Application Note

# Alcatel OmniPCX 4400 Rel. 8.0 using SIP via Cisco Unified Communications Manager–Session Manager Edition 8.5(1) to Cisco Unified Communications Manager 8.5(1)

**April 6, 2011 – Rev. 4**

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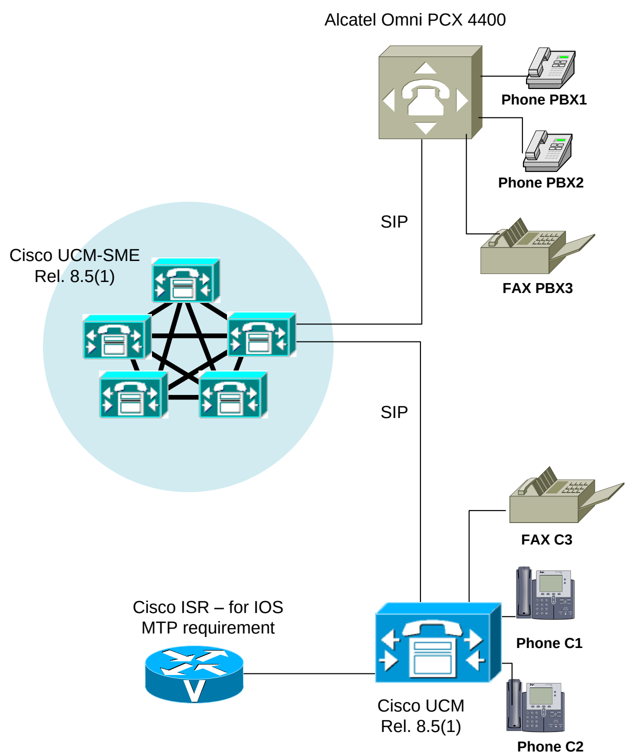
## Introduction

- This application note describes the necessary steps and configurations for connectivity between Alcatel OmniPCX 4400 Rel. 8.0, 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 Alcatel PBX via a Cisco UCM-SME using SIP trunks (between Cisco UCM-SME and Alcatel 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.

## Network Topology

Diagram of the network topology setup

**Figure 1.** Network Topology




## Capabilities

- Voice calls including supplementary services can be successfully established between endpoints controlled by the Alcatel PBX and endpoints controlled by the Cisco Unified Communications Manager. Fax calls can successfully be established when Cisco UCM fax end points initiate fax transmissions towards Alcatel fax end points. Fax calls initiated from Alcatel fax end points towards Cisco UCM fax end points fail. Please refer to the Limitations section for additional information.

## Limitations

- "Redirect by Application" checkbox must be enabled under the SIP Profile used by the SIP Trunk connecting Cisco UCM-SME to the Alcatel PBX (and a SIP Route Pattern must also be configured), in order for External Call Forwarding to work properly on Alcatel end points.
- Alcatel's implementation of SIP REFER w/Replaces does not work properly: the Refer-To header provided by the Alcatel PBX during local early-attended call transfers contains wrong Replaces dialog. Because of this issue, Cisco UCM-SME must be configured so as not to accept inbound Replaces headers.
- COLP, CONP, COLR and CONR features:
  - Alcatel Omni PCX 4400 with software release 8.0 restricts connected Name/Number delivery by sending Privacy: user on the 200 OK response message after call completion. Cisco UCM ignores any Privacy headers sent on SIP response messages, as per RFC3323 interpretation.
- Alerting Name:
  - Alcatel Omni PCX 4400 with software release 8.0 does not support Alerting Name feature support across SIP Trunk. Cisco UCM does support Alerting Name feature using "Remote-Party-Id" field. Since both systems do not interoperate with one another, both systems kept the dialed number on the phone.
- Attended Call Transfer:
  - Both systems support Attended Call Transfer feature where the transferor places the transferee on hold and calls the target. After conversing with the target, the transferor completes the transfer and drops out of both calls. The transferee is automatically taken off of hold and connected to the target. However, they are not able to update the phone displays properly after the transfer is completed. This is due to the differences between the two systems method of passing the name and number information across SIP Trunk.
- Early Attended Call Transfer:
  - Both systems support Early Attended Call Transfer feature but there are some interoperability issues with the Alcatel Omni PCX 4400 software with their SIP software stack. With Early Attended Transfer, the transferor places the original call on hold and calls the target. Upon hearing ring-back tone, the transferor transfers the call to the target and drops out of both calls. The transferee hears ring back while the target's phone is alerting. When the target answers, a connection is established between transferee and target.
  - One example of call transfer failed to complete is for Early Attended Local Call Transfer (where Alcatel phones are the transferor and the target phone). The call scenario is when Cisco UCM phone calls an Alcatel phone, and then the Alcatel phone performs early attended transfer to another Alcatel phone. During this call scenario, it appears as though the Alcatel Omni PCX is not sending Cisco UCM the correct "Replaces" header within the SIP Refer message. Because of this limitation, "Accept Replaces Header" in the SIP Trunk Security Profile assigned to the SIP trunk connecting Cisco UCM-SME to the Alcatel PBX must be disabled.
- Local Call Forwarding (CFU, CFB, and CFNA):
  - Both systems support Local Call Forwarding (CFU, CFB, and CFNA) features. Calls are forwarded properly and establish audio path. However, they are not able to update the phone display properly after the call is forwarded because the two systems have different methods of passing the name and number information.
- Network Call Forwarding (CFU, CFB, and CFNA):
  - There are interoperability issues between the two systems depending on the call flow.
  - For CFU and CFB call scenario where Alcatel station is the forwarding station, it required CISCO UCM to have the "Redirect by Application" checkbox enabled under the SIP Profile used by the SIP Trunk to the Alcatel PBX. For example, for the call flow where a Cisco UCM phone calls an Alcatel phone, and that phone is CFU or CFB back to another Cisco UCM phone, without the checkbox enabled, the call would fail. Trace analysis shows Cisco UCM sends out a regular SIP Invite message to Alcatel. Alcatel respond back with SIP 302 Moved Temporarily with Contact header sip:xxxx@yyyy, where xxxx = ext. number of forwarding target, and yyyy = IP address of the Alcatel PBX. Cisco UCM then send a new SIP Invite message to Alcatel based on the Contact header information. Alcatel responds with SIP 301 Moved Permanently with a different Contact header sip:xxxx@zzzz, where zzzz = IP address of Cisco UCM-SME. By enabling "Redirect by Application", Cisco UCM-SME will

perform digit analysis and will initiate a call to the forwarding target extension residing on Cisco UCM, using the Route Pattern configuration to properly route the call out to Cisco UCM.

- Call Conference:
  - Both systems support call conferencing using their local media resources. However, if Alcatel station is the conferencing party, local call conferences will work fine but network call conferences encounter one-way audio issues. For example, a network conference call where an Alcatel station conferences in a Cisco UCM station over the SIP trunk, one-way audio occurred between the conferenced-in party and the Alcatel station that initiated the conference. Trace analysis shows that the Alcatel PBX never sends a SIP INVITE message to signal the conferenced-in station to redirect media to the PBX's conference bridge resource after the call conference is established.
- No support for centralized voice messaging across the SIP Trunk. Cisco UCM uses SIP Diversion header to pass the redirect information across the SIP Trunk. However, Alcatel Omni PCX 4400 does not support any redirect information. Also noticed during testing is that the Alcatel PBX, upon receiving inbound SIP NOTIFY messages (sent as message waiting activation/deactivation updates for Alcatel stations), would respond with 200 OK response messages, but would not update the status of message waiting indicators on its stations.
- Cisco SIP phones hear double ringback (both local ringback and remote ringback playing simultaneously) when calling Alcatel phones. This is caused by the Alcatel PBX connecting media immediately after sending 180 Ringing message, causing the Cisco SIP phone to play local ringback while listening to the remote ringback provided by the Alcatel. This issue affects Cisco SIP phones using G.711 codec. Cisco SIP phones using G.729 codec are unaffected: although the Alcatel PBX still connects media to the Cisco SIP phone immediately, the media stream being sent only contains silence. This issue is currently under investigation. Cisco SCCP phones are unaffected by this issue.
- Fax transmissions from Cisco UCM to Alcatel fax end points can be successfully established using T.38 fax relay. Fax transmissions from Alcatel-controlled to Cisco-controlled fax end points fail. Trace analysis shows Cisco UCM sending mid-call INVITE to negotiate T.38 fax containing SDP with connection information = 0.0.0.0. This media attribute seems to be causing the Alcatel PBX not to properly connect the media after a second INVITE is sent by Cisco UCM with valid connection information. Fax Pass-Through is not supported by the Alcatel PBX.

## 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
- Alcatel Omni PCX 4400 PBX with INT-IP2 card
- Alcatel digital and IP phones

### Software Requirements

- Cisco Unified Communications Manager Release 8.5(1) - Session Manager Edition
- Cisco Unified Communications Manager Release 8.5(1) – Cisco UCM
- Alcatel software R8.0 (g1.503)

## Features Supported

- CLIP-Calling Line (Number) Identification Presentation (Refer to Limitations section)
- CLIR-Calling Line (Number) Identification Restriction (Refer to Limitations section)
- CNIP-Calling Name Identification Presentation (Refer to Limitations section)
- CNIR-Calling Name Identification Restriction (Refer to Limitations section)
- Alerting Name (Refer to Limitations section)

- Attended Call Transfer (Refer to Limitations section)
- Early Attended Call Transfer (Refer to Limitations section)
- CFU-Call Forwarding Unconditional  (Refer to Limitations section)
- CFB-Call Forwarding Busy  (Refer to Limitations section)
- CFNA-Call Forwarding No Answer  (Refer to Limitations section)
- COLP-Connected Line (Number) Identification Presentation  (Refer to Limitations section)
- COLR- Connected Line (Number) Identification Restriction  (Refer to Limitations section)
- CONP-Connected Name Identification Presentation  (Refer to Limitations section)
- CONR-Connected Name Identification Restriction   (Refer to Limitations section)
- Hold and Resume
- Conference Call  (Refer to Limitations section)
- DTMF-relay using RFC2833

### Features Not Supported

- Centralized Voicemail
- Call Completion (Callback; Automatic Callback)
- SIP Blind Call Transfer
- Inbound Fax to Cisco UCM  (Refer to Limitations section)

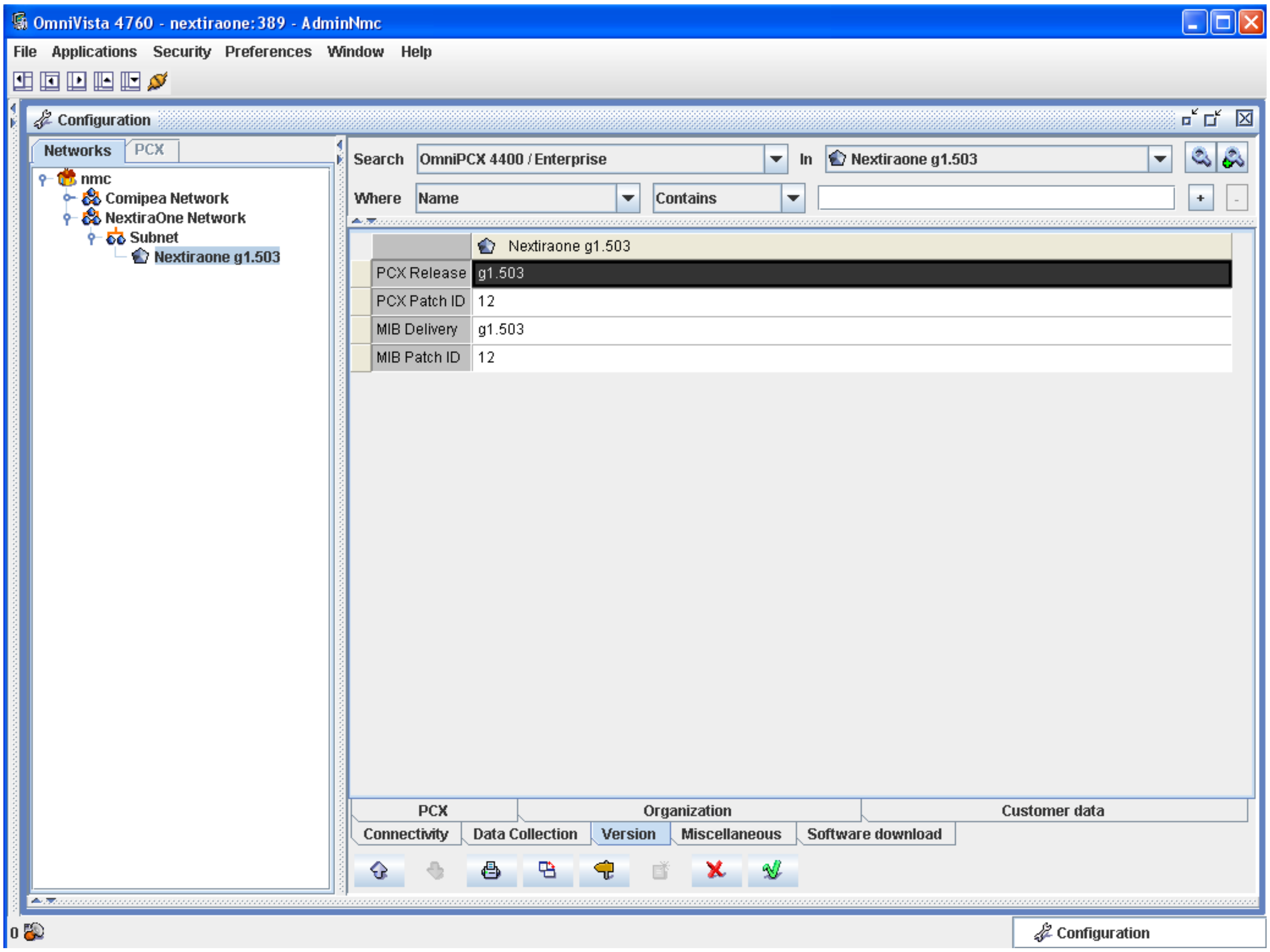
## Configuration

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

### Configurating the Alcatel Omni PCX 4400

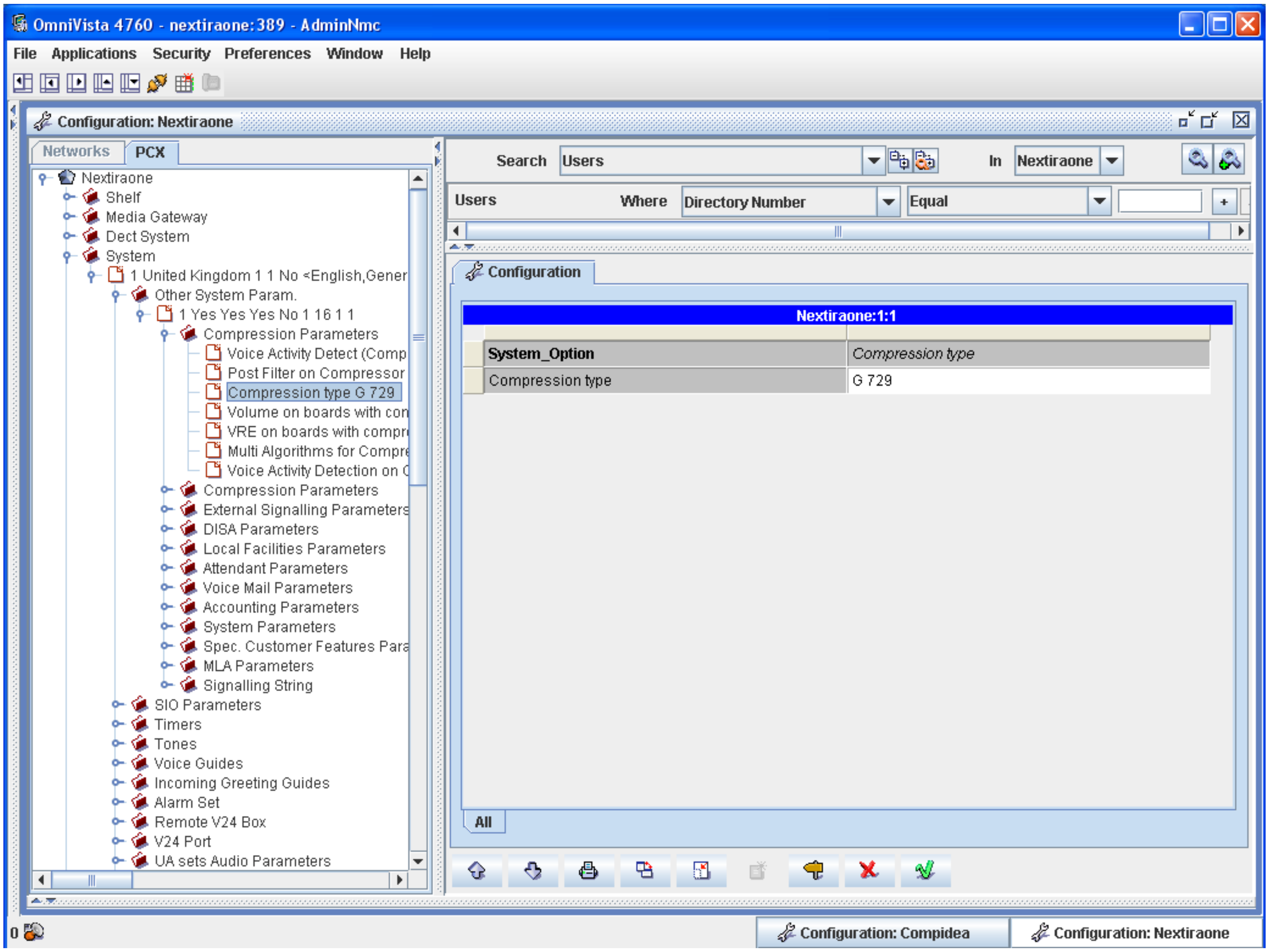
1. Alcatel Omni PCX 4400 Software Version and Hardware Configuration List
2. Configure System → Other System Parameters → Compression Type
3. Configure IP → IP Domain → Extra Domain Without Compression
4. Configure SIP Network: Translator → Network Routing Table
5. Configure SIP Trunk group
6. Configure T2 Trunk Group Type
7. Configure Virtual Access for SIP
8. Configure Alcatel SIP Gateway
9. Configure Alcatel SIP Proxy setting
10. Configure SIP External Gateway
11. Configure IP Parameters
12. Configure GF diversion on joining
13. Configure call routing (Translator) to Cisco UCM phone extensions
14. Configure Alcatel standard users

Alcatel Omni PCX 4400 Software Version and Hardware Configuration List:



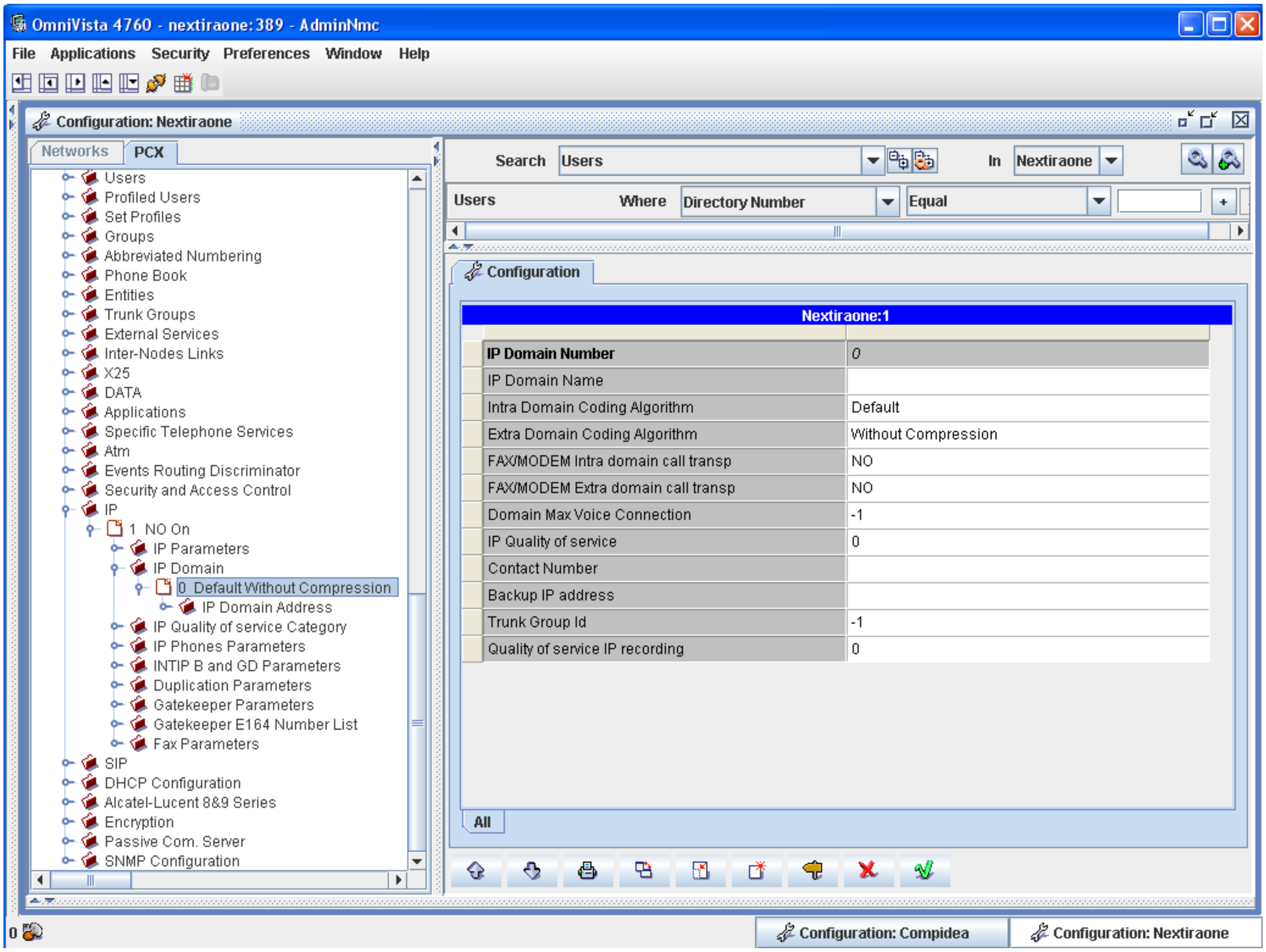
Configure System → Other System Parameters → Compression Type

- Ensure the Compression Type used by the system is set to G.729



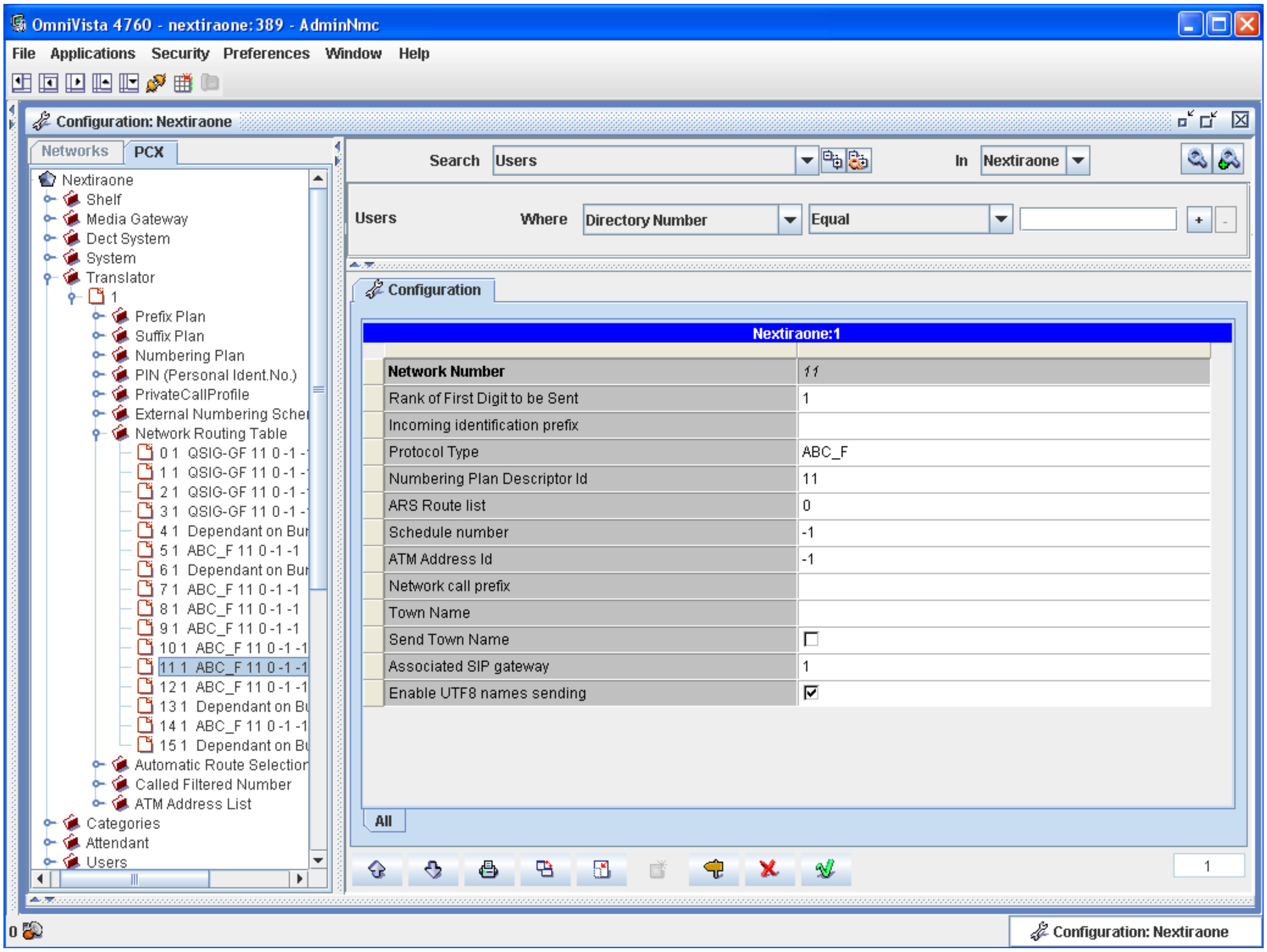
Configure IP → IP Domain → Extra Domain Coding Algorithm Without Compression

- Ensure the Extra Domain Coding Algorithm is set to “Without Compression”



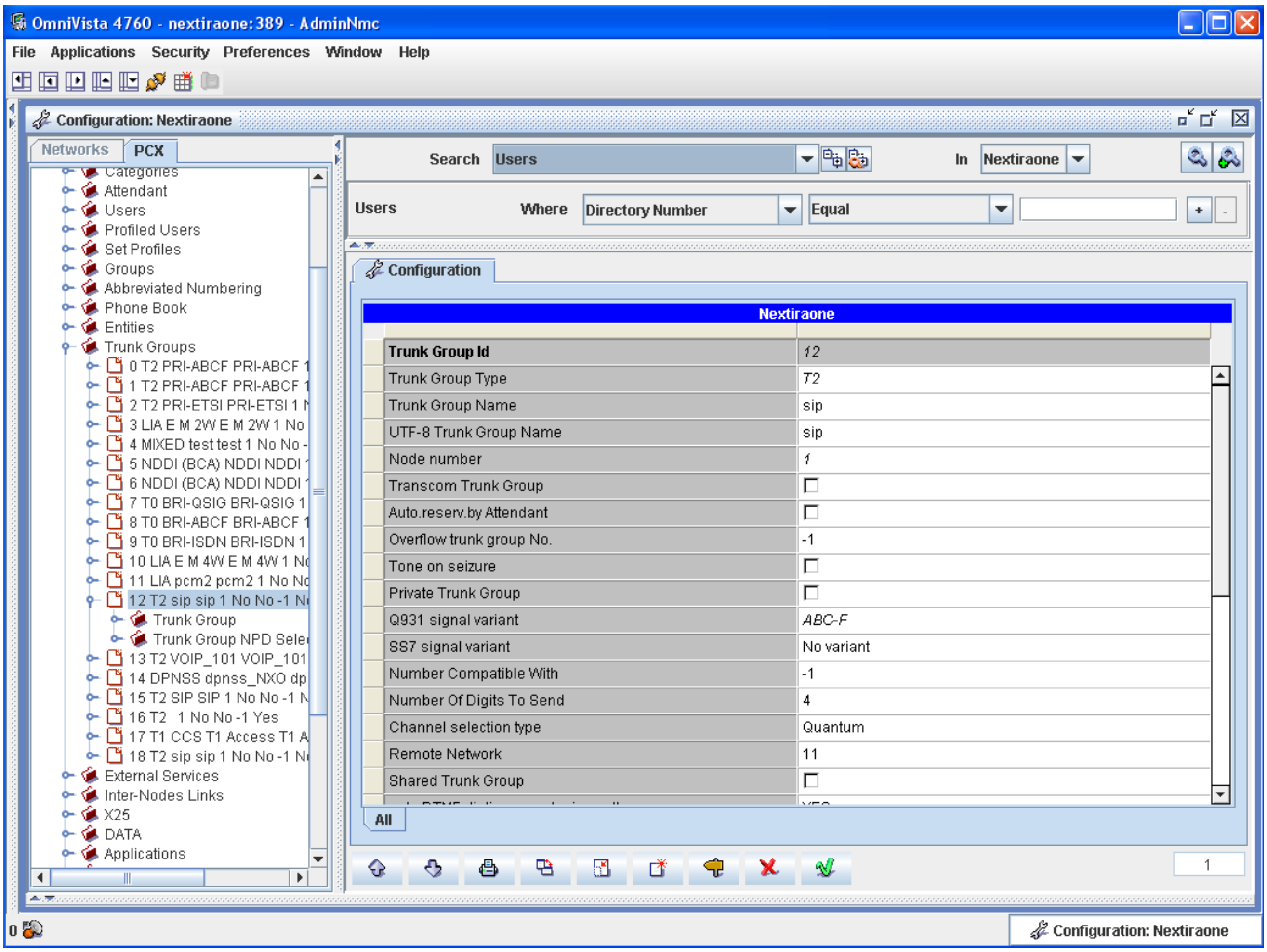
Configure SIP Network: Translator → Network Routing Table:

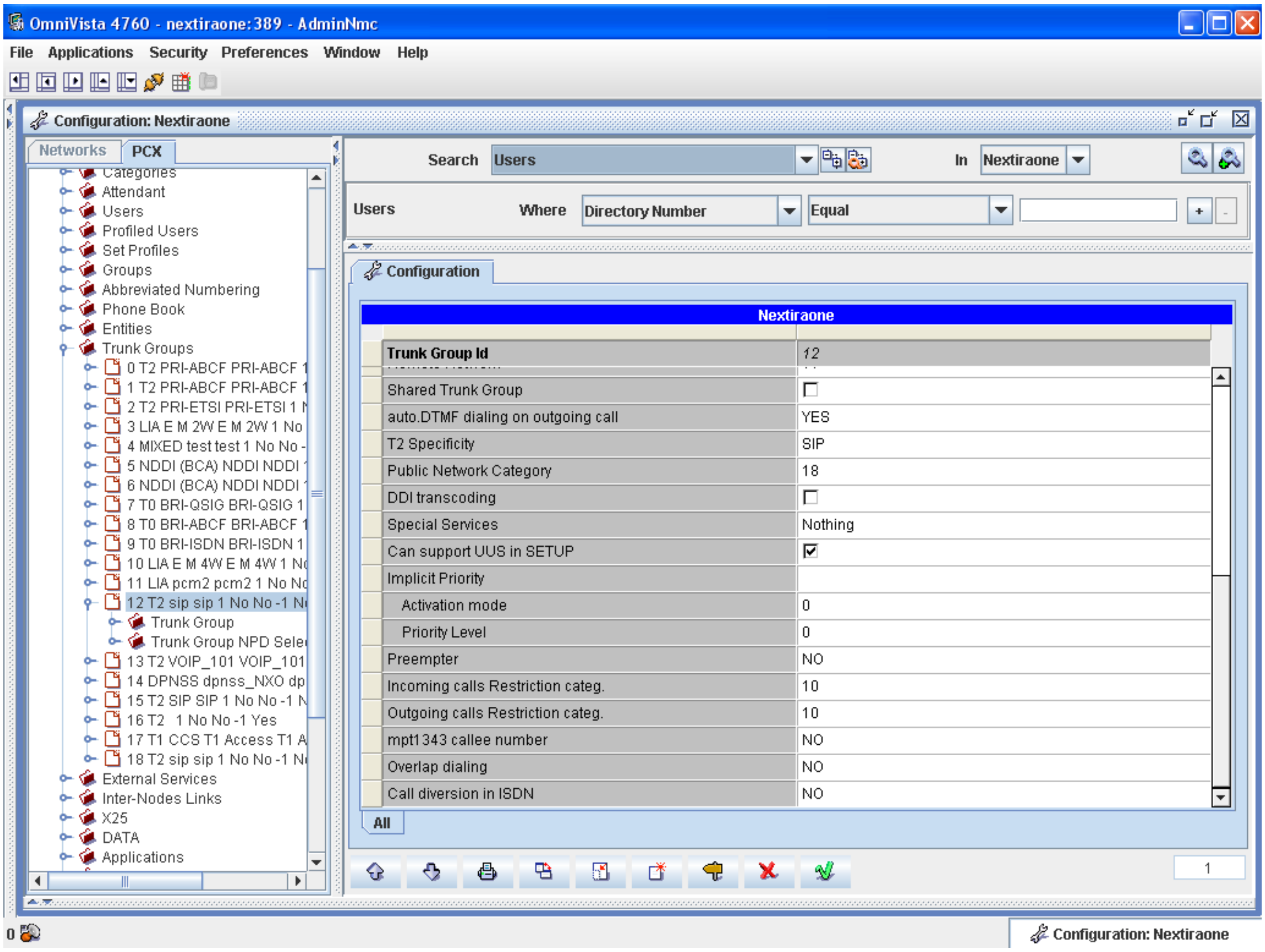
- Ensure the sub-network number used by SIP sets and SIP trunk group have the "Protocol Type = ABC_F"



Configure SIP Trunk group:

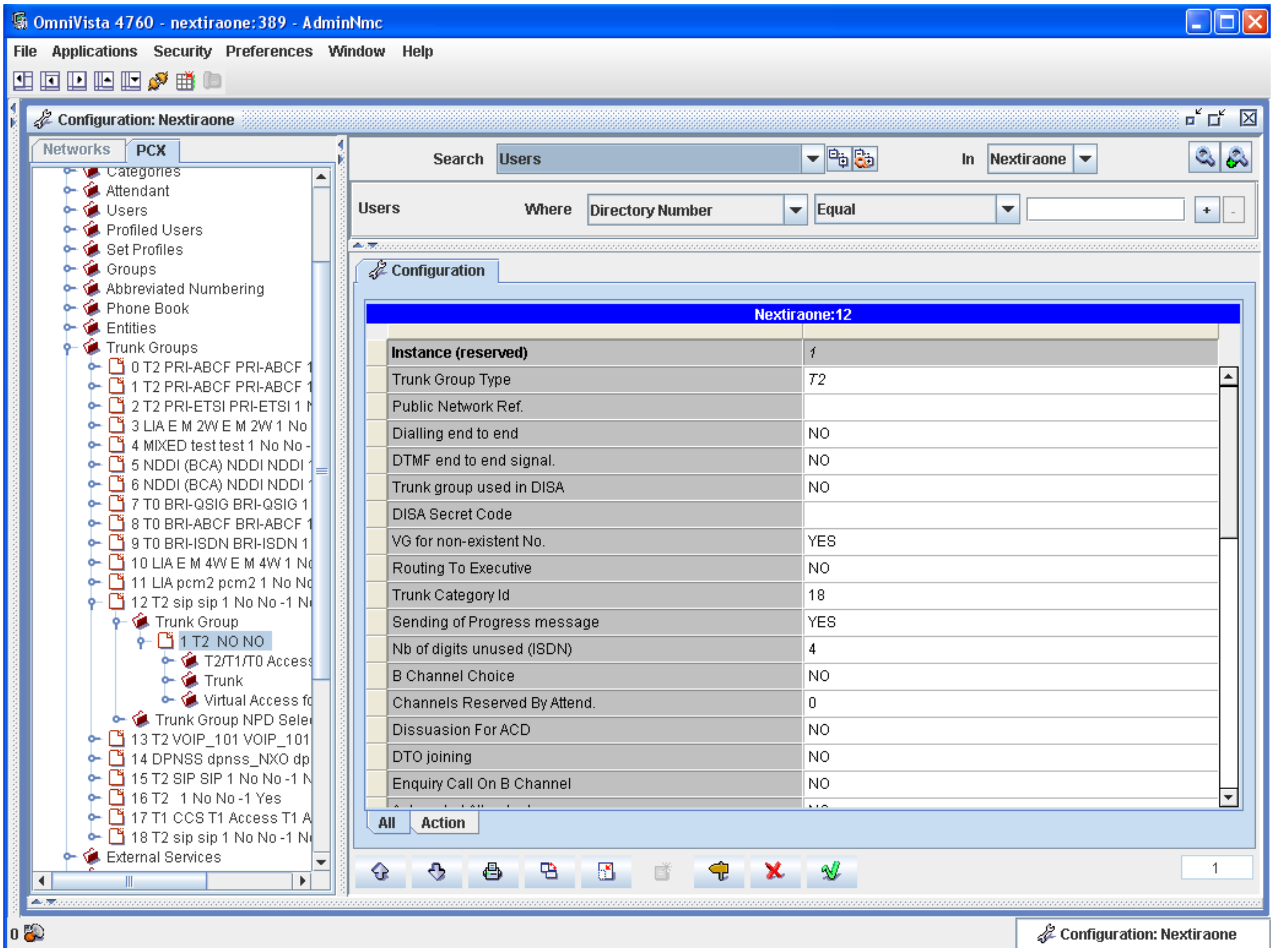
| | |
|---|---|
| Trunk Group ID: | Enter the trunk group number |
| Trunk Group Type: | T2 |
| Remote Network: | Enter the sub-network number associated with the trunk group. |
| Node number: | Enter the node number |
| Q931 signal variant: | Select ABC-F for the main SIP trunk group |
| T2 Specification: | SIP |
| Overlap dialing: | No |

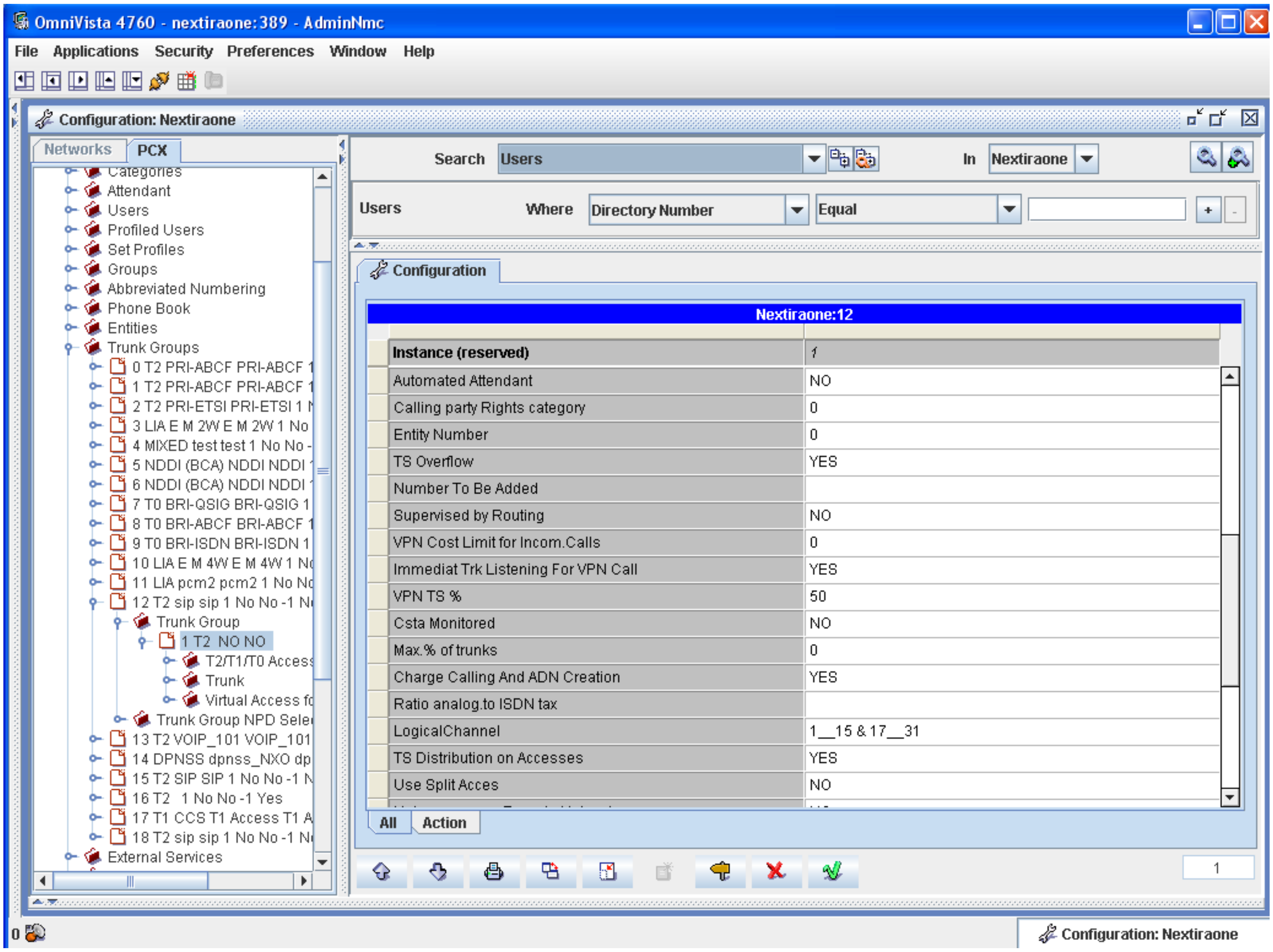



OmniVista 4760 - nextiraone:389 - AdminNmc
File Applications Security Preferences Window Help
Configuration: Nextiraone
Networks PCX
Categories
Attendant
Users
Profiled Users
Set Profiles
Groups
Abbreviated Numbering
Phone Book
Entities
Trunk Groups
0 T2 PRI-ABCF PRI-ABCF 1
1 T2 PRI-ABCF PRI-ABCF 1
2 T2 PRI-ETSI PRI-ETSI 1
3 LIA E M 2W E M 2W 1 No
4 MIXED test test 1 No No
5 NDDI (BCA) NDDI NDDI
6 NDDI (BCA) NDDI NDDI
7 T0 BRI-QSIG BRI-QSIG 1
8 T0 BRI-ABCF BRI-ABCF 1
9 T0 BRI-ISDN BRI-ISDN 1
10 LIA E M 4W E M 4W 1
11 LIA pcm2 pcm2 1 No
12 T2 sip sip 1 No No -1
Trunk Group
Trunk Group NPD Sele
13 T2 VOIP_101 VOIP_101
14 DPNSS dpnss_NXO dp
15 T2 SIP SIP 1 No No -1
16 T2 1 No No -1 Yes
17 T1 CCS T1 Access T1
18 T2 sip sip 1 No No -1
External Services
Inter-Nodes Links
X25
DATA
Applications
Search Users In Nextiraone
Users Where Directory Number Equal
Configuration
Nextiraone
Trunk Group Id 12
Shared Trunk Group
auto.DTMF dialing on outgoing call YES
T2 Specificity SIP
Public Network Category 18
DDI transcoding
Special Services Nothing
Can support UUS in SETUP
Implicit Priority
Activation mode 0
Priority Level 0
Preempter NO
Incoming calls Restriction categ. 10
Outgoing calls Restriction categ. 10
mpt1343 callee number NO
Overlap dialing NO
Call diversion in ISDN NO
All
1
0
Configuration: Nextiraone

Configure T2 Trunk Group Type:

IP Compression Type: G.711

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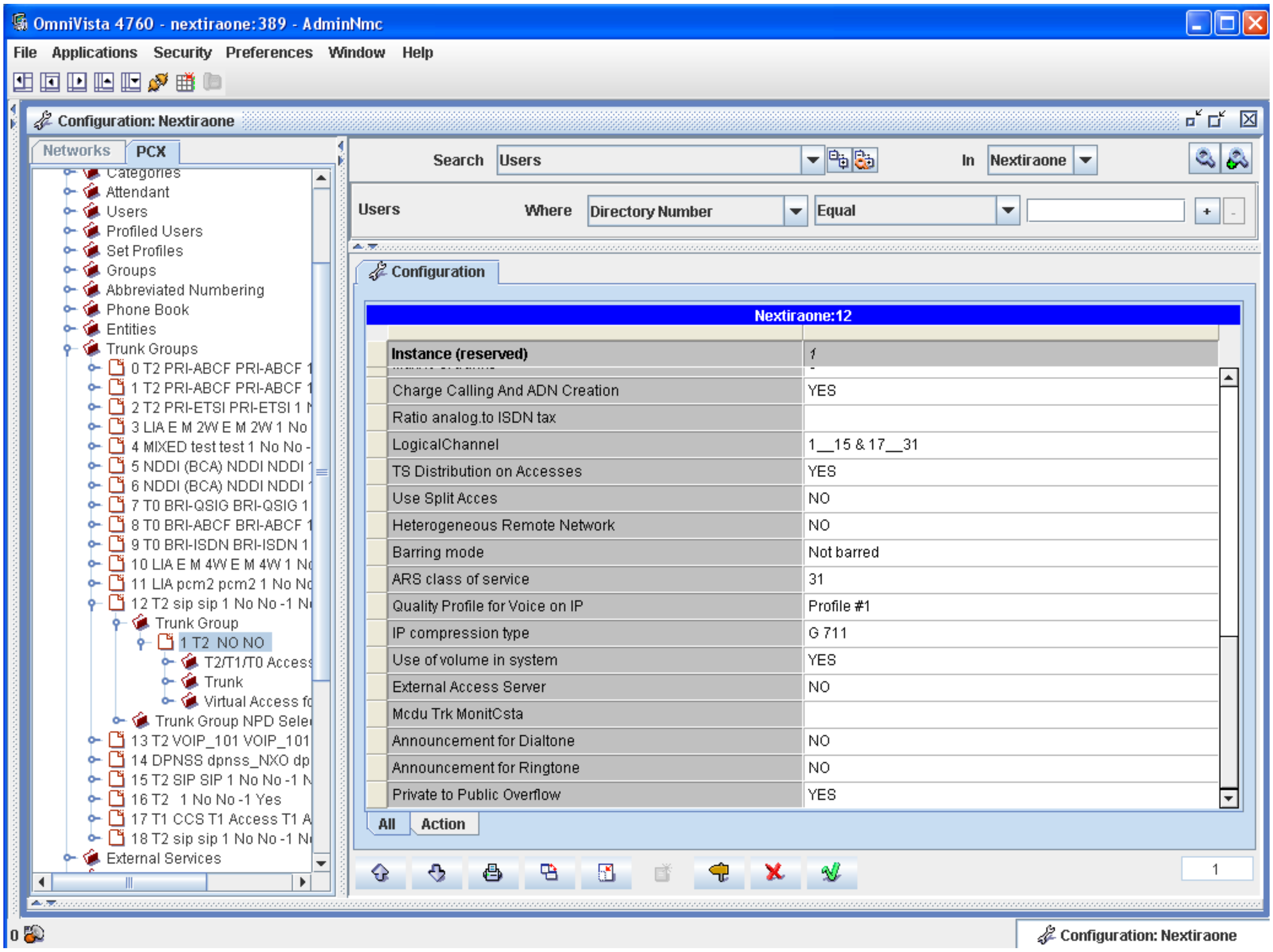
6 NDDI (BCA) NDDI NDDI 1
7 T0 BRI-QSIG BRI-QSIG 1
8 T0 BRI-ABCF BRI-ABCF 1
9 T0 BRI-ISDN BRI-ISDN 1
10 LIA E M 4W E M 4W 1 No
11 LIA pcm2 pcm2 1 No No
12 T2 sip sip 1 No No -1 No
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual Access fo
Trunk Group NPD Selec
13 T2 VOIP_101 VOIP_101
14 DPNSS dpnss_NXO dp
15 T2 SIP SIP 1 No No -1 N
16 T2 1 No No -1 Yes
17 T1 CCS T1 Access T1 A
18 T2 sip sip 1 No No -1 No
External Services
Search Users In Nextiraone
Users Where Directory Number Equal
Configuration
Nextiraone:12
Instance (reserved) 1
Automated Attendant NO
Calling party Rights category 0
Entity Number 0
TS Overflow YES
Number To Be Added
Supervised by Routing NO
VPN Cost Limit for Incom.Calls 0
Immediat Trk Listening For VPN Call YES
VPN TS % 50
Csta Monitored NO
Max.% of trunks 0
Charge Calling And ADN Creation YES
Ratio analog.to ISDN tax
LogicalChannel 1__15 & 17__31
TS Distribution on Accesses YES
Use Split Acces NO
All Action
1
0
Configuration: Nextiraone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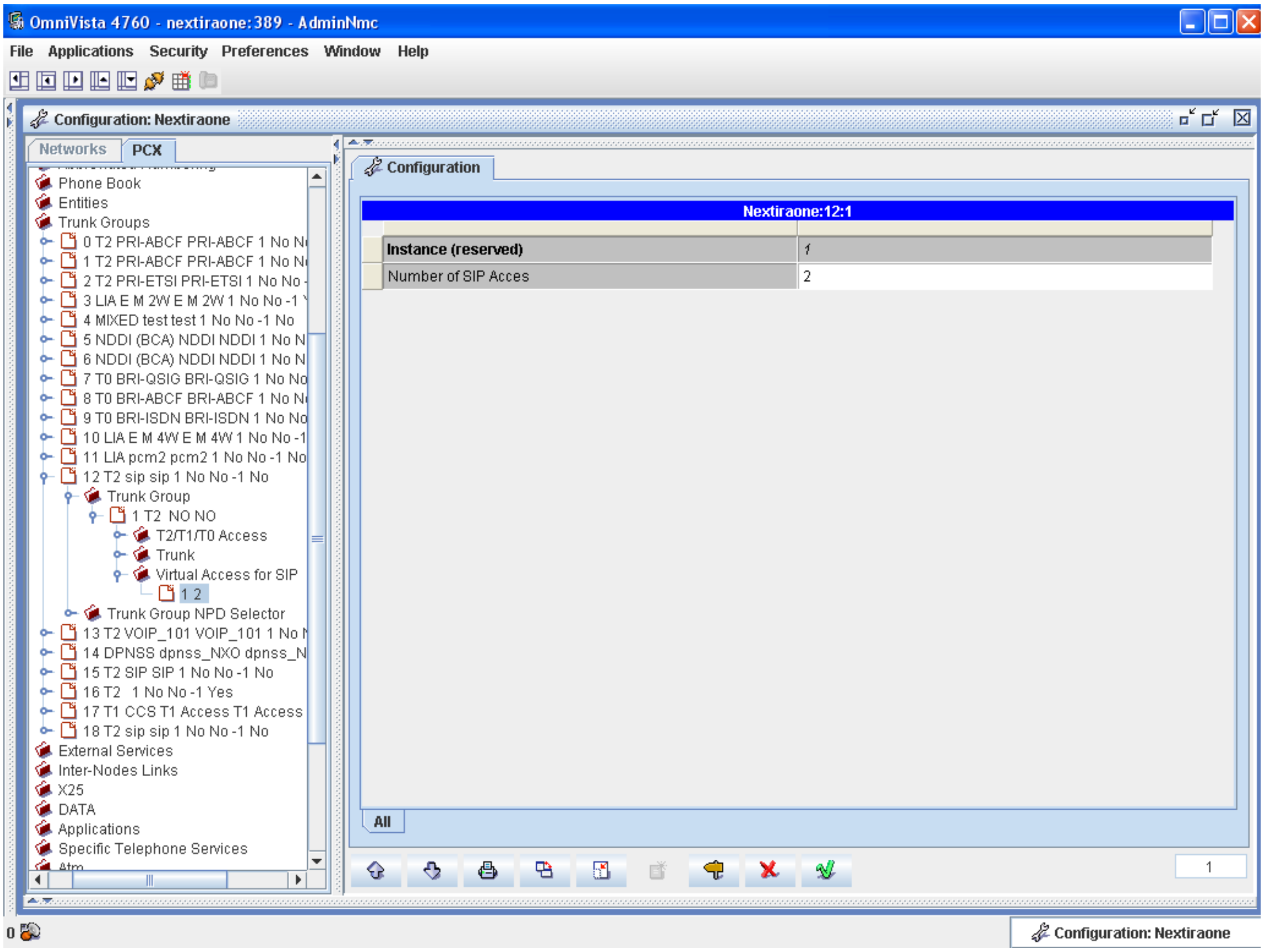
11 LIA pcm2 pcm2 1 No
12 T2 sip sip 1 No No -1
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual Access
Trunk Group NPD Sele
13 T2 VOIP_101 VOIP_101
14 DPNSS dpnss_NXO dp
15 T2 SIP SIP 1 No No -1
16 T2 1 No No -1 Yes
17 T1 CCS T1 Access T1
18 T2 sip sip 1 No No -1
External Services
Search Users In Nextiraone
Users Where Directory Number Equal
Configuration
Nextiraone:12
Instance (reserved) 1
Charge Calling And ADN Creation YES
Ratio analog.to ISDN tax
LogicalChannel 1__15 & 17__31
TS Distribution on Accesses YES
Use Split Acces NO
Heterogeneous Remote Network NO
Barring mode Not barred
ARS class of service 31
Quality Profile for Voice on IP Profile #1
IP compression type G 711
Use of volume in system YES
External Access Server NO
Mcdu Trk MonitCsta
Announcement for Dialtone NO
Announcement for Ringtone NO
Private to Public Overflow YES
All Action
1
Configuration: Nextiraone

Configure Virtual Access for SIP:

Number of SIP Access: When a SIP trunk group is created, a pair of accesses is automatically created.

Note: Two SIP accesses allow 60 simultaneous calls on the trunk group.



Configure Alcatel SIP Gateway:

This is Alcatel SIP Call Server configuration:

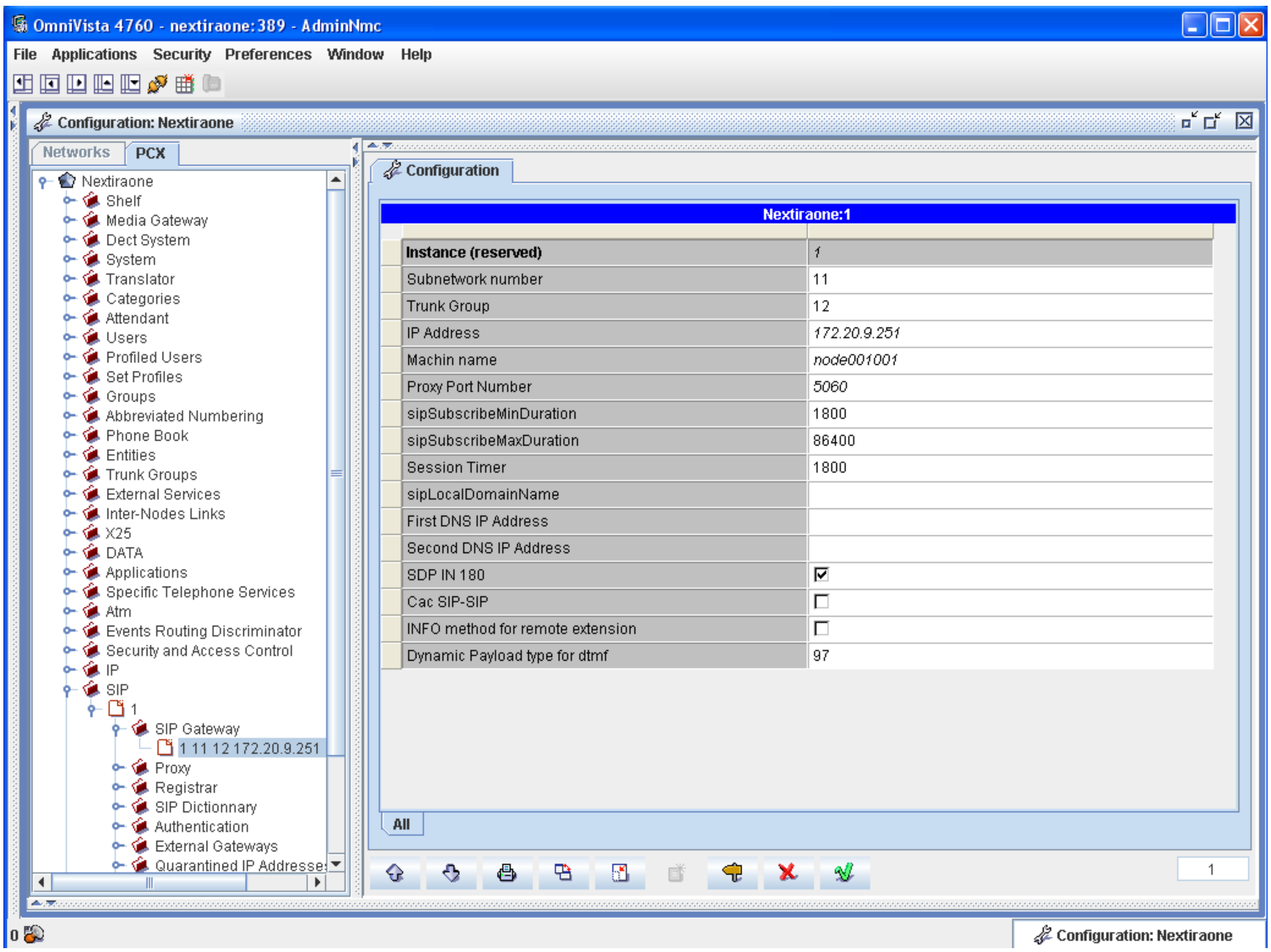
SIP Subnetwork: Enter the sub-network number used by SIP sets and SIP trunk group

SIP Trunk Group: Enter the SIP trunk group number

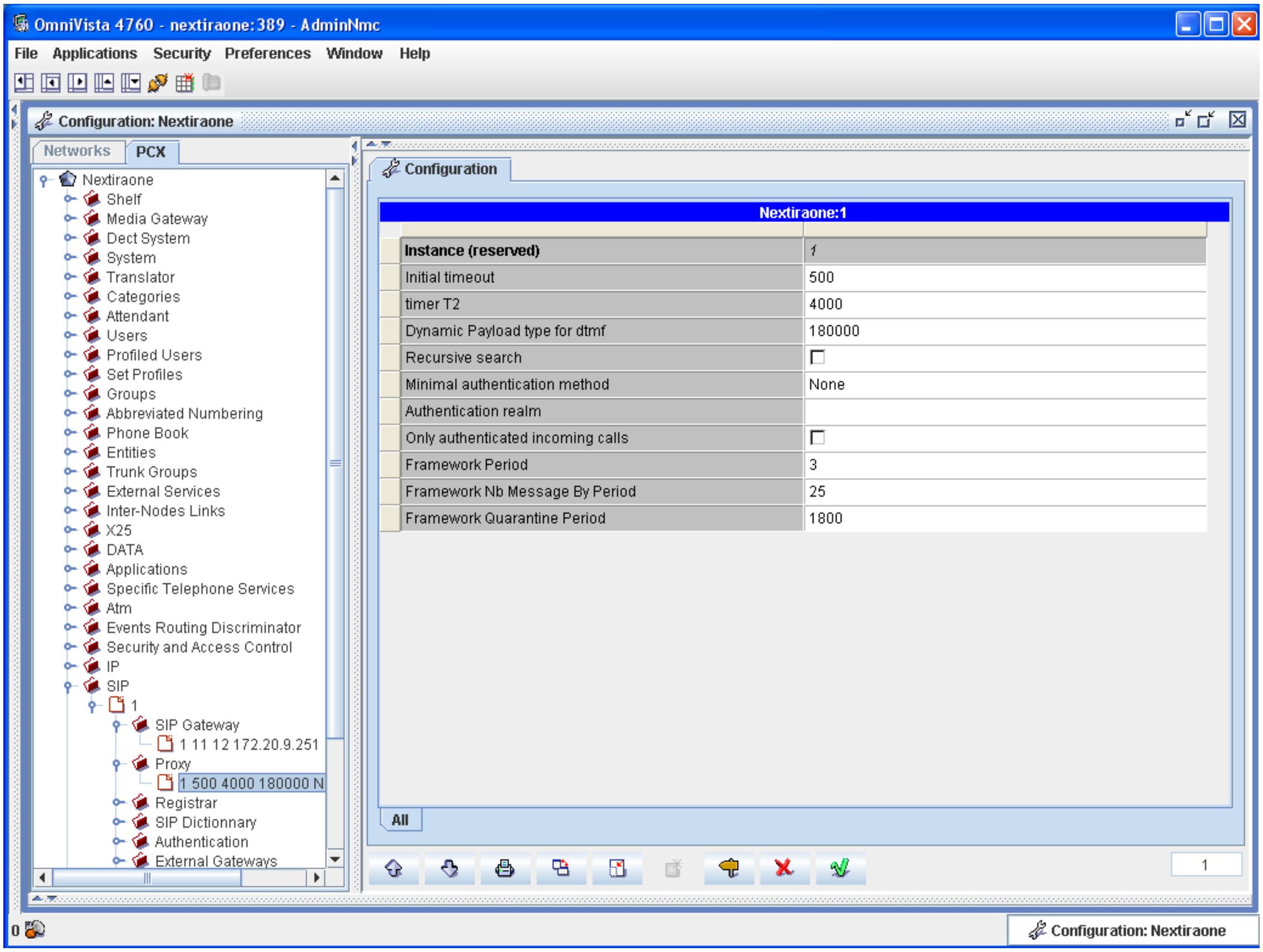
IP Address: Enter the IP Address of the Alcatel Call Server

SIP Port Number: Enter the TCP or UDP port number use for SIP signaling message

SIP Proxy Port Number: Enter the SIP Proxy TCP/UDP port number



Configure Alcatel SIP Proxy setting:



Configure SIP External Gateway:

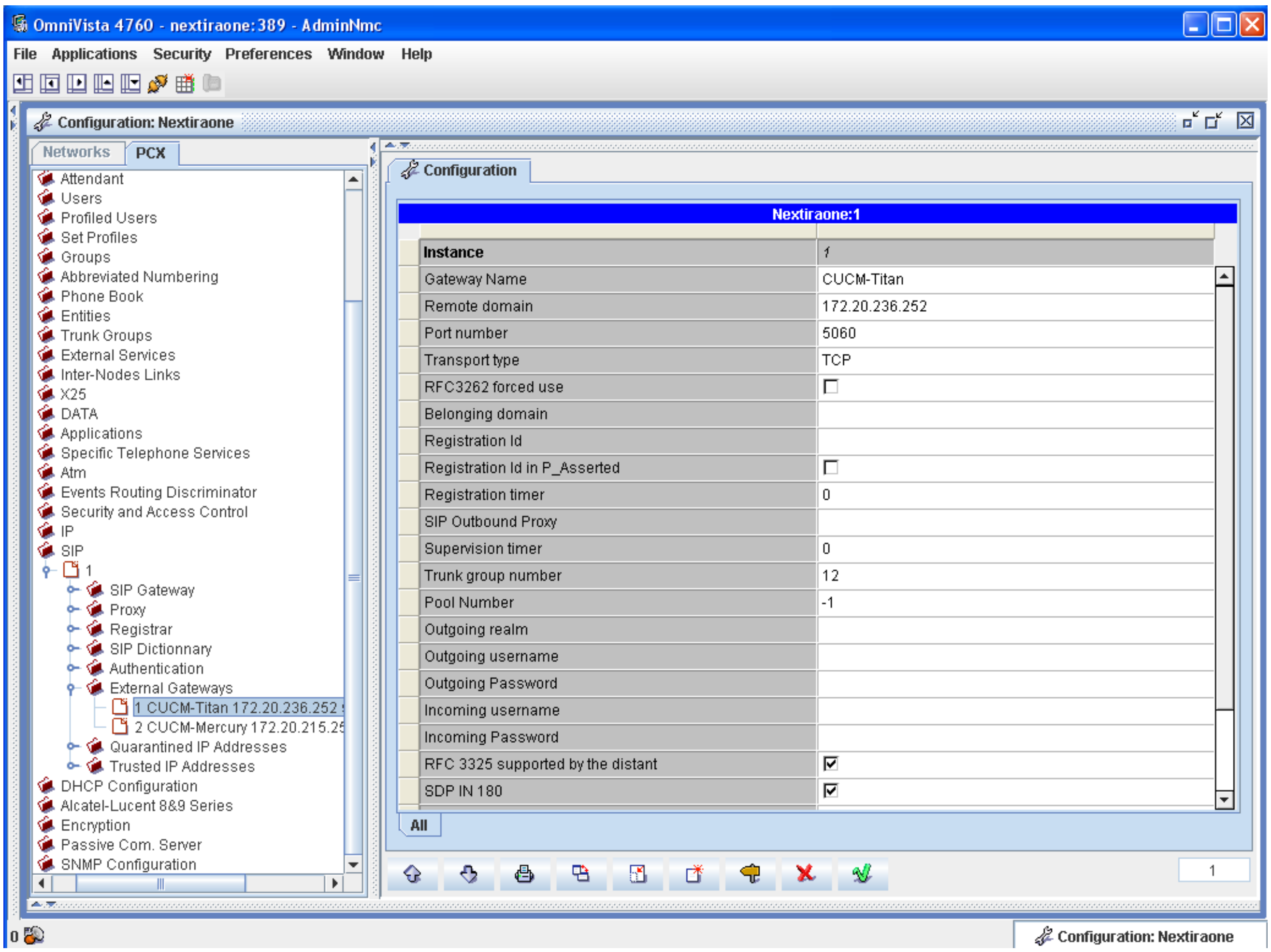
This is where the Cisco UCM-SME server is configured as an (external) SIP gateway

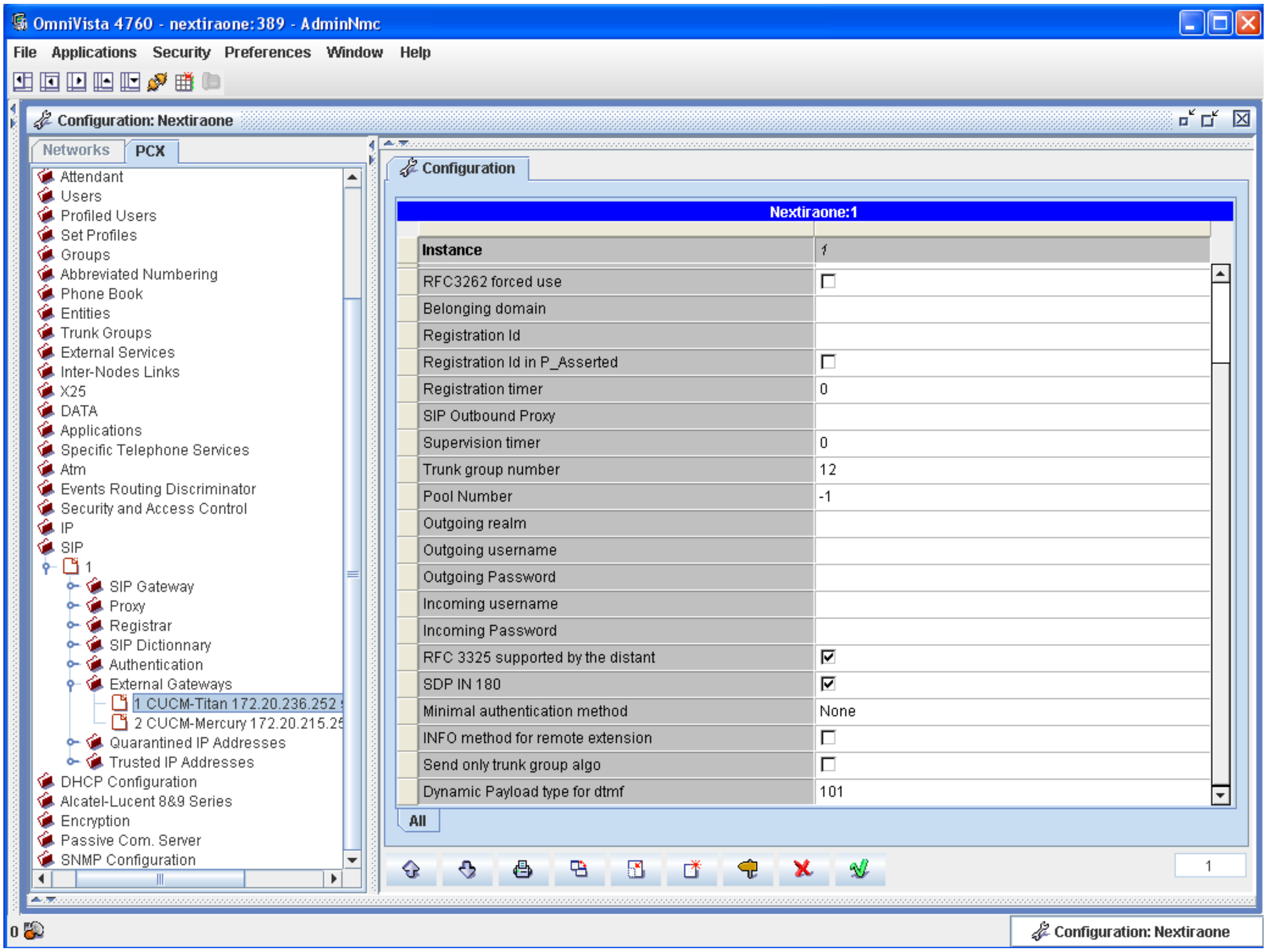
SIP Remote Domain: Enter the IP address or FQDN of Cisco UCM-SME server

SIP Port Number: Enter the TCP or UDP port number used by Cisco UCM-SME for SIP signaling message

SIP Transport type: Enter TCP or UDP as the transport protocol use for SIP signaling

Dynamic Payload Type for dtmf: Enter 101

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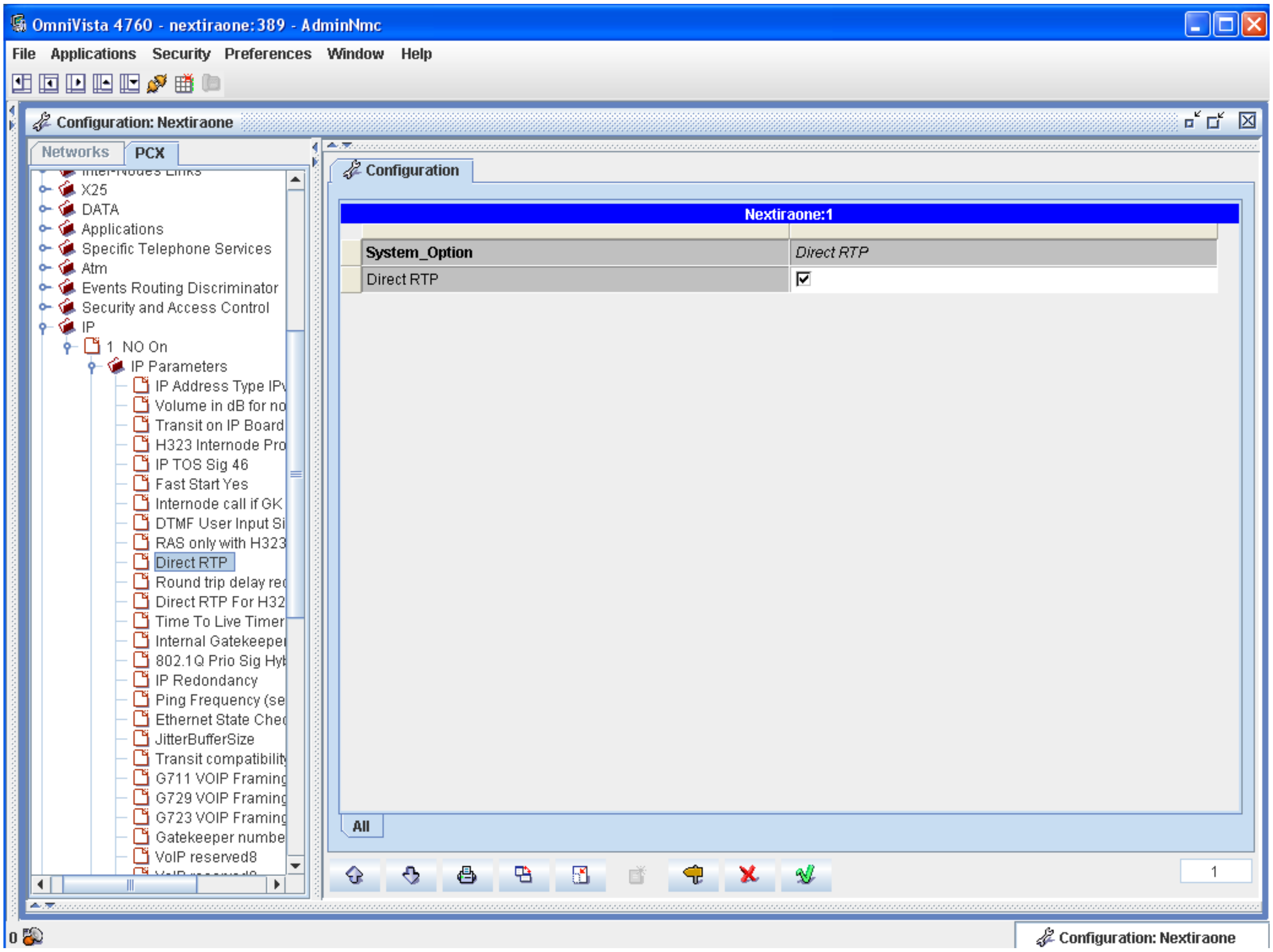
Profiled Users
Set Profiles
Groups
Abbreviated Numbering
Phone Book
Entities
Trunk Groups
External Services
Inter-Nodes Links
X25
DATA
Applications
Specific Telephone Services
Atm
Events Routing Discriminator
Security and Access Control
IP
SIP
1
SIP Gateway
Proxy
Registrar
SIP Dictionnary
Authentication
External Gateways
1 CUCM-Titan 172.20.236.252
2 CUCM-Mercury 172.20.215.25
Quarantined IP Addresses
Trusted IP Addresses
DHCP Configuration
Alcatel-Lucent 8&9 Series
Encryption
Passive Com. Server
SNMP Configuration
Configuration
Nextiraone:1
Instance 1
RFC3262 forced use
Belonging domain
Registration Id
Registration Id in P_Asserted
Registration timer 0
SIP Outbound Proxy
Supervision timer 0
Trunk group number 12
Pool Number -1
Outgoing realm
Outgoing username
Outgoing Password
Incoming username
Incoming Password
RFC 3325 supported by the distant
SDP IN 180
Minimal authentication method None
INFO method for remote extension
Send only trunk group algo
Dynamic Payload type for dtmf 101
All
1
0
Configuration: Nextiraone

Configure IP Parameters:

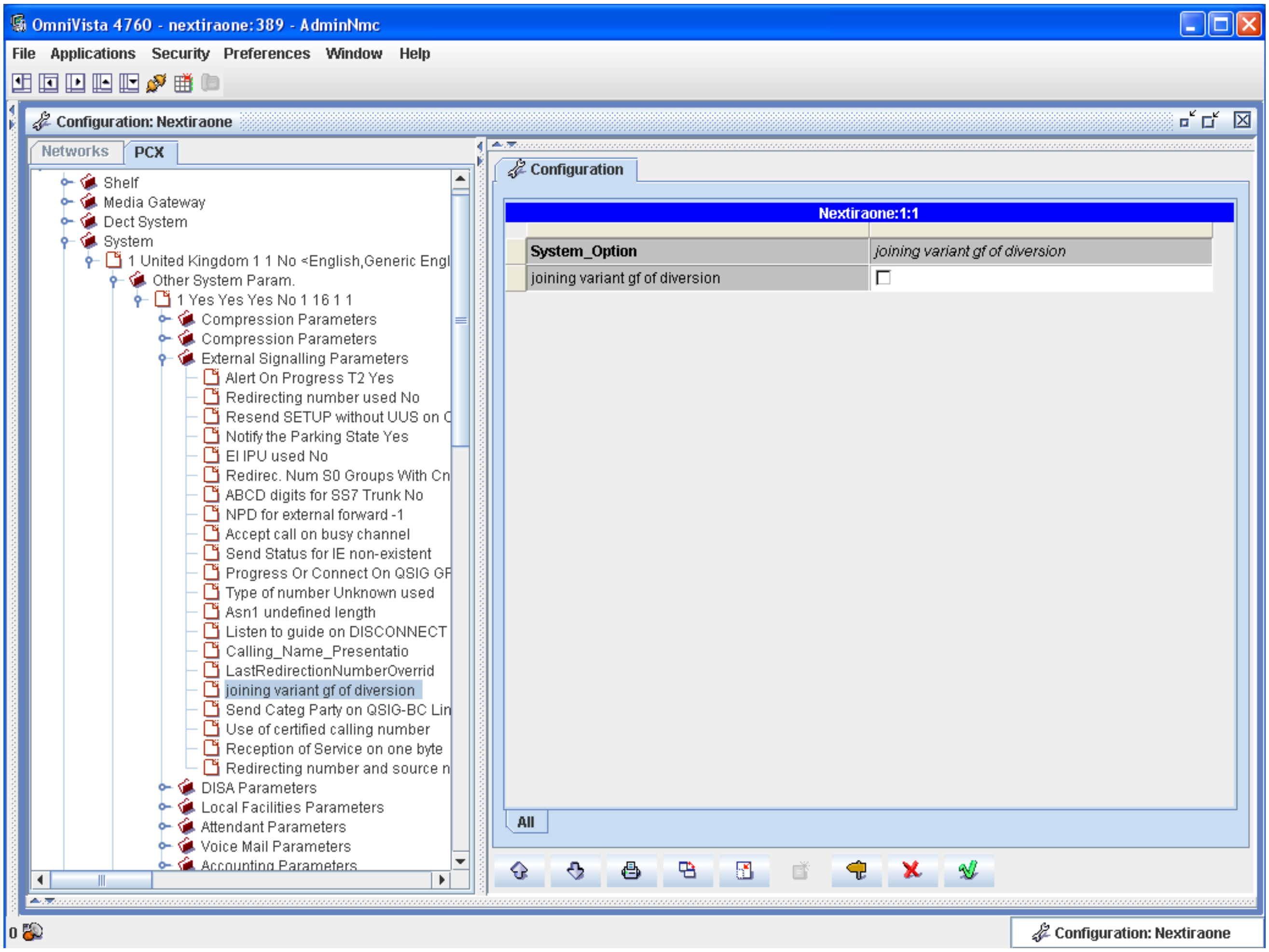
Direct RTP: enable the checkbox for "Direct RTP"



Configure GF diversion on joining:

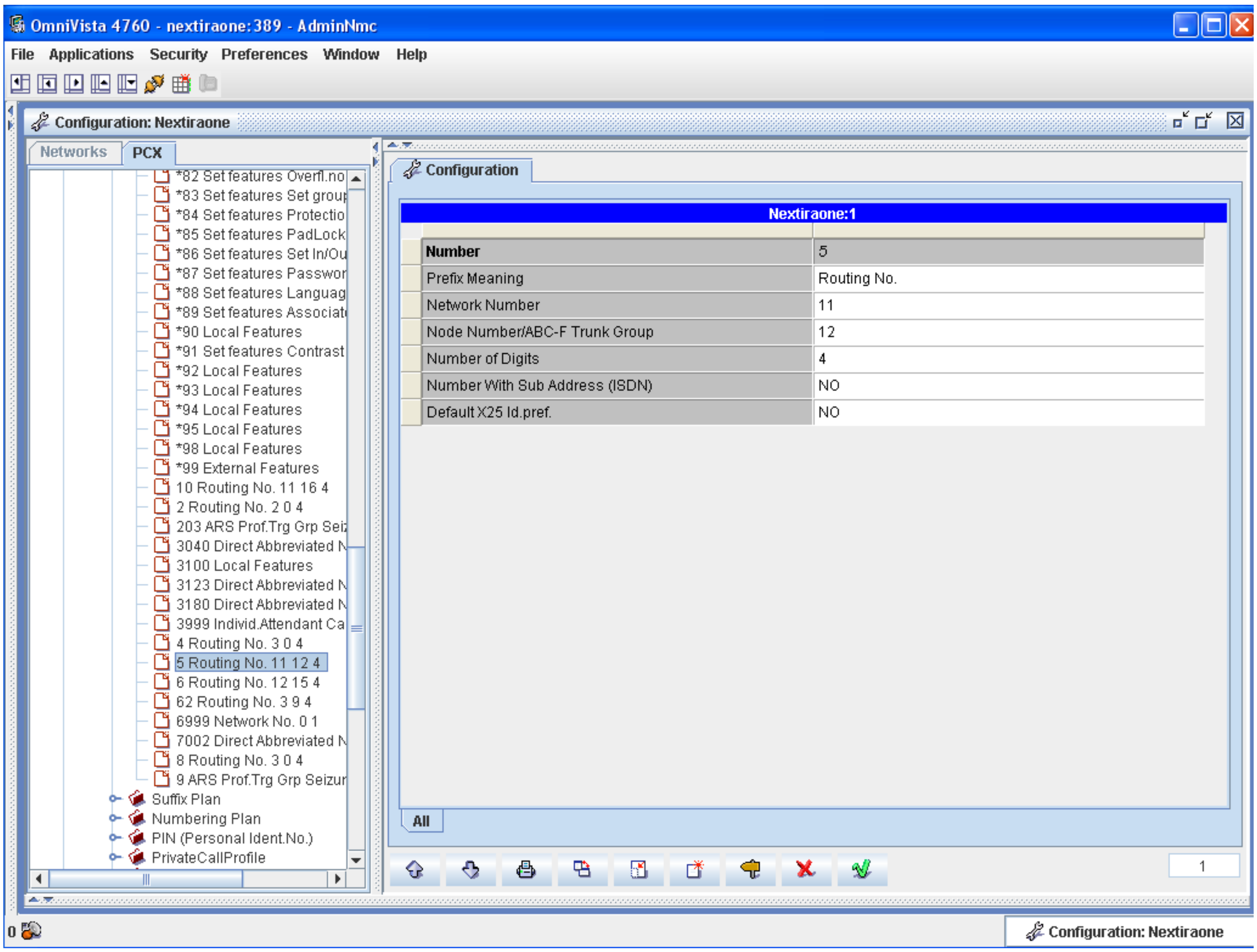
Disable the gf diversion on joining parameter by uncheck the box under the following:

Other System Parameter → External Signaling Parameters → gf diversion on joining



Configure call routing (Translator) to Cisco UCM phone extensions:

Select Translator → Prefix Plan → Create a new prefix for 5xxx to use SIP trunk group 12

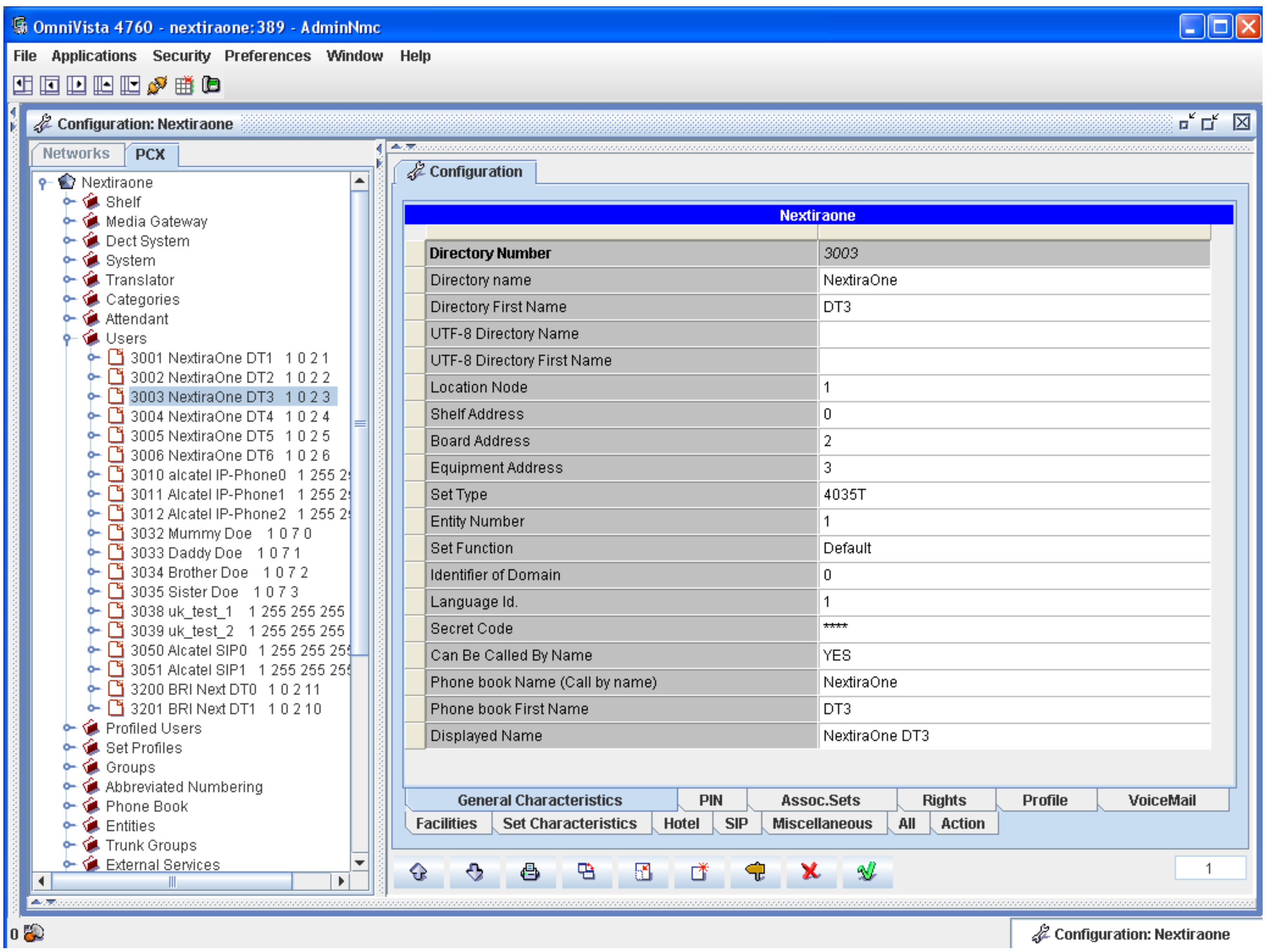


Configure Alcatel standard users (digital stations):

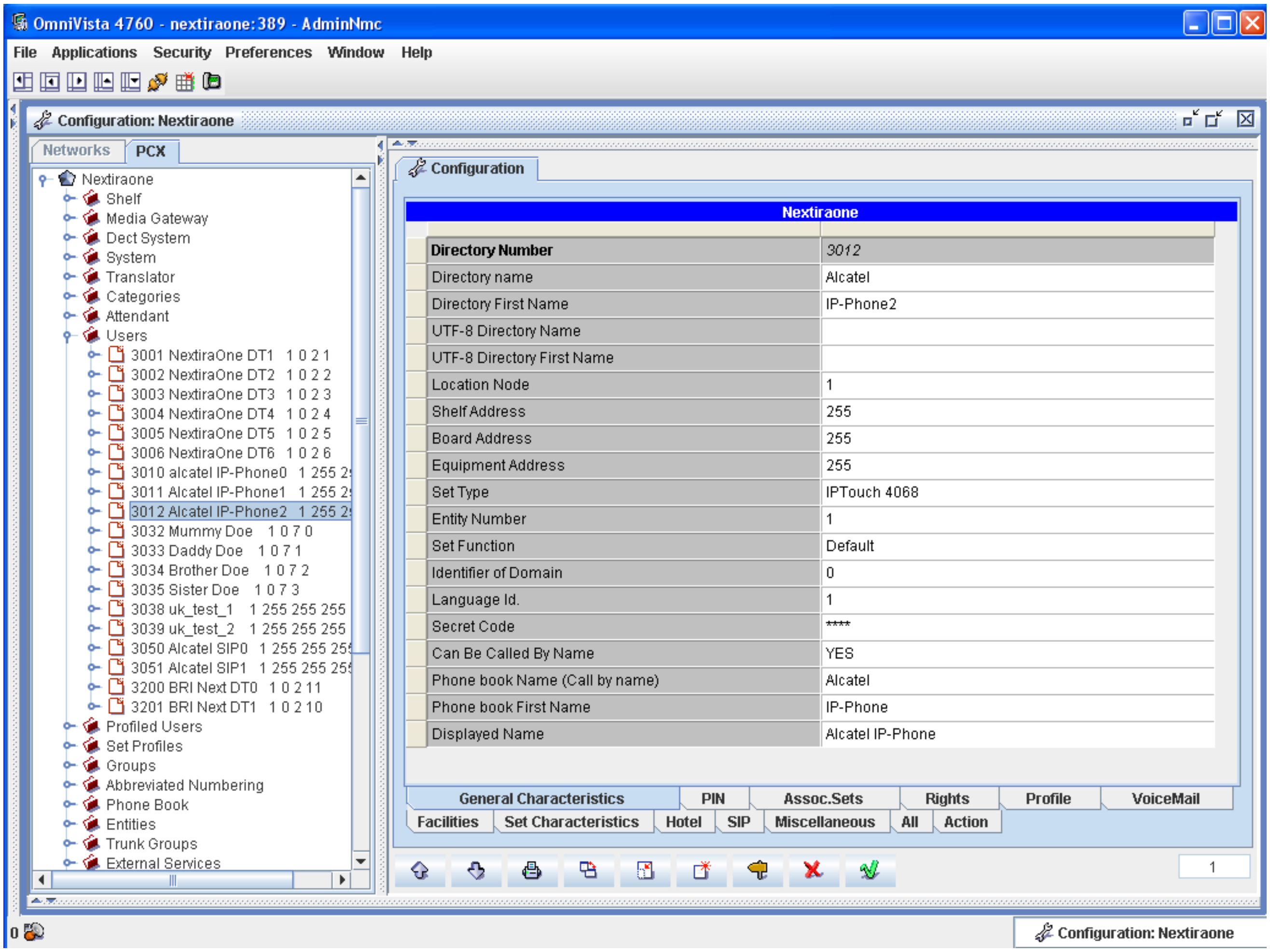
Select User → Create → Create a new user for the digital phone

For a standard user, the URL <username> and <domain> attributes are optional. They can be completed to make the set accessible to the SIP world by a specific SIP URL in form of username@domain type. If they are not configured, the URL is automatically constructed by the system from MAO system configuration data where the URL <domain> takes the SIP gateway IP address (or FQDN) as the default value and the URL <username> takes the set directory numbers as the default value. As an example, the digital phone set with DN = 3003 will have SIP URL = 3003@172.20.9.251 where 172.20.9.251 is the IP Address of the Alcatel SIP media gateway.

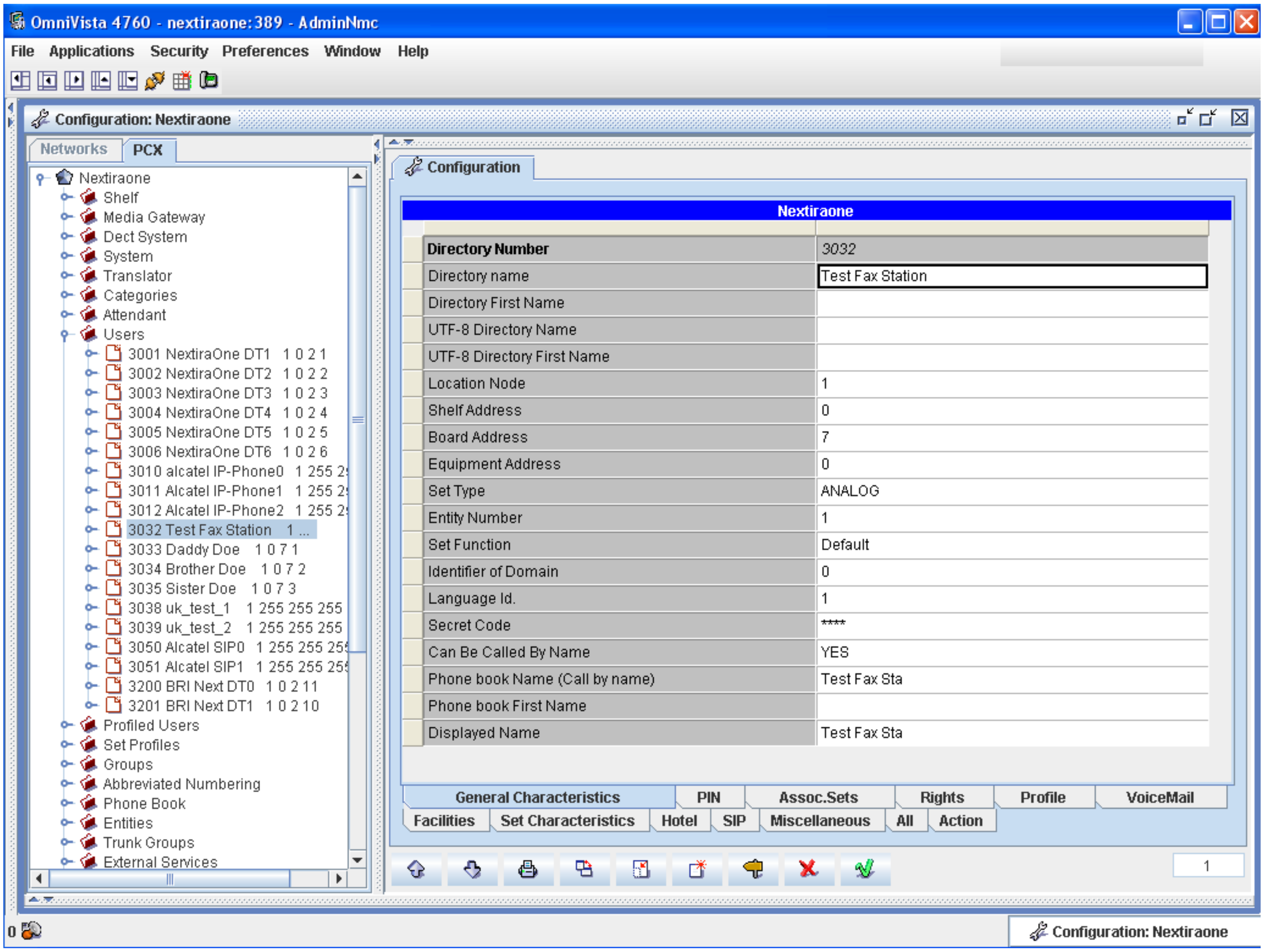
Alcatel 4035 digital phone ext. 3003



Alcatel IP Touch 4068 phone ext. 3012



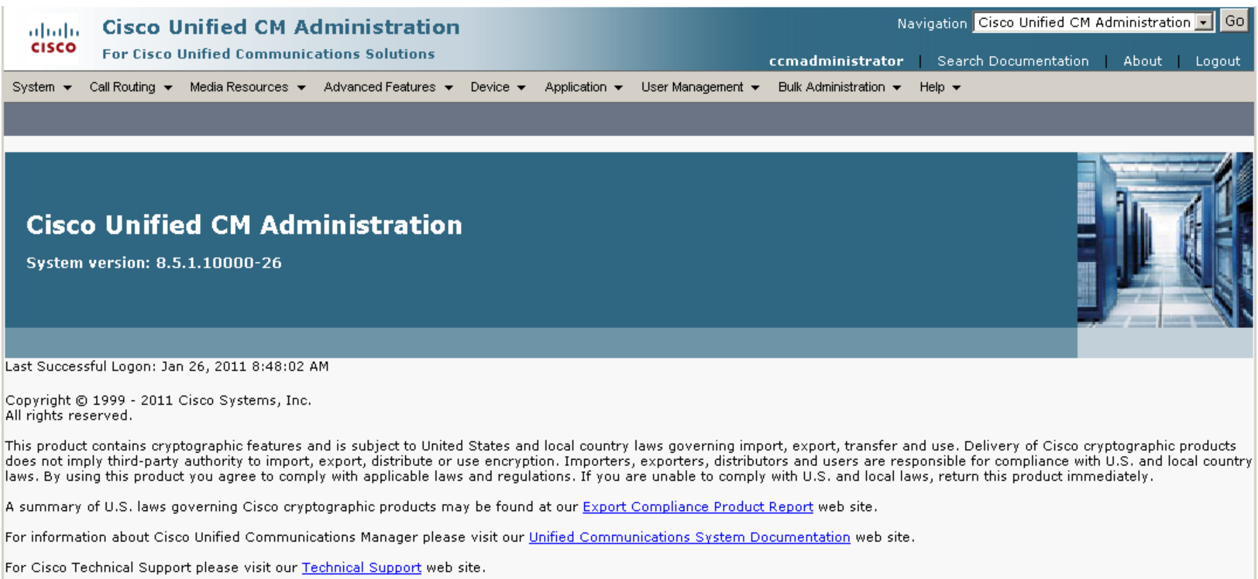
Alcatel analog station ext. 3032



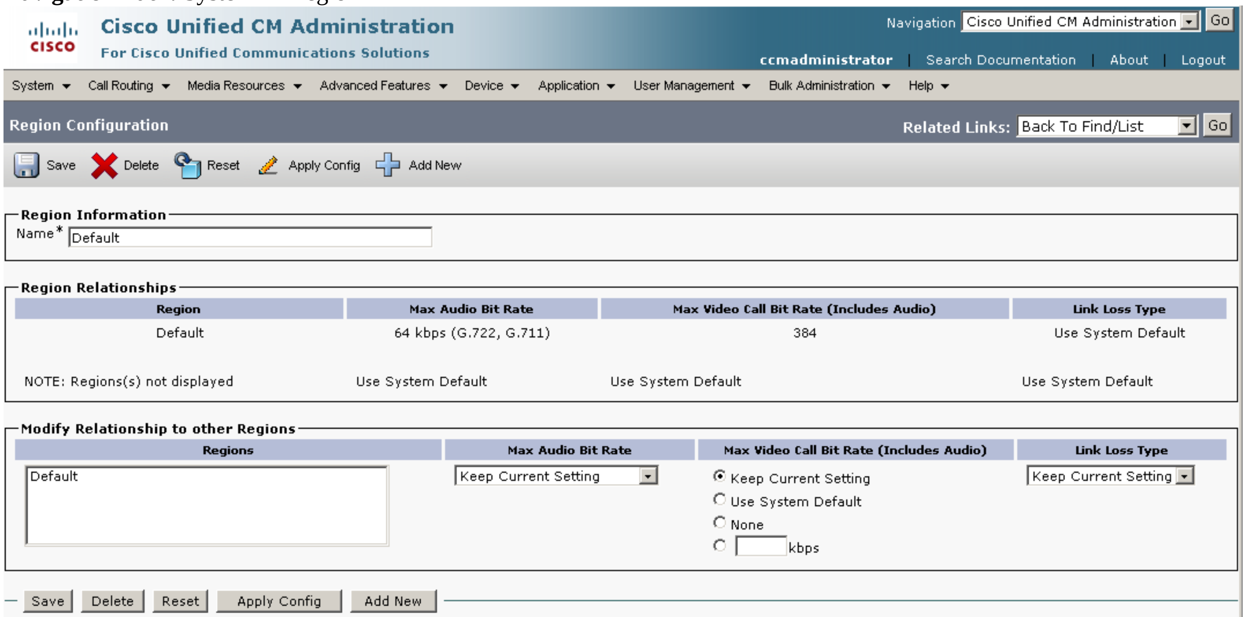
## Configuring the Cisco Unified Communications Manager – Session Manager Edition

1. Cisco Session Manager Version
2. Device Pool and Region mapping configuration
3. SIP Profile (used by SIP trunks) configuration
4. SIP Trunk Security Profile (used by SIP trunks) configuration
5. SIP trunk configuration to Alcatel PBX
6. SIP Trunk configuration to Cisco UCM
7. Route Pattern configuration to Alcatel PBX
8. Route Pattern configuration to Cisco UCM
9. SIP Route Pattern configuration to process redirect messages from Alcatel PBX

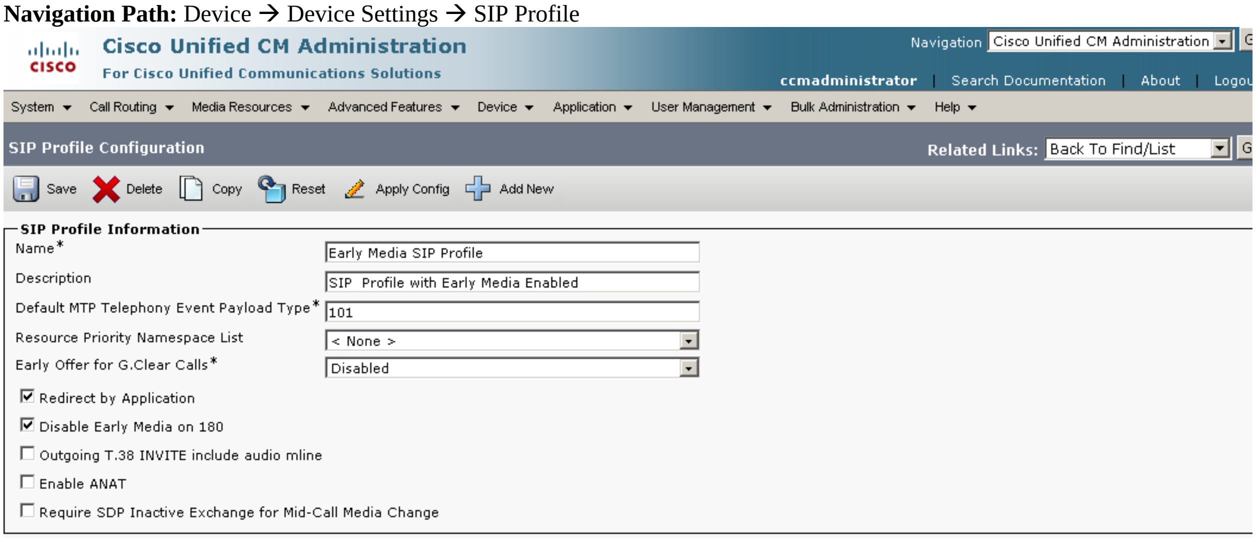
**Cisco Unified Communications Manager – Session Manager Edition software version**

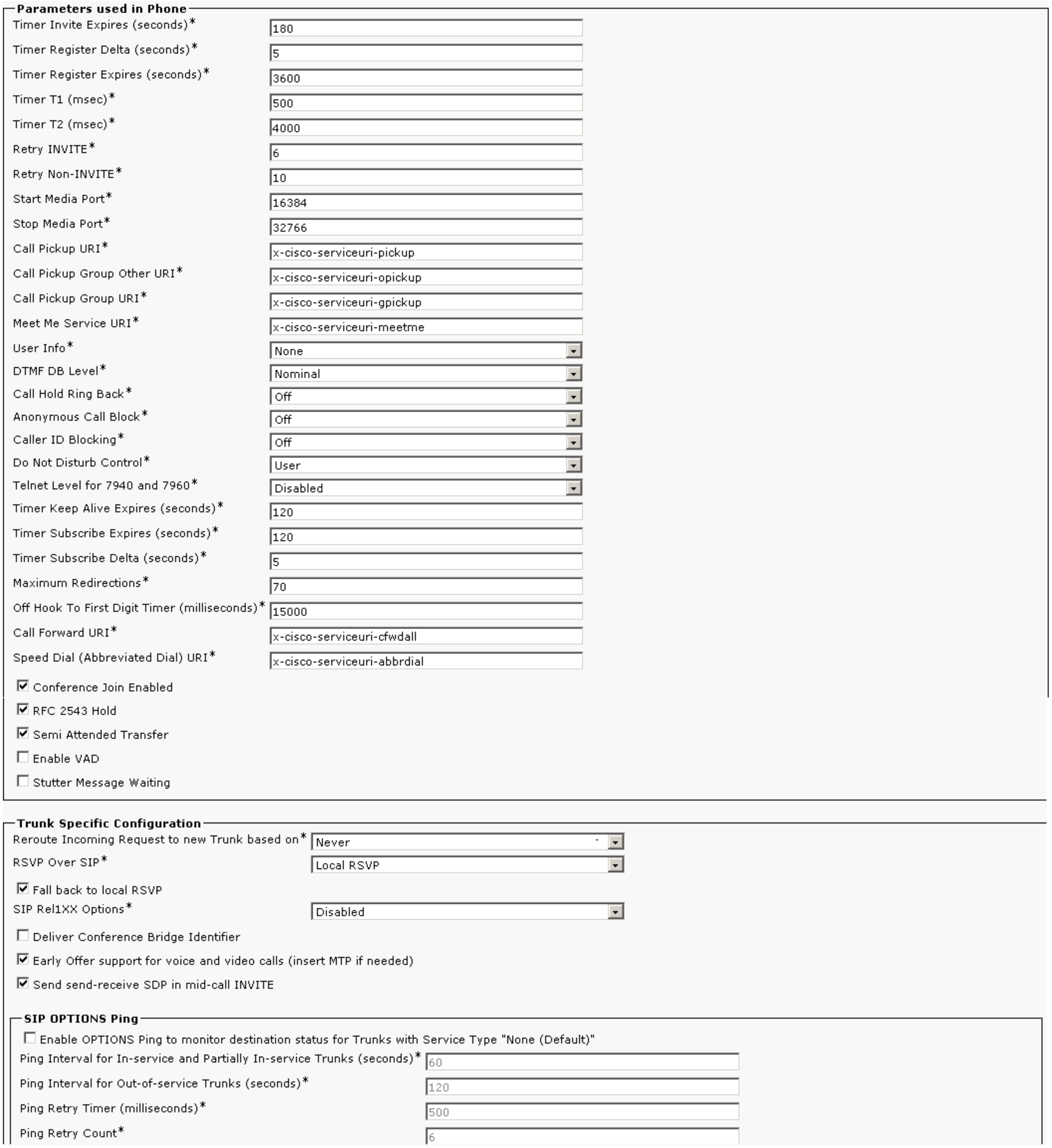


**Configuration of Device Pool to Region mapping**
**Navigation Path:** System → Region



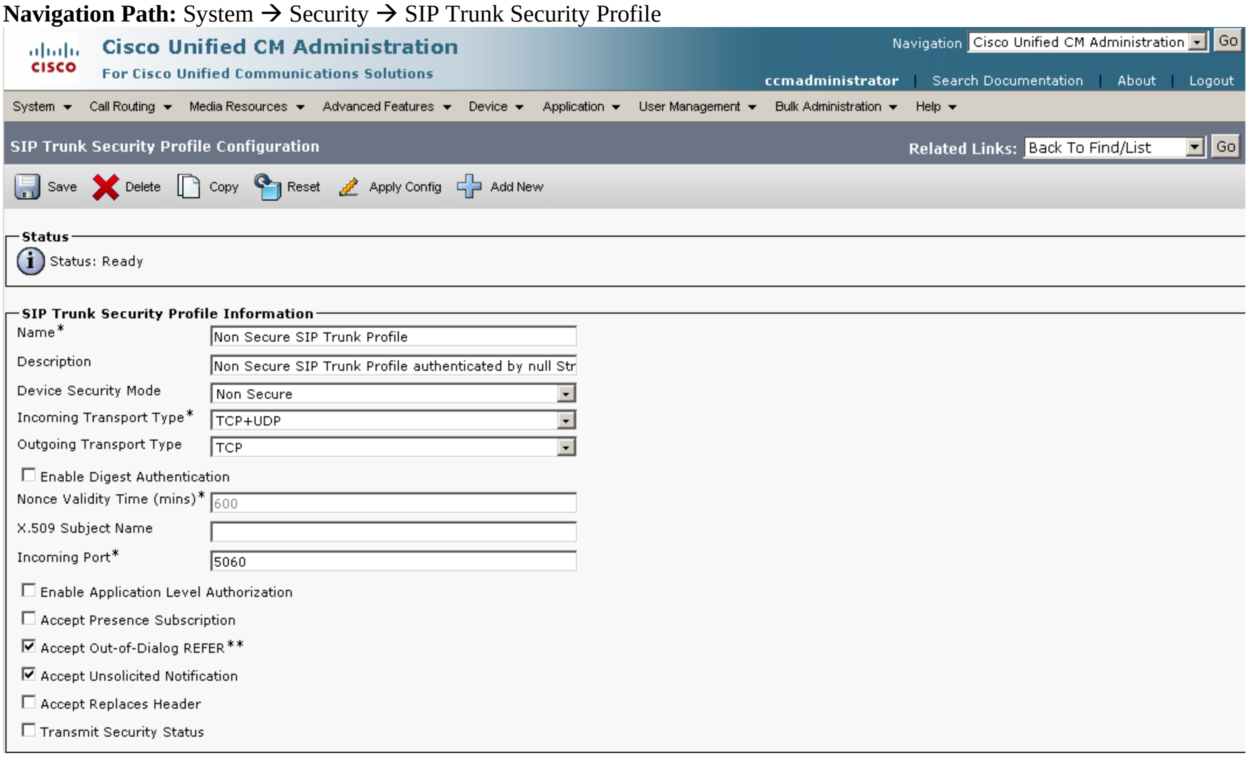

**Configuration of SIP Profile used by SIP trunks**
**Navigation Path:** Device → Device Settings → SIP Profile





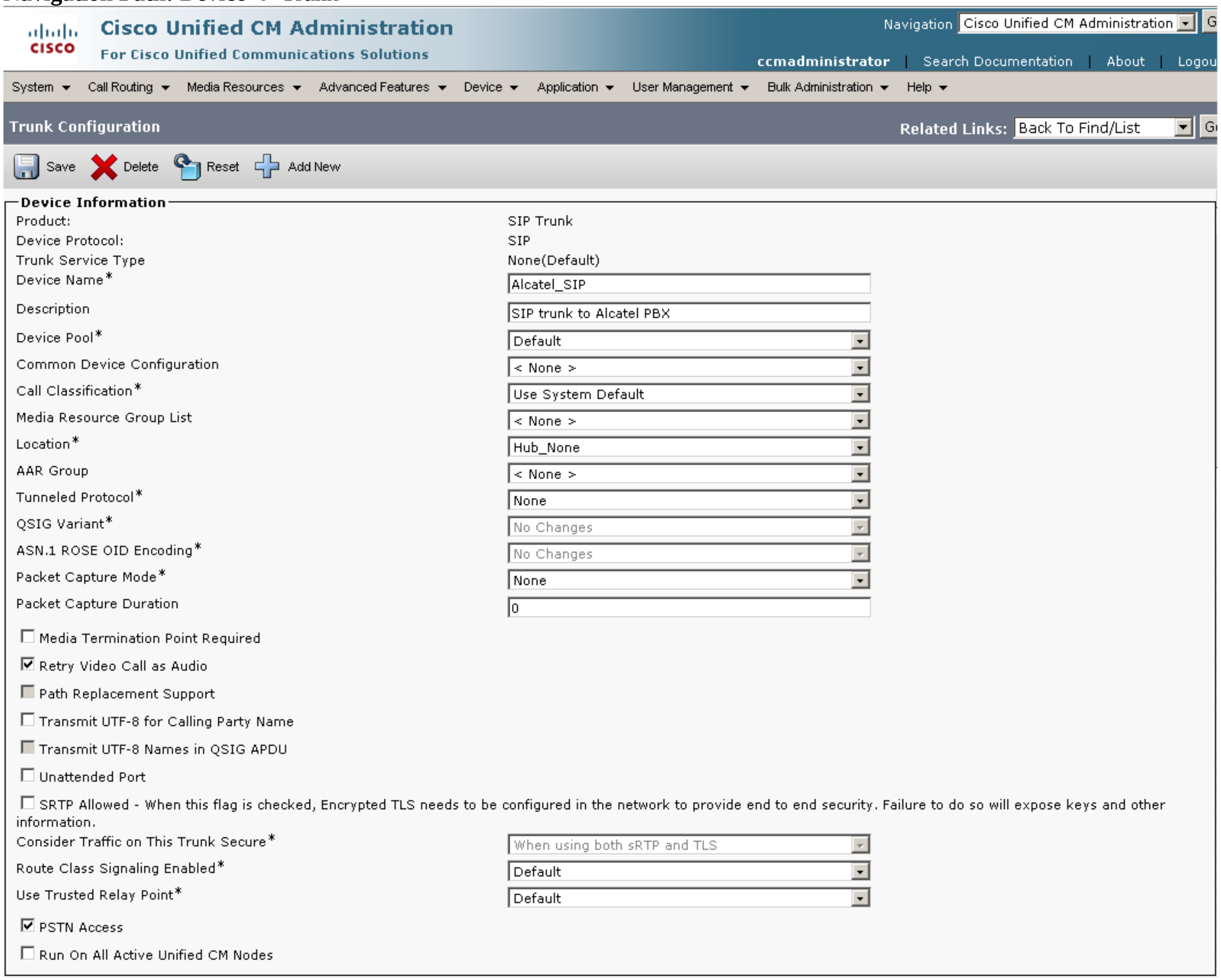

**Note:** The Alcatel PBX does not respond to OPTIONS messages in a reliable fashion (it ignores the first several attempts, but it will eventually responds with 483 Too Many Hops message, which Cisco UCM-SME accepts as a valid response to the OPTIONS ping). Because of this, it is recommended to either disable SIP OPTIONS ping, or shorten the Ping Interval timer forcing Cisco UCM-SME to generate more ping messages.

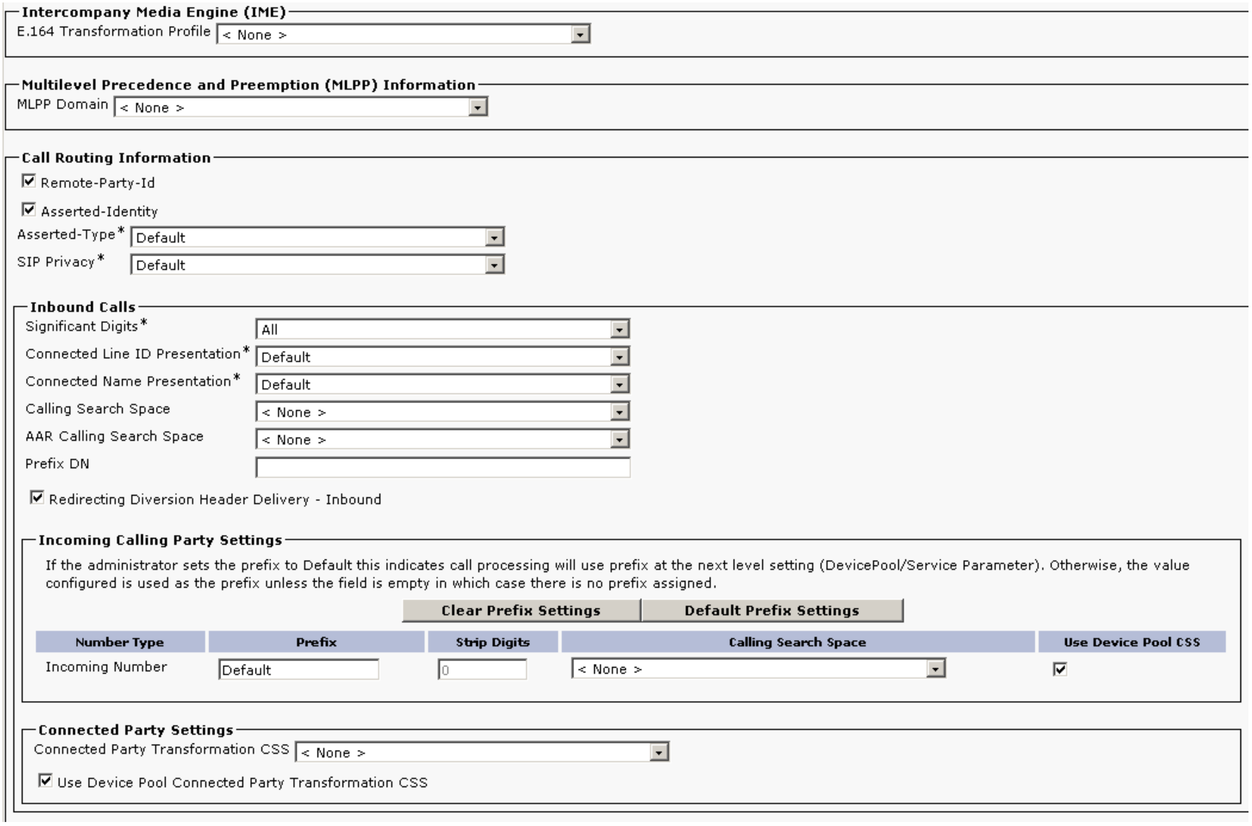
**Configuration of SIP Trunk Security Profile used by SIP trunks**
**Navigation Path:** System → Security → SIP Trunk Security Profile



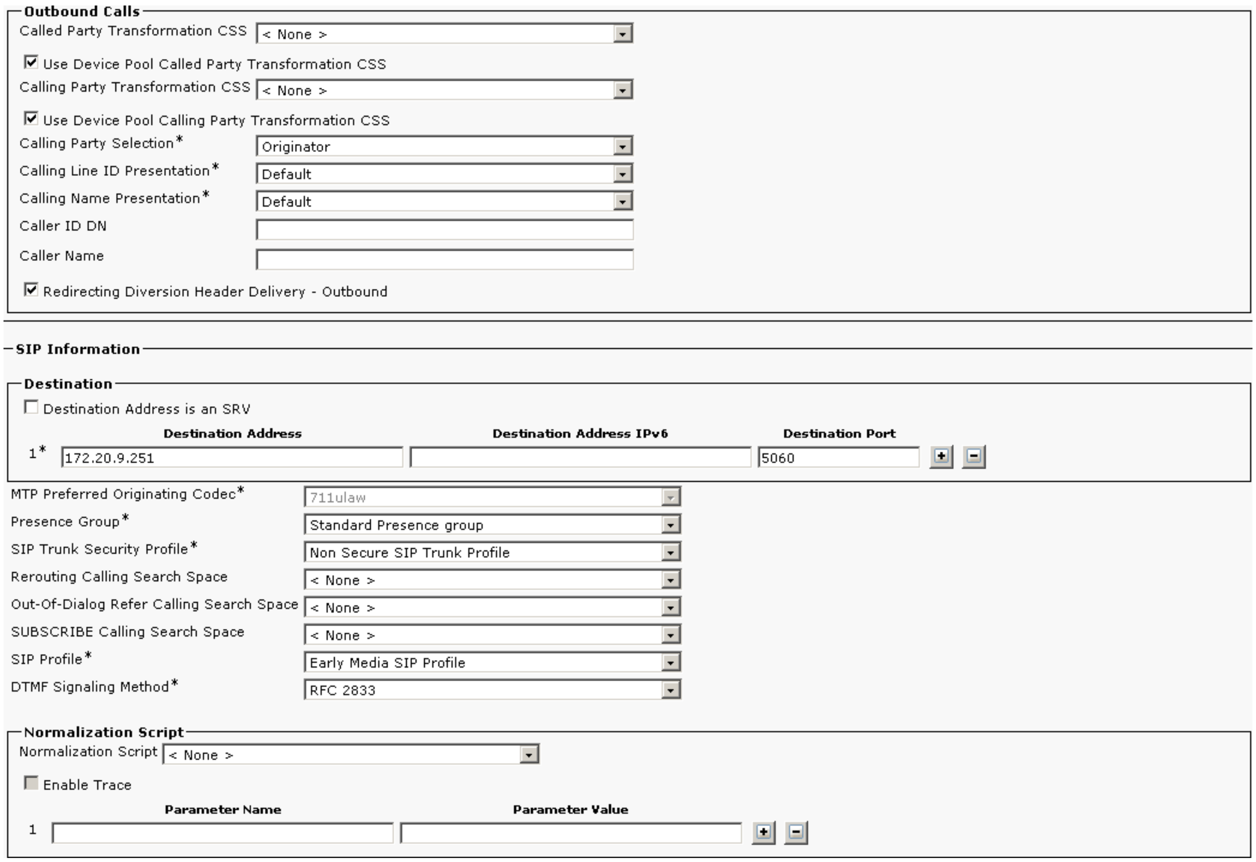

**Note:** Make sure parameter "Accept Replaces Header" is disabled. This will ensure proper functionality for local early-attended call transfers from Alcatel stations.

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

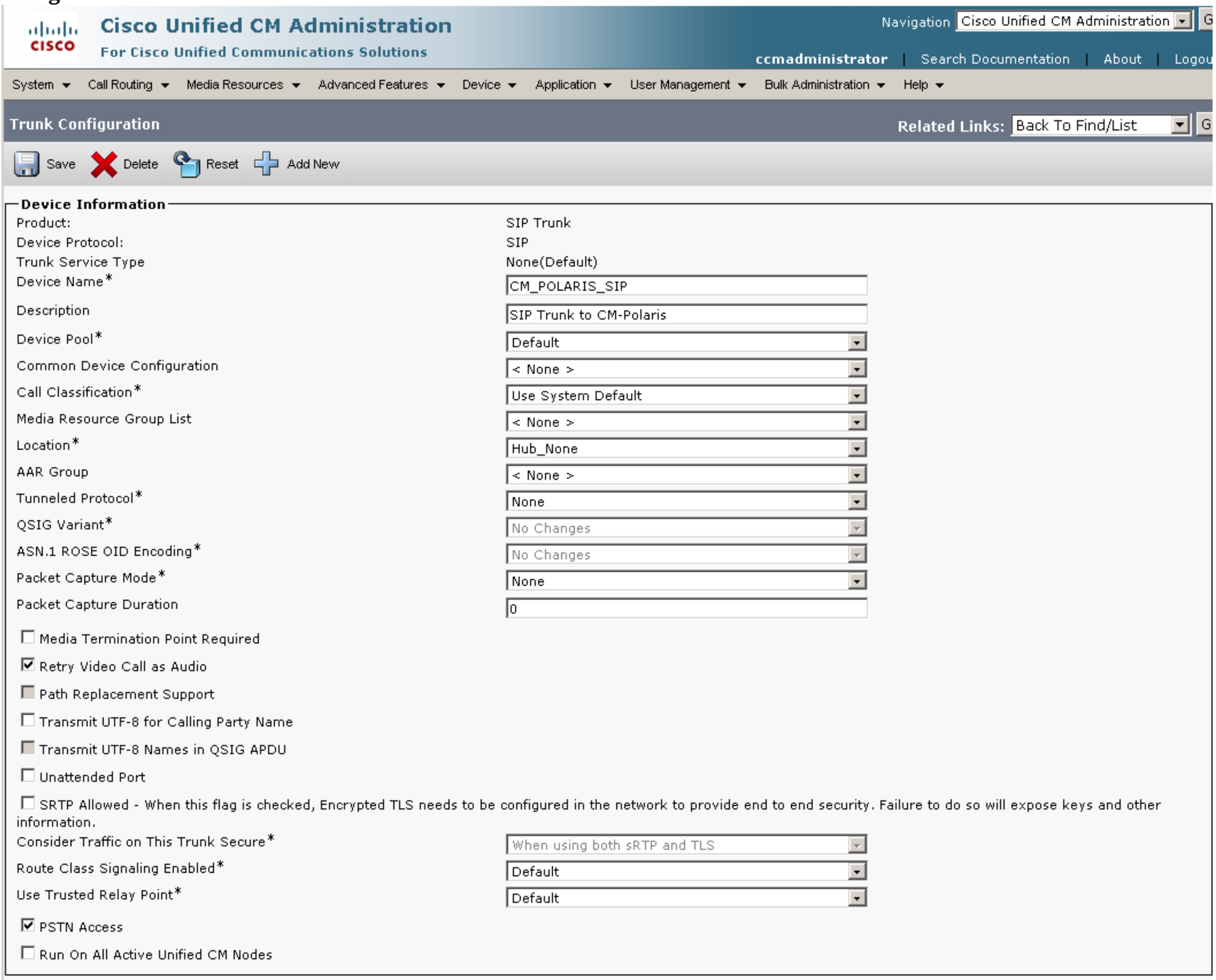


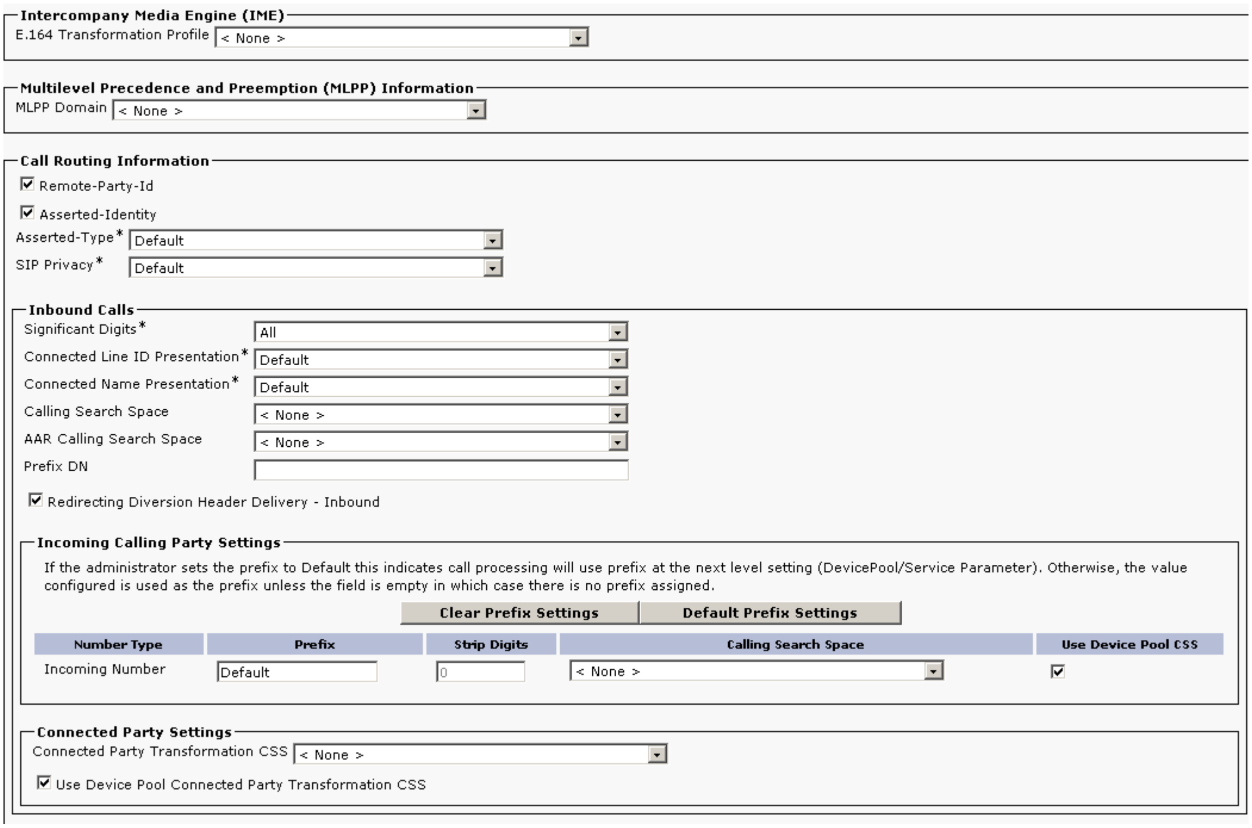


Intercompany Media Engine (IME)
E.164 Transformation Profile < None >
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >
Call Routing Information
Remote-Party-Id
Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default
Inbound Calls
Significant Digits* All
Connected Line 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*
172.20.9.251
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 SIP Profile
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
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 >

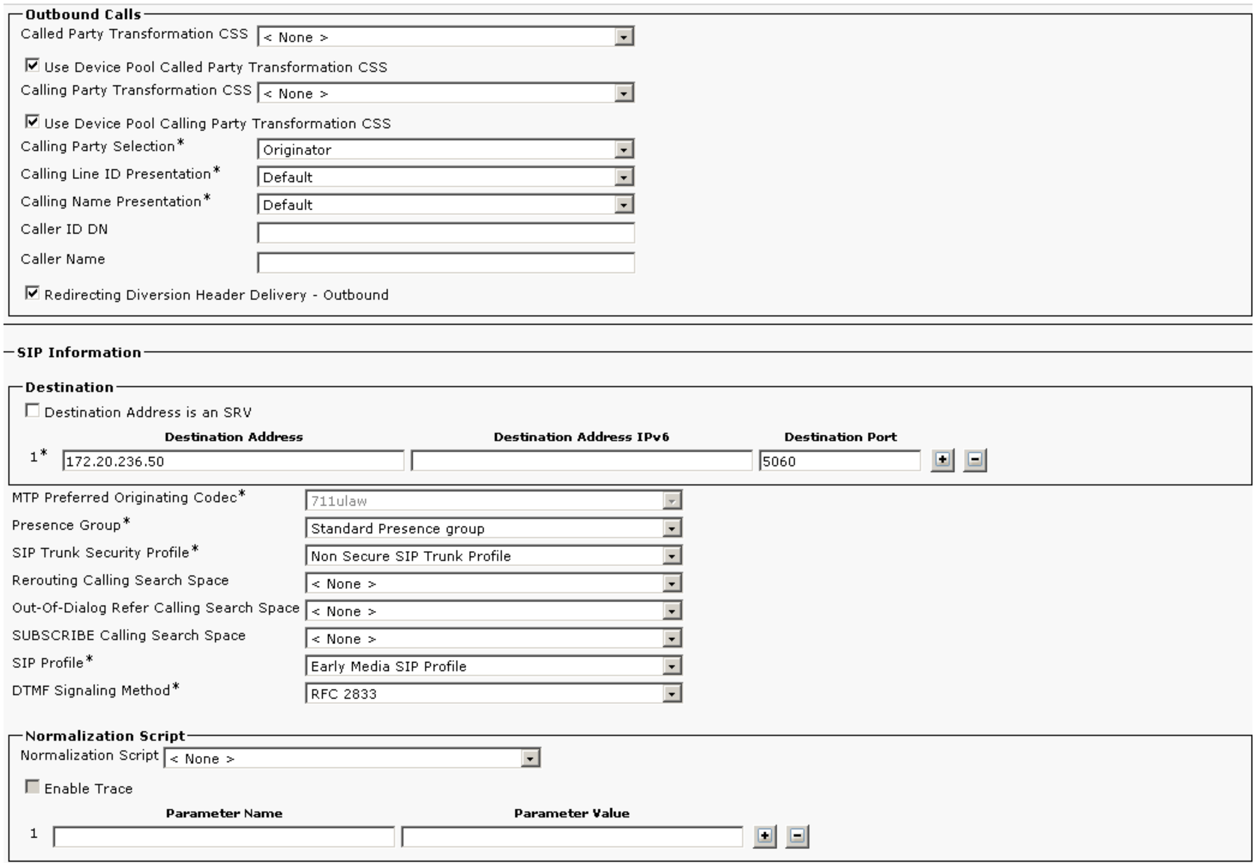
Call Routing Information
☑ Remote-Party-Id
☑ Asserted-Identity
Asserted-Type* Default
SIP Privacy* Default

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

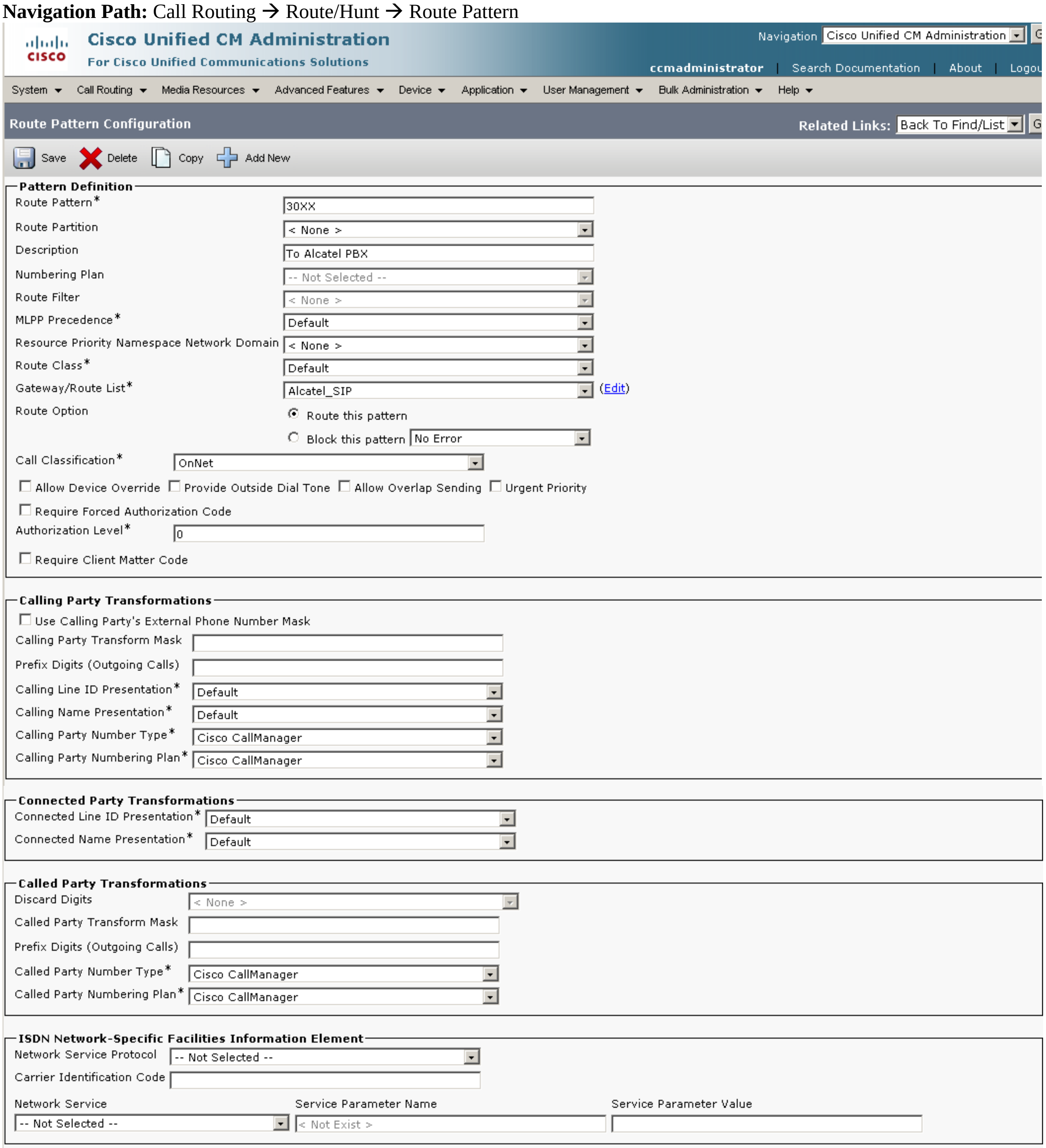
Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.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

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

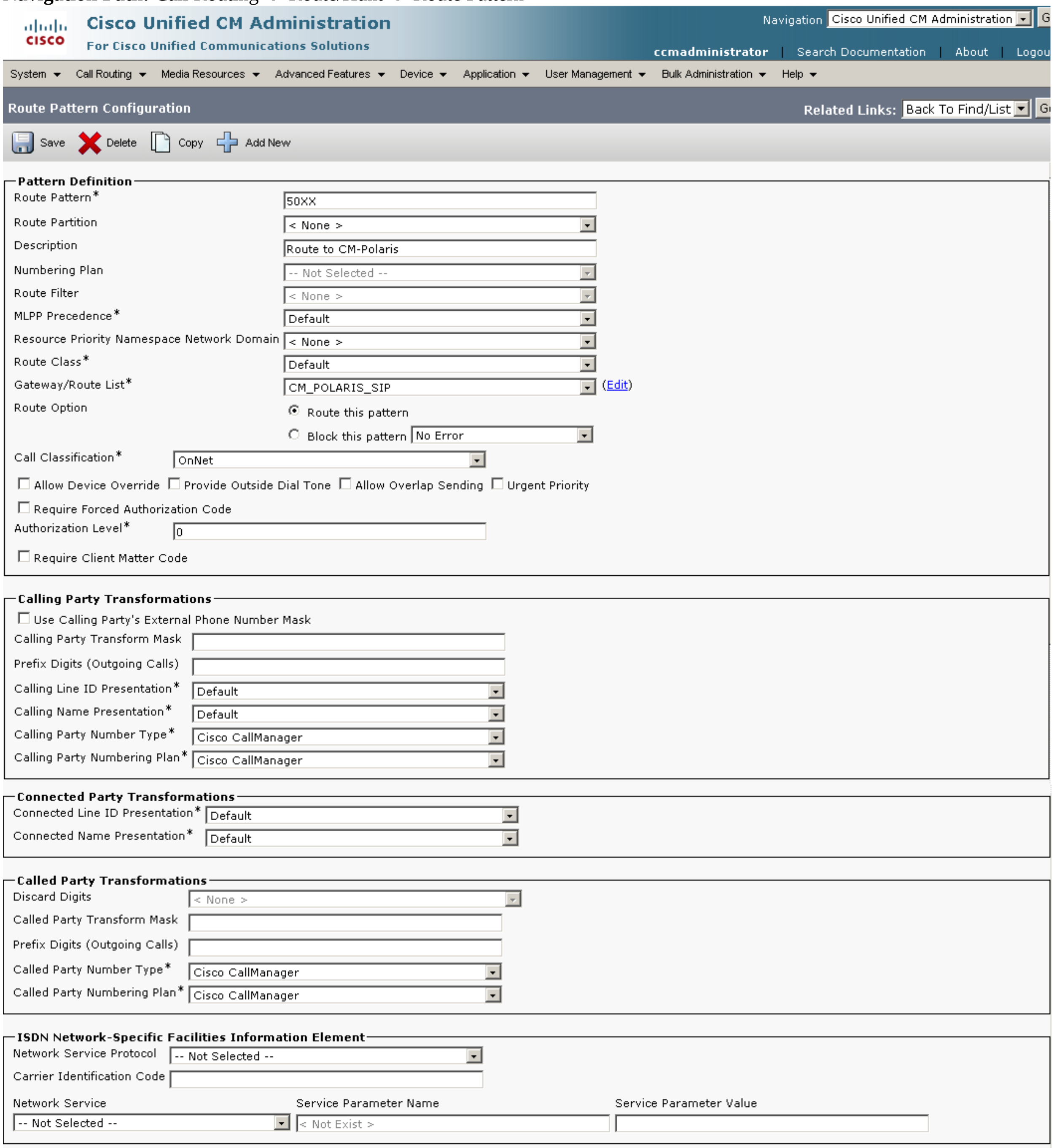
Connected Party Settings
Connected Party Transformation CSS < None >
☑ Use Device Pool Connected Party Transformation CSS


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 SIP Profile
DTMF Signaling Method*
RFC 2833
Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1

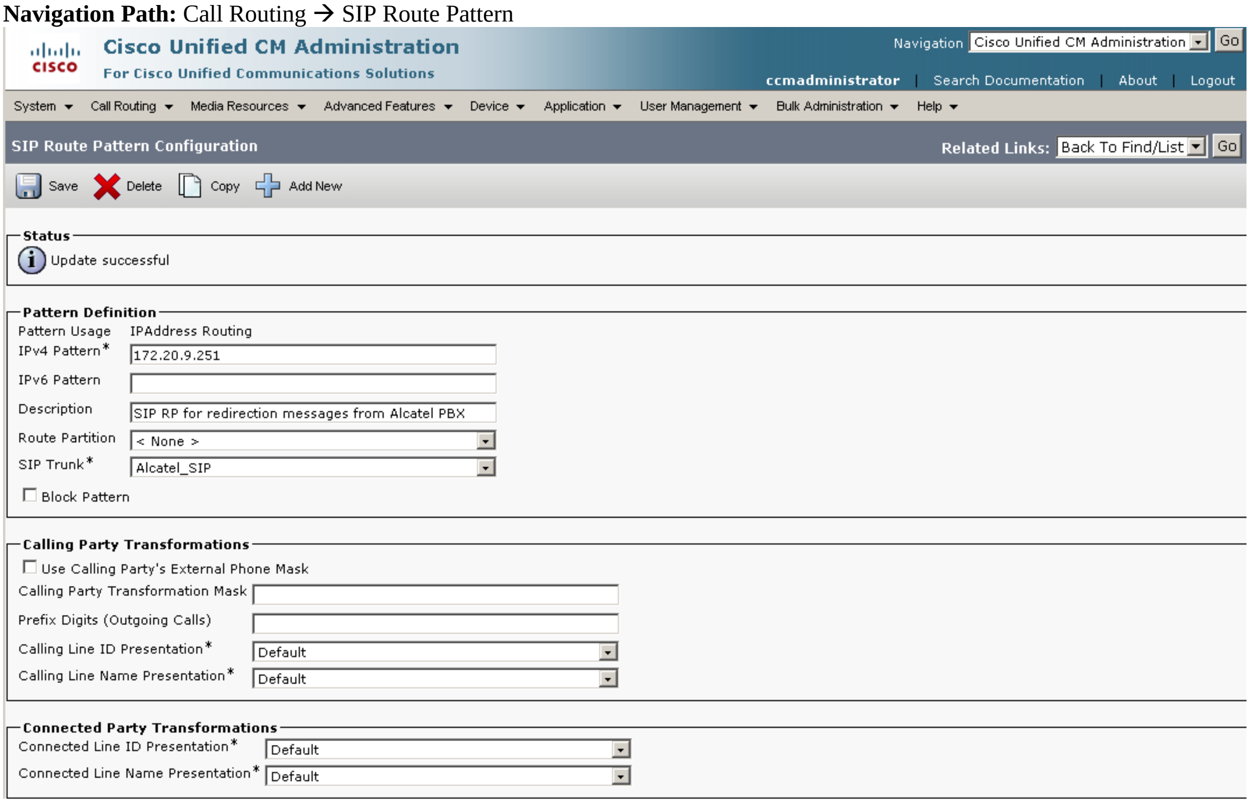
**Configuration of Route Patterns – To Alcatel PBX**
**Navigation Path:** Call Routing → Route/Hunt → Route Pattern



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



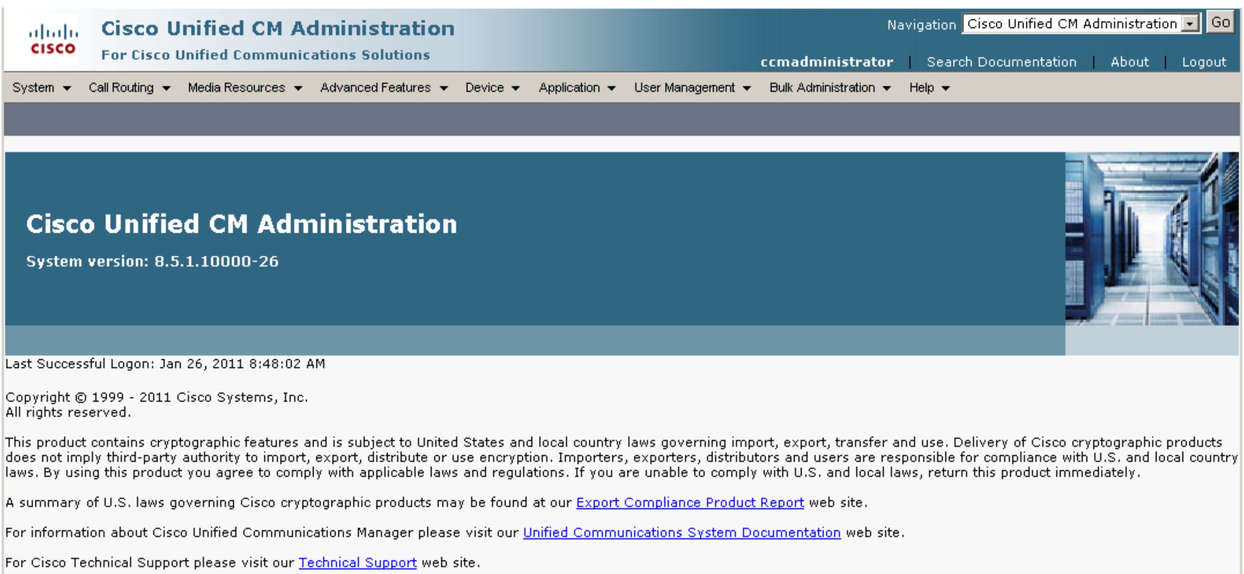
**Configuration of SIP Route Pattern for processing redirect messages from Alcatel PBX**
**Navigation Path:** Call Routing → SIP Route Pattern



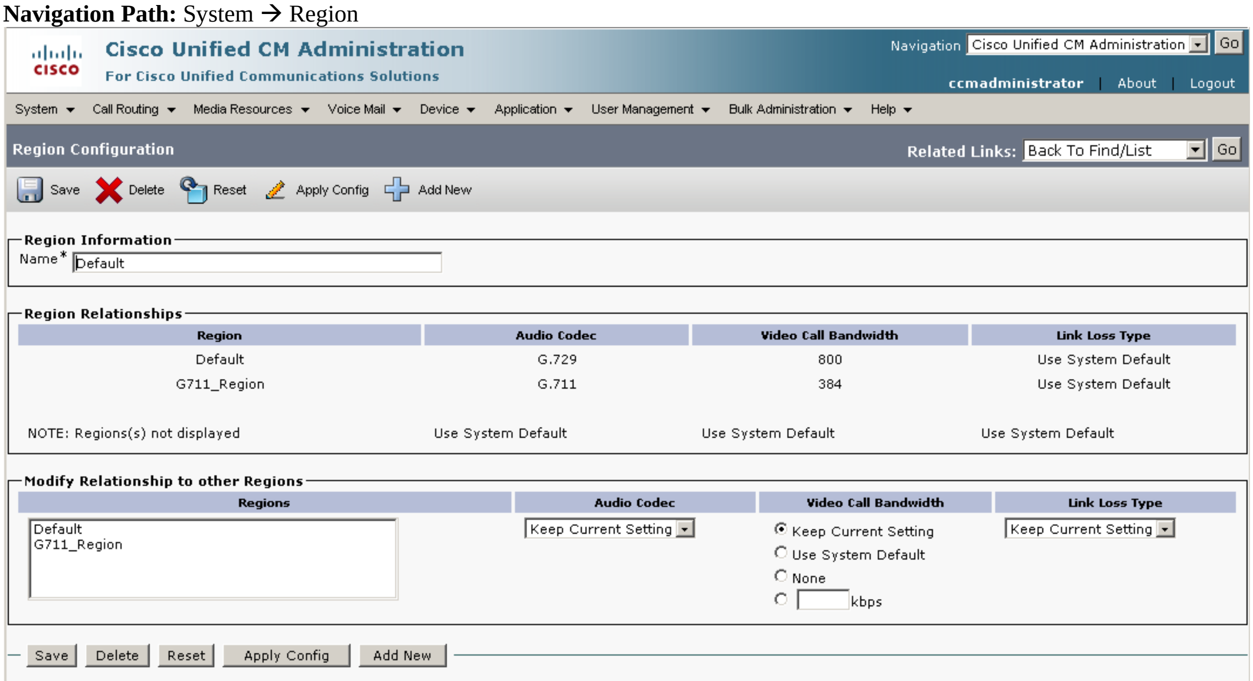
### Configuring the Cisco Unified Communications Manager

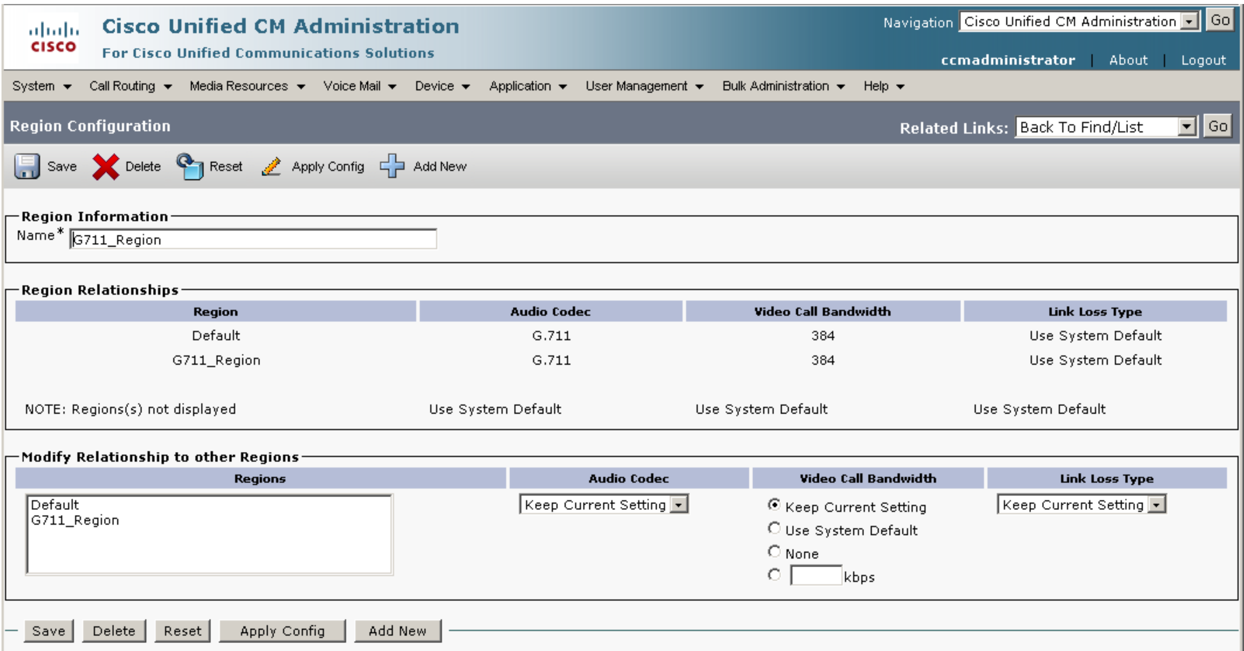
1. Cisco Unified Communications Manager Version
2. Device pool and Region mapping configuration
3. Conference Bridge configuration
4. Media Resource Group configuration
5. Media Resource Group List configuration
6. SIP Profile configuration
7. SIP Trunk to SME configuration
8. Route Pattern configuration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

**Cisco Unified Communications Manager version**

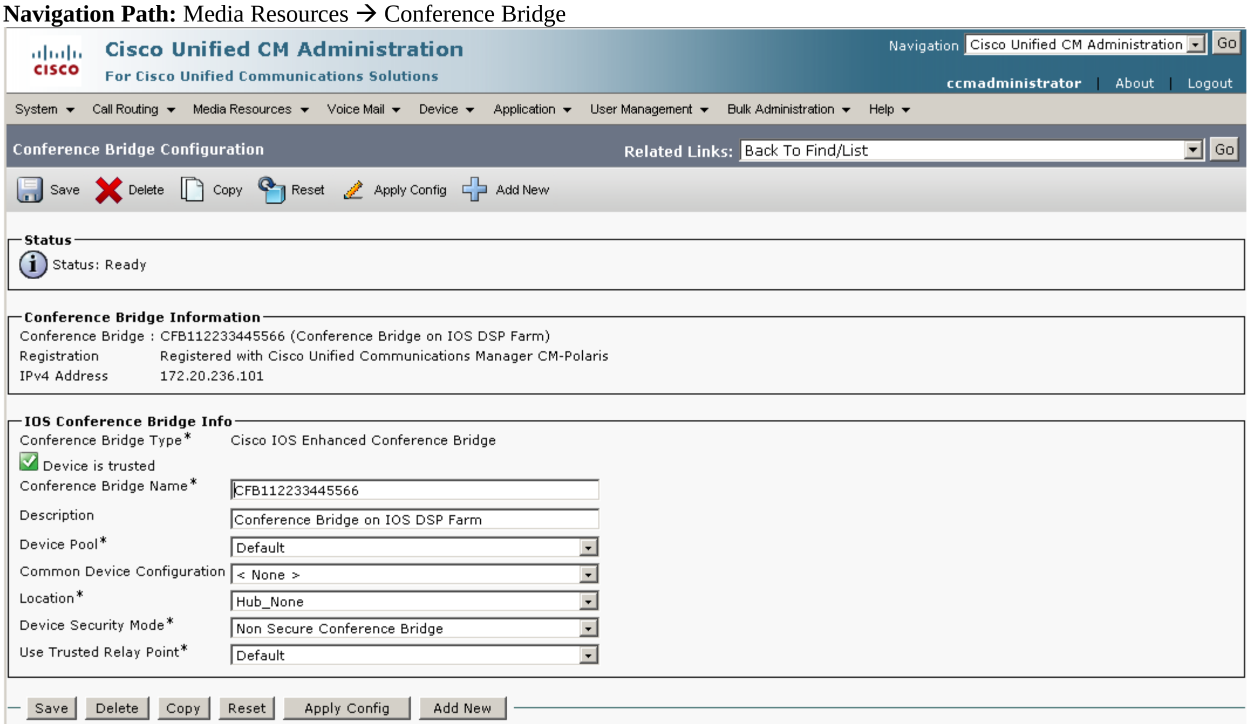


**Configuration of Device Pool to Region mapping**
**Navigation Path:** System → Region






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

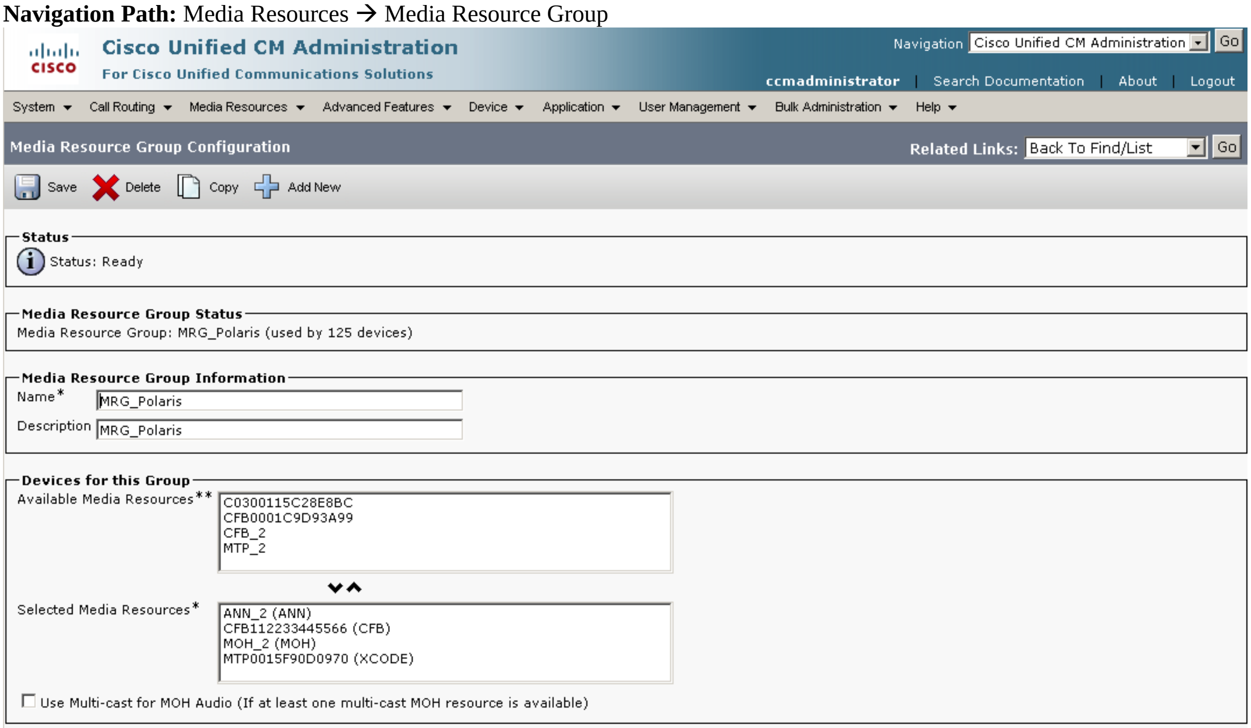


**Conference Bridge IOS configuration:**

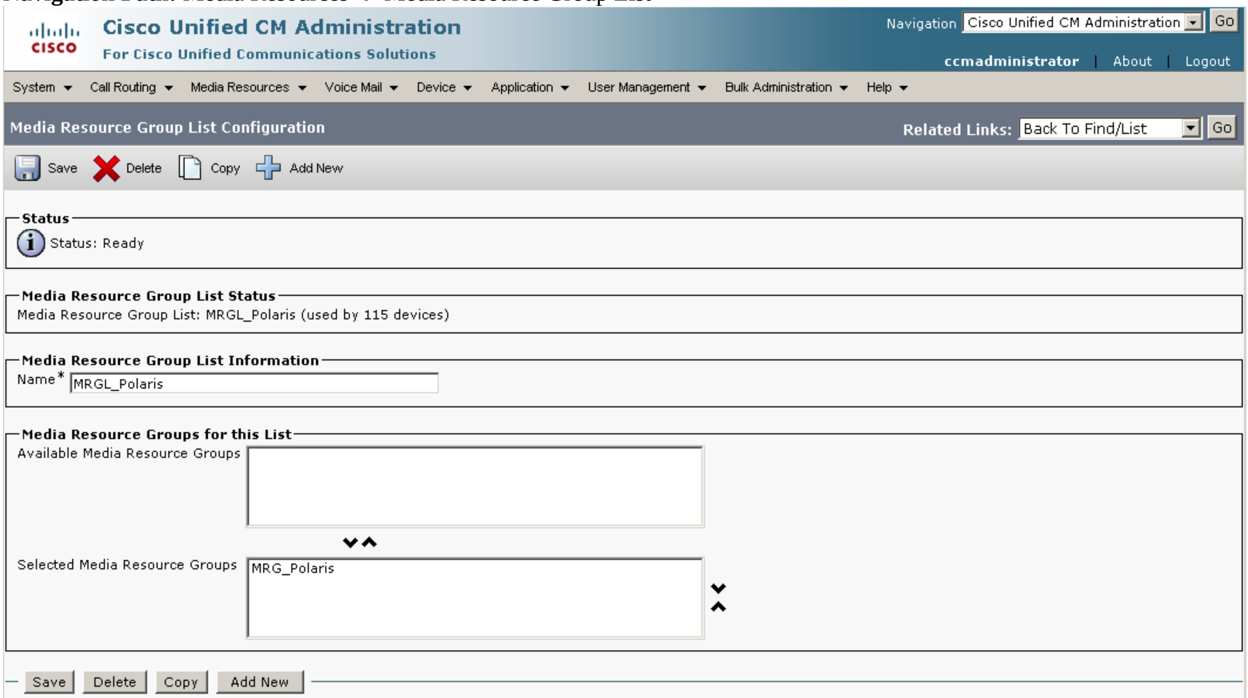
```
sccp local GigabitEthernet0/0
sccp ccm 172.20.236.50 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface GigabitEthernet0/0
 associate ccm 1
 priority 1
 associate profile 98 register cfb112233445566
!
dspfarm profile 98 conference
 codec g729r8
 codec g711ulaw
 maximum sessions 8
 associate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

**Configuration of Media Resource Group**

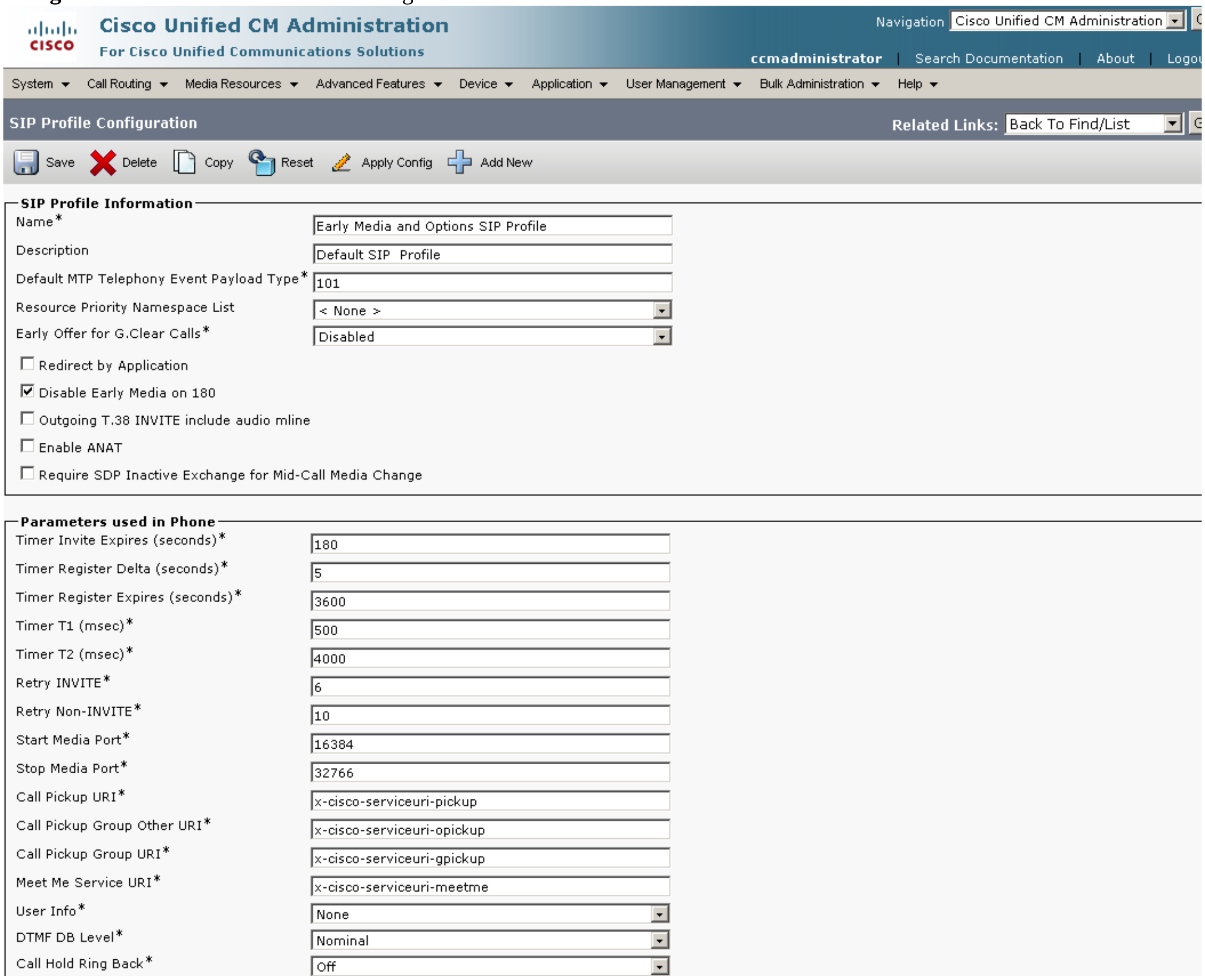
**Navigation Path:** Media Resources → Media Resource Group

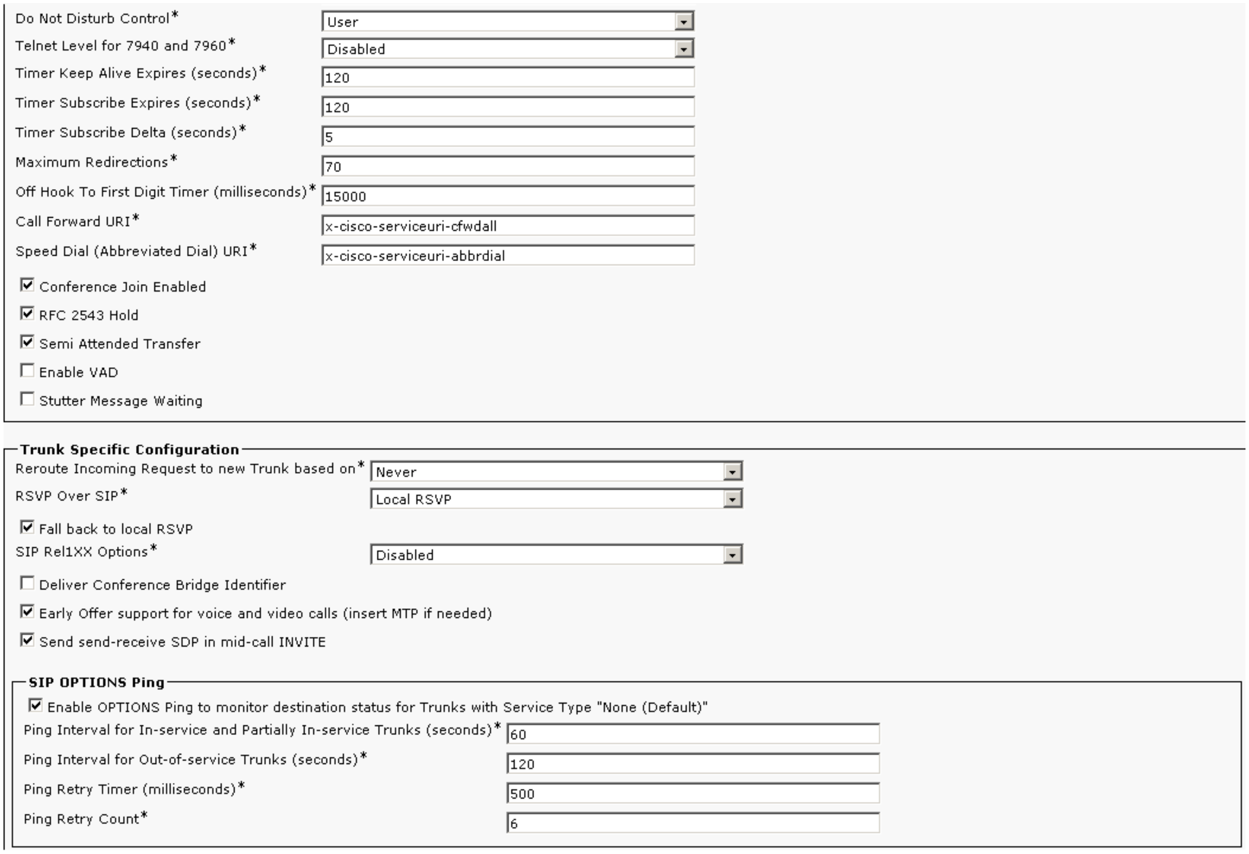


**Configuration of Media Resource Group List**
**Navigation Path:** Media Resources → Media Resource Group List

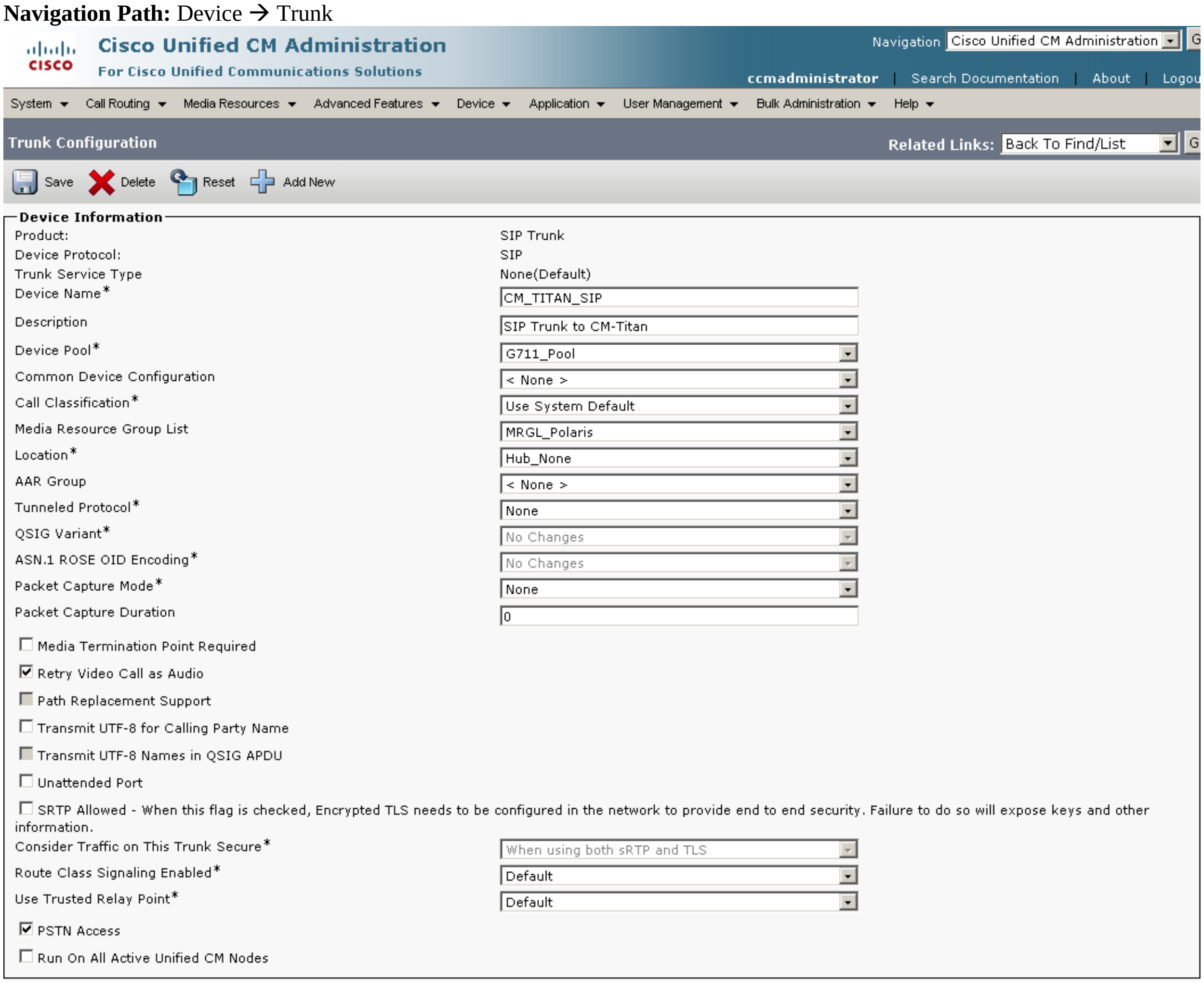


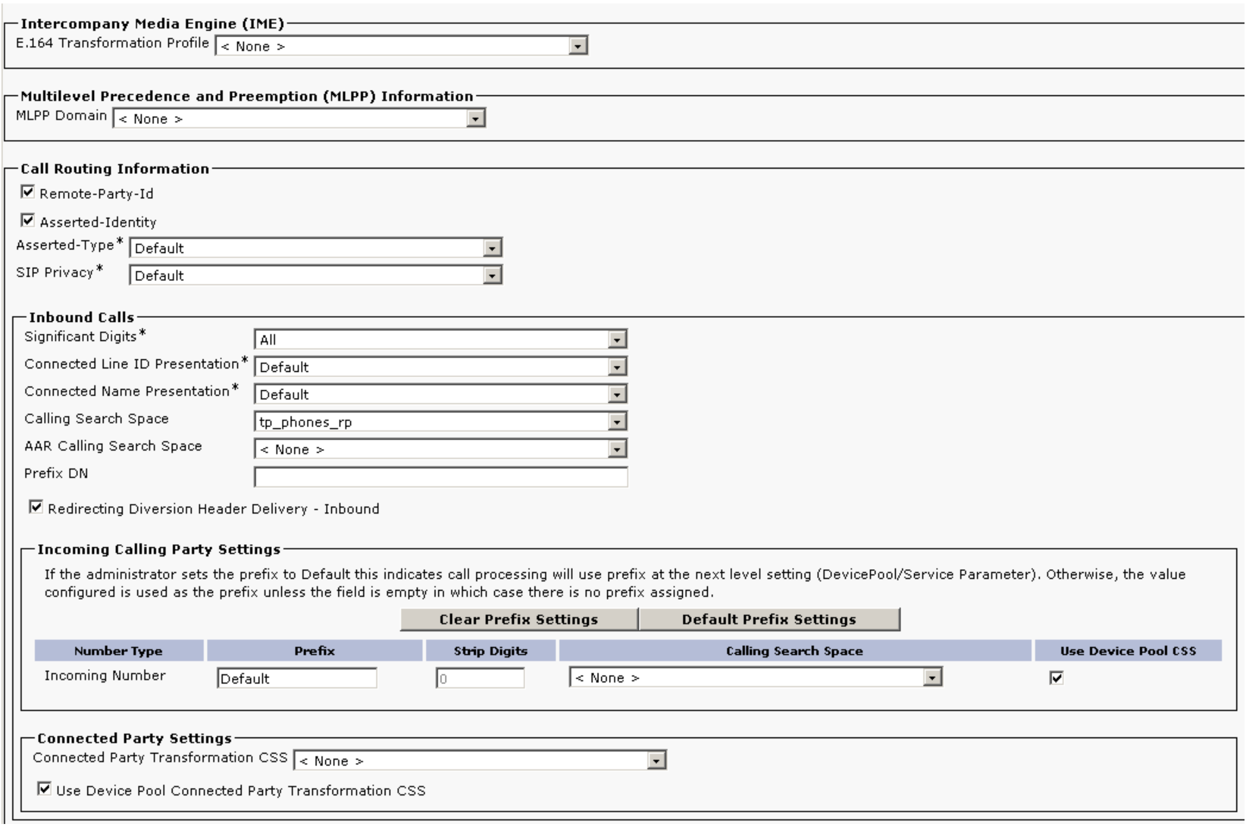
**Configuration of SIP Profile**
**Navigation Path:** Device → Device Settings → SIP Profile




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
Conference Join Enabled
RFC 2543 Hold
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
Fall back to local RSVP
SIP Rel1XX Options*
Disabled
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-receive SDP in mid-call INVITE
SIP OPTIONS Ping
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




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

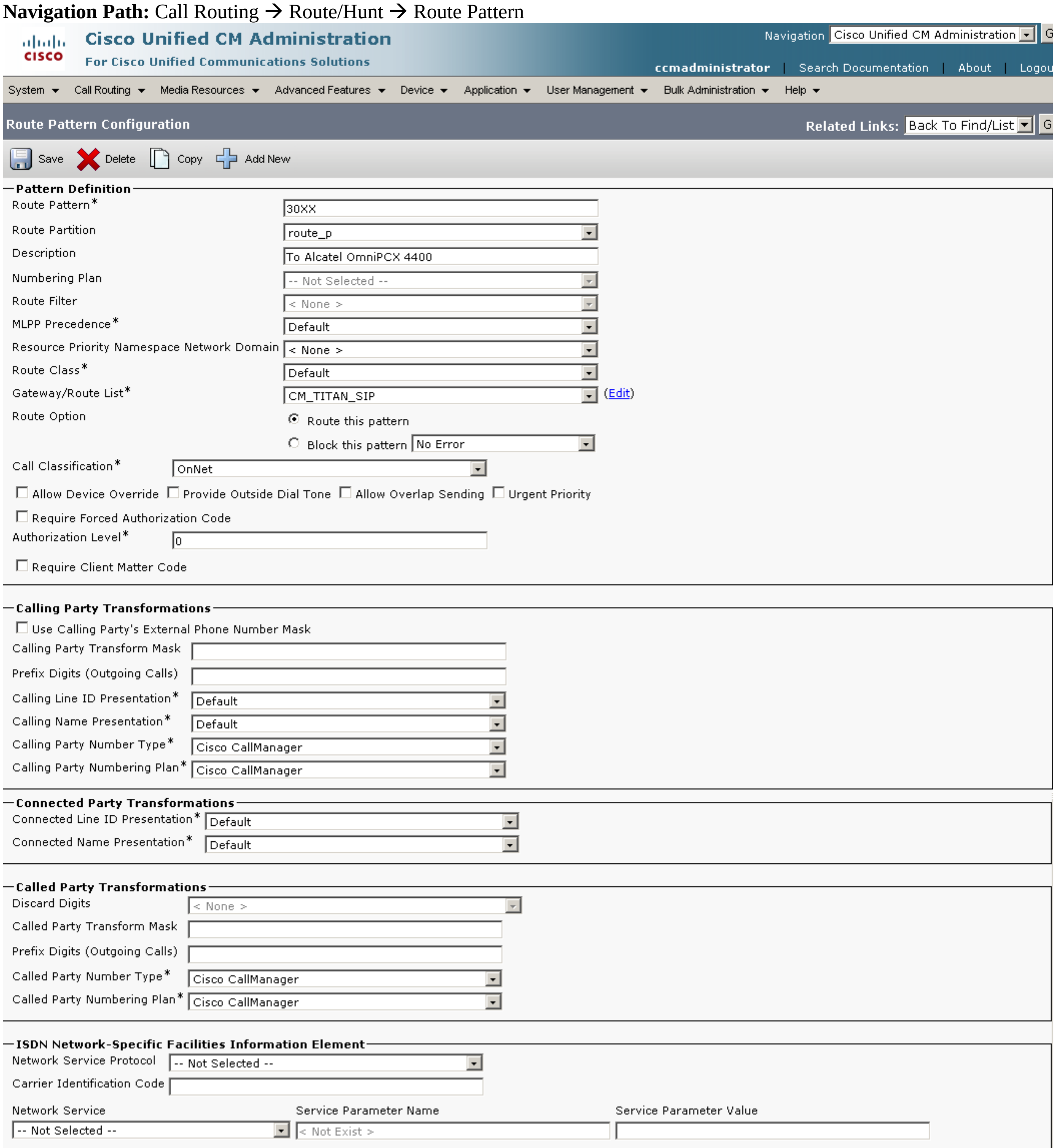
**Normalization Script**

Normalization Script: < None >

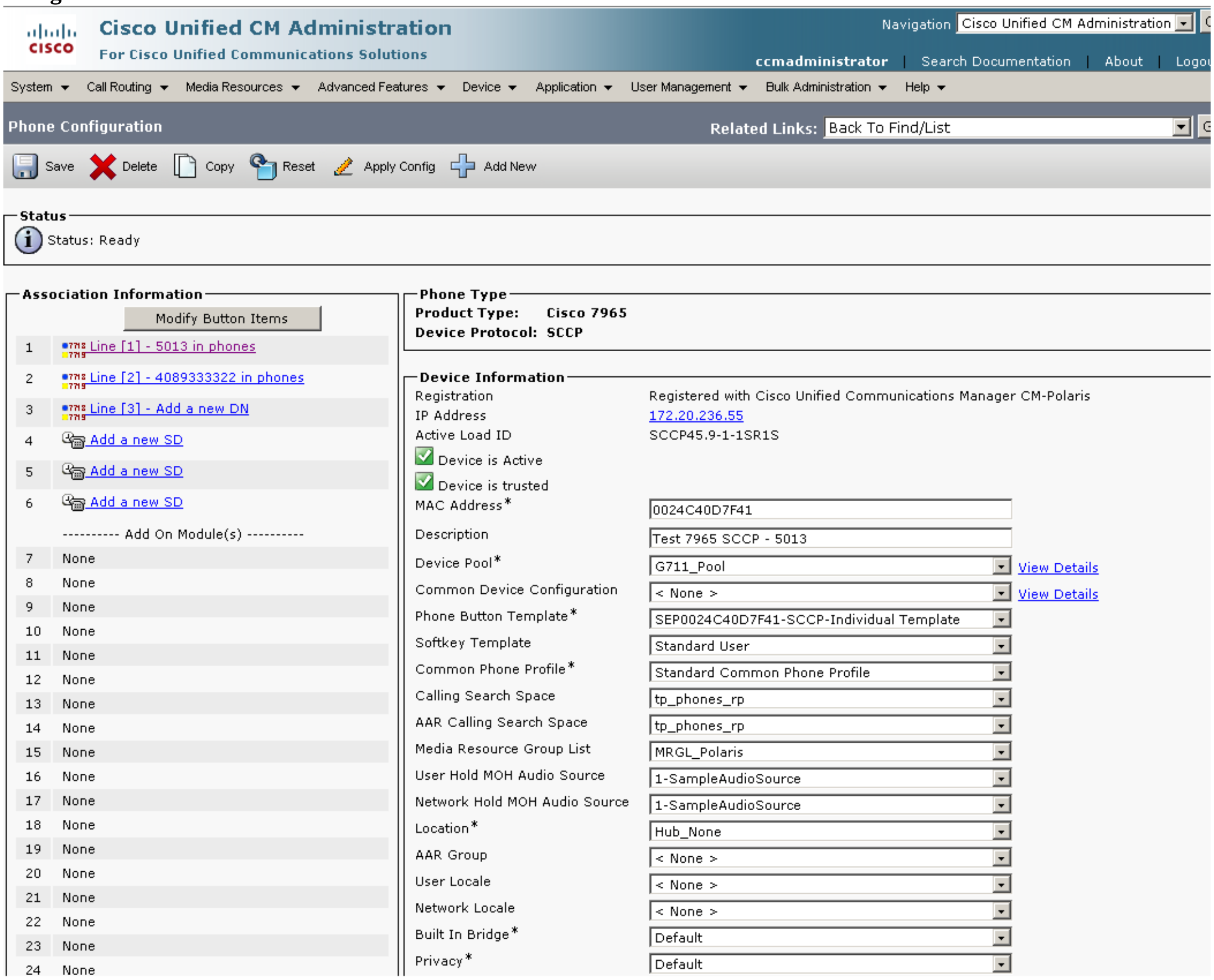
☐ Enable Trace

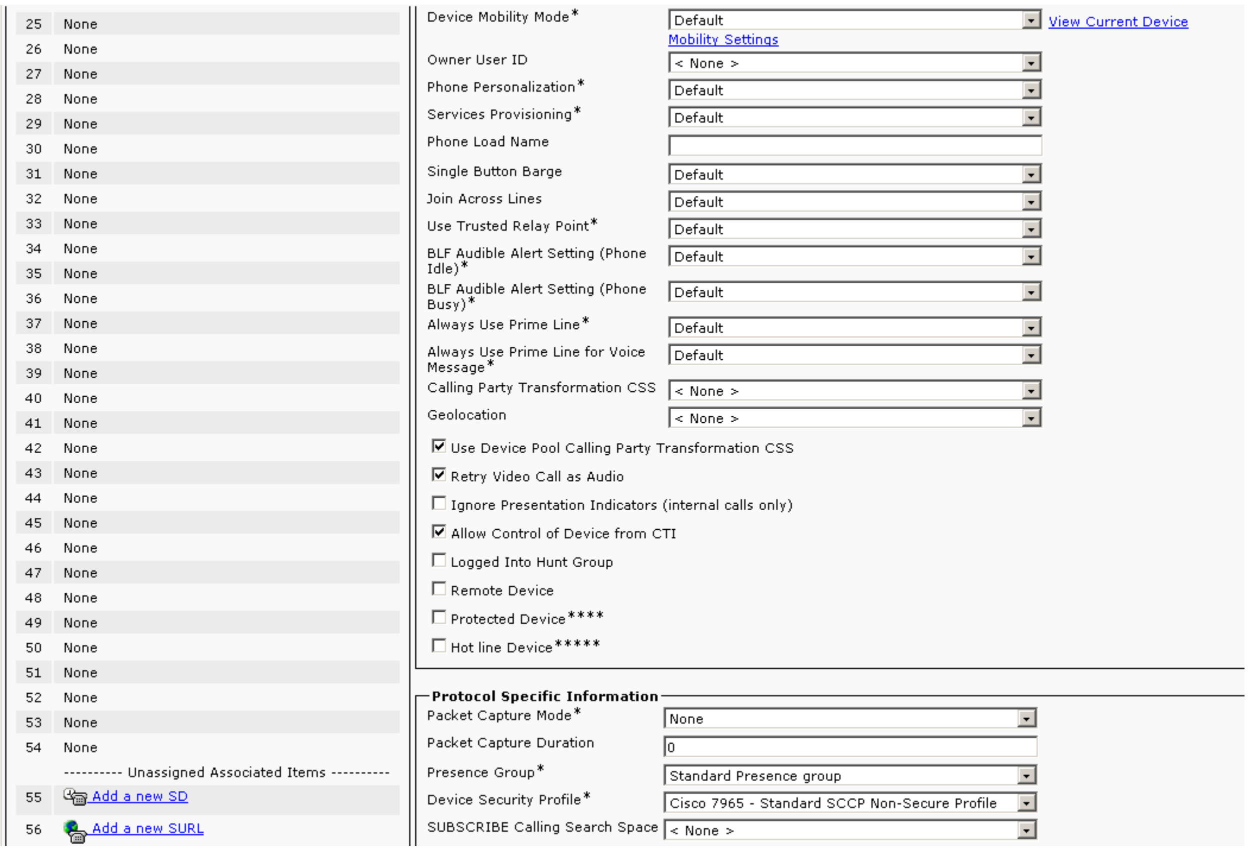
| | Parameter Name | Parameter Value |
|---|---|---|
| 1 | | |

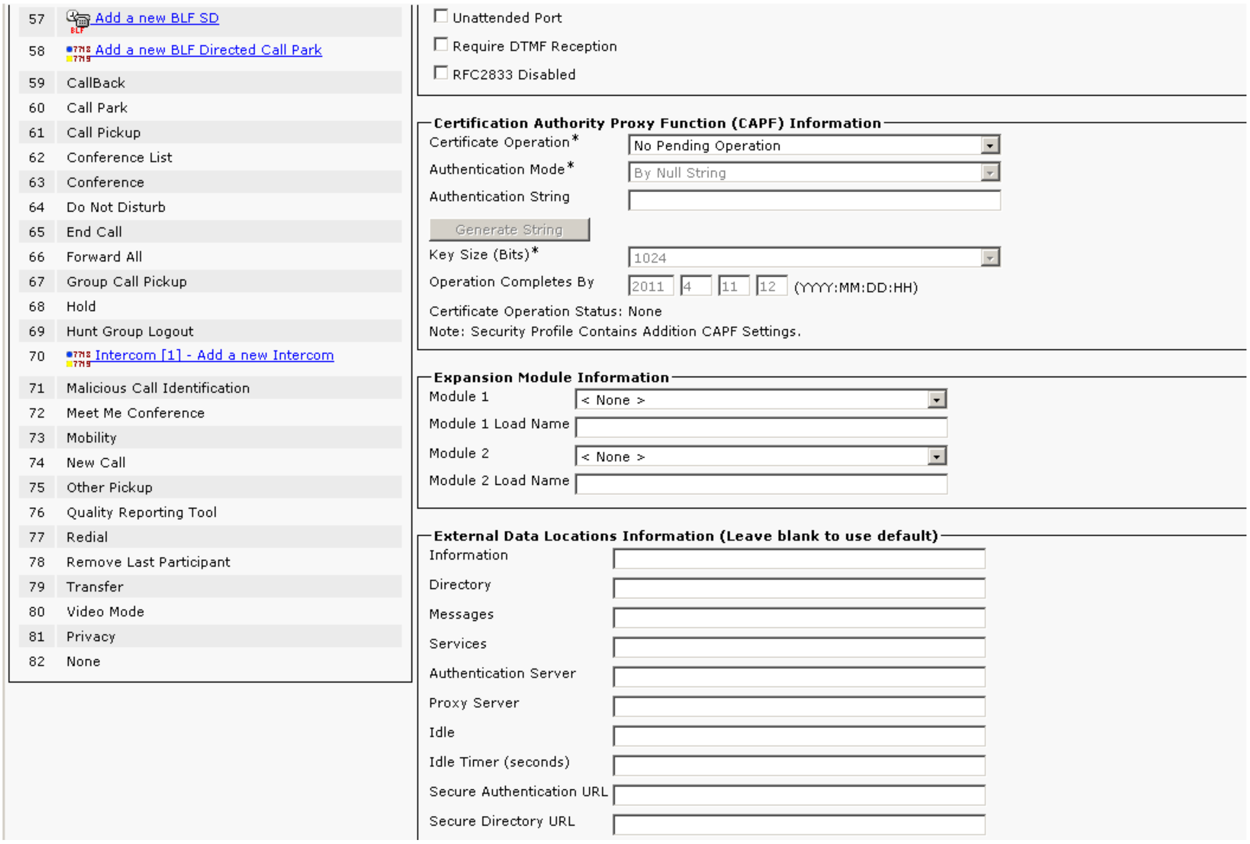
**Configuration of Route Pattern to Alcatel PBX through SME**
**Navigation Path:** Call Routing → Route/Hunt → Route Pattern



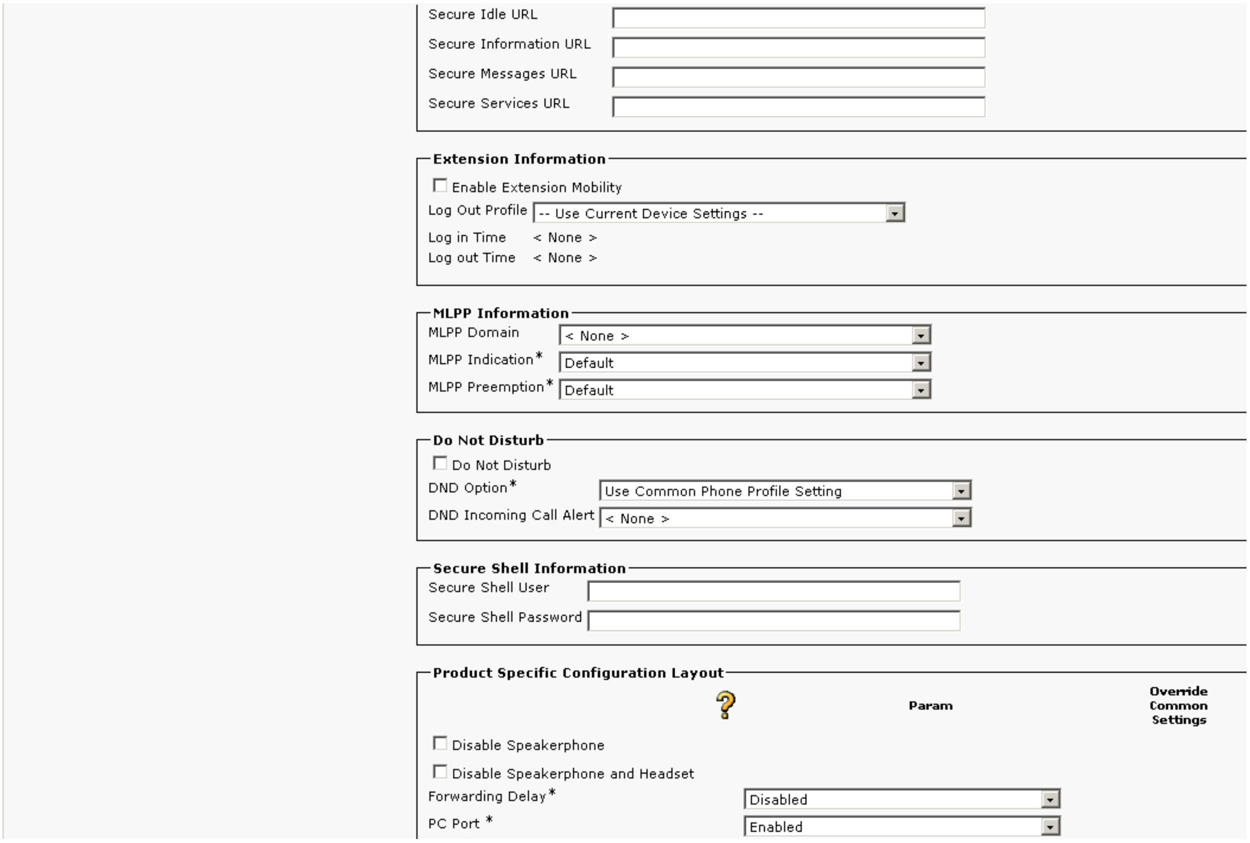
**Configuration of Cisco SCCP 7965 Phone**
**Navigation Path:** Device → Phone




25 None
26 None
27 None
28 None
29 None
30 None
31 None
32 None
33 None
34 None
35 None
36 None
37 None
38 None
39 None
40 None
41 None
42 None
43 None
44 None
45 None
46 None
47 None
48 None
49 None
50 None
51 None
52 None
53 None
54 None
---------- Unassigned Associated Items ----------
55 Add a new SD
56 Add a new SURL
Device Mobility Mode* Default
View Current Device Mobility Settings
Owner User ID < None >
Phone Personalization* Default
Services Provisioning* Default
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 >
Geolocation < 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****
Hot line Device*****
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
Device Security Profile* Cisco 7965 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space < None >

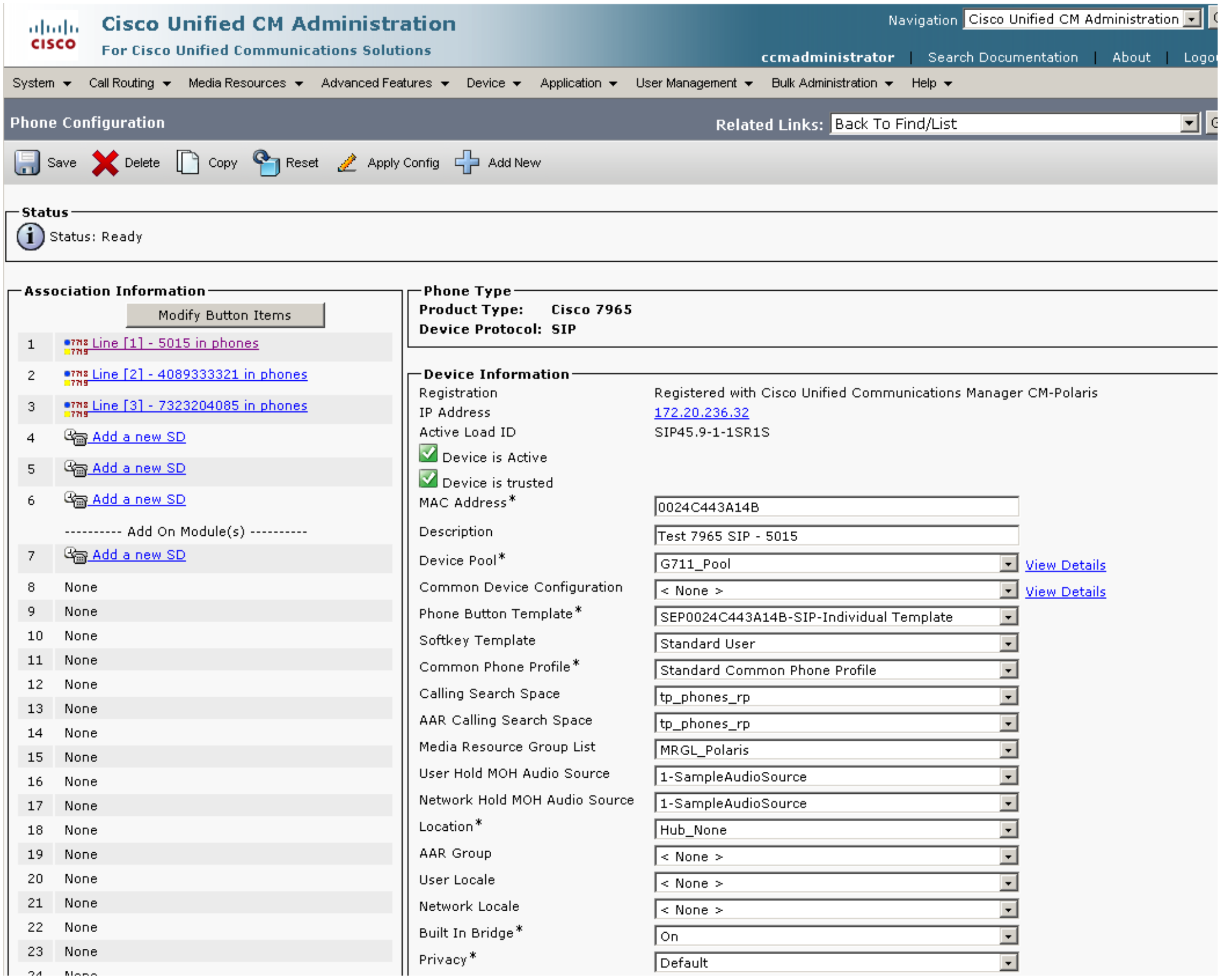


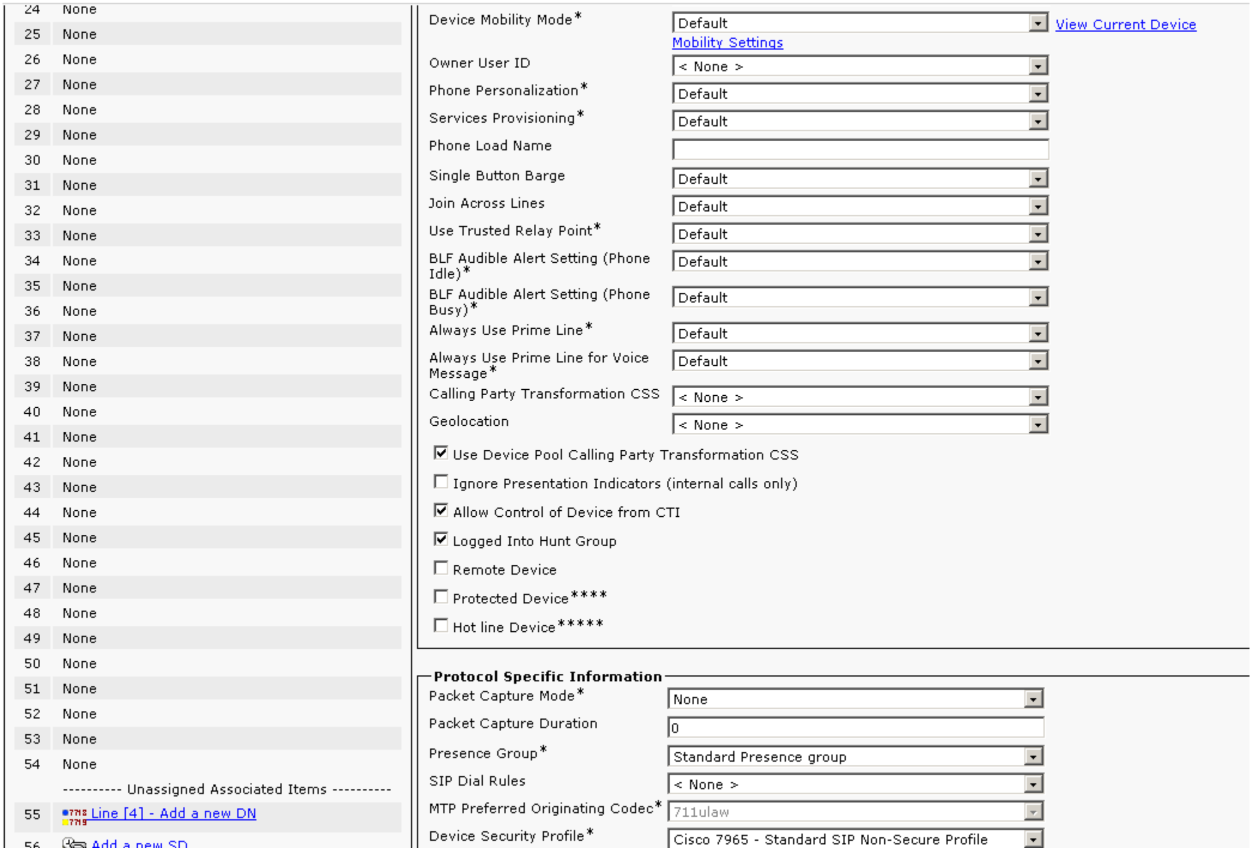
57 Add a new BLF SD
58 Add a new BLF Directed Call Park
59 CallBack
60 Call Park
61 Call Pickup
62 Conference List
63 Conference
64 Do Not Disturb
65 End Call
66 Forward All
67 Group Call Pickup
68 Hold
69 Hunt Group Logout
70 Intercom [1] - Add a new Intercom
71 Malicious Call Identification
72 Meet Me Conference
73 Mobility
74 New Call
75 Other Pickup
76 Quality Reporting Tool
77 Redial
78 Remove Last Participant
79 Transfer
80 Video Mode
81 Privacy
82 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
2011 4 11 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)
Secure Authentication URL
Secure Directory URL

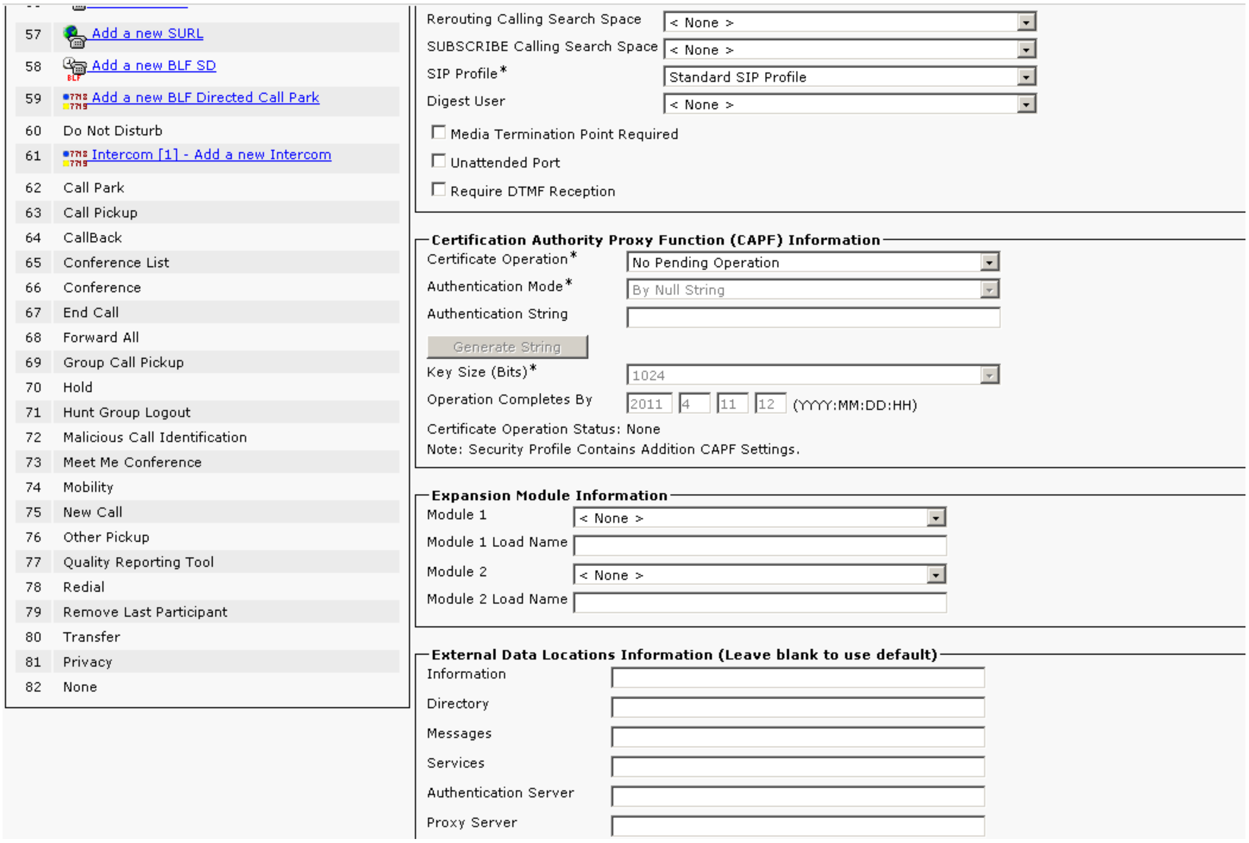

Secure Idle URL
Secure Information URL
Secure Messages URL
Secure Services URL
Extension Information
Enable Extension Mobility
Log Out Profile
-- Use Current Device Settings --
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain
< None >
MLPP Indication*
Default
MLPP Preemption*
Default
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
Param
Override Common Settings
Disable Speakerphone
Disable Speakerphone and Headset
Forwarding Delay*
Disabled
PC Port *
Enabled

| | | |
|---|---|---|
| Settings Access* | Enabled | ☐ |
| Gratuitous ARP* | Disabled | |
| PC Voice VLAN Access* | Enabled | |
| Video Capabilities* | Disabled | ☐ |
| Auto Line Select* | Disabled | |
| Web Access* | Disabled | ☐ |
| Days Display Not Active | Sunday<br>Monday<br>Tuesday | ☐ |
| Display On Time | 07:30 | ☐ |
| Display On Duration | 10:30 | ☐ |
| Display Idle Timeout | 01:00 | ☐ |
| 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 | | |
| Display On When Incoming Call* | Disabled | |
| 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 Headset* | Enabled | |
| Peer Firmware Sharing* | Enabled | ☐ |
| 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 | |
| Wireless Headset Hookswitch Control* | Disabled | |
| IPv6 Load Server | | ☐ |
| IPv6 Log Server | | |
| 802.1x Authentication* | User Controlled | ☐ |
| Detect Unified CM Connection Failure* | Normal | |
| Minimum Ring Volume* | 0-Silent | |
| HTTPS Server* | http and https Enabled | |
| Handset/Headset Monitor* | Enabled | |
| Enbloc Dialing* | Enabled | |
| Switch Port Remote Configuration* | Disabled | ☐ |
| PC Port Remote Configuration* | Disabled | ☐ |
| Automatic Port Synchronization* | Disabled | ☐ |

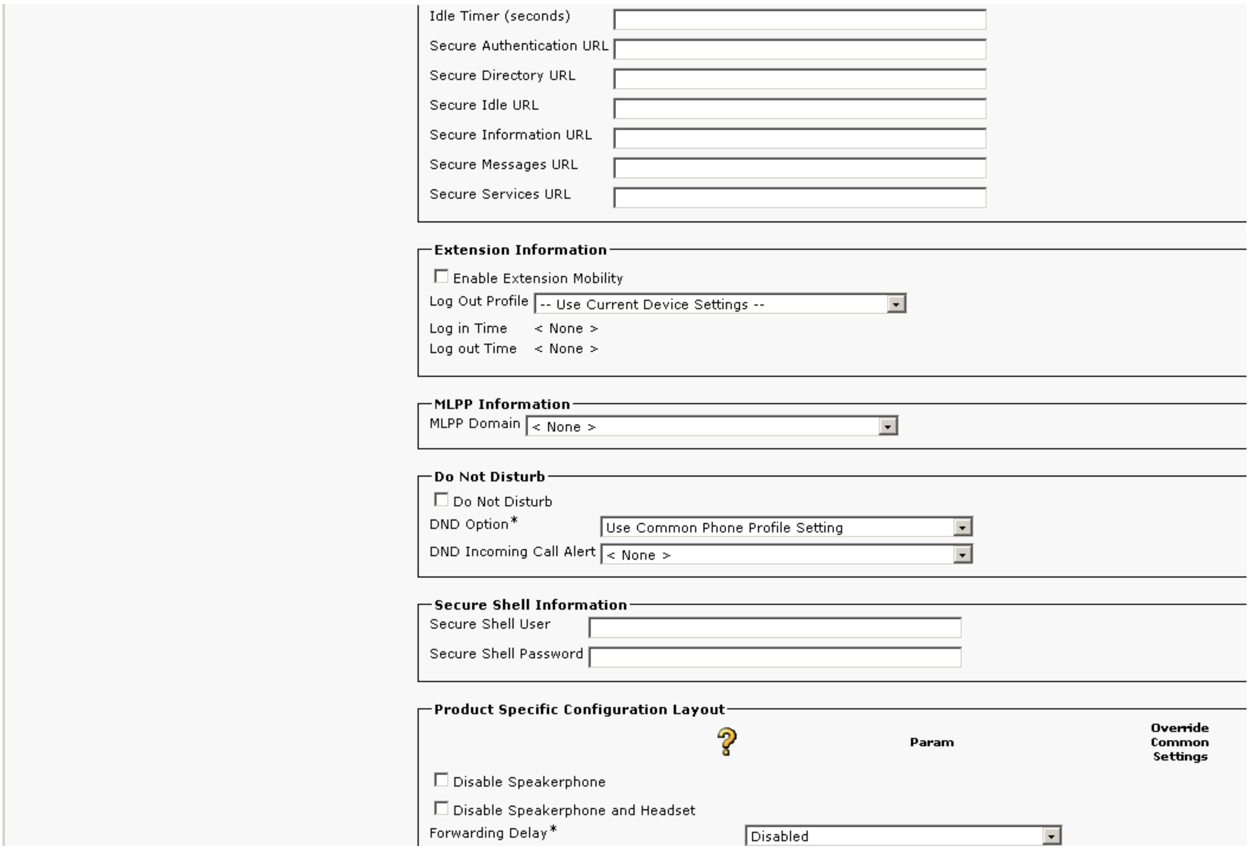
**Configuration of Cisco SIP 7965 Phone**
**Navigation Path:** Device → Phone




24 None
25 None
26 None
27 None
28 None
29 None
30 None
31 None
32 None
33 None
34 None
35 None
36 None
37 None
38 None
39 None
40 None
41 None
42 None
43 None
44 None
45 None
46 None
47 None
48 None
49 None
50 None
51 None
52 None
53 None
54 None
---------- Unassigned Associated Items ----------
55 Line [4] - Add a new DN
Add a new SD
Device Mobility Mode* Default
View Current Device Mobility Settings
Owner User ID < None >
Phone Personalization* Default
Services Provisioning* Default
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 >
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****
Hot line 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 7965 - Standard SIP Non-Secure Profile



57 Add a new SURL
58 Add a new BLF SD
59 Add a new BLF Directed Call Park
60 Do Not Disturb
61 Intercom [1] - Add a new Intercom
62 Call Park
63 Call Pickup
64 CallBack
65 Conference List
66 Conference
67 End Call
68 Forward All
69 Group Call Pickup
70 Hold
71 Hunt Group Logout
72 Malicious Call Identification
73 Meet Me Conference
74 Mobility
75 New Call
76 Other Pickup
77 Quality Reporting Tool
78 Redial
79 Remove Last Participant
80 Transfer
81 Privacy
82 None
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
2011 4 11 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 Timer (seconds)
Secure Authentication URL
Secure Directory URL
Secure Idle URL
Secure Information URL
Secure Messages URL
Secure Services URL

**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 >

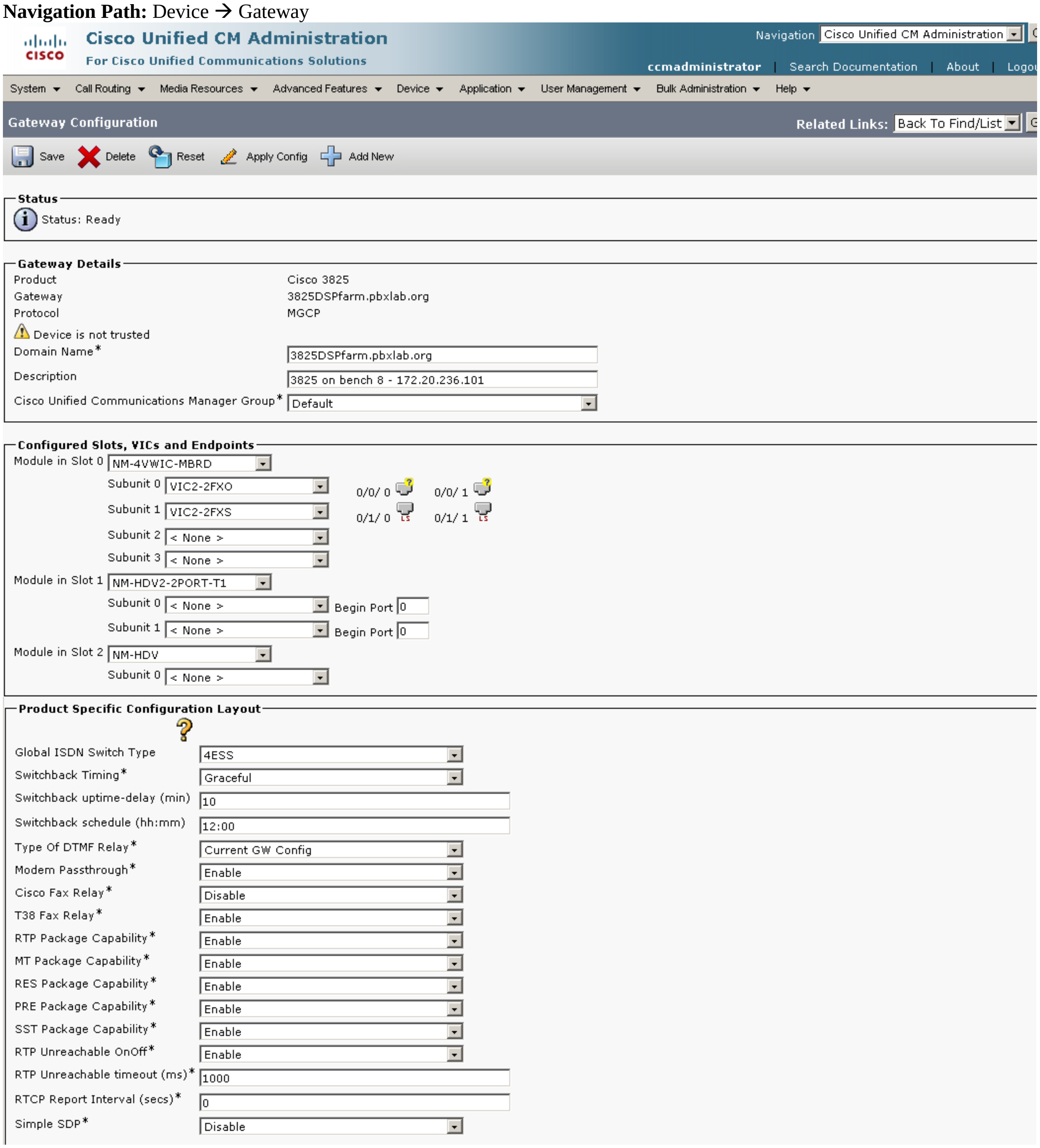
**Do Not Disturb**
Do Not Disturb
DND Option* Use Common Phone Profile Setting
DND Incoming Call Alert < None >

**Secure Shell Information**
Secure Shell User
Secure Shell Password

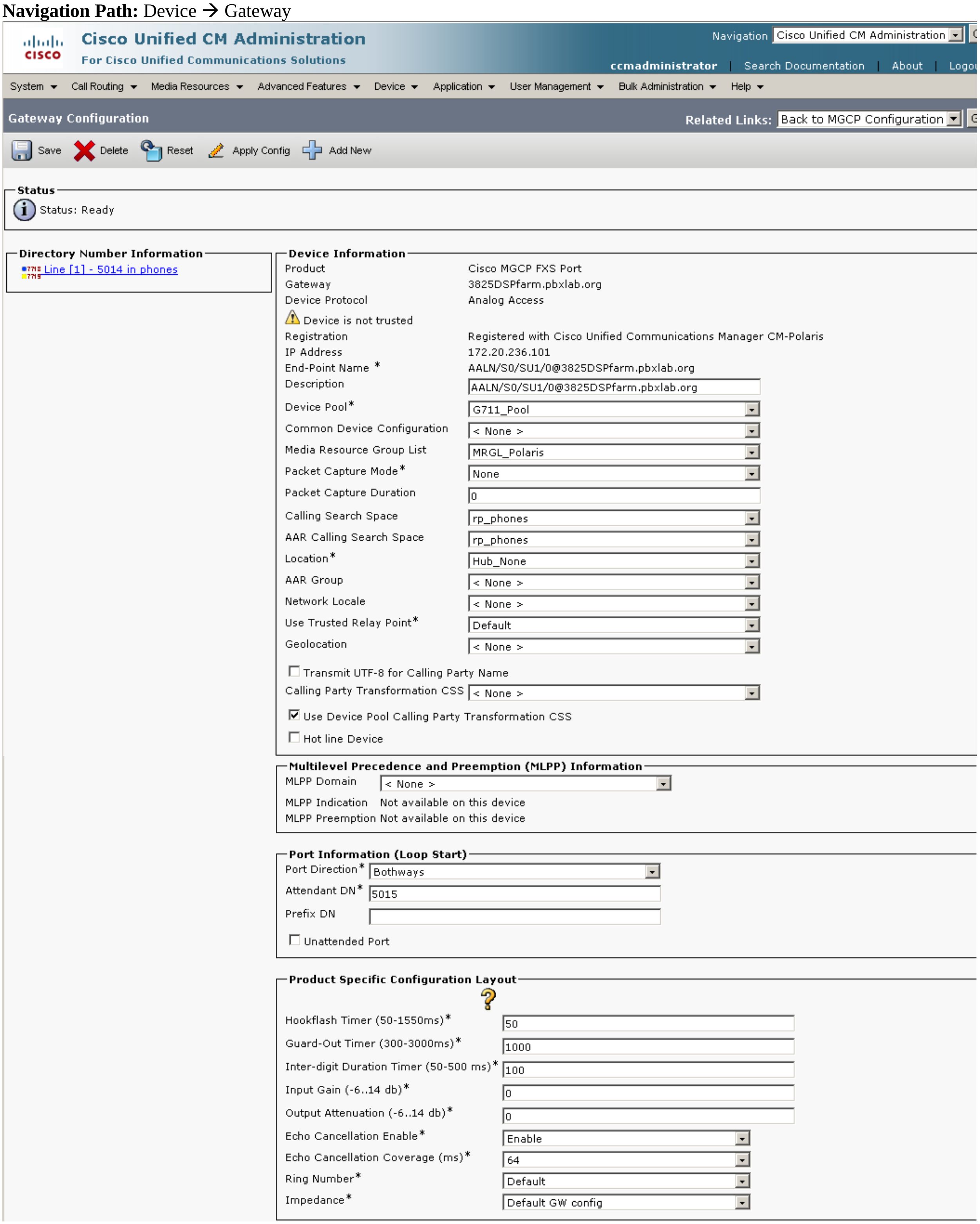
**Product Specific Configuration Layout**
Param
Override Common Settings
Disable Speakerphone
Disable Speakerphone and Headset
Forwarding Delay* Disabled

| | | |
|---|---|---|
| PC Port * | Enabled | |
| Settings Access* | Enabled | ☐ |
| Gratuitous ARP* | Disabled | |
| PC Voice VLAN Access* | Enabled | |
| Auto Line Select* | Disabled | |
| Web Access* | Disabled | ☐ |
| Days Display Not Active | Sunday<br>Monday<br>Tuesday | ☐ |
| Display On Time | 07:30 | ☐ |
| Display On Duration | 10:30 | ☐ |
| Display Idle Timeout | 01:00 | ☐ |
| 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 | | |
| Display On When Incoming Call* | Disabled | |
| 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 Headset* | Enabled | |
| Peer Firmware Sharing* | Enabled | ☐ |
| 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 | |
| Wireless Headset Hookswitch Control* | Disabled | |
| IPv6 Load Server | | ☐ |
| IPv6 Log Server | | |
| 802.1x Authentication* | User Controlled | ☐ |
| Detect Unified CM Connection Failure* | Normal | |
| Minimum Ring Volume* | 0-Silent | |
| HTTPS Server* | http and https Enabled | |
| Handset/Headset Monitor* | Disabled | |
| Enbloc Dialing* | Enabled | |
| Switch Port Remote Configuration* | Disabled | ☐ |
| PC Port Remote Configuration* | Disabled | ☐ |
| Automatic Port Synchronization* | Disabled | ☐ |

**Configuration of MGCP FAX Gateway**
**Navigation Path:** Device → Gateway



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



### Acronyms

| Acronym | Definitions |
|---|---|
| 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 |
| DNS | Domain Name Server |
| FQDN | Fully Qualified Domain Name |
| MWI | Message Waiting Indicator |
| PSTN | Public Switched Telephone Network |
| SIP | Session Initiated Protocol |

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**Corporate Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

**European Headquarters**

Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

**Americas Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

**Asia Pacific Headquarters**

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799


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