Application Note

Avaya S8500 Rel. 5.2.1 using SIP via Cisco Unified Communications Manager–Session Manager Edition 8.5(1) to Cisco Unified Communications Manager 8.5(1) and Cisco Unified Border Element (Enterprise Edition) Release 8 on ASR to Service Provider

February 3, 2011 – Initial Version

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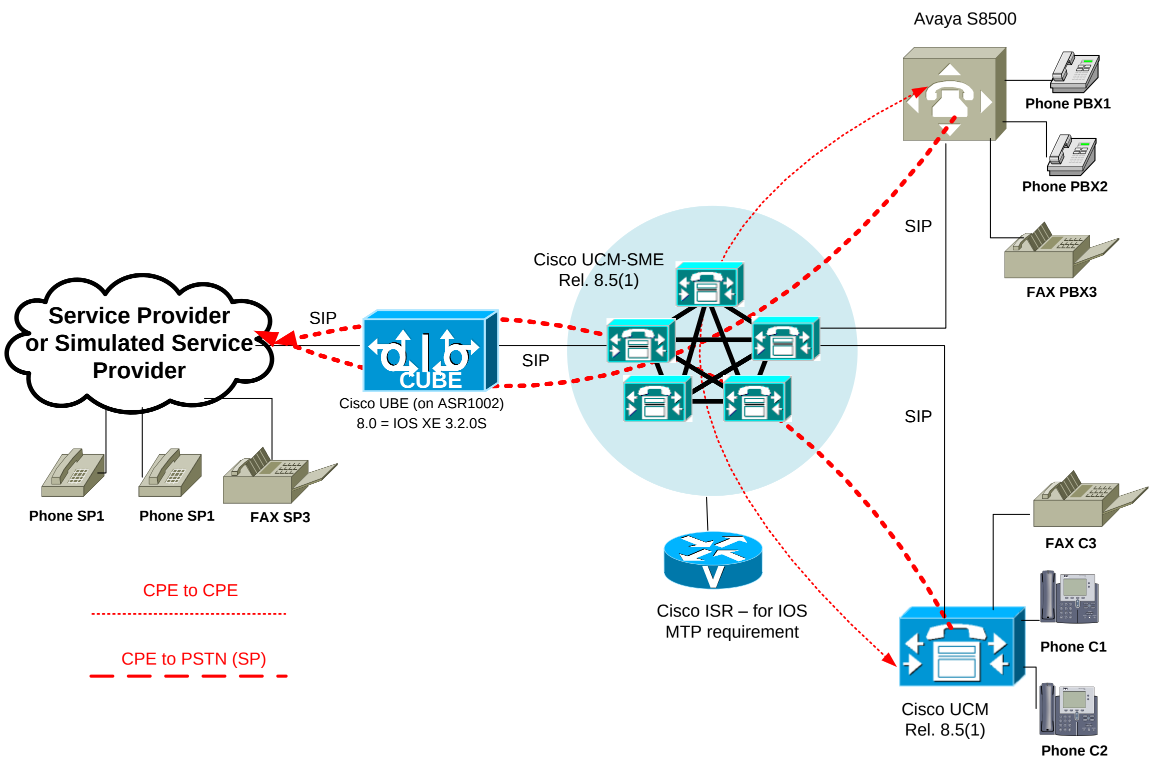
Introduction

- This application note describes the necessary steps and configurations for connectivity between Avaya S8500 release 5.2.1, and a Cisco Unified Communications Manager (Cisco UCM) version 8.5(1) with Cisco Unified Communications Manager-Session Management Edition (Cisco UCM-SME) Version 8.5(1).
- The network topology diagram (Figures 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager (Cisco UCM) Release 8.5(1) connected to the Avaya S8500 PBX via a Cisco UCM-SME using SIP trunks (between Cisco UCM-SME and Avaya PBX) and SIP trunks (between the Cisco UCM-SME and Cisco UCM). Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, fax transmission, and DTMF interworking. This test setup also includes a connection to a Service Provider (AT&T IP Flexible Reach), using SIP trunks. Cisco Unified Border Element (Cisco UBE) on ASR is used as a session border controller (SBC), providing demarcation, security, and interworking services between the customer's private network and the service provider's SIP network.
- During testing, a Cisco ASR1002 voice gateway was used to run the Cisco Unified Border Element features set. However other Cisco voice gateways can be used. The decision to choose the Cisco gateway model is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running Cisco UBE.
- Cisco 3900 Series Integrated Services Routers
- Cisco 2900 Series Integrated Services Routers
- Cisco 2800 Series Integrated Services Routers
- Cisco 3800 Series Integrated Services Routers
- Cisco AS5350XM Universal Gateway
- Cisco AS5400XM Universal Gateway
- Cisco ASR 1000 Series Aggregation Services Routers
- If additional guidance on the Cisco UBE is needed, please refer to the Cisco UBE section on the Cisco Interoperability Portal (www.cisco.com/go/interoperability).
- This configuration was tested using AT&T IP Flexible Reach SIP trunk service as the Service Provider. Results may vary based on Service Provider being used.

Network Topology

Figure 1. Basic Call Setup



Capabilities

- Voice/fax calls including supplementary services can be successfully established between endpoints controlled by the Avaya PBX and endpoints controlled by the Cisco Unified Communications Manager.
- Voice/fax calls including supplementary services can be successfully established between endpoints controlled by the Avaya PBX and the PSTN, using Cisco UBE on ASR as a session border controller.

Limitations

<u>Avaya PBX</u>

Centralized Avaya voicemail using QSIG integration to the Avaya PBX is not supported. SIP-to-QSIG interworking on the Avaya does not provide diversion information over the QSIG call leg. Centralized voicemail using Cisco Unity/Unity Connection integrated to Cisco UCM is supported, so long as Diversion header is passed to Cisco Unified Communications Manager. This can be achieved by enabling support of Diversion Header on the Avaya Communication Manager 5.0 SIP trunk group configuration form, or by using a SIP Normalization Script converting History-info headers into Diversion headers.

<u>Cisco UBE (Enterprise Edition)</u>

- Call scenarios involving inbound PSTN calls over SIP trunk, which are then early-attended transferred back out over the same SIP trunk to the PSTN, do not provide ringback tone to the caller. During testing, it was observed that Cisco UBE does not extend a 18X message towards the PSTN upon receiving it. Talk path is correctly established after the transfer target station is answered.

- Cisco ASR platforms currently do not officially support voice class codec configuration. Official support of that feature will be introduced with Cisco UBE Release 10.

Cisco UCM

- Cisco UCM does not natively support History-info headers. A SIP Normalization script must be applied to SIP trunk(s) to Avaya PBX in order to convert History-info (supported the Avaya PBX) to Diversion header (Cisco UCM-supported). This is required whenever a Cisco UCM-hosted centralized voicemail platforms (such as Unity Connection) is used.

System Components

Hardware Requirements

- Cisco MCS 7800 Unified Communications Manager Appliance
- 2 Cisco Unified IP phone 7960 configured as SCCP phones
- 2 Cisco Unified IP phone 7970 configured as SIP phones
- Avaya S8500 PBX
- Avaya digital and IP stations
- TN799DP C-LAN Circuit Pack
- TN2302AP Med-Pro Circuit Pack
- Cisco ASR 1000 (Cisco UBE-Enterprise)

Software Requirements

The following software is required:

- Cisco Unified Communications Manager Release 8.5(1) - Session Manager Edition
- Cisco Unified Communications Manager Release 8.5(1) – Cisco UCM
- Cisco IOS-XE Release 15.1(1)S (ASR 1000 used as Cisco UBE-Enterprise)
- Avaya Aura Communication Manager Release 5.2.1
- Avaya 9600 Series H.323 IP Phone firmware version 3.1.1
- Avaya TN799DP Firmware Vintage 39
- Avaya TN2302AP Firmware Vintage 121

Features

This section lists supported and unsupported features.

Features Supported

- Basic calls
- CLIP-Calling line (Number) identification presentation
- CLIR-Calling line (Number) identification restriction
- COLP-Connected line (Number) identification presentation
- COLR- Connected line (Number) identification restriction

- CNIP-Calling name identification presentation
- CNIR-Calling name identification restriction
- CONP-Connected name identification presentation
- CONR- Connected name identification restriction
- Consultation transfer – Local and Network/External
- Early Attended transfer – Local and Network/External
- Call forward Local – Unconditional, Busy and No reply (See Limitations section for details.)
- Call forward Network/External – Unconditional, Busy and No reply (See Limitations section for details.)
- DTMF interworking (using RFC 2833 DTMF relay)
- Fax transmissions (T.38 and G.711 pass-through)

Features Not Supported

- Centralized QSIG Voicemail hosted by Avaya PBX.

Configuration

This section contains configuration menus and commands and describes configuration sequences and tasks.

Configuring the Avaya S8500 PBX

1. Configure node-name IP table to include Cisco UCM-SME as a valid IP node.
2. Configure the ip-network-region to assign to the SIP trunk.
3. Configure the ip-codec-set to assign to ip-network-region used by the SIP trunk.
4. Add the new signaling group.
5. Add the new trunk group.
6. Add the new route pattern.
7. Configure AAR/ARS Table entries.
8. Configure Uniform Dialing Plan.
9. Configure ISDN Public/Unknown Numbering Table entry.

Software/Firmware Versions

```
list configuration software-versions

                              SOFTWARE VERSIONS

    SOFTWARE VERSION
       Memory Resident: R015x.02.1.016.4
         Disk Resident: R015x.02.1.016.4

    TRANSLATION DATE
       Memory Resident: 10:00 pm  TUE SEP 14, 2010
         Disk Resident: 10:00 pm  TUE SEP 14, 2010
      Disk Second Copy: good
```

```
list configuration carrier 1a

                              SYSTEM CONFIGURATION

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

01A00    POWER SUPPLY              655A
01A01    IP SERVER INTFC          TN2312BP HW06 FW050 01 02 03 04 05 06 07 08
01A02    CONTROL-LAN               TN799DP HW01 FW039 u  u  u  u  u  u  u  u
                                                      u  u  u  u  u  u  u  u
                                                      17
01A03    IP MEDIA PROCESSOR       TN2302AP HW20 FW121 01    03    05    07
01A04    DIGITAL LINE             TN2224B  000012     01 02 03 04 05 06 07 08
                                                      09 10 11 12 13 u  u  u
                                                      u  u  u  u  u  u  u  u
01A07    ANALOG LINE               TN746B  000002     01 02 03 04 05 06 07 u
                                                      u  u  u  u  u  u  u  u
```

System Parameters IP Options

```
display system-parameters ip-options                           Page   1 of   3
                          IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                    Packet Loss (%)     High: 40       Low: 15
                    Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10
                  Enable Voice/Network Stats? n
 RTCP MONITOR SERVER
         Default Server IP Address:    .   .   .
               Default Server Port: 5005
  Default RTCP Report Period(secs): 5

AUTOMATIC TRACE ROUTE ON
           Link Failure? y


 H.248 MEDIA GATEWAY                  H.323 IP ENDPOINT
  Link Loss Delay Timer (min): 5        Link Loss Delay Timer (min): 5
                                          Primary Search Time (sec): 75
                                  Periodic Registration Timer (min): 20
```

```
display system-parameters ip-options                           Page   2 of   3
                          IP-OPTIONS SYSTEM PARAMETERS

 Always use G.711 (30ms, no SS) for intra-switch Music-On-Hold? n
                      Force Phones and Gateways to Active LSPs? n

 IP DTMF TRANSMISSION MODE
   Intra-System IP DTMF Transmission Mode: rtp-payload
                     Inter-System IP DTMF: See Signaling Group Forms

 HYPERACTIVE MEDIA GATEWAY REGISTRATIONS
   Enable Detection and Alarms? n
```

```
display system-parameters ip-options                          Page   3 of   3
                           IP-OPTIONS SYSTEM PARAMETERS

 SNMP PARAMETERS
    Download Flag? n
    Community String:

 SOURCE ADDRESSES
        1.                                   4.
        2.                                   5.
        3.                                   6.


 SERVICES DIAL PAD PARAMETERS

        Download Flag? n
        Password: *
```

IP Nodes

```
list node-names all                                                   Page   1

                         NODE NAMES

Type      Name               IP Address
IP        ACME-SBC           172.20.110.105
IP        ATT_CUBE_ASR       172.20.110.154
IP        CCM4.1             172.20.231.254
IP        CM-EUROPA          172.20.236.254
IP        CM-NEPTUNE         172.20.236.2
IP        CM-POLARIS         172.20.236.50
IP        CM-TITAN           172.20.236.252
IP        CM-VANLAB01        172.20.170.254
IP        CME_bench_8        172.20.236.249
IP        CUCMExpress        172.20.228.254
IP        Carole_SME         172.20.150.141
IP        CecilyGW           172.20.174.30
IP        Gateway001         172.20.212.1
IP        Mubarik_UCM        10.105.1.1
IP        Nortel_CS104       172.30.11.100
IP        Sarath_CUBE        172.20.109.203
```

```
Type      Name               IP Address
IP        TACIPIPGW          172.16.243.218
IP        TFTP               172.20.2.181
IP        TonyB-CUBE         172.20.8.103
IP        TonyBGW            172.20.8.26
IP        VERIZON_SBC        172.20.110.152
IP        VERIZON_SBC_ASR    172.20.110.151
IP        avayasip1          172.20.212.254
IP        clan1              172.20.212.253
IP        clan1serverb       172.20.213.253
IP        default            0.0.0.0
IP        medpro1            172.20.212.252
IP        procr              172.20.212.200
IP        sarat_sess_mgr     172.20.109.252
```

IP Network Region

```
display ip-network-region 1                                          Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: lab.com
    Name: CiscoLAB
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                   RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

```
display ip-network-region 1                                     Page   2 of  19
                               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)    H.323 SECURITY PROFILES
 1                                    1    challenge
 2                                    2
 3                                    3
 4                                    4
 5
 6                                    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
```

```
display ip-network-region 1                                     Page   3 of  19

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

IP Codec Set

```
display ip-codec-set 3                                          Page   1 of   2

                           IP Codec Set

    Codec Set: 3

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A           n          2         20
 2: G.711MU          n          2         20
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

```
display ip-codec-set 3                                          Page   2 of   2

                           IP Codec Set

                               Allow Direct-IP Multimedia? n



                Mode                Redundancy
    FAX         t.38-standard        0
    Modem       off                  0
    TDD/TTY     off                  0
    Clear-channel  n                 0
```

Note: The ip-codec-set configuration above is assigned to SIP trunk(s) using codecs G.729a, G.711Mulaw, and T.38 fax relay. If the Service Provider does not support T.38 fax relay, a trunk using G.711 codec is required, with FAX Mode set to "off".

Signaling Group

```
display signaling-group 445
                              SIGNALING GROUP

 Group Number: 445             Group Type: sip
                         Transport Method: tcp
  IMS Enabled? n







   Near-end Node Name: clan1                   Far-end Node Name: CM-TITAN
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 1
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? y
         Enable Layer 3 Test? n                      Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Trunk Group

```
display trunk-group 445                                        Page   1 of  21
                              TRUNK GROUP

Group Number: 445                  Group Type: sip          CDR Reports: y
  Group Name: SIP trunk to CM-TITAN       COR: 1        TN: 1        TAC: 817
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n

                                                       Signaling Group: 445
                                                     Number of Members: 4
```

```
display trunk-group 445                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                Digital Loss Group: 18
                       Preferred Minimum Session Refresh Interval(sec): 600
```

```
display trunk-group 445                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                           Maintenance Tests? y


                     Numbering Format: public
                                                  UUI Treatment: service-provider

                                                   Replace Restricted Numbers? y
                                                  Replace Unavailable Numbers? y






 Show ANSWERED BY on Display? n
```

```
display trunk-group 445                                          Page   4 of  21
                                PROTOCOL VARIATIONS

                          Mark Users as Phone? n
                  Prepend '+' to Calling Number? n
            Send Transferring Party Information? n

                        Send Diversion Header? y
                      Support Request History? y
                 Telephone Event Payload Type: 101
```

```
display trunk-group 445                                          Page   5 of  21
                                   TRUNK GROUP
                                       Administered Members (min/max):    1/4
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:    4

      Port             Name
  1: T00118            SIP trunk
  2: T00119            SIP trunk
  3: T00120            SIP trunk
  4: T00121            SIP trunk
  5:
  6:
  7:
  8:
  9:
 10:
 11:
 12:
 13:
 14:
 15:
```

Route Pattern

```
display route-pattern 445                                       Page   1 of   3
                    Pattern Number: 445 Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 445  1                    3                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                          Subaddress
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

Note: The Route Pattern configuration above is assigned to AAR code 217, used to route calls to Cisco UCM's 4-digit extensions. As you can notice, route pattern 445 "strips" the first 3 digits of the routing number (AAR code 217) before the INVITE message is transmitted over the SIP trunk.

```
display route-pattern 446                                       Page   1 of   3
                    Pattern Number: 446 Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 445  1                    0                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                          Subaddress
 1: y y y y y n  n            rest                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

Note: The Route Pattern configuration above is assigned to ARS analysis table entry 1408, used to route calls to the SP using 1+10-digit dialing. As you can notice, route pattern 446 does not perform any "digit stripping" before the INVITE message is transmitted over the SIP trunk, thus using all 11 digits.

AAR/ARS Analysis

```
display aar analysis 217                                        Page   1 of   2
                           AAR DIGIT ANALYSIS TABLE
                                 Location: all           Percent Full:    2

          Dialed             Total    Route    Call   Node   ANI
          String            Min  Max  Pattern  Type   Num    Reqd
      217                   7    7    445      aar           n
      218                   7    7    18       aar           n
      219                   7    7    15       aar           n
      221                   7    7    11       aar           n
      222                   7    7    21       aar           n
      224                   7    7    14       aar           n
      225                   4    4    13       aar           n
      226                   7    7    13       aar           n
      227                   7    7    21       aar           n
      228                   7    7    44       aar           n
      229                   7    7    7        aar           n
      255                   7    7    555      aar           n
      256                   7    7    556      aar           n
      257                   7    7    557      aar           n
      3                     7    7    999      aar           n
```

Note: AAR code 217 is used to route calls to Cisco UCM extensions over Route Pattern 445. This code will be used in the Uniform Dialplan table, and will be prefixed onto the 4-digit numbers assigned to Cisco UCM stations. As stated previously, Route Pattern 445 will "strip" the AAR code, leaving only the dialed 4-digit number in the outbound INVITE message.

display ars analysis 1408 Page 1 of 2

ARS DIGIT ANALYSIS TABLE

Location: all Percent Full: 2

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
1408	11	11	446	natl		n
140812352	11	11	111	natl		n
1408123804	11	11	111	natl		n
1408123806	11	11	24	natl		n
1408525	11	11	12	natl		n
1408526	11	11	555	fnpa		n
1408527	11	11	12	natl		n
1408555	11	11	14	natl		n
1408577	11	11	446	natl		n
141	11	11	deny	fnpa		n
1415222	11	11	24	natl		n
142	11	11	deny	fnpa		n
143	11	11	deny	fnpa		n
144	11	11	deny	fnpa		n
145	11	11	deny	fnpa		n

Note: Dial String 1408 is used to route calls to the SP. After dialing 9 (ARS access code) plus 1408-nxx-xxxx, all 11 digits are included in the outbound INVITE message (parameter "Call Type" must be set to "natl") and the call is routed over Route Pattern 446. This Route Pattern is configured not to "strip" any leading digits.

Uniform Dialing Plan

display uniform-dialplan 5013 Page 1 of 2

UNIFORM DIAL PLAN TABLE

Percent Full: 0

Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
5013	4	0	217	aar	n	
5014	4	0	217	aar	n	
5015	4	0	217	aar	n	
5017	4	0	228	aar	n	
5018	4	0	217	aar	n	
5019	4	0	217	aar	n	
5050	4	0	213	aar	n	
5051	4	0	226	aar	n	
506	4	0	211	aar	n	
5060	4	0	228	aar	n	
508	4	0	217	aar	n	
5081	4	0	217	aar	n	
5082	4	0	217	aar	n	
5085	4	0	217	aar	n	
5088	4	0	217	aar	n	
5099	4	0	224	aar	n	

ISDN Public/Unknown Numbering Plan

```
display public-unknown-numbering 1                              Page   1 of   2
                        NUMBERING - PUBLIC/UNKNOWN FORMAT
                                            Total
Ext Ext            Trk       CPN            CPN
Len Code           Grp(s)    Prefix         Len
                                                     Total Administered: 17
 4  2                                        4          Maximum Entries: 9999
 4  3                                        4
 4  4                                        4
 4  5                                        4
 4  6                                        4
 4  7                                        4
 4  4001           1-5       650414          10
 4  4004           1         650414          10
 4  4114           112       4089336168      10
 4  4114           445       7323204084      10
 4  4123           12        4089336169      10
 4  4123           112       4089336169      10
 4  4124           12        4089336170      10
 4  4124           445       7323204084      10
 4  4149           12        408933          10
```

Note: The table above is used to define numbering plans to be used on ISDN/SIP calls. In the example above, 4-digit extensions in the 5XXX range are used on Cisco UCM, while 4-digit extensions in the 4XXX range are used by the Avaya PBX. Also note the digit transformation performed for extensions 4114 and 4124: when calls are sent over trunk group 445 (trunk group used to connect the PBX to Cisco UCM-SME), ext. 4124 is "transformed" into phone number 732-320-4084 (AT&T IP Flexible Reach DID number).

Station Configuration

```
display station 4124                                            Page   1 of   5
                                    STATION

Extension: 4124                         Lock Messages? n                BCC: 0
     Type: 9630                         Security Code: *                 TN: 1
     Port: S00029                     Coverage Path 1:                  COR: 1
     Name: 9630 test phone            Coverage Path 2:                  COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19      Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 4124
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n



                                             Customizable Labels? y
```

```
display station 4124                                            Page   2 of   5
                                    STATION
FEATURE OPTIONS
           LWC Reception: spe          Auto Select Any Idle Appearance? n
          LWC Activation? y                     Coverage Msg Retrieval? y
  LWC Log External Calls? n                                Auto Answer: none
             CDR Privacy? n                           Data Restriction? n
   Redirect Notification? y                 Idle Appearance Preference? n
 Per Button Ring Control? n               Bridged Idle Line Preference? n
   Bridged Call Alerting? n                   Restrict Last Appearance? y
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n           Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                     EC500 State: disabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
     MWI Served User Type:                      Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? y
                                                   Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 4124         Always Use? n IP Audio Hairpinning? y
```

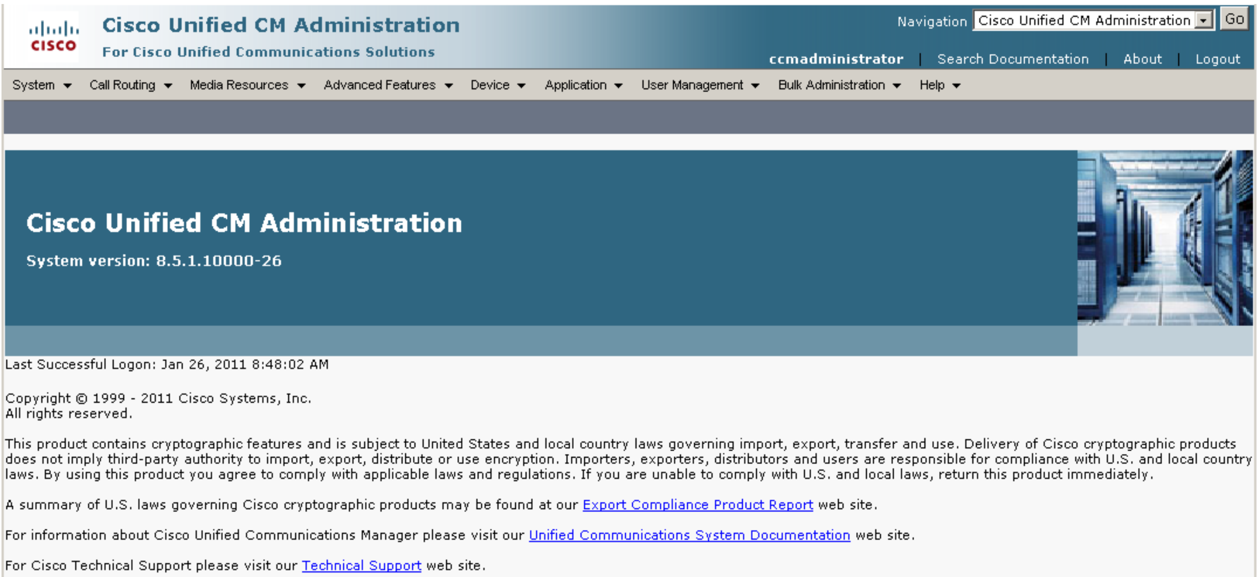
```
display station 4124                                            Page   3 of   5
                                     STATION

          Conf/Trans on Primary Appearance? n
Bridged Appearance Origination Restriction? n


       Call Appearance Display Format: disp-param-default
IP Phone Group ID:

                               ENHANCED CALL FORWARDING
                                        Forwarded Destination         Active
 Unconditional For Internal Calls To:                                    n
                   External Calls To:                                    n
          Busy For Internal Calls To:                                    n
                   External Calls To:                                    n
      No Reply For Internal Calls To:                                    n
                   External Calls To:                                    n

            SAC/CF Override: n
```

```
display station 4124                                            Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                             Headset? n
       Jack:                                             Speaker? n
      Cable:                                            Mounting: d
      Floor:                                         Cord Length: 0
   Building:                                           Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:



BUTTON ASSIGNMENTS
 1: call-appr                            5: auto-cback
 2: call-appr                            6: call-park
 3: call-fwd    Ext:                     7:
 4: cfwd-bsyda Ext:                      8:

    voice-mail Number:
```

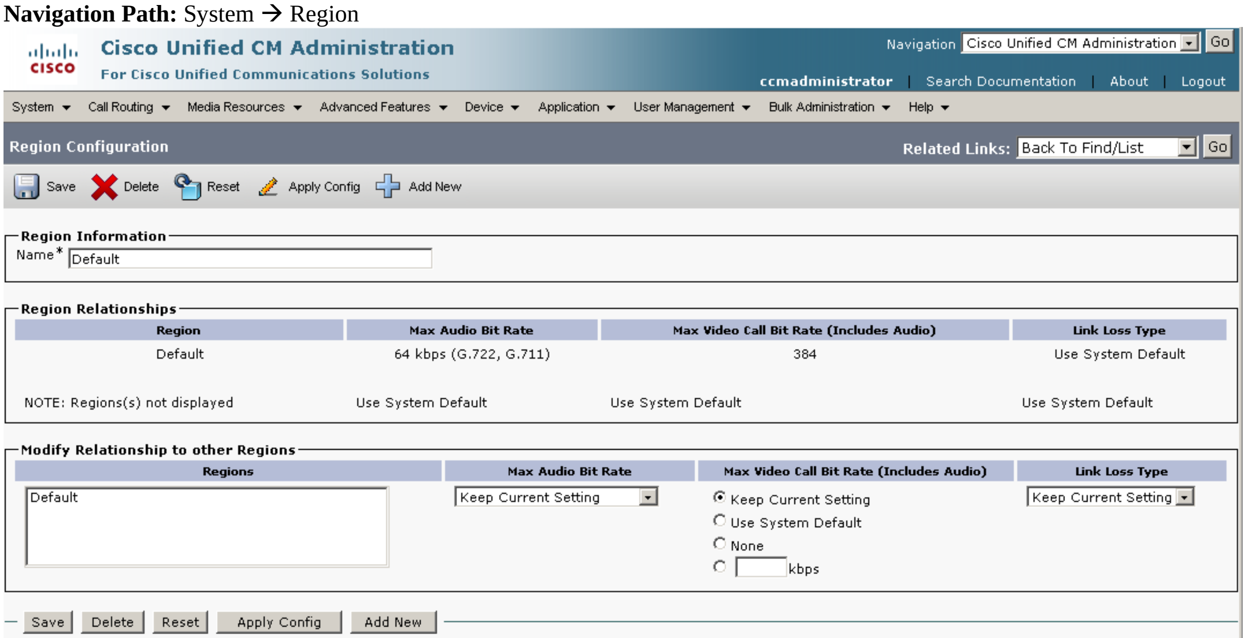
Configuring the Cisco Unified Communications Manager – Session Manager Edition

1. Cisco Session Manager Version
2. Device Pool and Region mapping configuration
3. SIP profile (used by SIP trunks) configuration
4. SIP Normalization Script (used by SIP trunk to Avaya PBX) configuration
5. SIP trunk configuration to SP
6. SIP trunk configuration to Avaya PBX
7. SIP Trunk configuration to Cisco UCM
8. Route Pattern configuration to Avaya PBX
9. Route Pattern configuration to Cisco UCM
10. Route Pattern configuration to SP

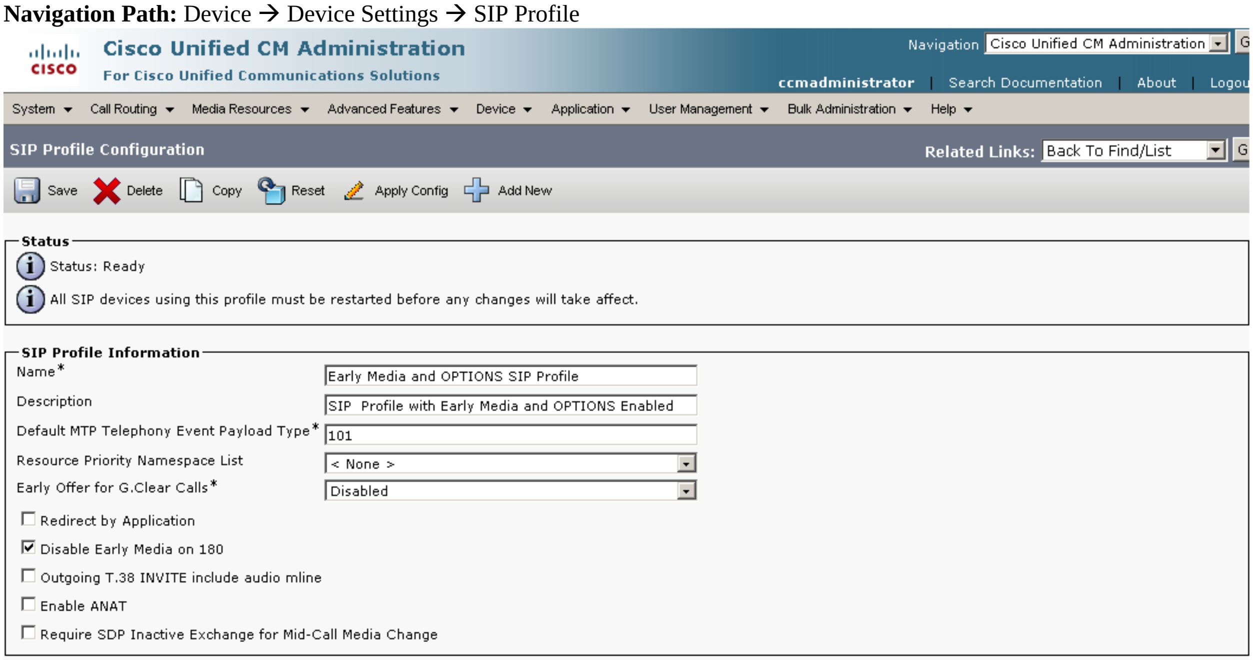
Cisco Unified Communications Manager – Session Manager Edition software version



Configuration of Device Pool to Region mapping
Navigation Path: System → Region



Configuration of SIP Profile used by SIP trunks
Navigation Path: Device → Device Settings → SIP Profile



Parameters used in Phone

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

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

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP

- [x] Fall back to local RSVP

SIP Rel1XX Options*	Disabled

- [ ] Deliver Conference Bridge Identifier
- [x] Early Offer support for voice and video calls (insert MTP if needed)
- [x] Send send-receive SDP in mid-call INVITE

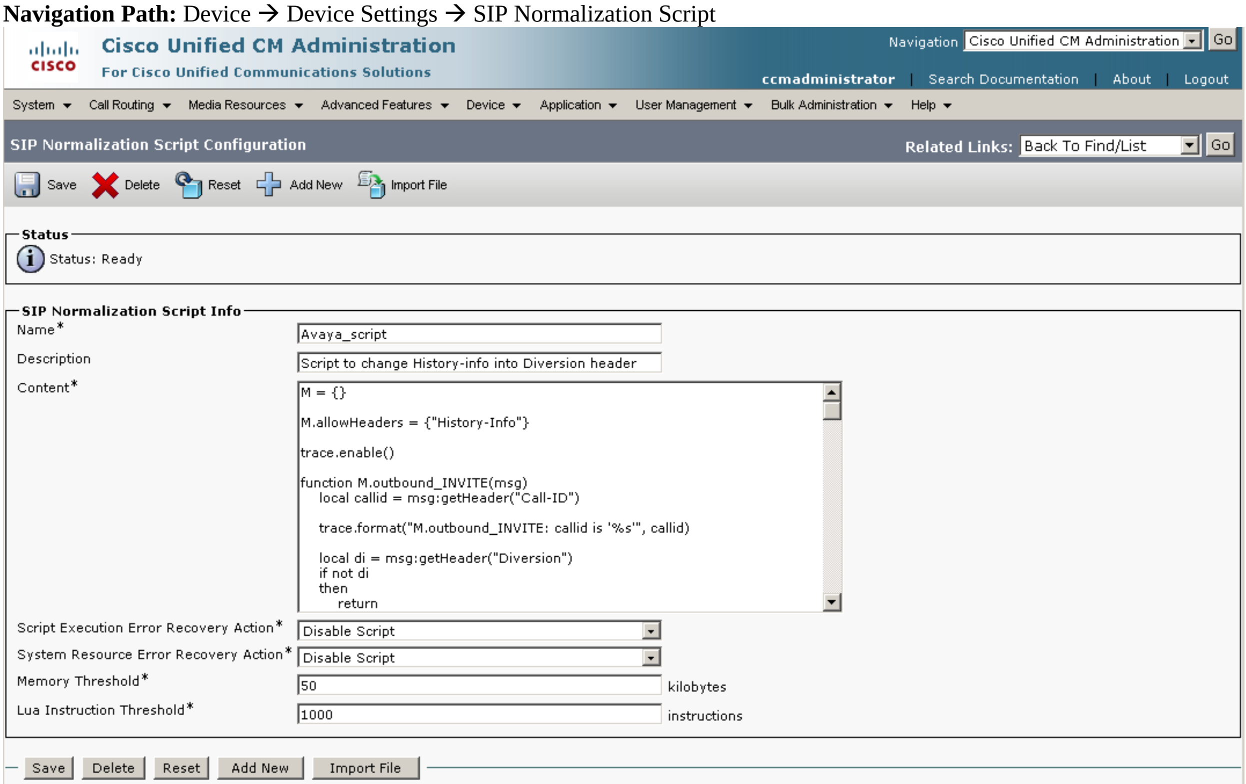
SIP OPTIONS Ping

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

Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

Save | Delete | Copy | Reset | Apply Config | Add New

Configuration of SIP Normalization Script (used by SIP trunk to Avaya PBX)
Navigation Path: Device → Device Settings → SIP Normalization Script



Note: Avaya supports SIP History-Info header when providing call forward (diversion) information over SIP trunks. Cisco Unified Communications Manager normally uses Diversion header. Because of this, a SIP Normalization script is required in order to convert History-Info headers into Diversion headers. This is useful whenever Cisco Unity/Unity Connection centralized voicemail (integrated with Cisco Unified Communications Manager) is used to support both Avaya and Cisco end users. The full content of the SIP Normalization Script is captured below:

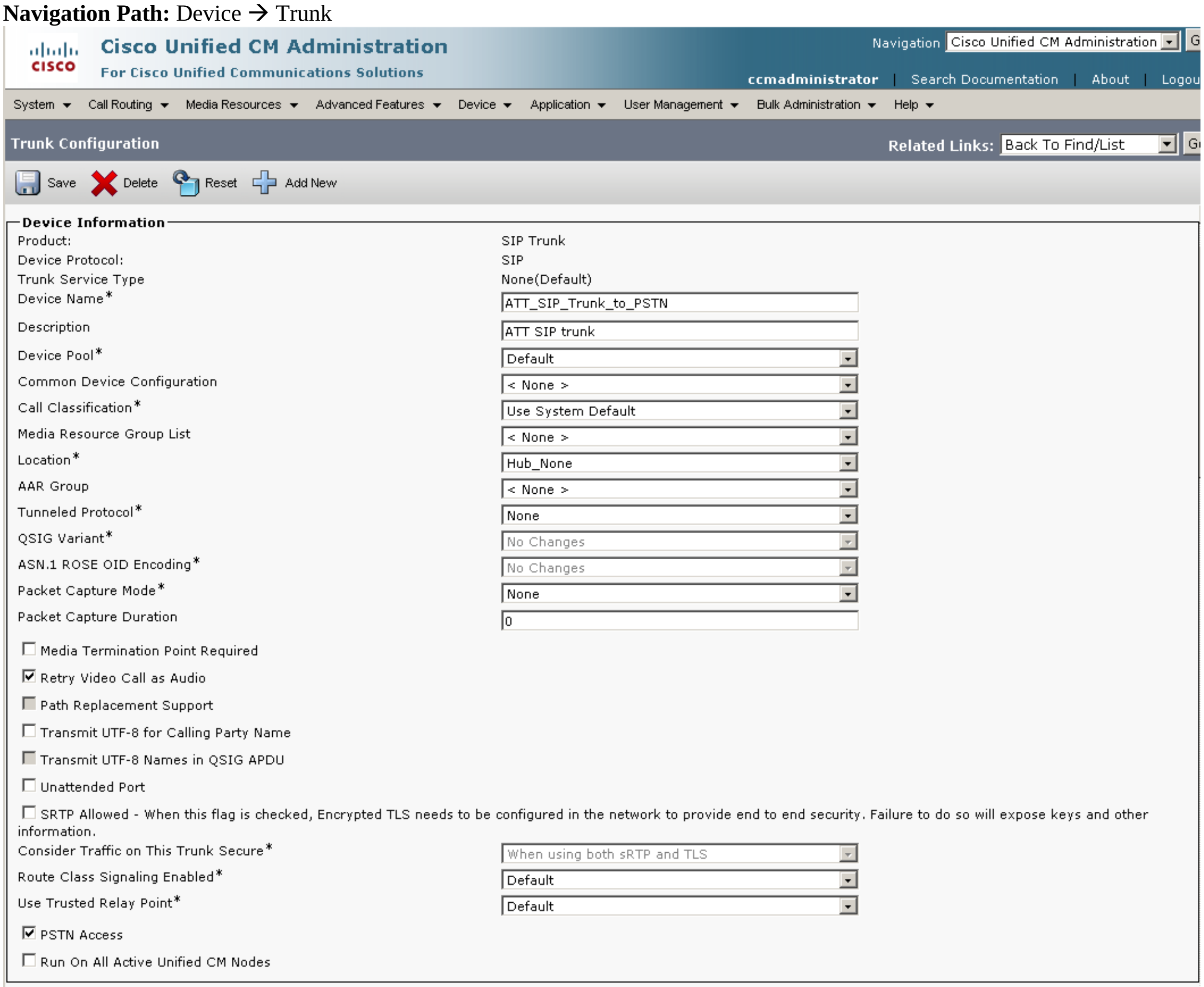
```
M = {}

M.allowHeaders = {"History-Info"}

trace.enable()

function M.outbound_INVITE(msg)
   local callid = msg:getHeader("Call-ID")

   trace.format("M.outbound_INVITE: callid is '%s'", callid)

   local di = msg:getHeader("Diversion")
   if not di
   then
      return
   end

   msg:convertDiversionToHI()

   msg:removeHeader("Diversion")

   local historyInfos = msg:getHeaderValues("History-Info")
```

```
    msg:removeHeader("History-Info")

    local newHi = ""

    for i, hi in ipairs(historyInfos)
    do
       local main_header = string.match(hi, '(.*)?') or string.match(hi, "(.*)>;index=(.*)")
       local embed_header = string.match (hi, '?Reason=sip(.*)>')
       local index = string.match(hi, '>;index=(.*)')

       local hiNext = historyInfos[i + 1]
       local indexNext = string.match(hiNext or "", '>;index=(.*)')

       trace.format("main_header is '%s'", main_header or "nil")

       if i == 1
       then
          local firstHi = string.format("%s>;index=%s", main_header, index)
          firstHi = string.gsub(firstHi, "@(.*):%d+", "@%1")
          msg:addHeader("History-Info", firstHi)
       end

       if embed_header
       then
          trace.format("embed_header is '%s'", embed_header)
          embed_header = string.gsub(embed_header, "unconditional", "Moved Temporarily")
          embed_header = string.gsub(embed_header, ";", "%%3B")
          embed_header = string.gsub(embed_header , "=", "%%3D")
          embed_header = string.gsub(embed_header, "\"", "%%22")
          embed_header = string.gsub(embed_header, " ", "%%20")

          embed_header = string.format("?Reason=SIP%s%s", embed_header, "&Reason=Redirection%3Bcause%3DCFI")
       end

       -- Get rid of the port number
       main_header = string.gsub(main_header, "@(.*):%d+", "@%1")

       if not indexNext
       then
          local left, right = string.match(index, "(%d+)%.(%d+)")
          indexNext = string.format("%s.%s", left + 1, right)
       end

       hi = string.format("%s%s>;index=%s", main_header, embed_header or "", indexNext)
       msg:addHeader("History-Info", hi)
    end
end

local HiCauseToDiversion = { }
HiCauseToDiversion["302"] = "unconditional"
HiCauseToDiversion["486"] = "user-busy"
HiCauseToDiversion["408"] = "no-answer"
HiCauseToDiversion["480"] = "deflection"
HiCauseToDiversion["487"] = "deflection"
HiCauseToDiversion["503"] = "unavailable"
HiCauseToDiversion["404"] = "unknown"

function convertHIToDiversion(msg)
    local historyInfos = msg:getHeaderValues("History-Info")
```

```
   for i, hi in ipairs(historyInfos)
   do
      hi = string.gsub(hi, "%%3B", ";")
      hi = string.gsub(hi, "%%3D", "=")
      hi = string.gsub(hi, "%%22", "\"")
      hi = string.gsub(hi, "%%20", " ")

      -- Reason=SIP;cause=302;text="Moved Temporarily"

      local uri, reason, cause, text = string.match(hi, "<(sip:.*@.*)?Reason=(SIP);cause=(%d+);text=(\".*\")")

      trace.format("hi: uri '%s', reason '%s', cause '%s', text '%s'", uri or "nil", reason or "nil", cause or "nil", text or "nil")

      if reason == "SIP"
      then
         local dReason = HiCauseToDiversion[cause] or "unknown"

         local diversion = string.format("<%s>;reason=\"%s\"", uri, dReason)
         msg:addHeader("Diversion", diversion)
      end
   end
end

function M.inbound_INVITE(msg)
   local callid = msg:getHeader("Call-ID")

   trace.format("M.inbound_INVITE: callid is '%s'", callid)

   local hist = msg:getHeader("History-Info")
   local di  = msg:getHeader("Diversion")

   if hist
   then
      local context = msg:getContext()
      if context
      then
         context["History-Info"] = hist
      end

      if not di
      then
         convertHIToDiversion(msg)
      end
   end

   local di  = msg:getHeader("Diversion")

   if di
   then
      trace.format(" -- found Diversion header")
      msg:removeHeader("History-Info")
      -- replace unknown to unconditional
      di = string.gsub(di, "unknown", "unconditional")
      msg:modifyHeader("Diversion", di)
   end
end

--[[
```

```
function M.outbound_ANY_INVITE(msg)
    local context = msg:getContext()
    if context
    then
        msg:addHeader("History-Info", context["History-Info"])
    end
end
--]]

return M
```

Configuration of SIP trunks to PSTN
Navigation Path: Device → Trunk



Intercompany Media Engine (IME)

E.164 Transformation Profile: < None >

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain: < None >

Call Routing Information

☑ Remote-Party-Id

☑ Asserted-Identity

Asserted-Type*: Default

SIP Privacy*: Default

Inbound Calls

Significant Digits*: All

Connected Line ID Presentation*: Default

Connected Name Presentation*: Default

Calling Search Space: < None >

AAR Calling Search Space: < None >

Prefix DN:

☑ Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	☑

Connected Party Settings

Connected Party Transformation CSS: < None >

☑ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Caller ID DN:

Caller Name:

☑ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.110.154		5060

MTP Preferred Originating Codec*: 711ulaw

Presence Group*: Standard Presence group

SIP Trunk Security Profile*: Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: Early Media and OPTIONS SIP Profile

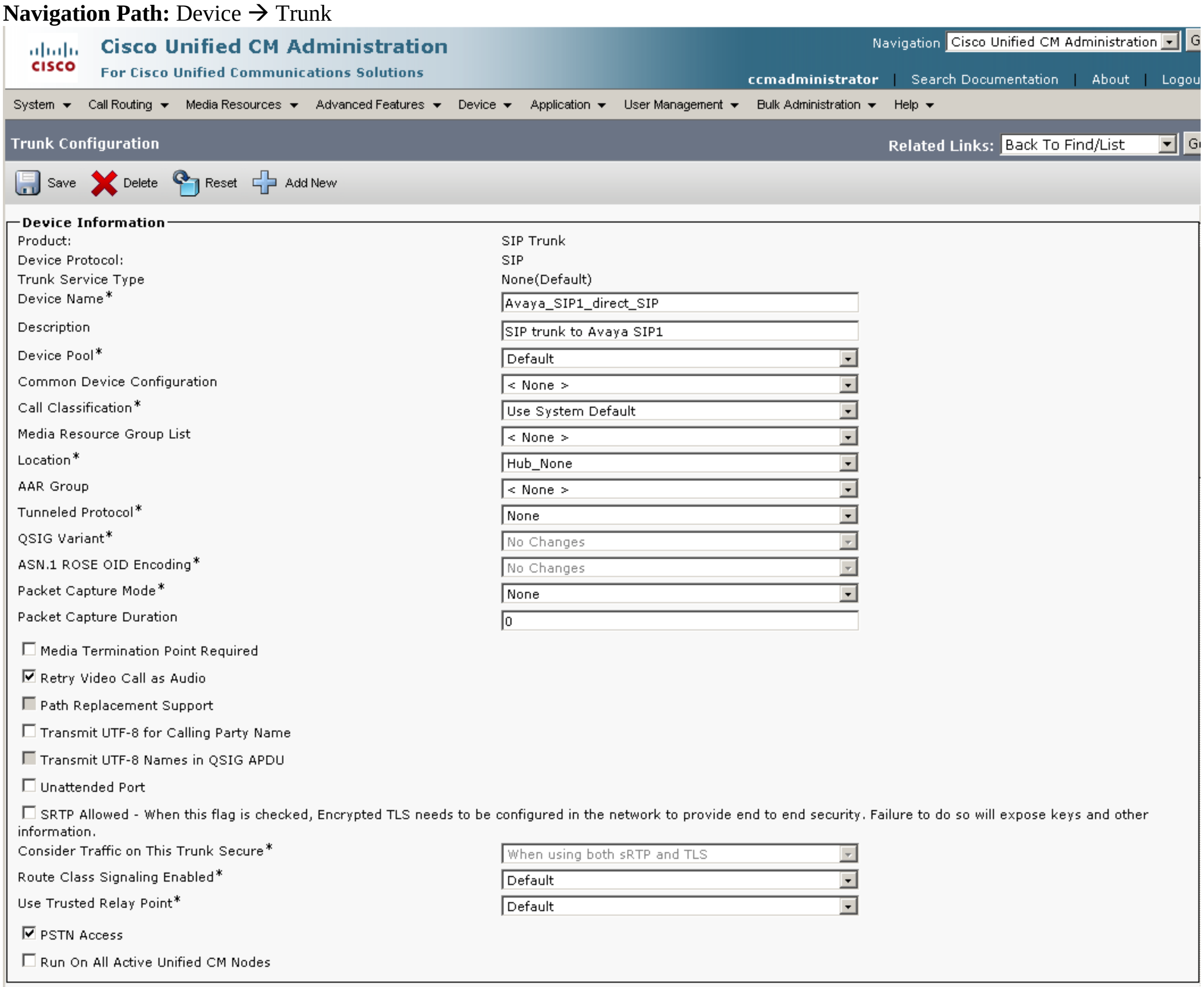
DTMF Signaling Method*: RFC 2833

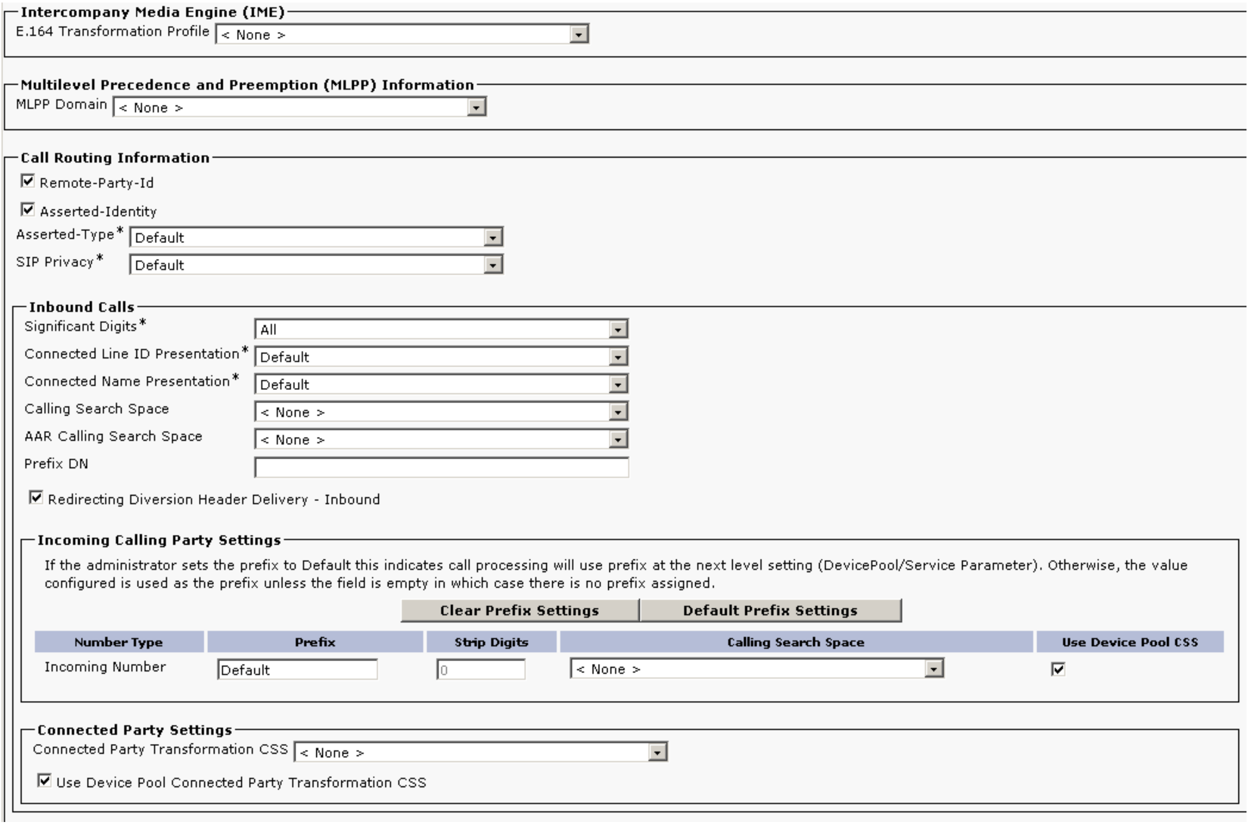
Normalization Script

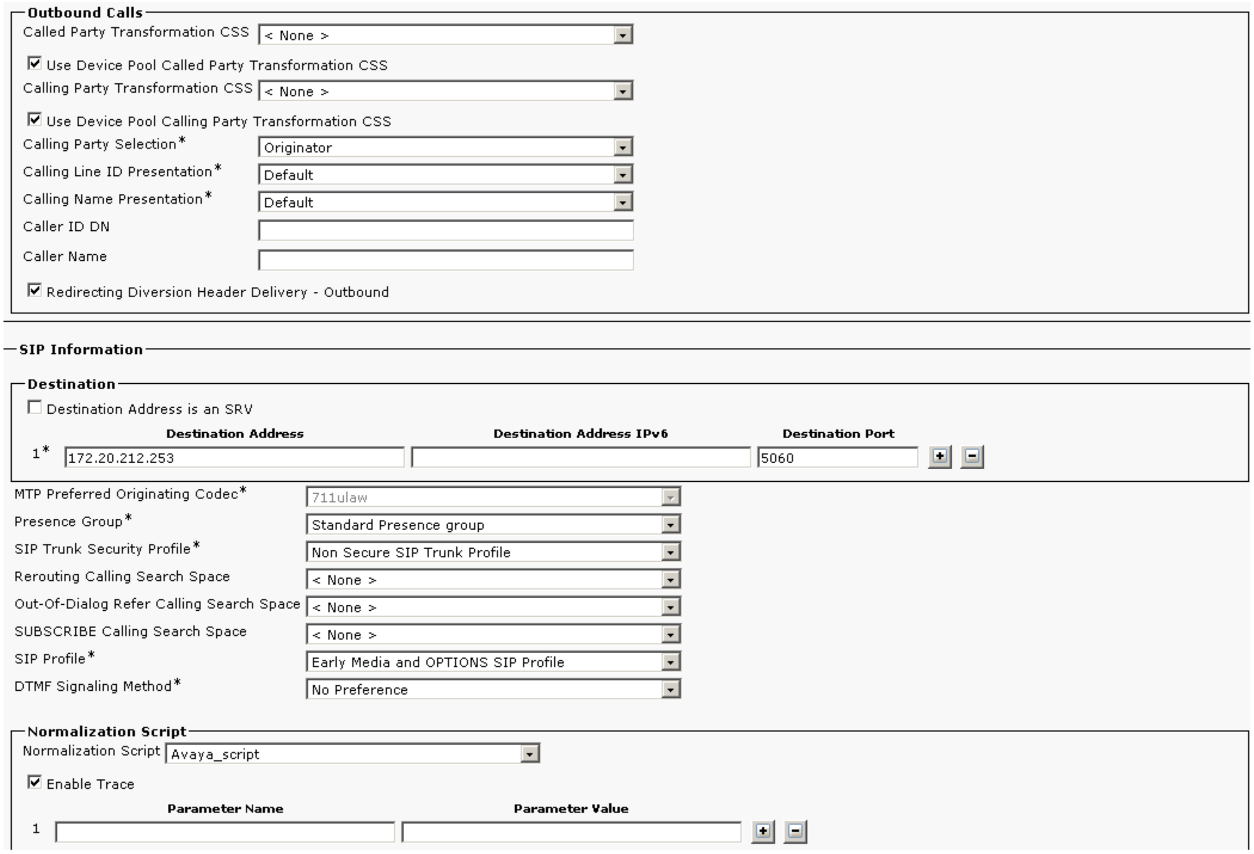
Normalization Script: < None >

☐ Enable Trace

Configuration of SIP trunk to Avaya PBX
Navigation Path: Device → Trunk




Intercompany Media Engine (IME)
E.164 Transformation Profile
< None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain
< None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type*
Default
SIP Privacy*
Default
Inbound Calls
Significant Digits*
All
Connected Line ID Presentation*
Default
Connected Name Presentation*
Default
Calling Search Space
< None >
AAR Calling Search Space
< None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS



Outbound Calls

Called Party Transformation CSS: < None >

☑ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

☑ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Caller ID DN:

Caller Name:

☑ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.212.253		5060

MTP Preferred Originating Codec*: 711ulaw

Presence Group*: Standard Presence group

SIP Trunk Security Profile*: Non Secure SIP Trunk Profile

Rerouting Calling Search Space: < None >

Out-Of-Dialog Refer Calling Search Space: < None >

SUBSCRIBE Calling Search Space: < None >

SIP Profile*: Early Media and OPTIONS SIP Profile

DTMF Signaling Method*: No Preference

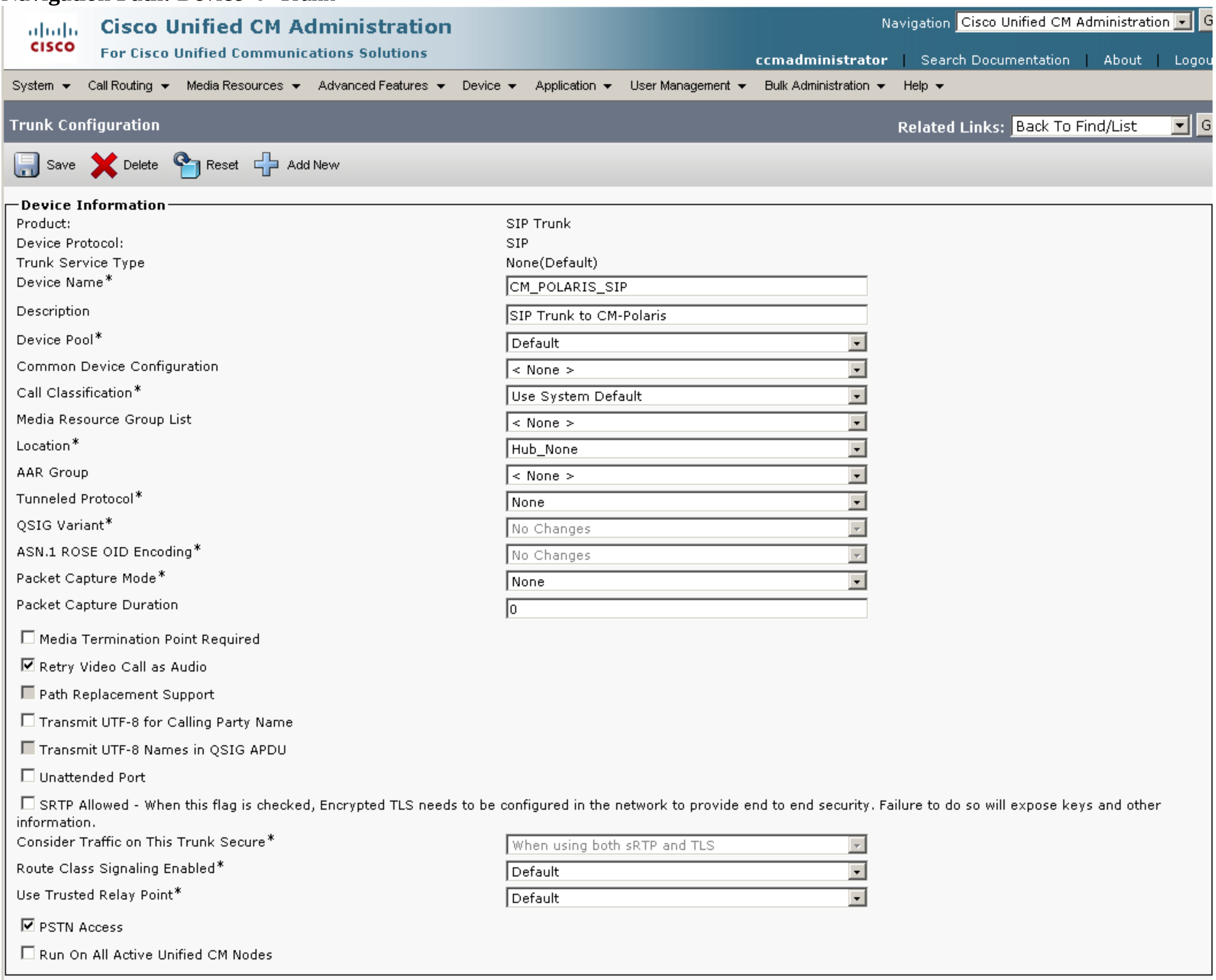
Normalization Script

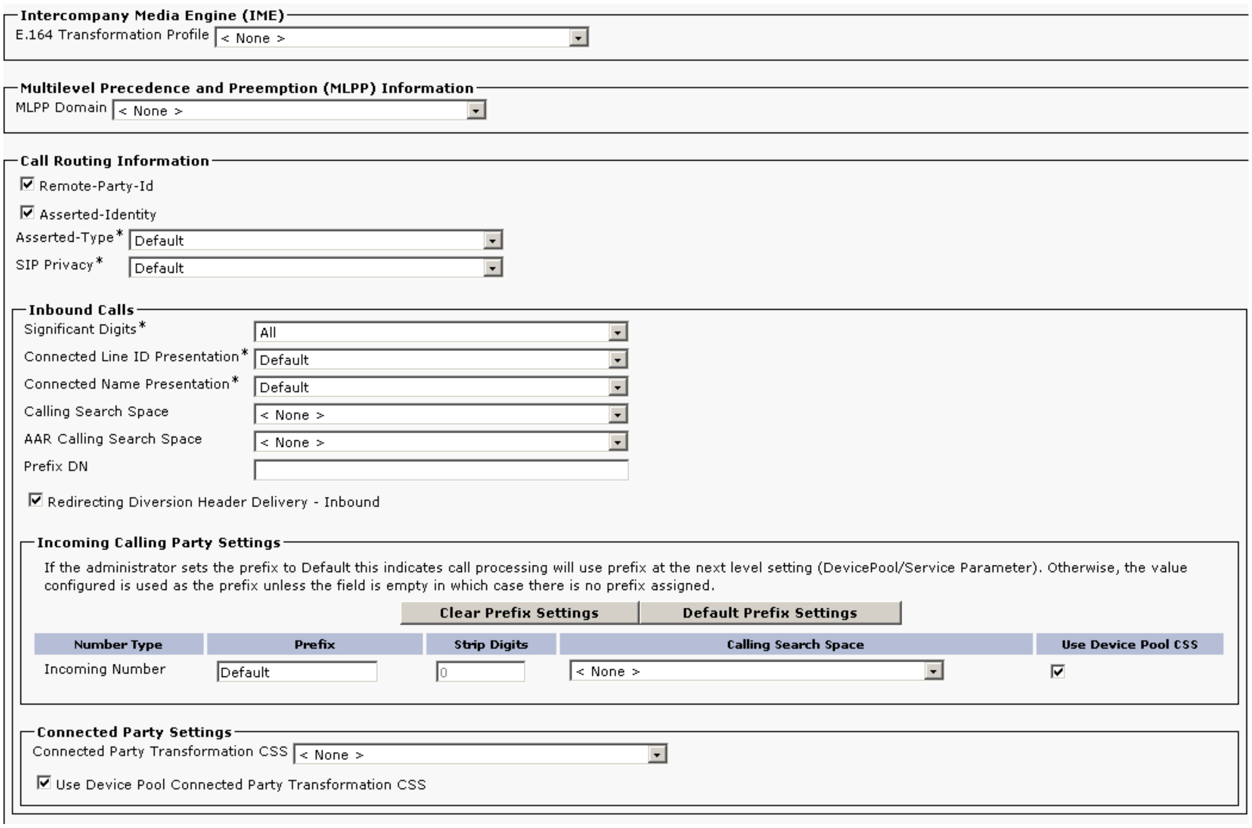
Normalization Script: Avaya_script

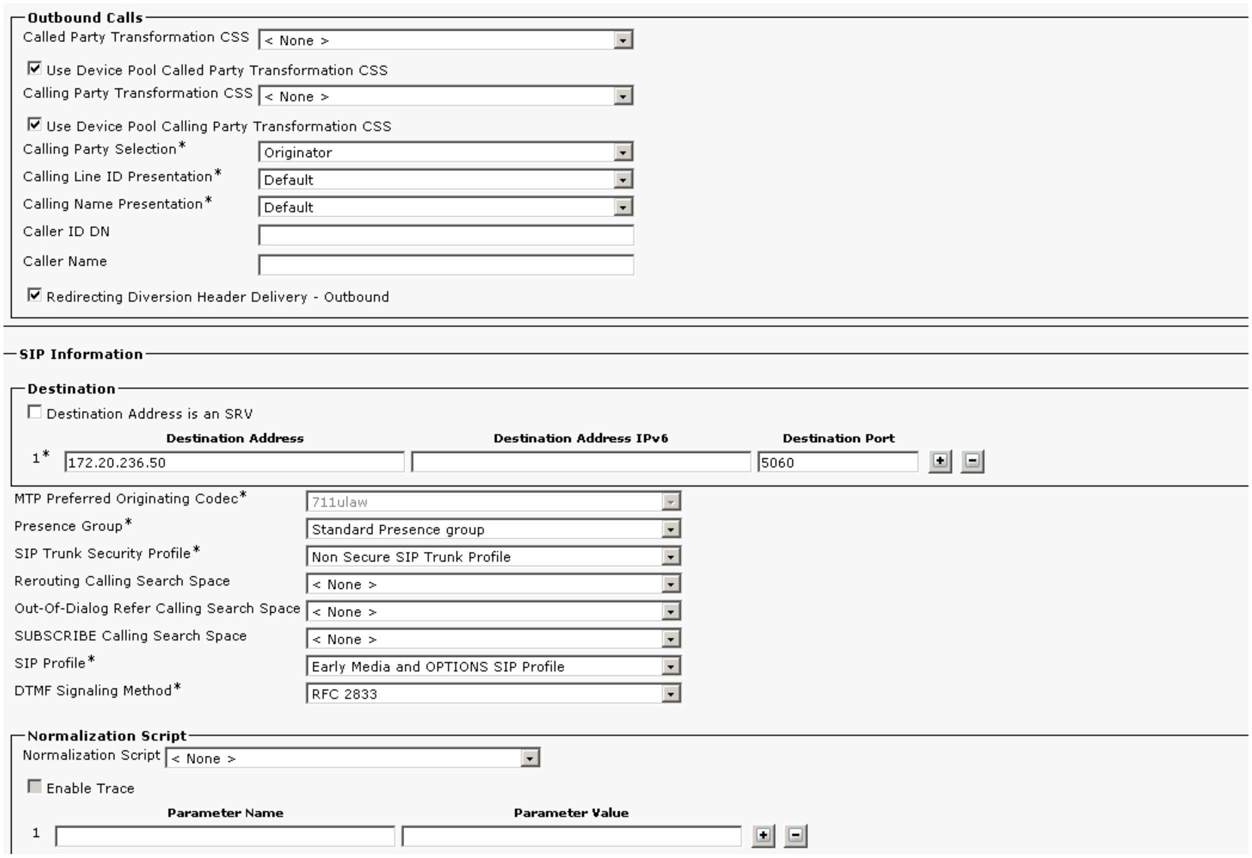
☑ Enable Trace

	Parameter Name	Parameter Value
1		

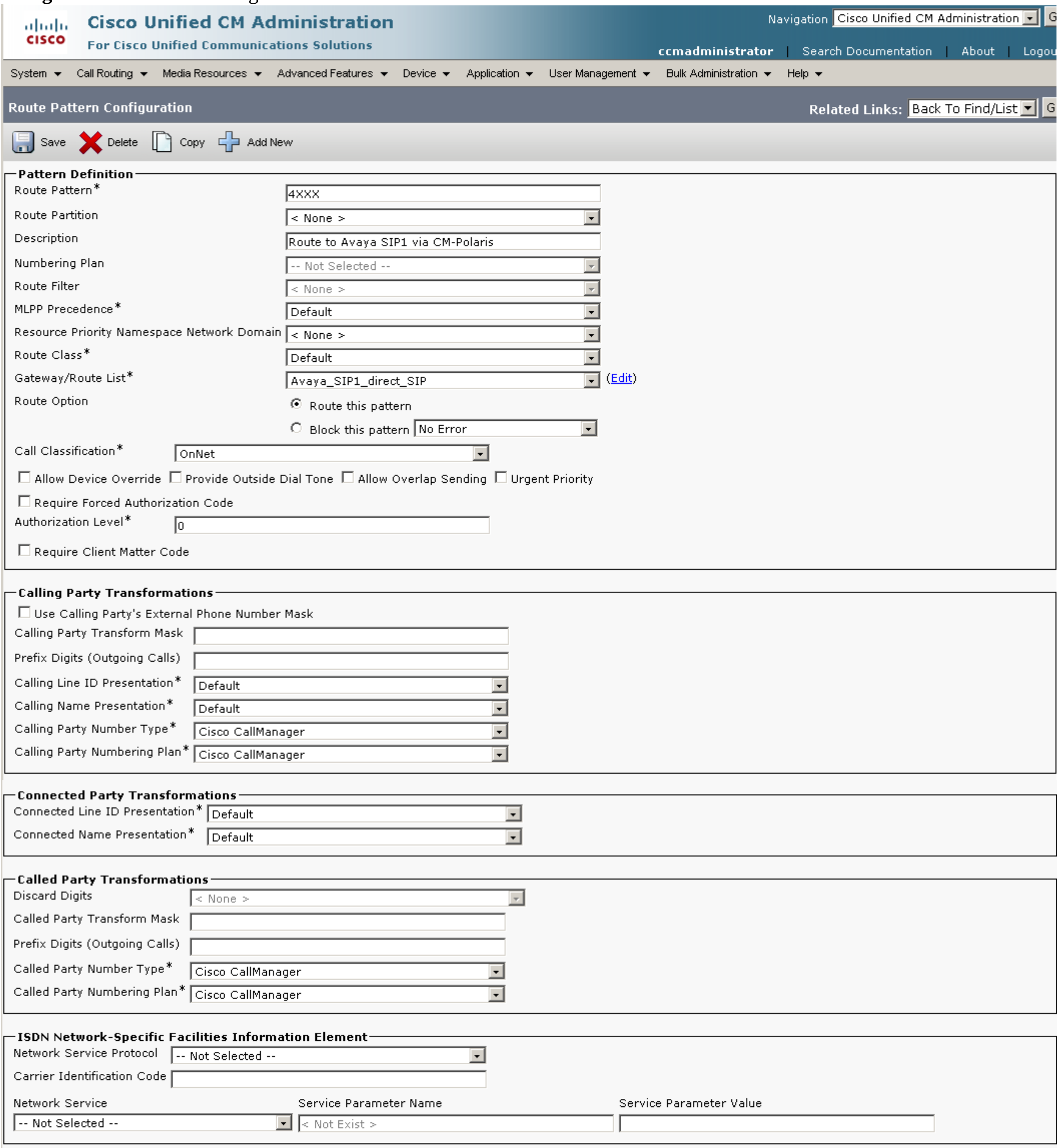
Configuration of SIP trunk to Cisco UCM
Navigation Path: Device → Trunk

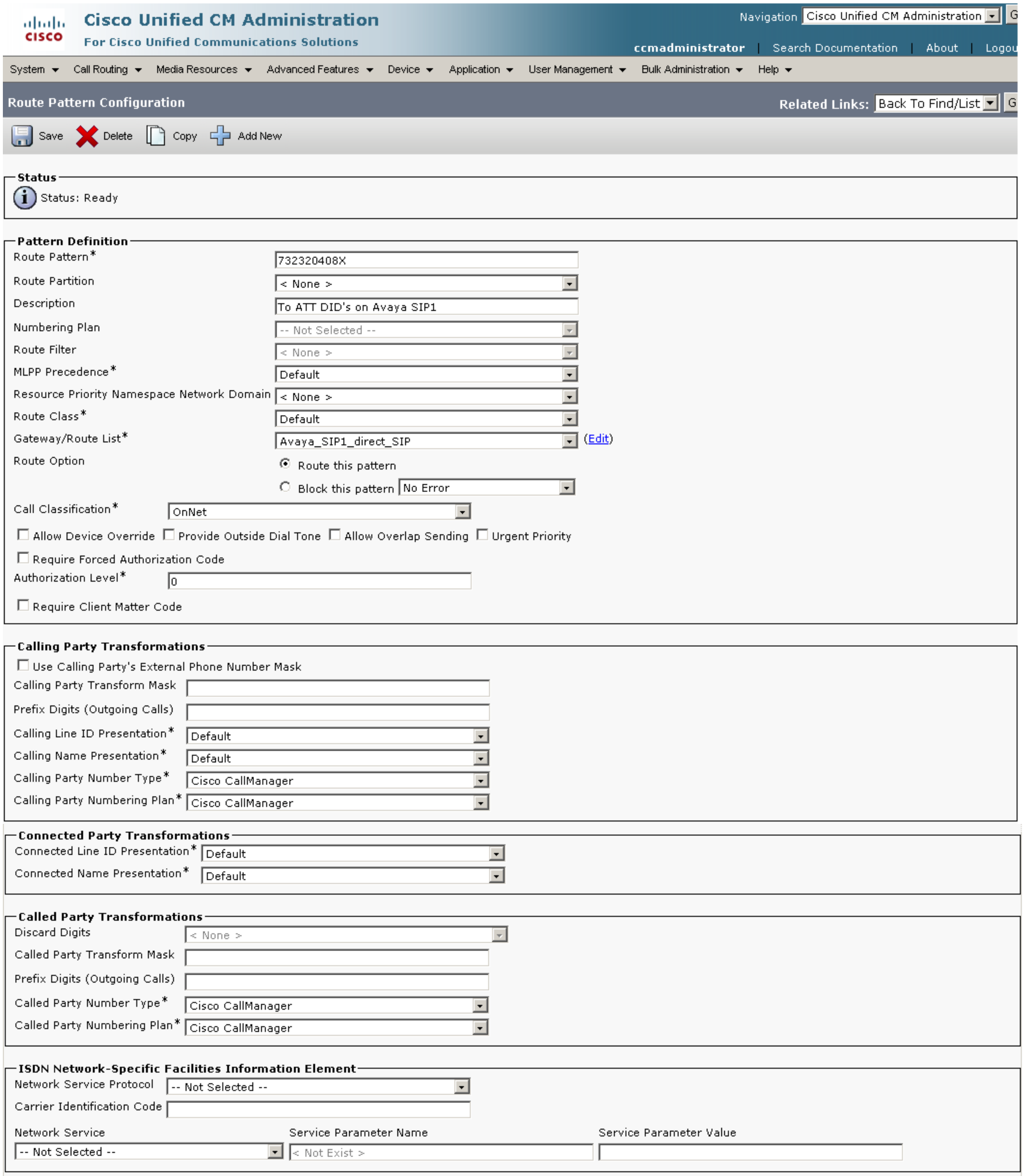



Intercompany Media Engine (IME)
E.164 Transformation Profile < None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS

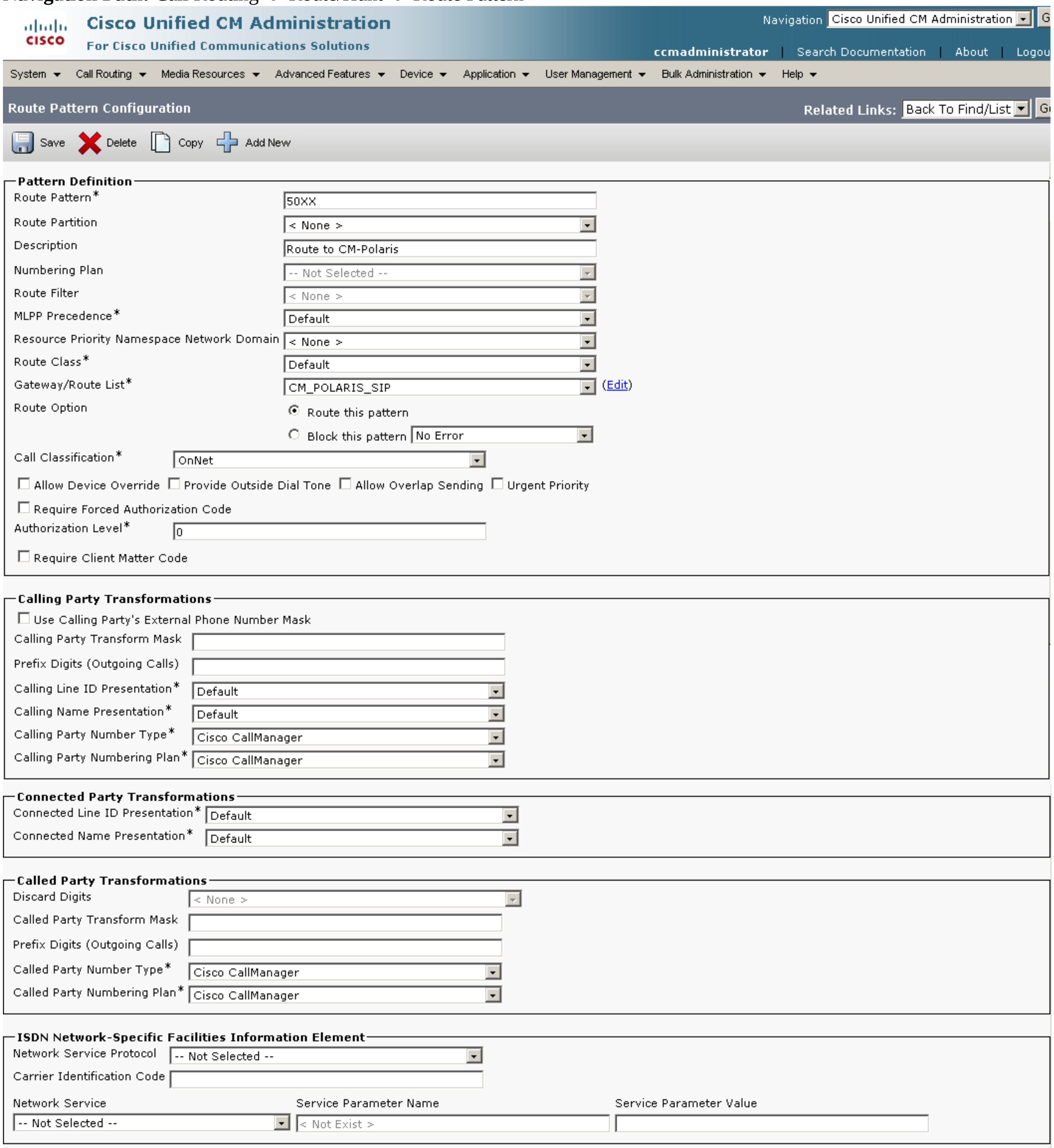

Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Caller ID DN
Caller Name
Redirecting Diversion Header Delivery - Outbound
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
172.20.236.50
5060
MTP Preferred Originating Codec*
711ulaw
Presence Group*
Standard Presence group
SIP Trunk Security Profile*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
Early Media and OPTIONS SIP Profile
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1

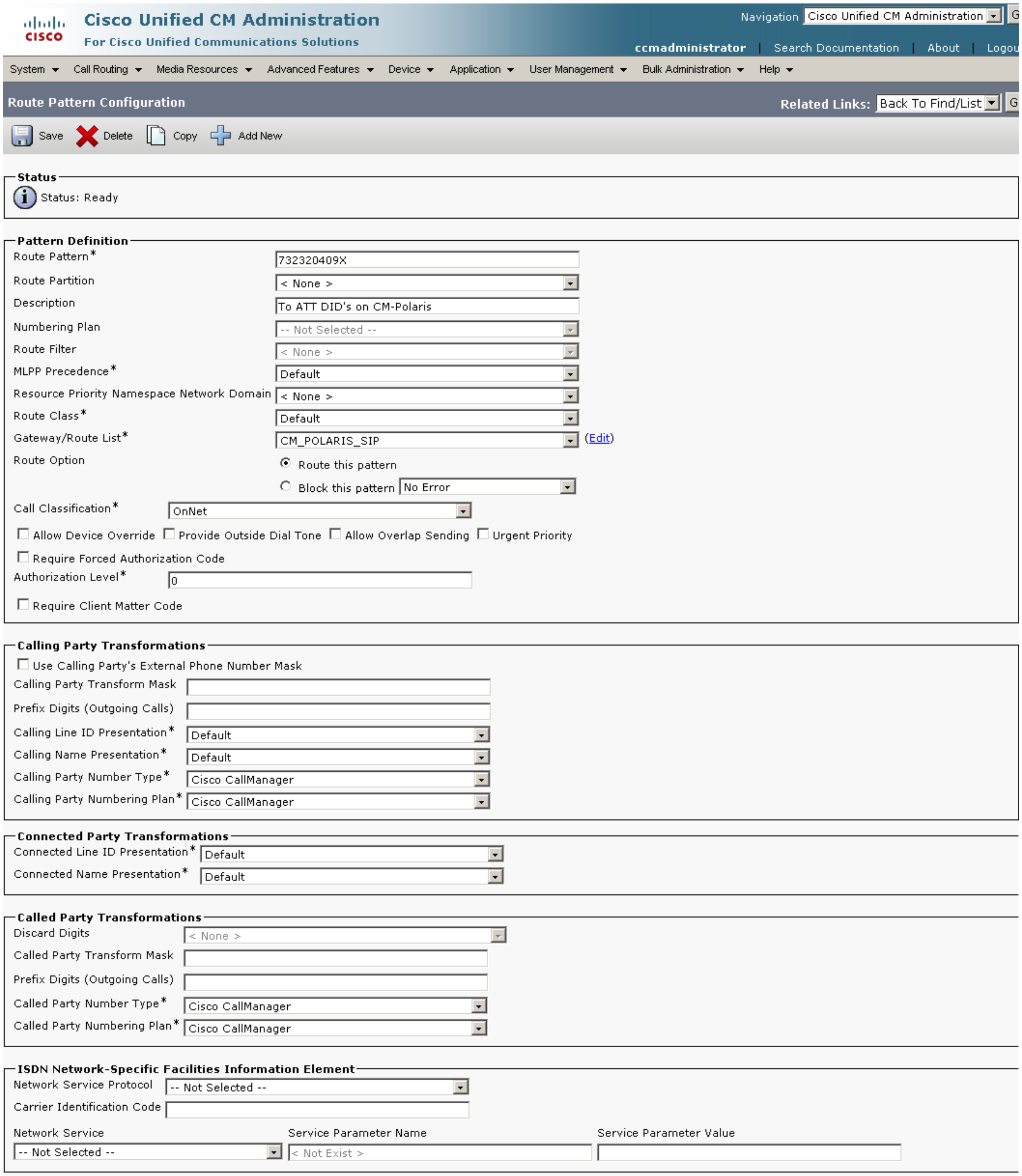
Configuration of Route Patterns – To Avaya PBX
Navigation Path: Call Routing → Route/Hunt → Route Pattern



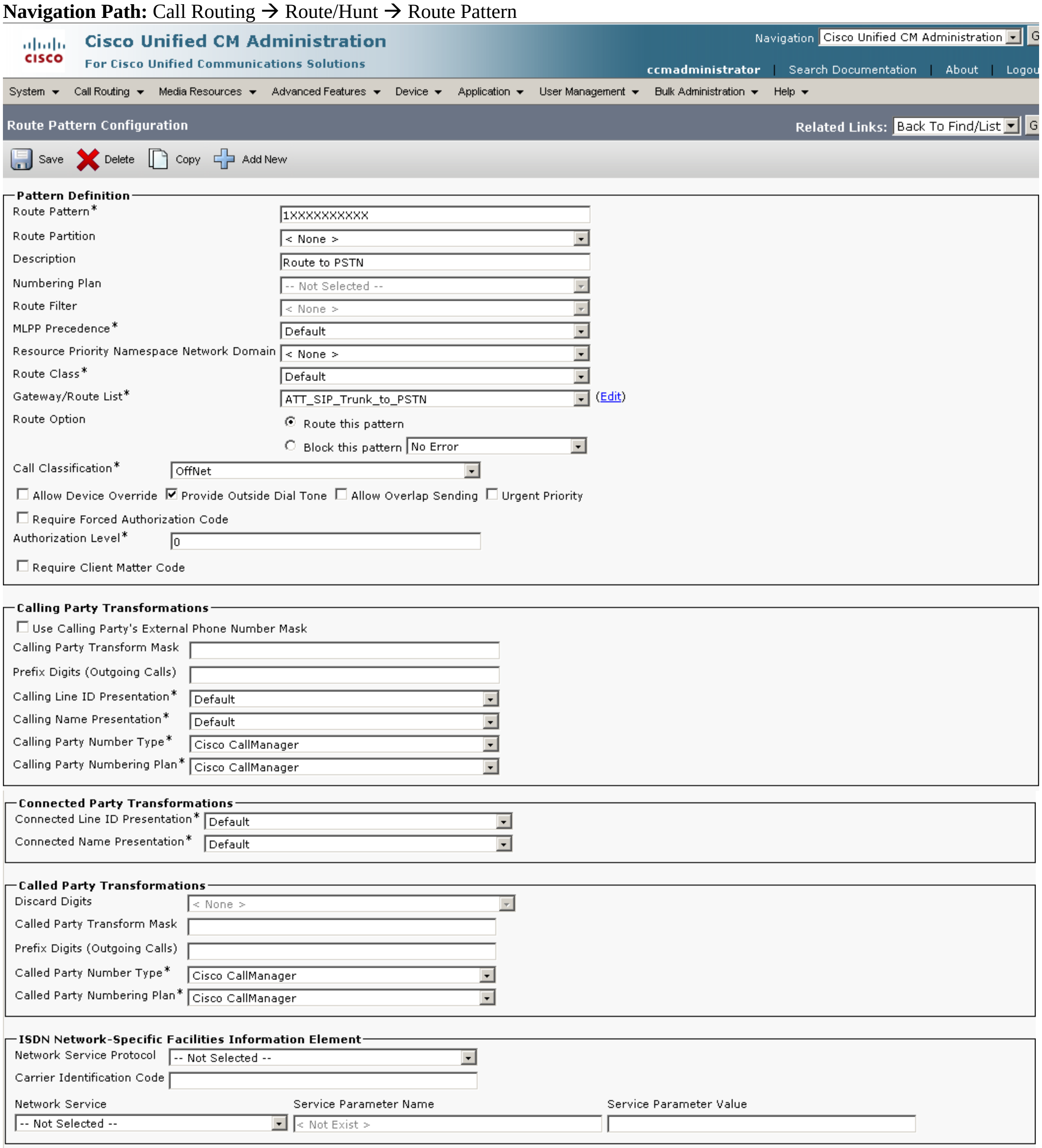

Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
ccmadministrator | Search Documentation | About | Logou
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Route Pattern Configuration
Related Links: Back To Find/List
Save Delete Copy Add New
Status
Status: Ready
Pattern Definition
Route Pattern* 732320408X
Route Partition < None >
Description To ATT DID's on Avaya SIP1
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* Avaya_SIP1_direct_SIP (Edit)
Route Option
Route this pattern
Block this pattern No Error
Call Classification* OnNet
Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Called Party Transformations
Discard Digits < None >
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 -- Not Selected --
Service Parameter Name < Not Exist >
Service Parameter Value

Configuration of Route Patterns – To Cisco UCM
Navigation Path: Call Routing → Route/Hunt → Route Pattern




Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Navigation Cisco Unified CM Administration
ccmadministrator | Search Documentation | About | Logou
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Route Pattern Configuration
Related Links: Back To Find/List
Save Delete Copy Add New
Status
Status: Ready
Pattern Definition
Route Pattern* 732320409X
Route Partition < None >
Description To ATT DID's on CM-Polaris
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Resource Priority Namespace Network Domain < None >
Route Class* Default
Gateway/Route List* CM_POLARIS_SIP (Edit)
Route Option
Route this pattern
Block this pattern No Error
Call Classification* OnNet
Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
Require Forced Authorization Code
Authorization Level* 0
Require Client Matter Code
Calling Party Transformations
Use Calling Party's External Phone Number Mask
Calling Party Transform Mask
Prefix Digits (Outgoing Calls)
Calling Line ID Presentation* Default
Calling Name Presentation* Default
Calling Party Number Type* Cisco CallManager
Calling Party Numbering Plan* Cisco CallManager
Connected Party Transformations
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Called Party Transformations
Discard Digits < None >
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 >

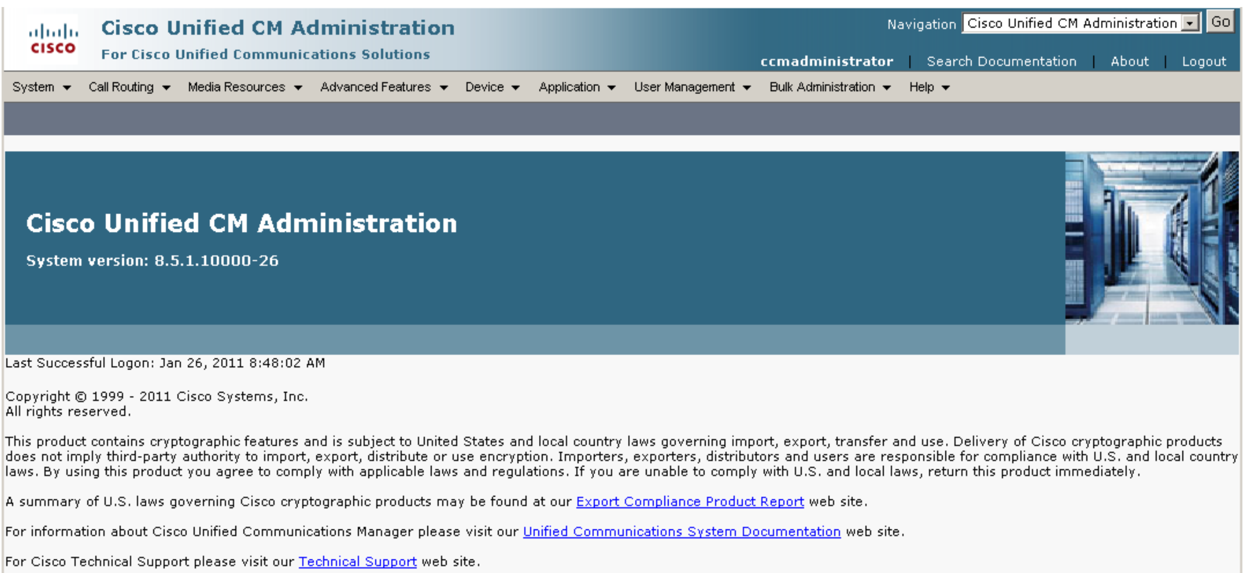
Configuration of Route Patterns – To PSTN
Navigation Path: Call Routing → Route/Hunt → Route Pattern



Configuring the Cisco Unified Communications Manager

1. Cisco Unified Communications Manager Version
2. Service Parameters configuration
3. Device pool and Region mapping configuration
4. Conference Bridge configuration
5. Media Resource Group configuration
6. Media Resource Group List configuration
7. SIP Profile configuration
8. SIP Trunk to SME configuration
9. Route Pattern configuration to SP
10. Route Pattern configuration to Avaya
11. Cisco IP Phone 7960 SCCP Configuration
12. Cisco IP Phone 7960 SIP Configuration
13. MGCP Fax gateway configuration

Cisco Unified Communications Manager version



Configuration of Service Parameters – Cisco CallManager

Navigation Path: System → Service Parameters

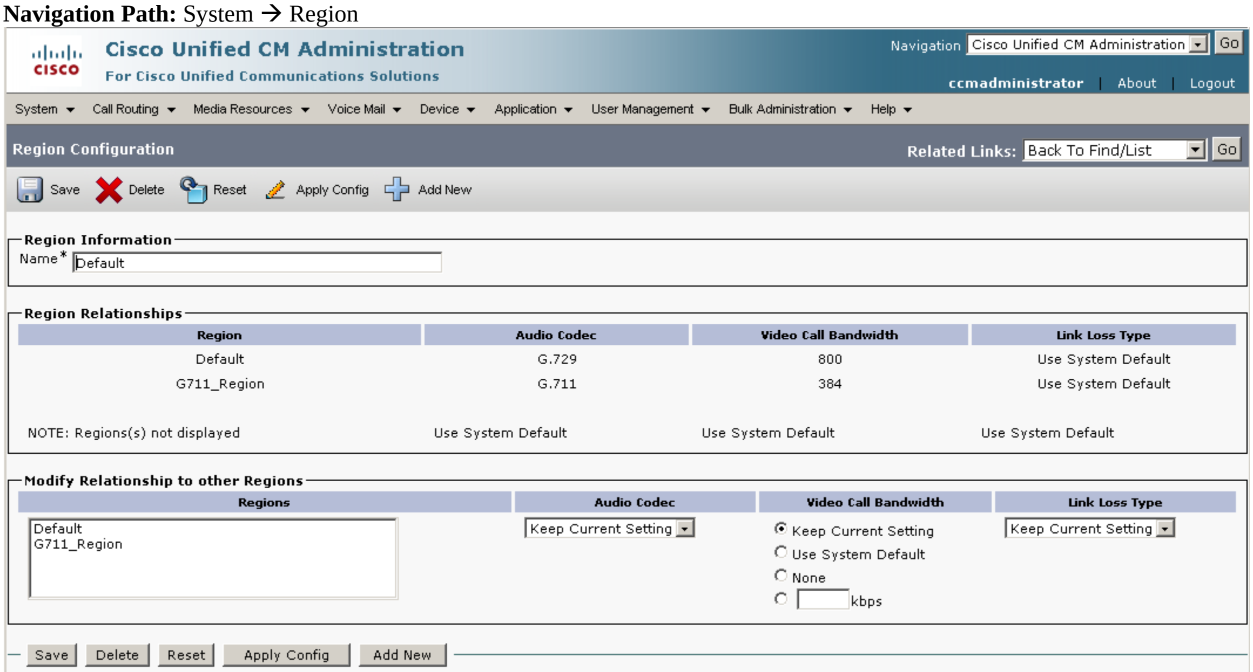
Clusterwide Parameters (Service)

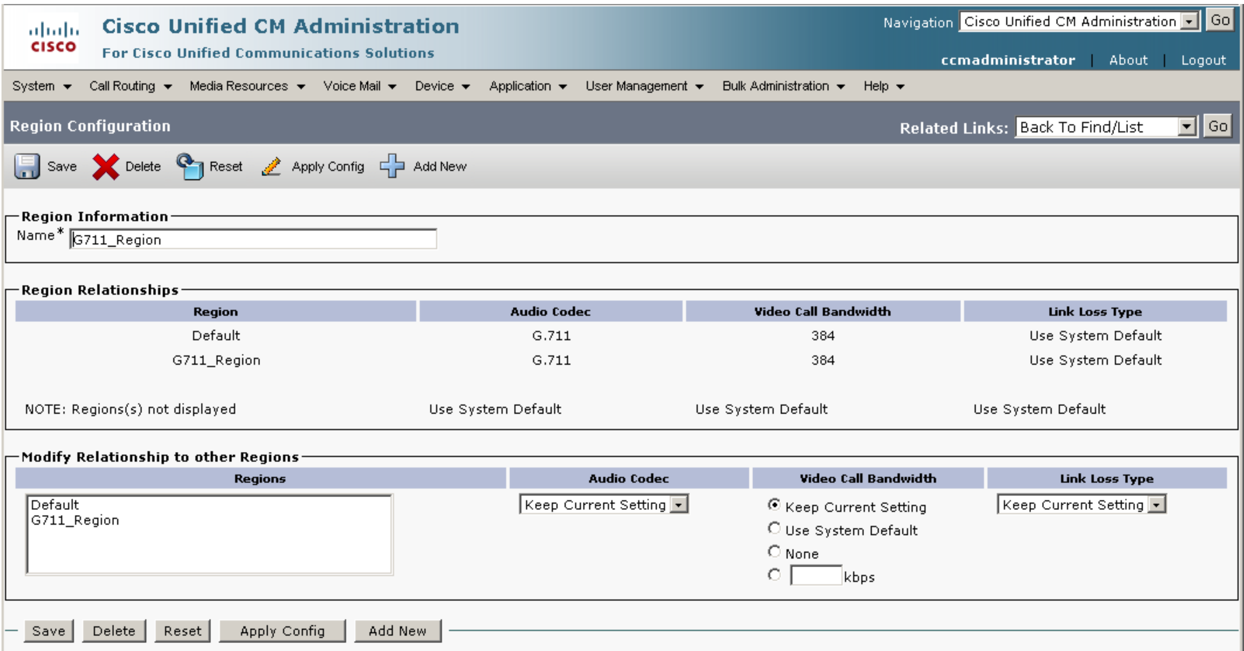
Parameter	Value	Default
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95
Intercluster Capabilities Mismatch Timer *	1000	1000
Silence Suppression *	False	False
Silence Suppression for Gateways *	False	False
Strip G.729 Annex B (Silence Suppression) from Capabilities *	False	False

Note: Service Parameter "Duplex Streaming Enabled" must be set to "True" in order to successfully provide MoH/Ringback to Avaya IP phones (H.323) and outside (PSTN) callers when calls are placed on hold and/or transferred from Cisco UCM stations.

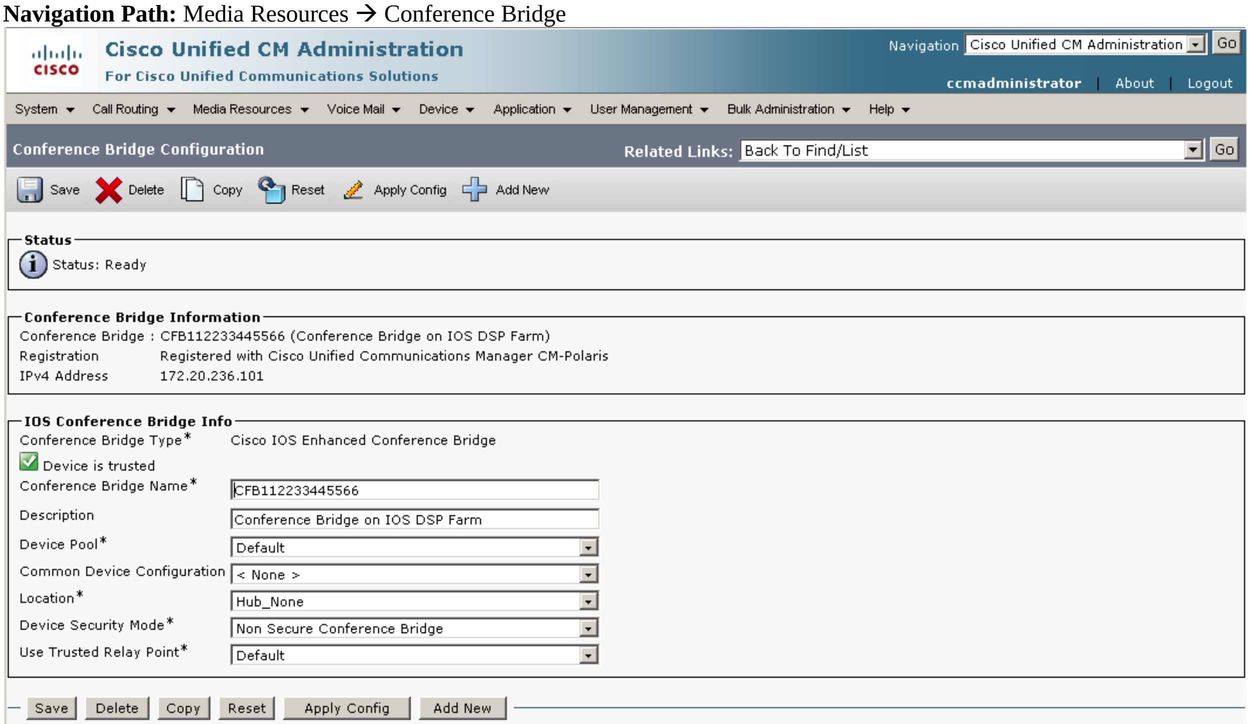
Configuration of Device Pool to Region mapping

Navigation Path: System → Region





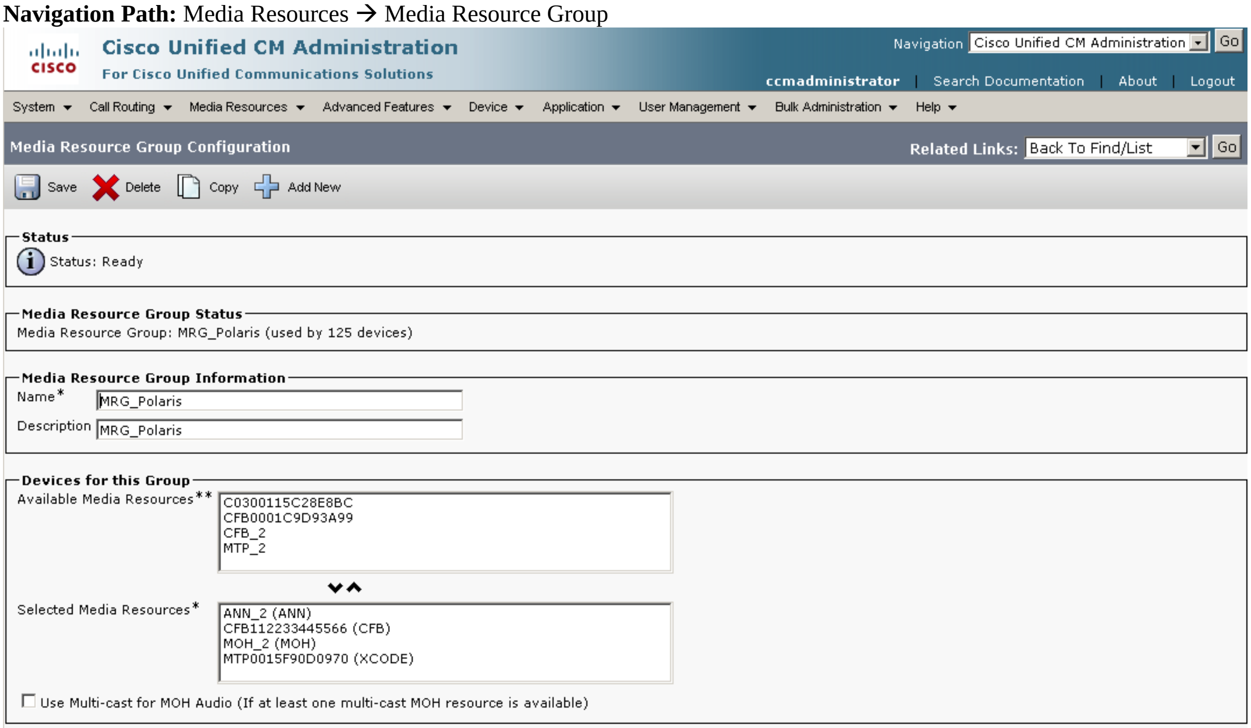
Configuration of Conference Bridge
Navigation Path: Media Resources → Conference Bridge



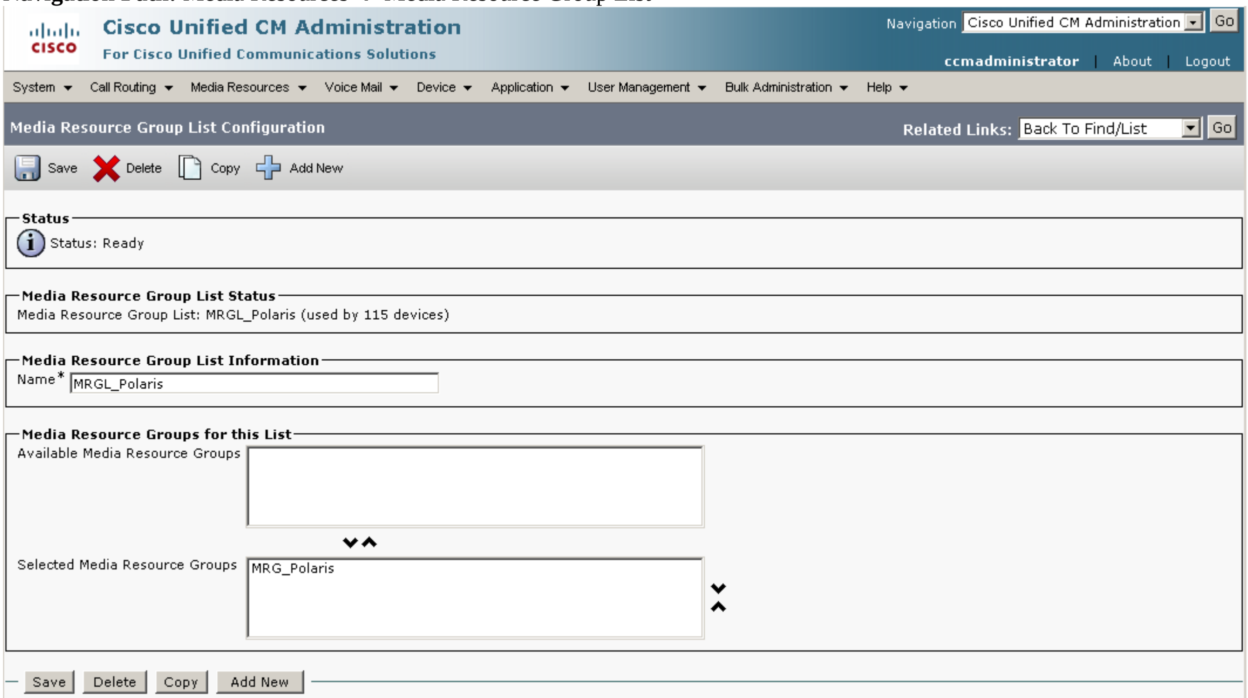
Conference Bridge IOS configuration:

```
sccp local GigabitEthernet0/0
sccp ccm 172.20.236.50 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface GigabitEthernet0/0
 associate ccm 1
 priority 1
 associate profile 98 register cfb112233445566
!
dspfarm profile 98 conference
 codec g729r8
 codec g711ulaw
 maximum sessions 8
 associate application SCCP
```

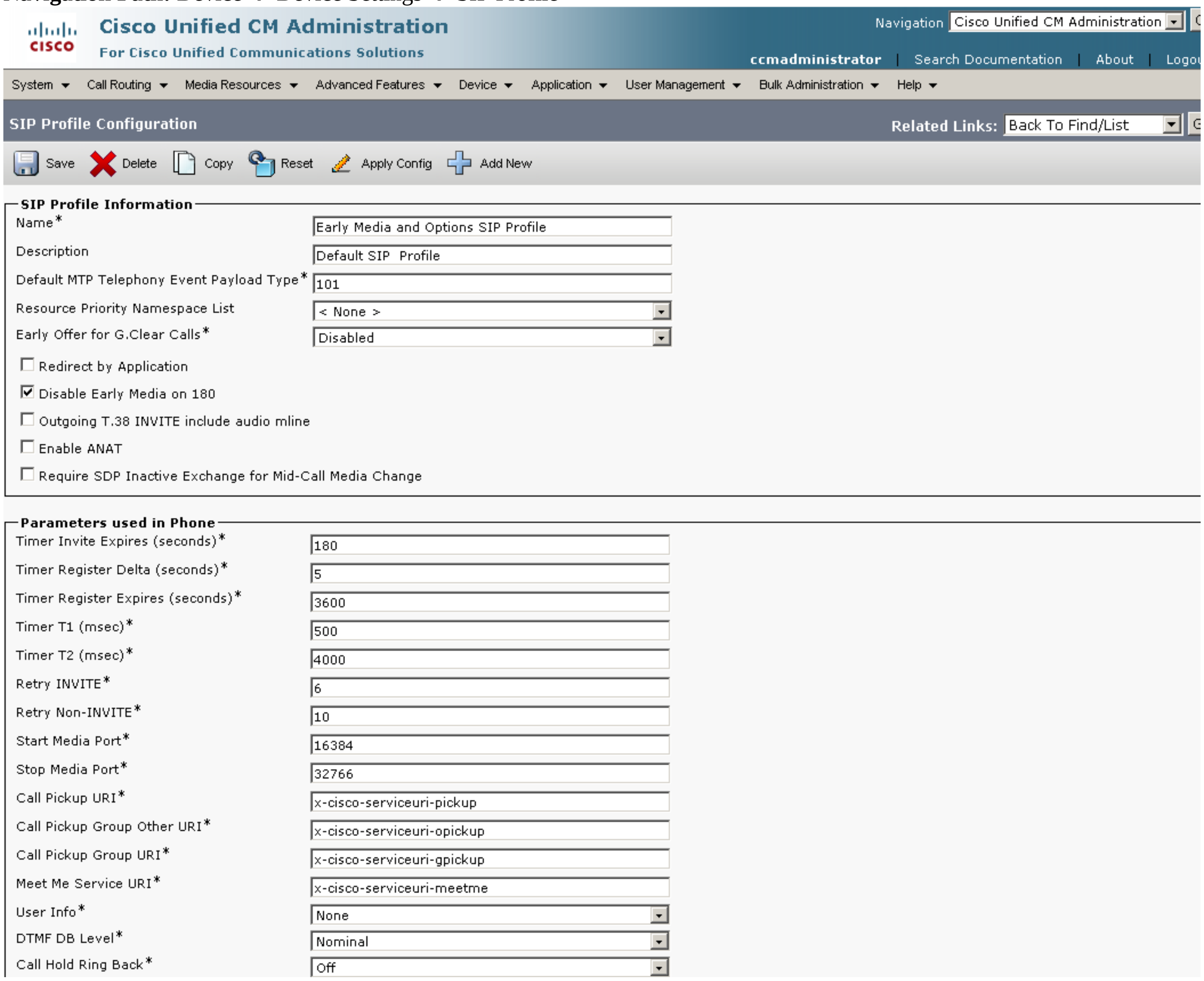
Configuration of Media Resource Group
Navigation Path: Media Resources → Media Resource Group

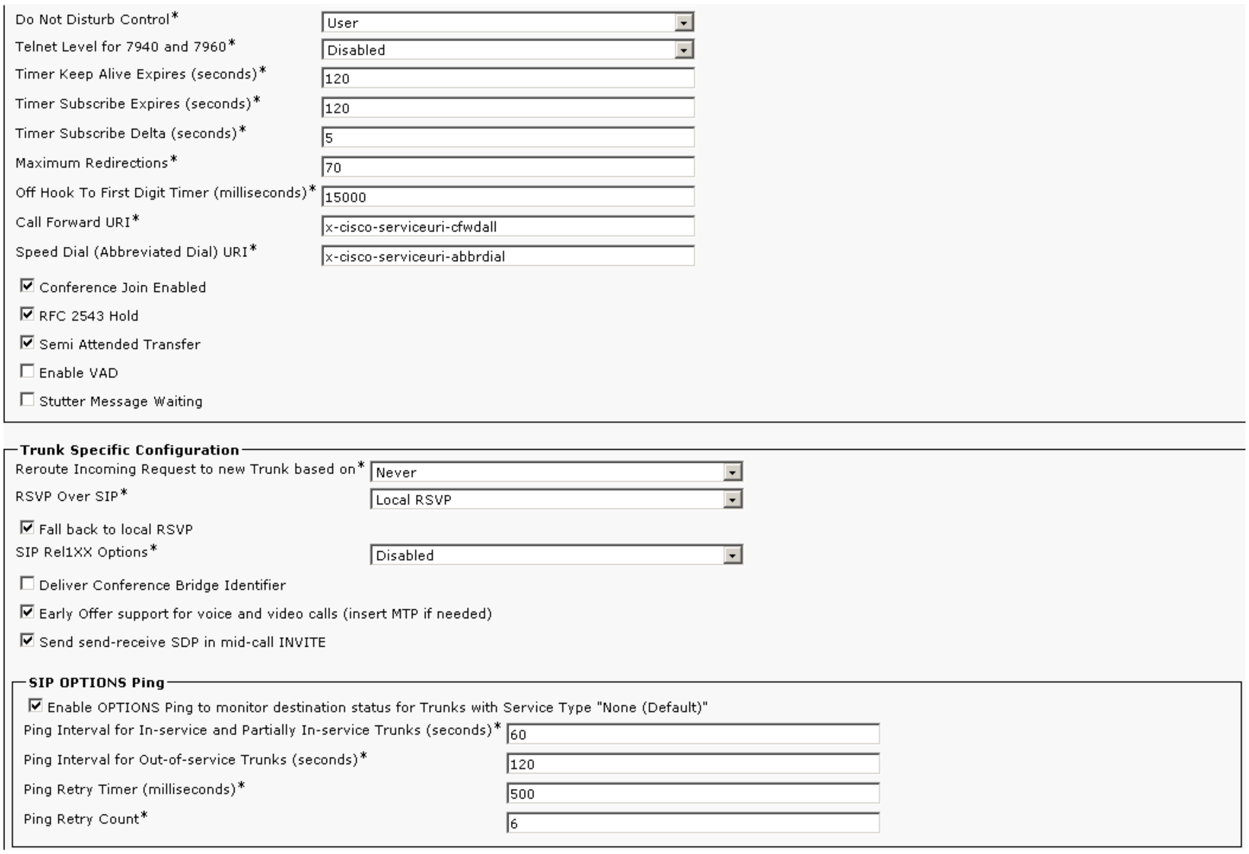


Configuration of Media Resource Group List
Navigation Path: Media Resources → Media Resource Group List



Configuration of SIP Profile
Navigation Path: Device → Device Settings → SIP Profile





Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

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

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP

- [x] Fall back to local RSVP

SIP Rel1XX Options*	Disabled

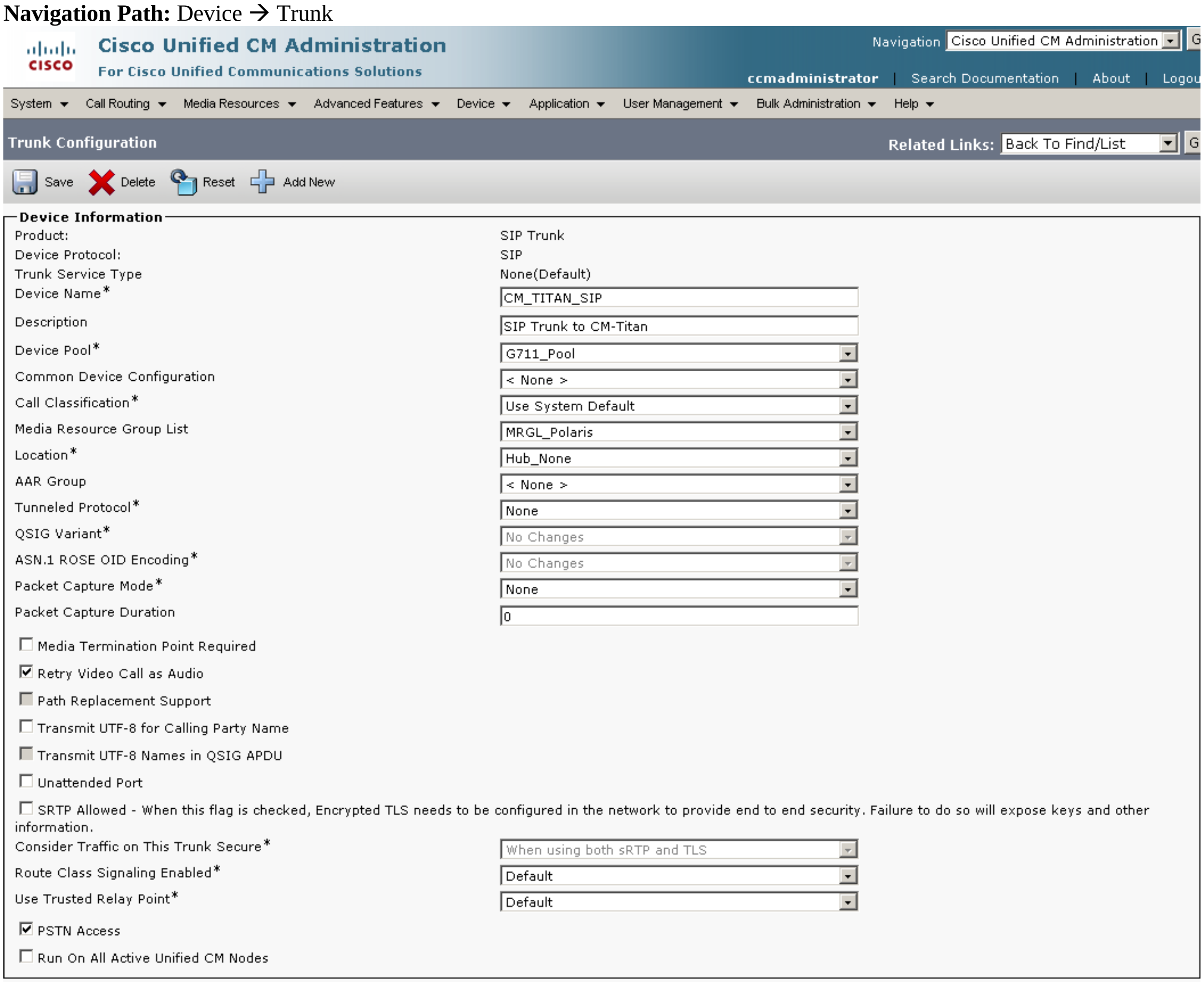
- [ ] Deliver Conference Bridge Identifier
- [x] Early Offer support for voice and video calls (insert MTP if needed)
- [x] Send send-receive SDP in mid-call INVITE

SIP OPTIONS Ping

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

Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

Configuration of SIP Trunk to SME
Navigation Path: Device → Trunk



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Trunk Configuration
Related Links: Back To Find/List

Save | Delete | Reset | Add New

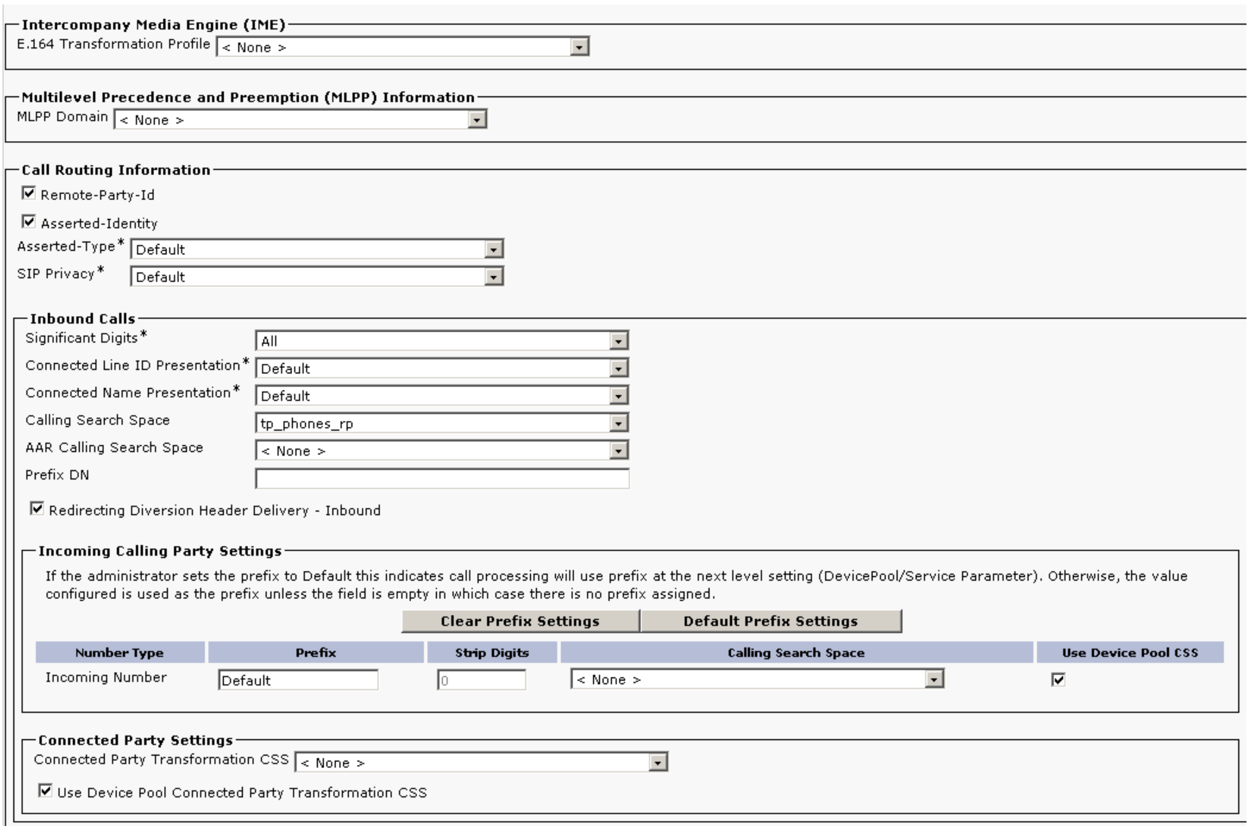
Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CM_TITAN_SIP
Description	SIP Trunk to CM-Titan
Device Pool*	G711_Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Polaris
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [x] PSTN Access
- [ ] Run On All Active Unified CM Nodes


Intercompany Media Engine (IME)
E.164 Transformation Profile < None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space tp_phones_rp
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
Clear Prefix Settings
Default Prefix Settings
Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS < None >
Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS	< None >
☑ Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
☑ Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
☑ Redirecting Diversion Header Delivery - Outbound	

SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	172.20.236.252		5060

MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	SIP Trunks
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Early Media and Options SIP Profile
DTMF Signaling Method*	RFC 2833

Normalization Script

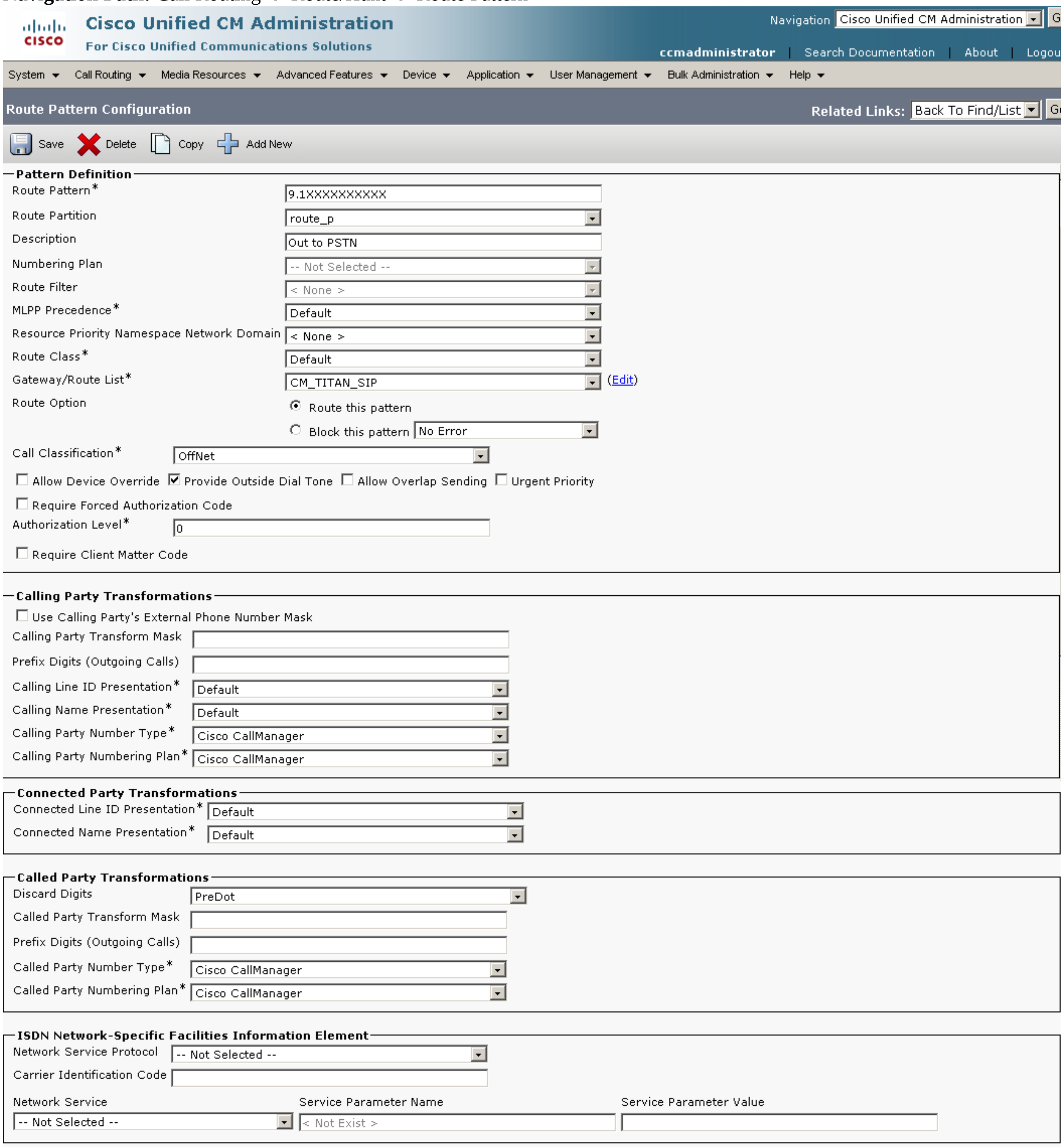
Normalization Script: < None >

☐ Enable Trace

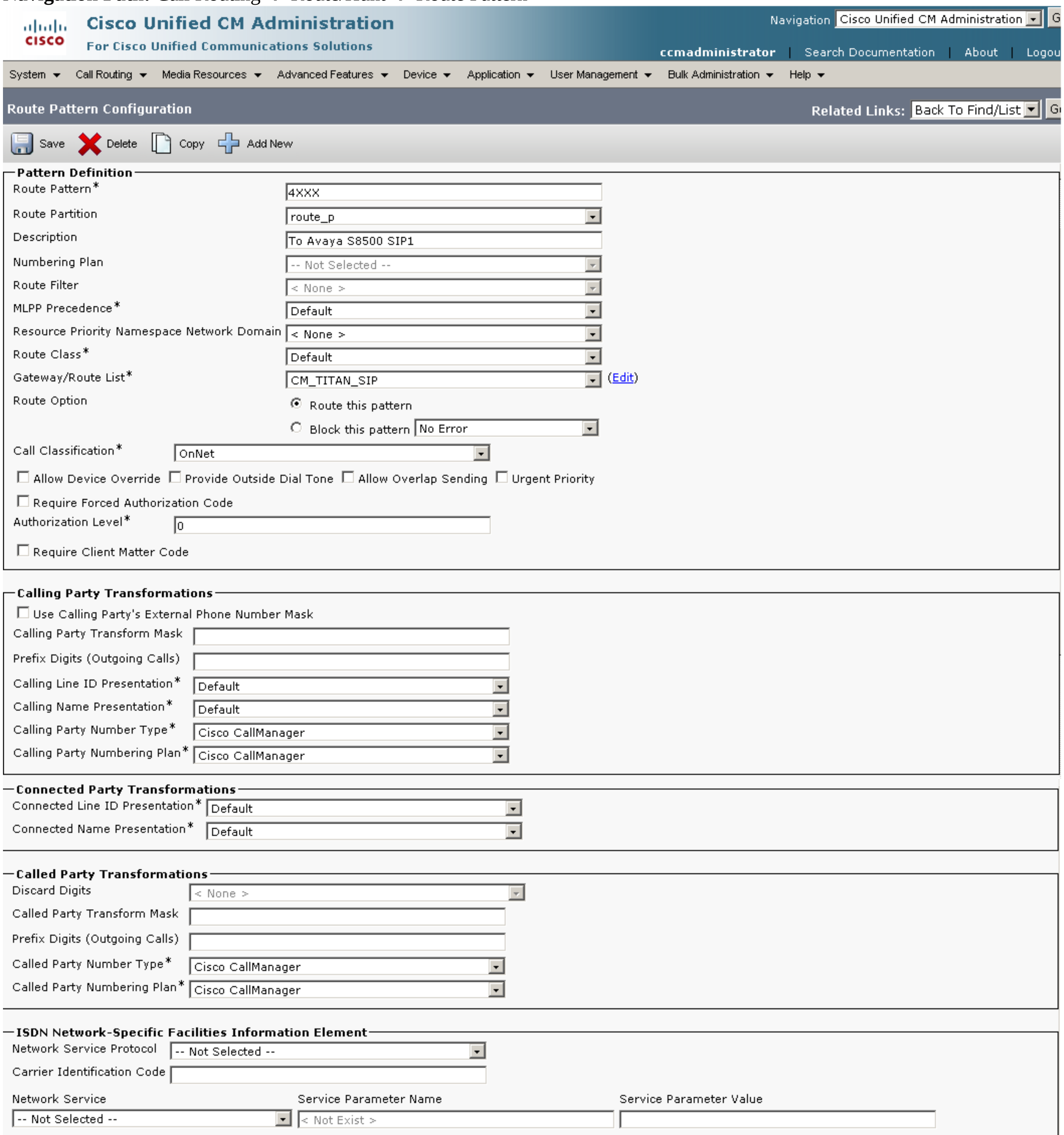
	Parameter Name	Parameter Value
1		

Configuration of Route Pattern to SP through SME

Navigation Path: Call Routing → Route/Hunt → Route Pattern



Configuration of Route Pattern to Avaya PBX through SME
Navigation Path: Call Routing → Route/Hunt → Route Pattern



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Route Pattern Configuration
Related Links: Back To Find/List

Save | Delete | Copy | Add New

Pattern Definition

Route Pattern*	4XXX
Route Partition	route_p
Description	To Avaya S8500 SIP1
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CM_TITAN_SIP (Edit)
Route Option	(•) Route this pattern
	( ) Block this pattern No Error
Call Classification*	OnNet
Authorization Level*	0

☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

Called Party Transformations

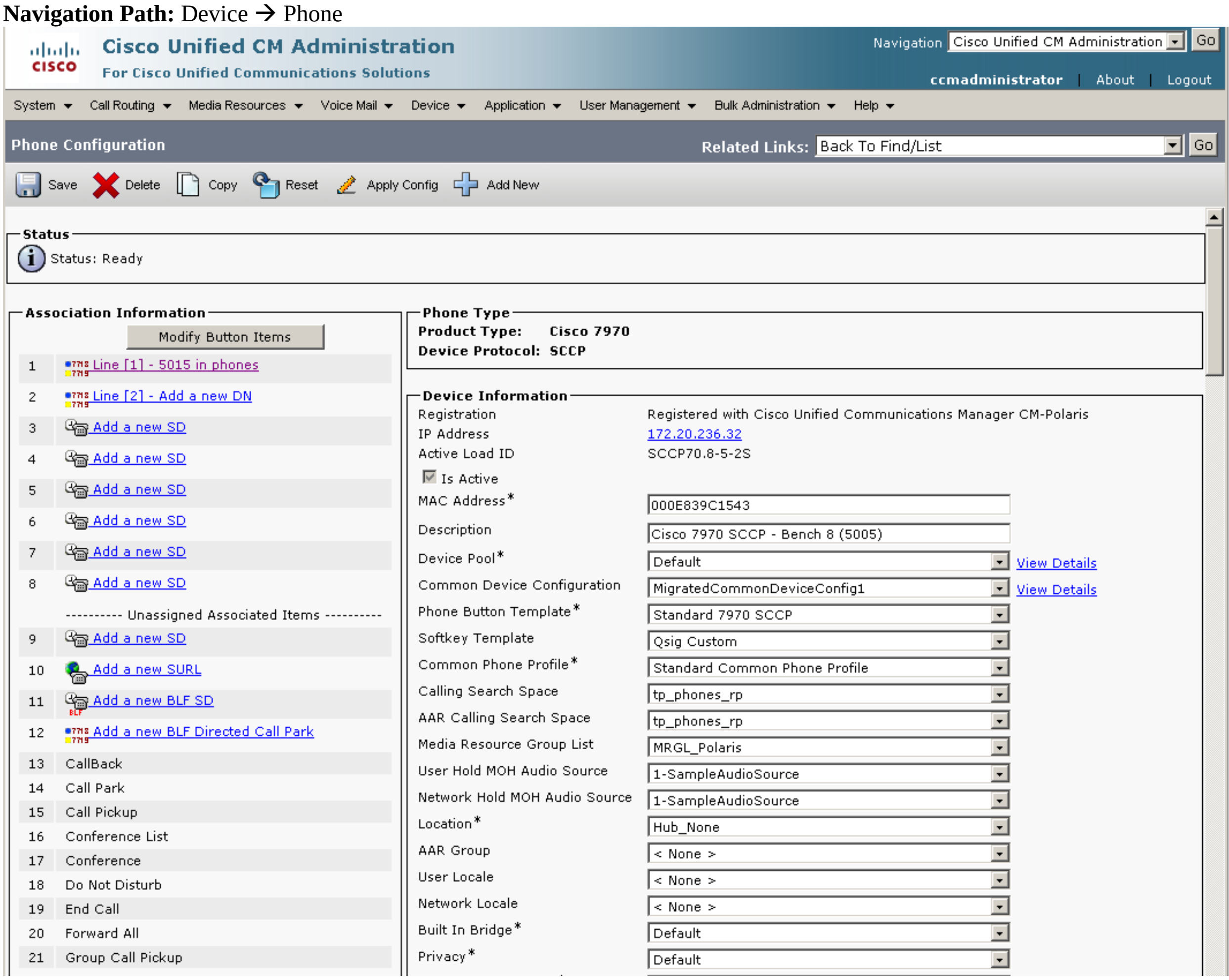
Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

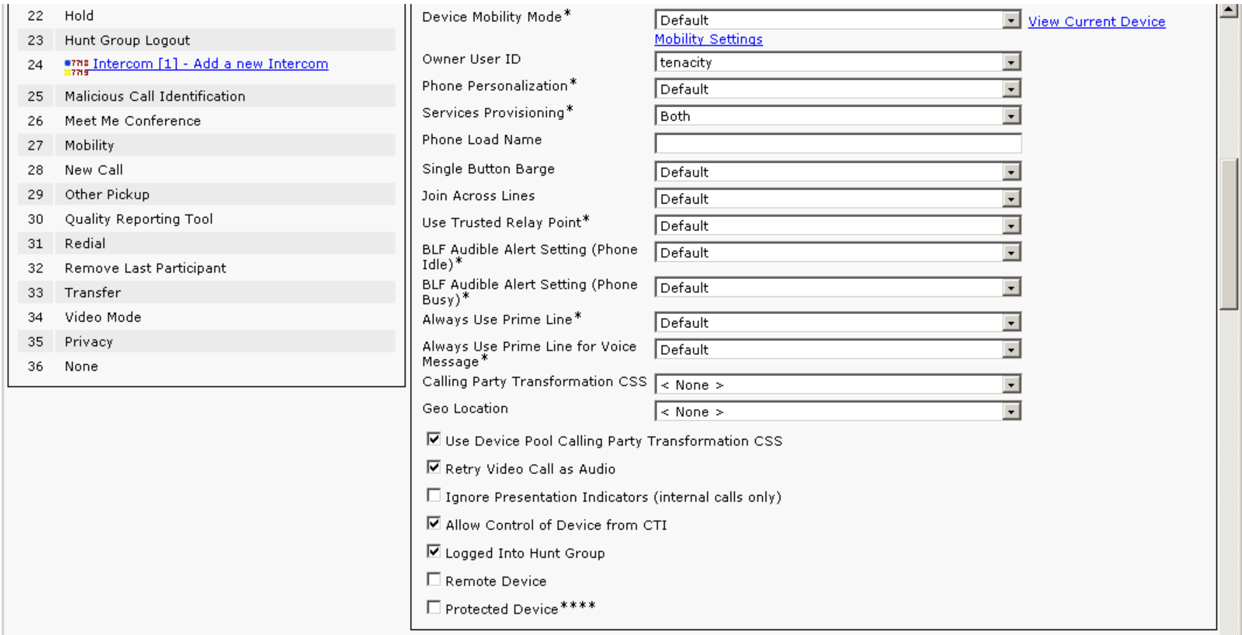
ISDN Network-Specific Facilities Information Element

Network Service Protocol	-- Not Selected --
Carrier Identification Code	

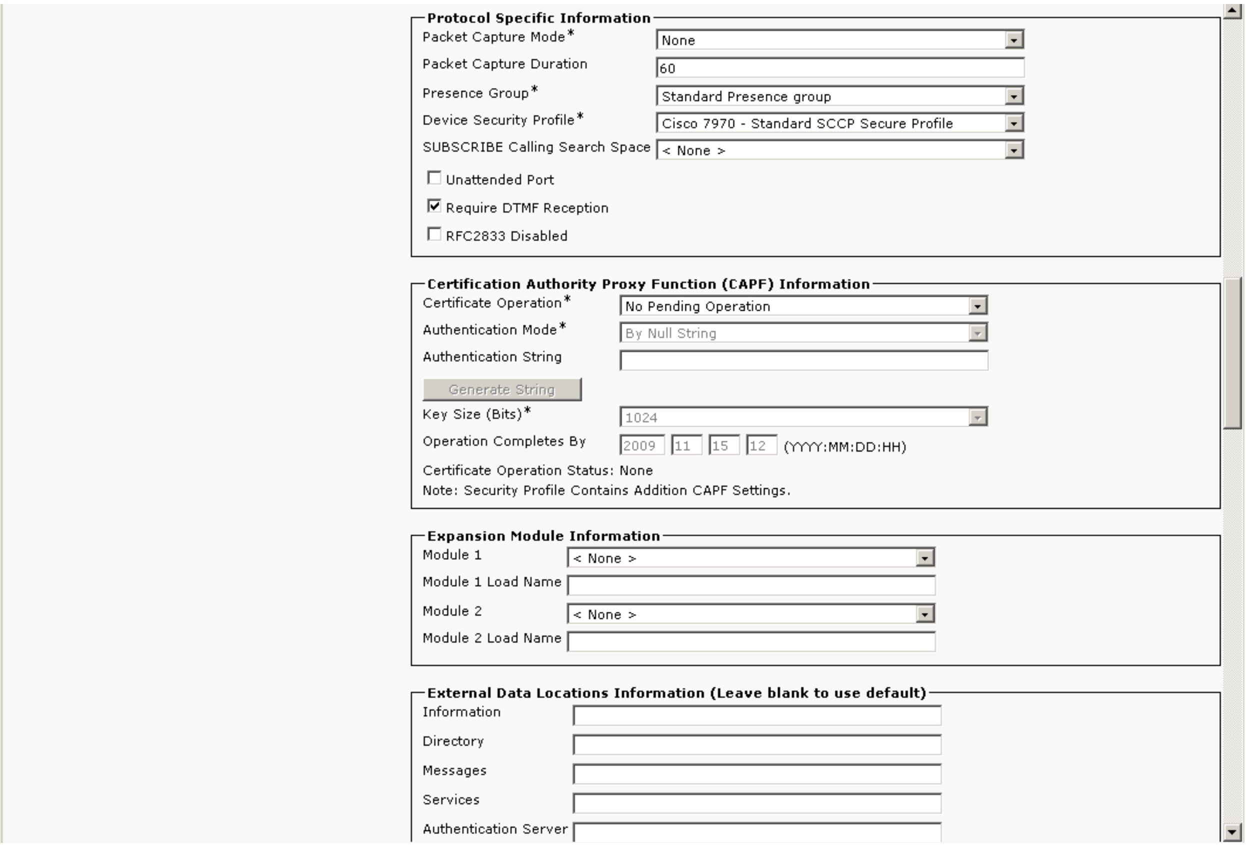
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

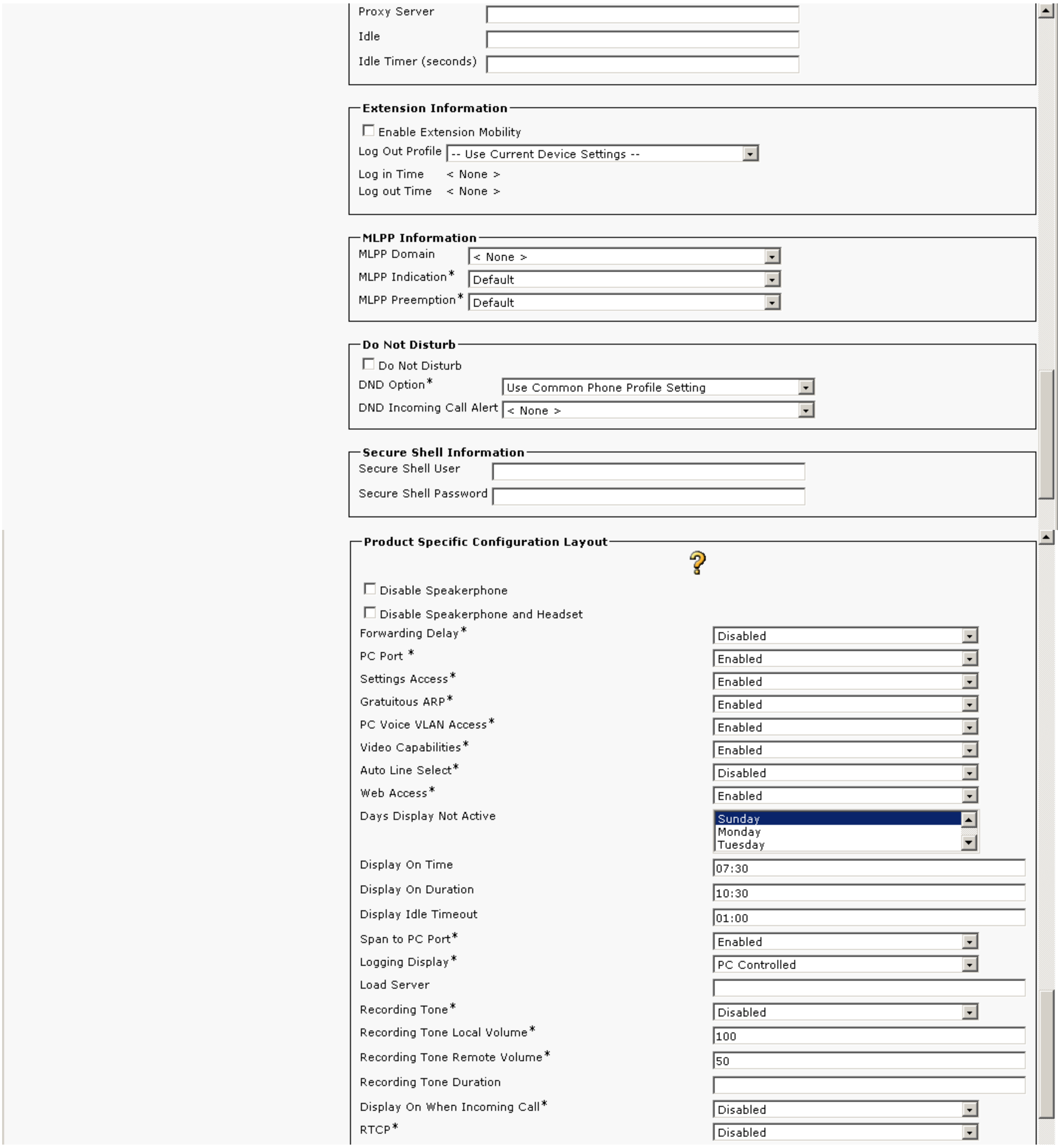
Configuration of Cisco SCCP 7970 Phone
Navigation Path: Device → Phone

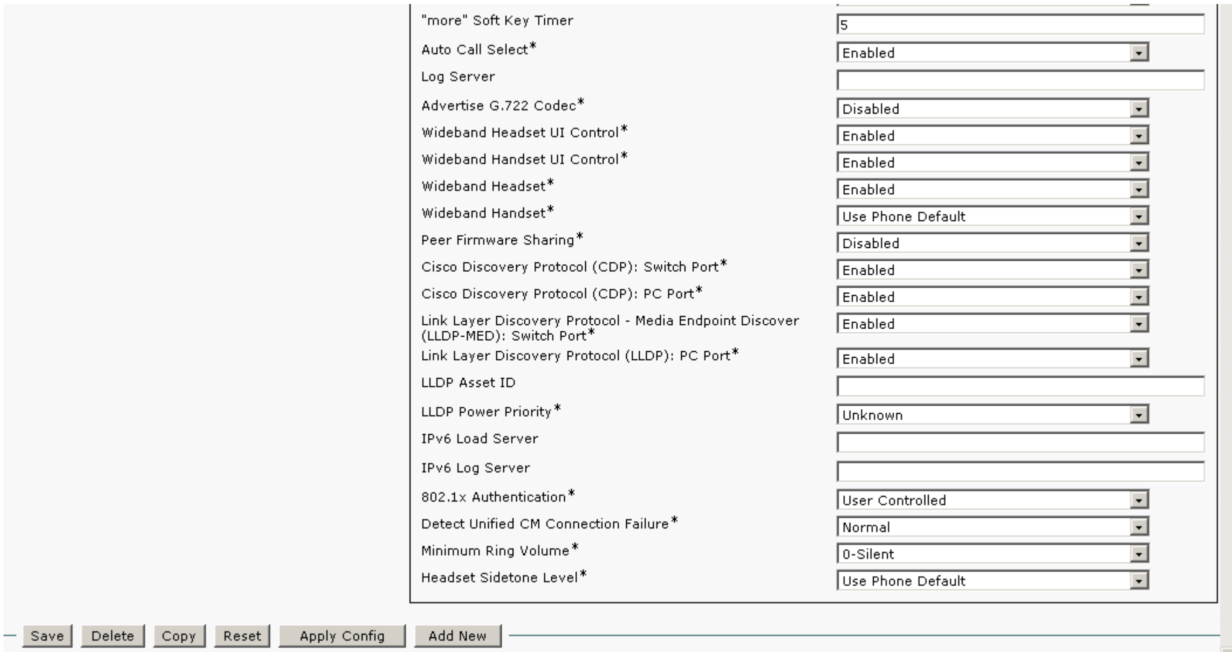




22 Hold
23 Hunt Group Logout
24 Intercom [1] - Add a new Intercom
25 Malicious Call Identification
26 Meet Me Conference
27 Mobility
28 New Call
29 Other Pickup
30 Quality Reporting Tool
31 Redial
32 Remove Last Participant
33 Transfer
34 Video Mode
35 Privacy
36 None
Device Mobility Mode*
Default
View Current Device
Mobility Settings
Owner User ID
tenacity
Phone Personalization*
Default
Services Provisioning*
Both
Phone Load Name
Single Button Barge
Default
Join Across Lines
Default
Use Trusted Relay Point*
Default
BLF Audible Alert Setting (Phone Idle)*
Default
BLF Audible Alert Setting (Phone Busy)*
Default
Always Use Prime Line*
Default
Always Use Prime Line for Voice Message*
Default
Calling Party Transformation CSS
< None >
Geo Location
< None >
Use Device Pool Calling Party Transformation CSS
Retry Video Call as Audio
Ignore Presentation Indicators (internal calls only)
Allow Control of Device from CTI
Logged Into Hunt Group
Remote Device
Protected Device****


Protocol Specific Information
Packet Capture Mode*
None
Packet Capture Duration
60
Presence Group*
Standard Presence group
Device Security Profile*
Cisco 7970 - Standard SCCP Secure Profile
SUBSCRIBE Calling Search Space
< None >
Unattended Port
Require DTMF Reception
RFC2833 Disabled
Certification Authority Proxy Function (CAPF) Information
Certificate Operation*
No Pending Operation
Authentication Mode*
By Null String
Authentication String
Generate String
Key Size (Bits)*
1024
Operation Completes By
2009 11 15 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.
Expansion Module Information
Module 1
< None >
Module 1 Load Name
Module 2
< None >
Module 2 Load Name
External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server

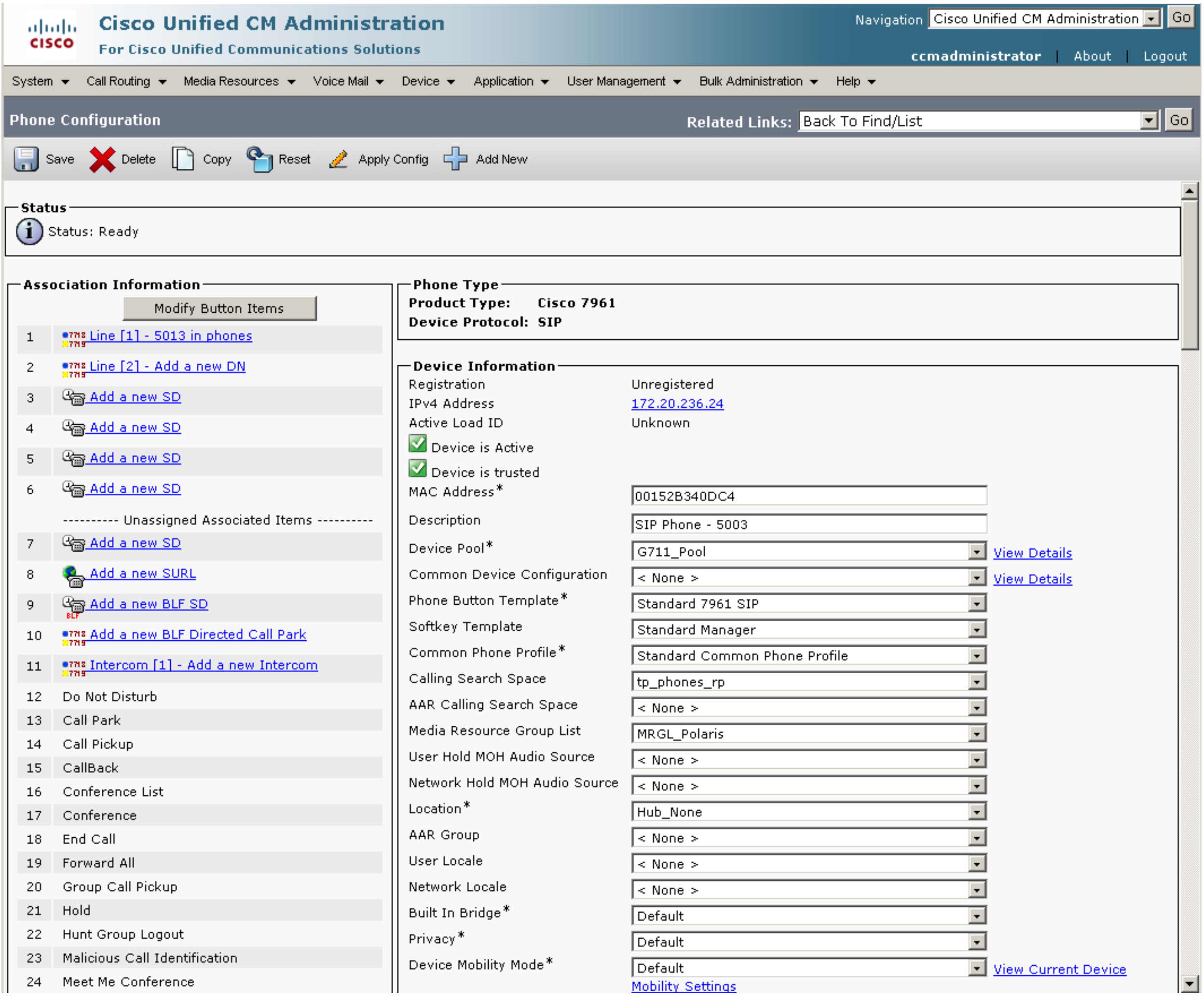

Proxy Server
Idle
Idle Timer (seconds)
Extension Information
Enable Extension Mobility
Log Out Profile -- Use Current Device Settings --
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain < None >
MLPP Indication* Default
MLPP Preemption* Default
Do Not Disturb
Do Not Disturb
DND Option* Use Common Phone Profile Setting
DND Incoming Call Alert < None >
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration Layout
Disable Speakerphone
Disable Speakerphone and Headset
Forwarding Delay* Disabled
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Enabled
PC Voice VLAN Access* Enabled
Video Capabilities* Enabled
Auto Line Select* Disabled
Web Access* Enabled
Days Display Not Active Sunday Monday Tuesday
Display On Time 07:30
Display On Duration 10:30
Display Idle Timeout 01:00
Span to PC Port* Enabled
Logging Display* PC Controlled
Load Server
Recording Tone* Disabled
Recording Tone Local Volume* 100
Recording Tone Remote Volume* 50
Recording Tone Duration
Display On When Incoming Call* Disabled
RTCP* Disabled

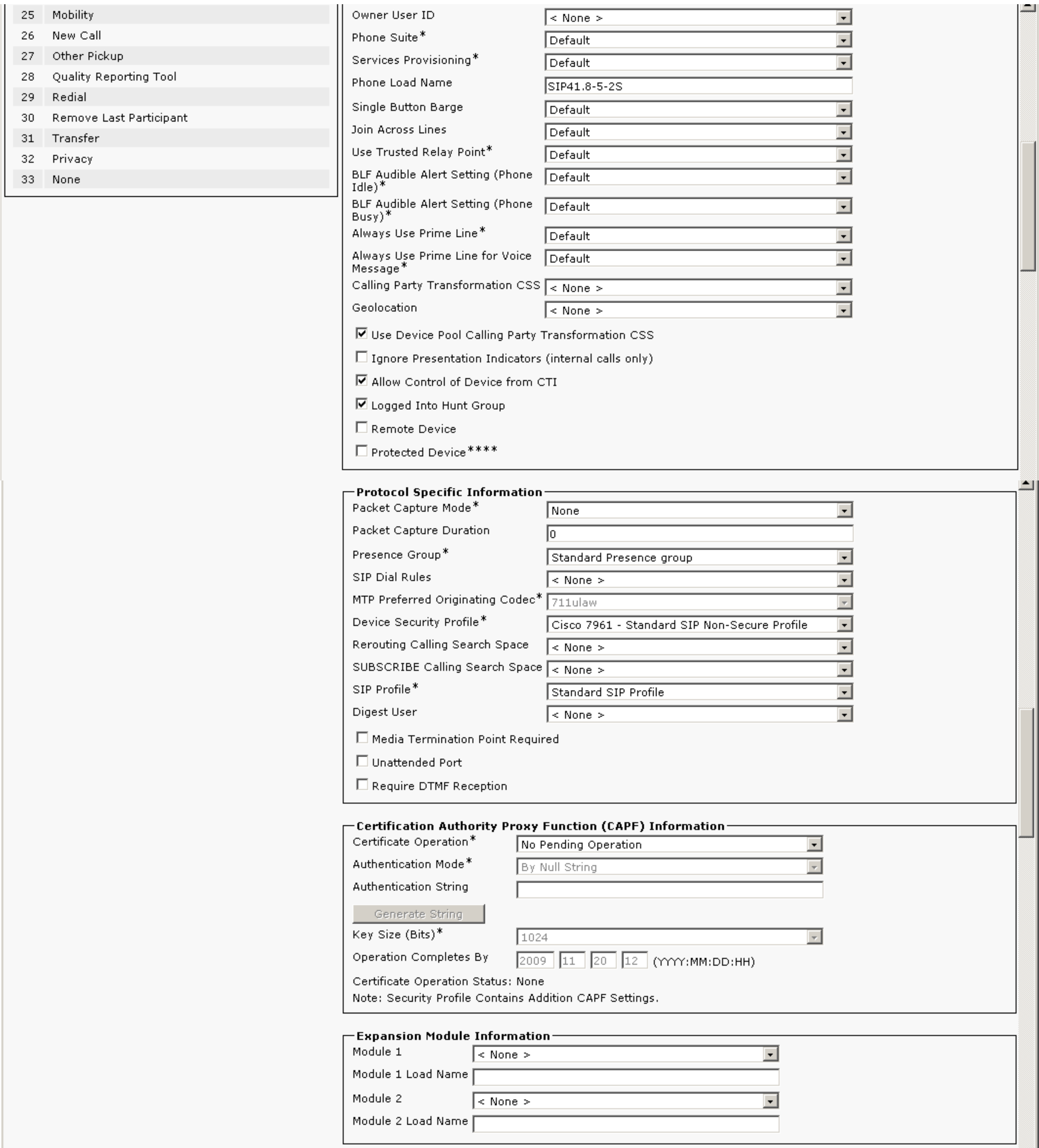


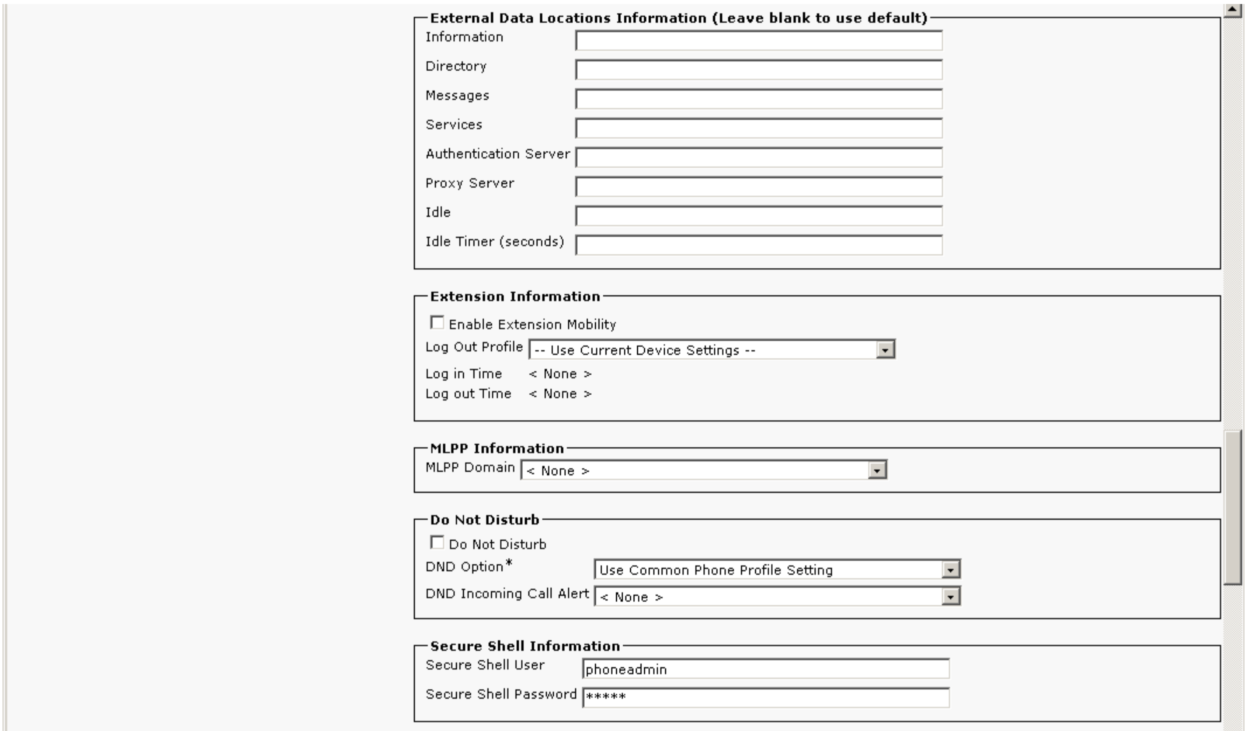
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Disabled
Wideband Headset UI Control*	Enabled
Wideband Handset UI Control*	Enabled
Wideband Headset*	Enabled
Wideband Handset*	Use Phone Default
Peer Firmware Sharing*	Disabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Use Phone Default

Save | Delete | Copy | Reset | Apply Config | Add New

Configuration of Cisco SIP 7961 Phone
Navigation Path: Device → Phone




25 Mobility
26 New Call
27 Other Pickup
28 Quality Reporting Tool
29 Redial
30 Remove Last Participant
31 Transfer
32 Privacy
33 None
Owner User ID < None >
Phone Suite* Default
Services Provisioning* Default
Phone Load Name SIP41.8-5-2S
Single Button Barge Default
Join Across Lines Default
Use Trusted Relay Point* Default
BLF Audible Alert Setting (Phone Idle)* Default
BLF Audible Alert Setting (Phone Busy)* Default
Always Use Prime Line* Default
Always Use Prime Line for Voice Message* Default
Calling Party Transformation CSS < None >
Geolocation < None >
Use Device Pool Calling Party Transformation CSS
Ignore Presentation Indicators (internal calls only)
Allow Control of Device from CTI
Logged Into Hunt Group
Remote Device
Protected Device****
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* Cisco 7961 - Standard SIP Non-Secure Profile
Rerouting Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Standard SIP Profile
Digest User < None >
Media Termination Point Required
Unattended Port
Require DTMF Reception
Certification Authority Proxy Function (CAPF) Information
Certificate Operation* No Pending Operation
Authentication Mode* By Null String
Authentication String
Generate String
Key Size (Bits)* 1024
Operation Completes By 2009 11 20 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.
Expansion Module Information
Module 1 < None >
Module 1 Load Name
Module 2 < None >
Module 2 Load Name


External Data Locations Information (Leave blank to use default)
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Extension Information
Enable Extension Mobility
Log Out Profile -- Use Current Device Settings --
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain < None >
Do Not Disturb
Do Not Disturb
DND Option* Use Common Phone Profile Setting
DND Incoming Call Alert < None >
Secure Shell Information
Secure Shell User phoneadmin
Secure Shell Password *****

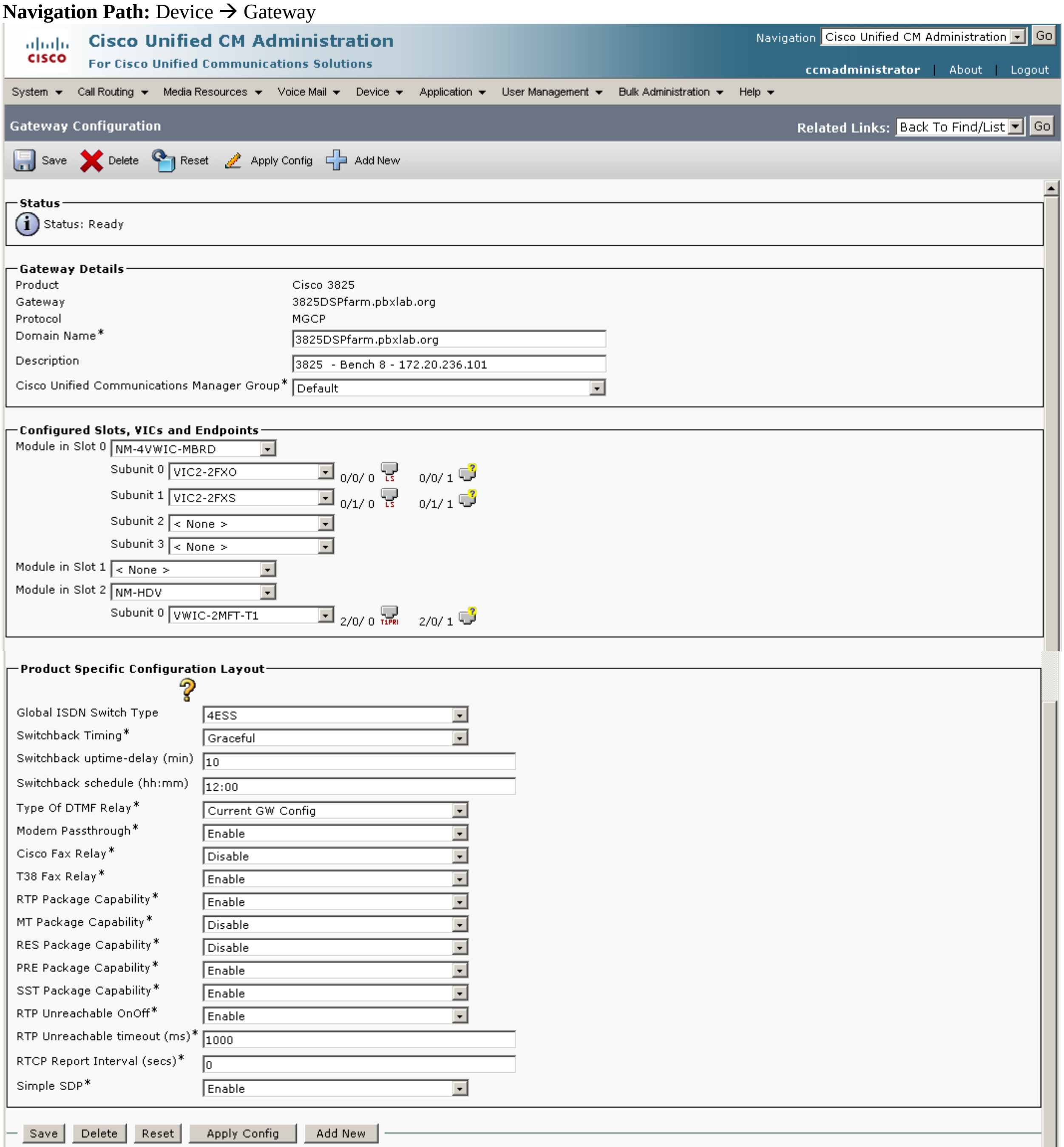
Product Specific Configuration Layout

☐ Disable Speakerphone

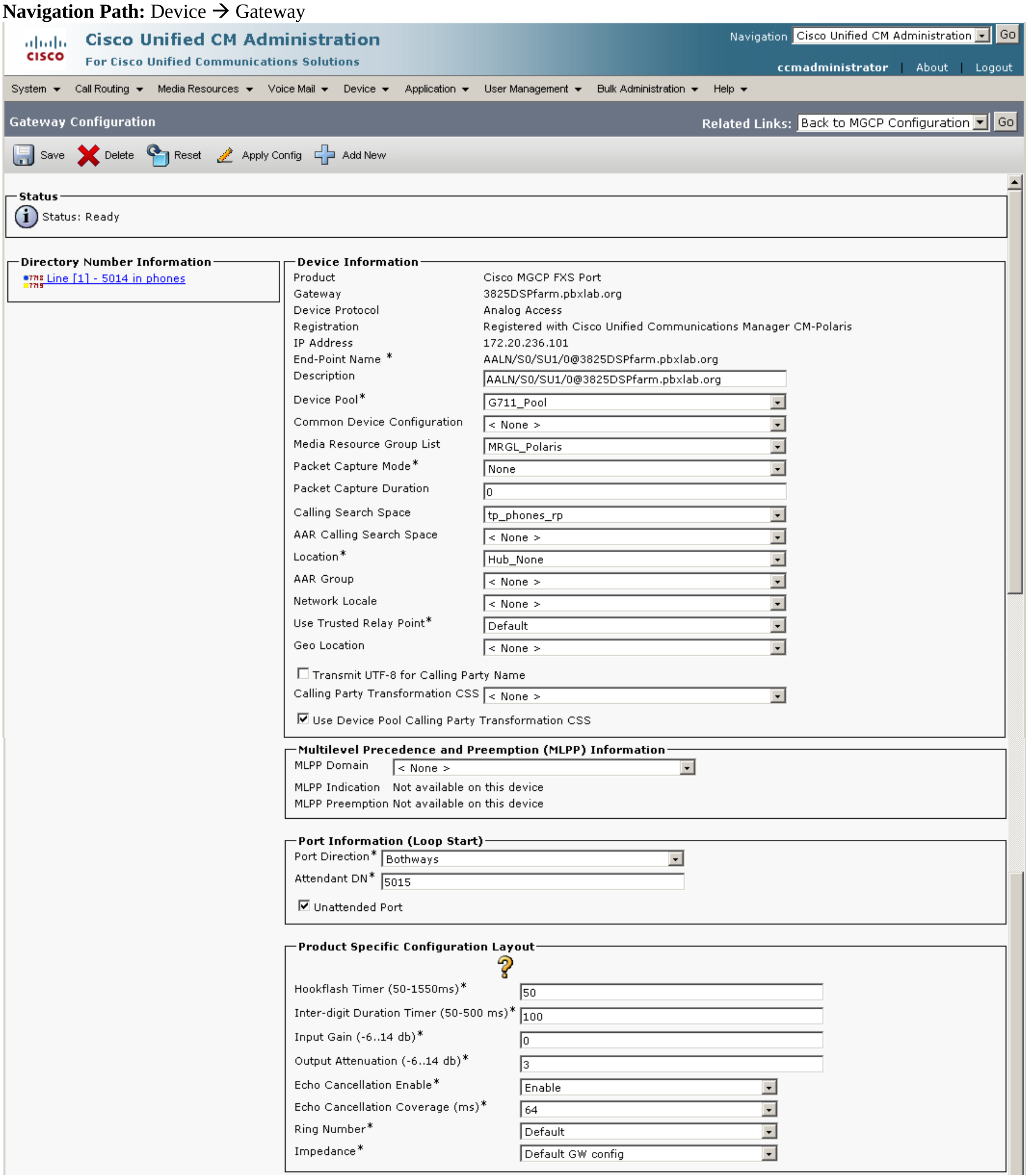
☐ Disable Speakerphone and Headset

PC Port *	Enabled
Settings Access*	Enabled
Gratuitous ARP*	Disabled
PC Voice VLAN Access*	Enabled
Auto Line Select*	Disabled
Web Access*	Enabled
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Handset UI Control*	Enabled
Wideband Headset*	Enabled
Wideband Handset*	Use Phone Default
Peer Firmware Sharing*	Disabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
Display Refresh Rate*	Normal
IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Use Phone Default
Enbloc Dialing*	Enabled

Configuration of MGCP FAX Gateway
Navigation Path: Device → Gateway



Configuration of MGCP FAX Gateway Analog Endpoint
Navigation Path: Device → Gateway



Configuring the Cisco UBE - Enterprise

```
CUBE-ASR1K_ATT#sho version
Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISEK9-M), Version
15.1(1)S, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Mon 22-Nov-10 12:18 by mcpre


ROM: IOS-XE ROMMON

CUBE-ASR1K_ATT uptime is 4 weeks, 19 hours, 10 minutes
Uptime for this control processor is 4 weeks, 19 hours, 12 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprisek9.03.02.00.S.151-1.S.bin"

cisco ASR1002 (2RU) processor with 1710464K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:.

Configuration register is 0x2102


CUBE-ASR1K_ATT#show running
Building configuration...

Current configuration : 8661 bytes
!
! Last configuration change at 17:18:57 UTC Tue Feb 1 2011
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CUBE-ASR1K_ATT
!
boot-start-marker
boot system bootflash:asr1000rp1-adventerprisek9.03.02.00.S.151-1.S.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
logging buffered 300000000
enable password cisco
!
no aaa new-model
!
!
```

```
!
ip source-route
!
!
!
no ip domain lookup
!
!
!
!
!
multilink bundle-name authenticated
!
voice service voip
 address-hiding
 allow-connections sip to sip
 redirect ip2ip
 no supplementary-service sip refer
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[1]
 h323
 sip
  header-passing error-passthru
  early-offer forced
  asserted-id pai[2]
  midcall-signaling passthru
  g729 annexb-all
!
!
voice class codec 1[3]
 codec preference 1 g729r8 bytes 20
 codec preference 2 g711ulaw bytes 160
!
!
!
redundancy
 mode none
!
!
interface GigabitEthernet0/0/0
 description Inside Interface
 ip address 172.20.110.154 255.255.255.0
 negotiation auto
!
interface GigabitEthernet0/0/1
 description connection to ATT Network
 ip address 70.xxx.xxx.xxx 255.255.255.248
 negotiation auto
!
interface GigabitEthernet0/0/2
 no ip address
 negotiation auto
```

[1] This command enables router to perform T.38 fax relay. To change fax protocol to pass-through using G.711mulaw, the command has to be changed to "fax protocol pass-through g711ulaw"
[2] This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai)
[3] This command configures the codec preference to be assigned to dial-peers. Alternatively, single codec's can be configured into individual dial-peers

```
!
interface GigabitEthernet0/0/3
 no ip address
 negotiation auto
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
!
!
no ip http server
no ip http secure-server
ip route 172.20.0.0 255.255.0.0 172.20.110.1
ip route 207.xxx.xxx.xxx 255.255.255.255 70.xxx.xxx.xxx
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 172.20.110.1
!
control-plane
!
!
!
dial-peer voice 100 voip
 description outgoing to AT&T Flexible Reach
 destination-pattern 1T
 session protocol sipv2
 session target ipv4:xxx.xxx.xxx.xxx
 voice-class codec 1
 dtmf-relay rtp-nte[4]
 fax-relay sg3-to-g3
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[1]
!
dial-peer voice 101 voip
 description incoming from AT&T to SME
 destination-pattern 732320408[45]
 session protocol sipv2
 session target ipv4:172.20.236.252
 incoming called-number 732320408[45]
 voice-class codec 1
 voice-class sip asymmetric payload dtmf[5]
 dtmf-relay rtp-nte
 fax-relay sg3-to-g3
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[1]
!
!
!
sip-ua
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 exec-timeout 5 0
 password cisco
```

---

[4] This command enables DTMF digit passing using RTP NTE (RFC2833) to calls matching this dial-peer

[5] This command enables the dial-peer to support asymmetric payload types for DTMF interworking

```
 login
!
exception data-corruption buffer truncate
end
```

Acronyms

Acronym	Definitions
ANF-PR	Additional Network Feature Path Replacement
AOC	Advice-of-charge. Information element is sent with the connection setup information for incoming Euro-ISDN connections. The AOC IE is used for call charge calculation.
Cisco UCM	Cisco Unified Communications Manager
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
MWI	Message Waiting Indicator
PSTN	Public Switched Telephone Network
SP	Service Provider

Important Information

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