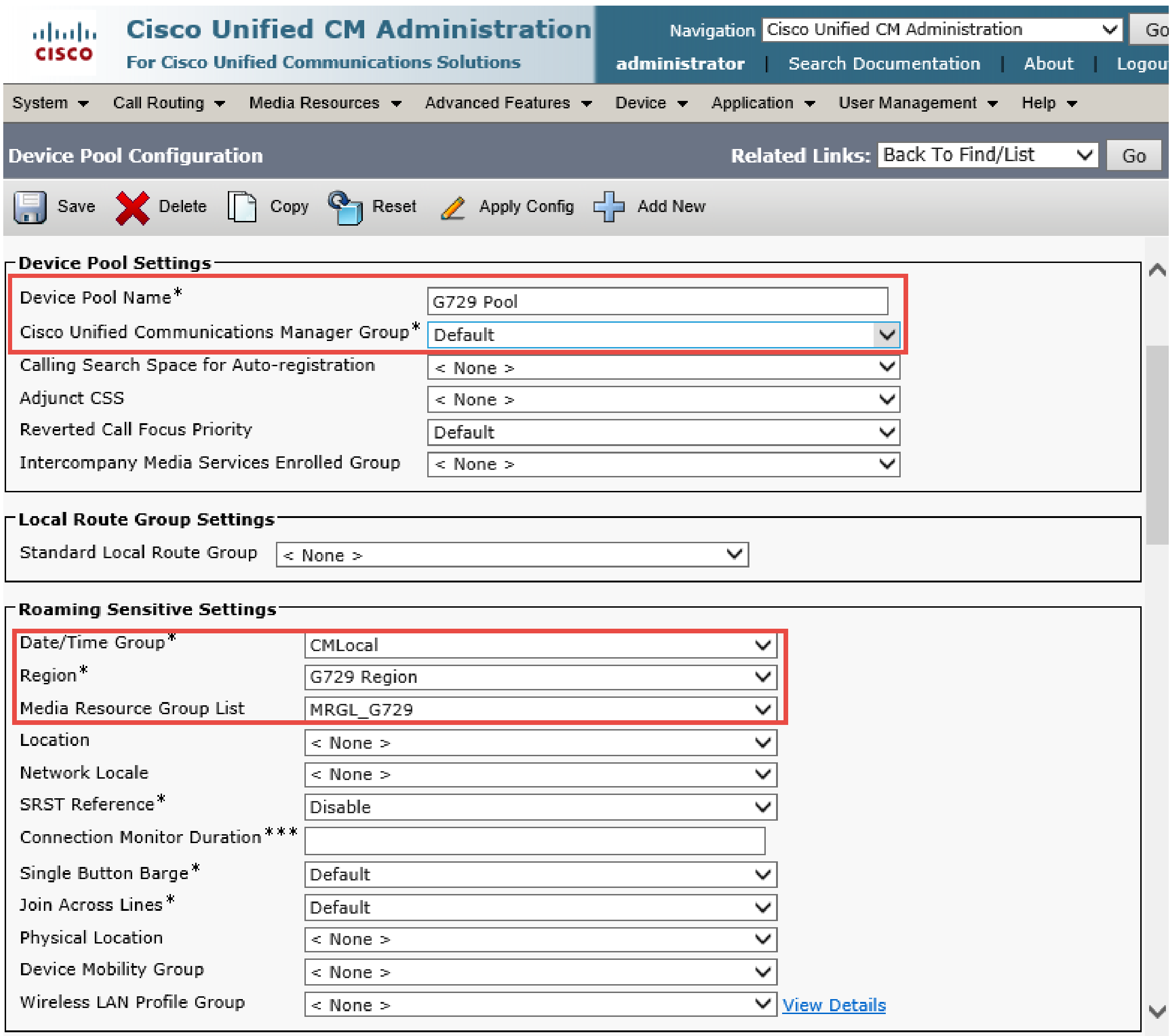
Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group* = CMLocal

Set Region* =G729 Region. This is used in this example

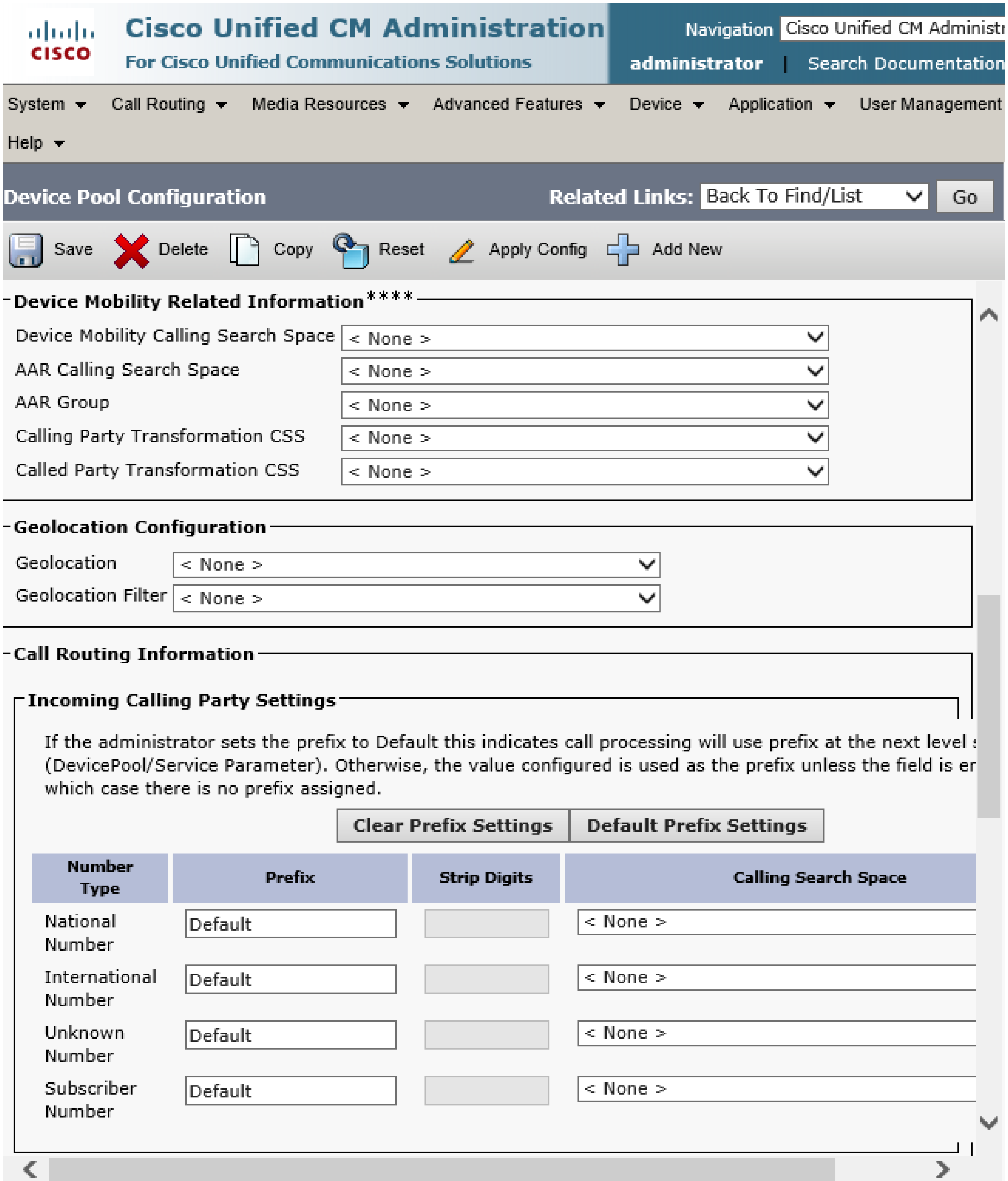
Set Media Resource Group List =MRGL_G729. This is used in this example.

All other values are default



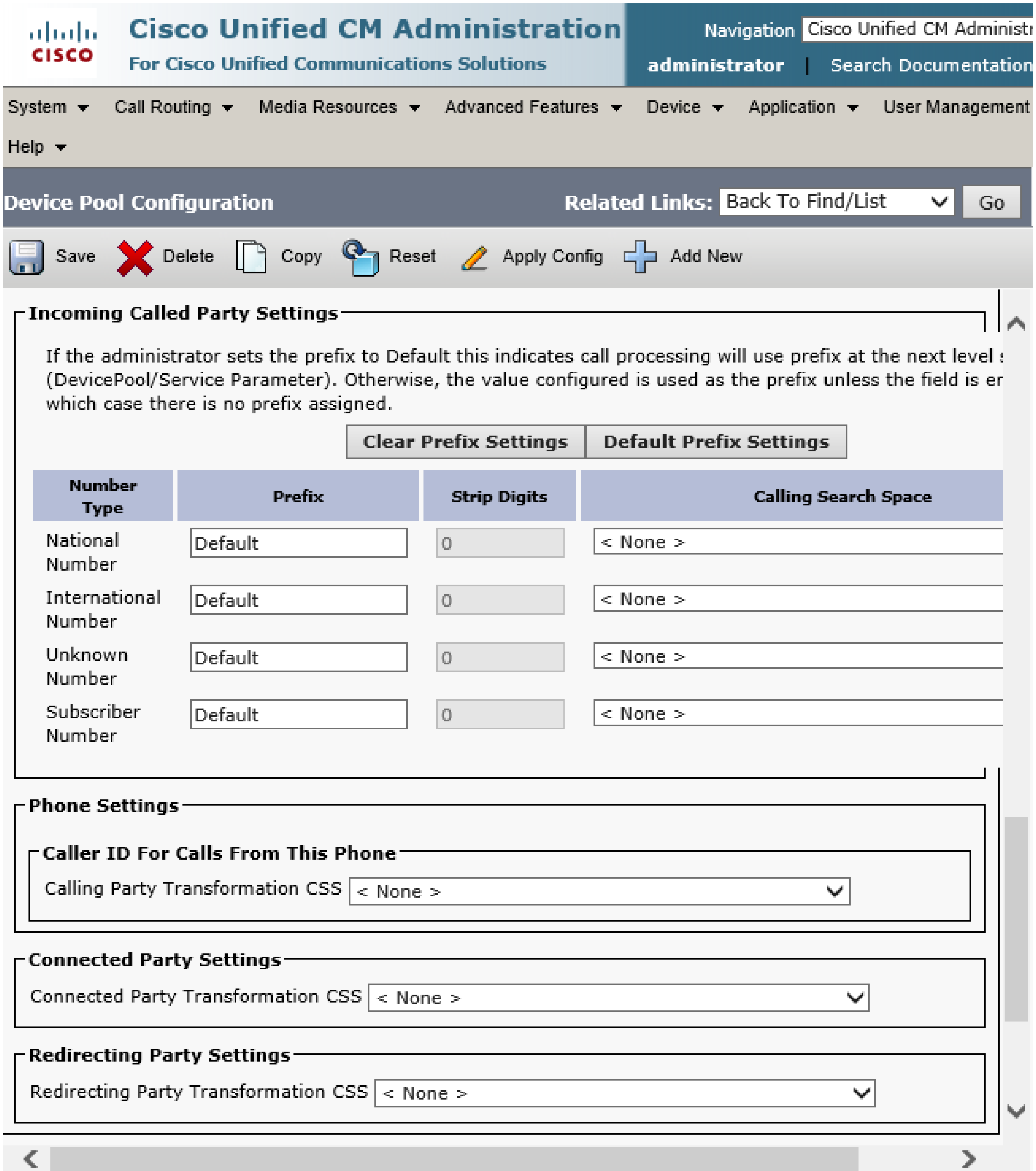
Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.



Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.



Acronyms

Acronym	Definition
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
Cisco UCM	Cisco Unified Communications Manager
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
MRGL	Media Resource Group List
MTP	Media Termination Point
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

Important Information

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