Application Note

Alcatel Omni PCX 4400 Release 6.0 with SIP Trunk to Cisco Unified CallManager Release 5.0

December 13, 2006 Initial Release

Table of Contents

Introduction

The purpose of this document is to detail the steps and configuration necessary for Cisco Unified CallManager 5.0 to interoperate with the Alcatel Omni PCX 4400 running software release 6.0 via SIP Trunk. It also includes information on interoperability issues, features and limitation with this type of integration.

Network Topology

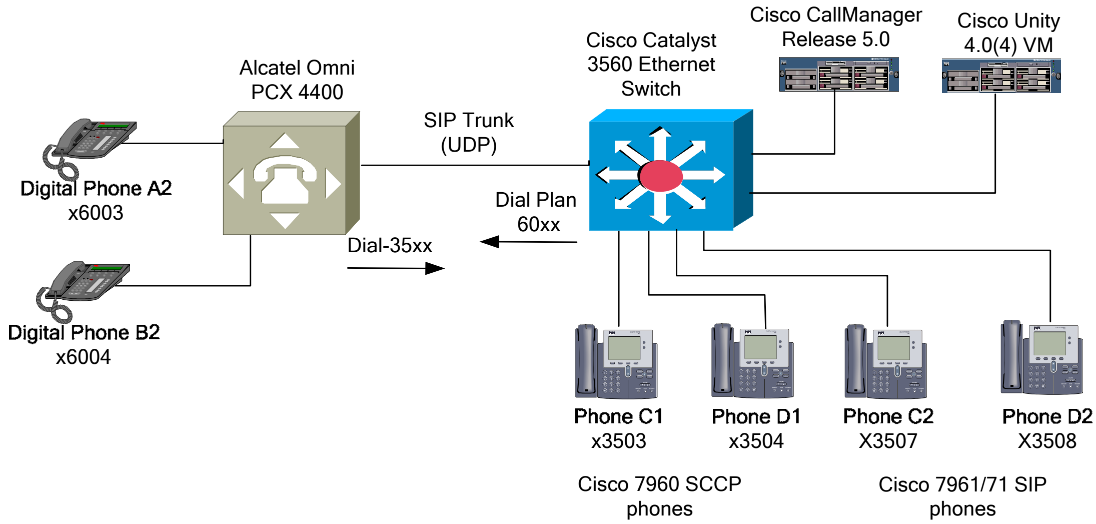
Cisco Unified

Boxes wrong

Switch

PBX

Figure 1. Network Topology



Limitations

Alcatel Omni PCX 4400 acts a SIP Proxy server whereas Cisco Unified CallManager (CUCM) acts as a SIP Back-to-Back User-Agent (B2B-UA).

"Media Termination Point Required" check box must be enabled on the CUCM SIP Trunk for basic call to work. Without MTP check box on the SIP trunk, CUCM is performing "delay-media" and Alcatel call server reject the call with a SIP 488 error message. It seems like Alcatel does not support SIP delay media connection, therefore, the "Media Termination Point Required" checkbox must be enabled under the SIP Trunk configuration in order for the two systems to interoperate successfully.

"Redirect by Application" checkbox must be enable under the SIP Profile used by the SIP Trunk in order for External Call Forwarding to work properly.

For CLIP and CNIP features:

- Alcatel Omni PCX 4400 with software release 6.0 does not support passing the name and number information across the Public SIP trunk using "P-Asserted-Id" or "Remote-Party-Id" fields. CUCM on the other hand, does support the feature using "Remote-Party-Id" field to pass the name and number information across the SIP Trunk. As a result of the differences, both systems will use the information from the SIP INVITE From header as the caller information.

For CLIR and CNIR features:

- Alcatel Omni PCX 4400 with software release 6.0 does not support Calling Name and Number Restriction across the Public SIP Trunks using "P-Asserted-Id" with "privacy" or "Remote-Party-Id" with "privacy" fields. CUCM does support these features using the "Remote-Party-Id" and "privacy" fields.

For COLP, CONP, COLR and CONR features:

- Alcatel Omni PCX 4400 with software release 6.0 does not support Connected Name and Number presentation and/or restriction across the Public SIP Trunk. CUCM does support these features using “Remote-Party-Id” and “privacy” fields.

For Alerting Name:

- Alcatel Omni PCX 4400 with software release 6.0 does not support Alerting Name feature support across Public SIP Trunk. CUCM 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.

For SIP Blind Call Transfer:

- Both Alcatel Omni PCX 4400 phones and Cisco Unified CallManager TNP phones (7961, 7970, 7971 and 7911) do not support SIP Blind Call Transfer. With SIP Blind Call Transfer, the transferor places the original call on hold and dials the target. The transferor then uses SIP signaling to redirect the transferee to the target. There is no call made to the target prior to transfer. The timing of when the transferor drops out of the call depends on the transferor's implementation of this feature, but most likely the drop occurs when the transferor is notified that the redirect operation was accepted.

For 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.

For 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 CUCM SCCP phone (3503) calls Alcatel digital phone (6003) and 6003 perform early attended transfer to another Alcatel digital phone (6004). From an external sniffer trace capture, it looks like there is an Alcatel software issue with not sending CUCM the right dialog to be replaced within the SIP Refer message. To Alcatel phone 6003, there are 2 dialogs. One is from the SCCP 3503 to Alcatel 6003 which is D1. The other one is Alcatel 6003 to Alcatel 6004 which is D2. The issue is Alcatel 6003 send a SIP Refer w/replaces header to SCCP 3503 for the D1 dialog (to replace itself). It should have sent a SIP Refer w/replaces header to replace Alcatel 6004 (D2 dialog). Since Alcatel phone 6003 sends Refer w/replaces to CUCM with D1 (instead of D2), CUCM software logic think this dialog is its own dialog and thus reject this Refer/replaces call. If Alcatel phone 6003 would have send to CUCM a SIP Refer w/replaces with D2, CUCM would have send a SIP Invite w/replaces with D2 to Alcatel PBX via the SIP trunk to replace Alcatel 6003 D1 dialog.
- Another example of call transfer failed to complete is for Early Attended Network Call transfer (where Alcatel phone is the transferor and CUCM phones are the calling party and target phone). The call scenario is when CUCM SCCP phone (3503) calls Alcatel digital phone (6003) and 6003 perform early attended network call transfer back to another CUCM SCCP digital phone (3504). Alcatel send a SIP Refer message with an incorrect Refer-To header. In the SIP Refer message, the Refer-To header has "sip:SIP_2@172.20.9.250". The host portion of the SIP URL has "SIP_2" which is the SIP trunk group name configured on the Alcatel Onmi PCX Call Server. It should have been populated with the CUCM transferred-to party phone information instead. This issue might be related to Alcatel software release defect id: XTSce61919 which indicate that their software SIP stack should use re-INVITE method instead of Refer message for Public SIP Trunk. This Public SIP Trunk feature will be available with their next software release.

For 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.

For 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 CUCM to have the "Redirect by Application" checkbox enabled under the SIP Profile used by the SIP Trunk to the Alcatel Call Server. For example, for the call

flow where CUCM SCCP phone (3503) calls Alcatel digital phone (6003) and 6003 perform CFU or CFB back to another CUCM SCCP phone (3504), without the checkbox enabled, the call would fail. Analysis of the sniffer trace capture for the call shows CUCM sends out a regular SIP Invite message to Alcatel. Alcatel respond back with SIP 302 Move Temporarily with Contact header "sip:3504@172.20.9.250". CUCM then send a new SIP Invite message to Alcatel based on the Contact header information. Alcatel again respond back with SIP 301 Move Permanently with a different Contact header "sip:3504@172.20.150.251". CUCM perform digit analysis on the information in the contact header and send out another SIP INVITE to itself. CCM then failed the call with SIP 500 Internal Server Error message because incoming SIP Invite request came from a source address that doesn't match a configured SIP trunk in the CUCM database and thus the call will get rejected. To resolved this issue, we need to either add a SIP trunk on the CUCM to itself so that it would pass the source address validation or enable/check the "Redirect by Application" checkbox under the SIP Profile used by the SIP Trunk to the Alcatel PBX. With the "Redirect by Application" checkbox enabled, CUCM uses a different application layer which has the necessary information and is smart enough to do a "CUCM internal" join call without the need to do CUCM to CUCM SIP Invite hairpin call.

- For CFNA call scenario where Alcatel station is the forwarding device, one way audio is encountered. The call flow is CUCM SCCP phone (3503) calls Alcatel digital phone (6003) and 6003 perform CFNA back to another CUCM SCCP phone (3504). This issue occurred independent of whether the "Redirect by Application" checkbox is enabled or not. Audio path work fine in the direction of forwarded-to party (3504) to the calling party (3503) but not vice versa. From the sniffer trace capture for the call, it showed that Alcatel sends different RTP port in the SDP section of the SIP 180 Ringing message vs the SIP 200 OK message. This is not legal per SIP RFC3261 where it's state the following "If the initial offer is in an INVITE, the answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE. For this specification, that is only the final 2xx response to that INVITE. That same exact answer MAY also be placed in any provisional responses sent prior to the answer. The UAC MUST treats the first session description it receives as the answer, and MUST ignore any session descriptions in subsequent responses to the initial INVITE."
- For CFU, CFB and CFNA call scenarios where CUCM phone is the forwarding device, one way audio is encountered but for a different reason. As an example, for the call flow where Alcatel phone (6003) call CUCM phone (3503) and 3503 perform a CFNA back to another Alcatel phone (6004), one-way audio occurred. Audio path work fine in the direction of the calling party (6003) to the forwarded-to party (6004) but not the other way. This one-way audio issue depends on when the forwarded-to party answered the call. If the forwarded-to party answers the call on the first ring, then audio works fine in both direction. If the forwarded-to party answers the call after the 2nd ring, then one-way audio occurred. From the sniffer trace capture and further analysis, it was determined that the root cause of the issue is due to ICMP port unreachable errors received from the Alcatel PBX on the original call leg during the alerting state of the call forwarding. For the call which the forwarded-to party answers the call after the 2nd ring, there were ICMP port unreachable error messages sent by the Alcatel PBX to the CUCM. After a certain amount of ICMP port unreachable error message received within a certain time frame, CUCM will stop transmitting the RTP packets toward the Alcatel PBX, therefore this led to the one way audio issue. We are not sure as to why Alcatel sends ICMP port unreachable during the call forwarding timeframe.

For Call Conference:

- Both systems support call conferencing using their local media resources. However, if Alcatel station is the conferencing party, local conference will work fine but network conference encounter one-way audio issue. For example, a network conference call where Alcatel station conference in a CUCM station via the SIP trunk, one-way audio occurred between the conference-in party and the rest of the other parties. Analysis of the sniffer trace capture showed when the Alcatel station perform the conference, Alcatel send out a SIP INVITE message to CUCM with SDP parameter "a=sendonly". Alcatel did not sent any additional SIP signaling message to change the SDP parameter to "a=sendrecv" for the call leg. As a result, one way audio occurred.

No support for centralized voice messaging across the SIP Trunk. CUCM uses SIP Diversion header to pass the redirect information across the SIP Trunk. However, Alcatel Omni PCX 4400 does not support SIP Diversion header. Therefore, without the redirect information, Centralized Voice Messaging will not work.

No support for MWI- Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk. CUCM uses SIP Notify message with SDPinfo Message Waiting=yes/no for MWI notification. Alcatel Omni PCX 4400 does not support MWI across their SIP Trunk and will not interpret those SIP signaling messages.

RFC2833 - Dynamic RTP Payload Type for DTMF-relay :

- There is an interoperability issue with Alcatel Omni PCX 4400 regarding the RFC2833-Dynamic RTP Payload Type for DTMF-relay feature. For outbound call, Alcatel Omni PCX 4400 does not advertise the support for RFC2833 to CUCM. Therefore, when digits are pressed on the Alcatel digital station, Alcatel media gateway pass the DTMF tones via in-band within the RTP packets using the voice codec negotiated. Cisco Unified CallManager or Cisco Unity currently does not support the passing of DTMF digits in-band via the voice codec. The DTMF digits will be treated the same as the caller voice stream and will not be interpret as DTMF events. For inbound call, if the incoming SIP INVITE message contained SDP parameter for the support of

RFC2833 DTMF-relay event, Alcatel Omni PCX 4400 will support it. However, Alcatel does not acknowledge this support back to the originator device. As a result, the originating side assumes RFC2833 DTMF-relay feature is not supported since there was no acknowledge back. Therefore, any RFC2833 DTMF-relay event packets send by Alcatel will be treated as regular voice stream packet and not DTMF-relay digits. In summary, since RFC2833-Dynamic RTP Payload Type for DTMF-relay feature was not properly negotiated by both side, Alcatel should not have send out digits via RFC2833.

System Components

Hardware Requirements

Cisco Unified CallManager MCS -7835H server,

Unity server MCS-7835H

Catalyst switch 3560

Cisco IP Phones 7970, 7971, and 7960

Alcatel Omni PCX 4400 PBX with INT-IP2 card

Alcatel digital phone (4035)

Software Requirements

Cisco Unified CallManager Release 5.0.4

Cisco Unity Release 4.0(4)

Alcatel software R6.0 (f1.602)

c3560-i5-mz.122-20.EX.bin

Features

Features Supported

CLIP-Calling Line (Number) Identification Presentation (Please see the Limitation section)

CLIR-Calling Line (Number) Identification Restriction (Please see the Limitation section)

CNIP-Calling Name Identification Presentation (Please see the Limitation section)

CNIR-Calling Name Identification Restriction (Please see the Limitation section)

Alerting Name (Please see the Limitation section)

Attended Call Transfer (Please see the Limitation section)

Early Attended Call Transfer (Please see the Limitation section)

CFU-Call Forwarding Unconditional (Please see the Limitation section)

CFB-Call Forwarding Busy (Please see the Limitation section)

CFNA-Call Forwarding No Answer (Please see the Limitation section)

COLP-Connected Line (Number) Identification Presentation (Please see the Limitation section)

COLR- Connected Line (Number) Identification Restriction (Please see the Limitation section)

CONP-Connected Name Identification Presentation (Please see the Limitation section)

CONR-Connected Name Identification Restriction (Please see the Limitation section)

Hold and Resume

Conference Call (Please see the Limitation section)

DTMF-relay using RFC2833 (Please see the Limitation section)

Features Not Supported

MWI- Message Waiting Indication (lamp ON, lamp OFF) across the SIP Trunk

Call Completion (Callback; Automatic Callback)

Blind Call Transfer

Callback feature via the SIP trunk

Configuration

Configuration Sequence and Tasks

Alcatel Call Server Configuration:

1. Alcatel Omni PCX 4400 Software Version and Hardware Configuration List
2. Configure SIP Network: Translator → Network Routing Table
3. Configure SIP Trunk group
4. Configure T2 Trunk Group Type
5. Configure Virtual Access for SIP
6. Configure Alcatel SIP Gateway
7. Configure Alcatel SIP Proxy setting
8. Configure SIP External Gateway
9. Configure IP Parameters
10. Configure GF diversion on joining
11. Configure call routing (Translator) to Cisco CallManager phone extensions
12. Configure Alcatel standard users (digital stations)

Cisco Unified CallManager:

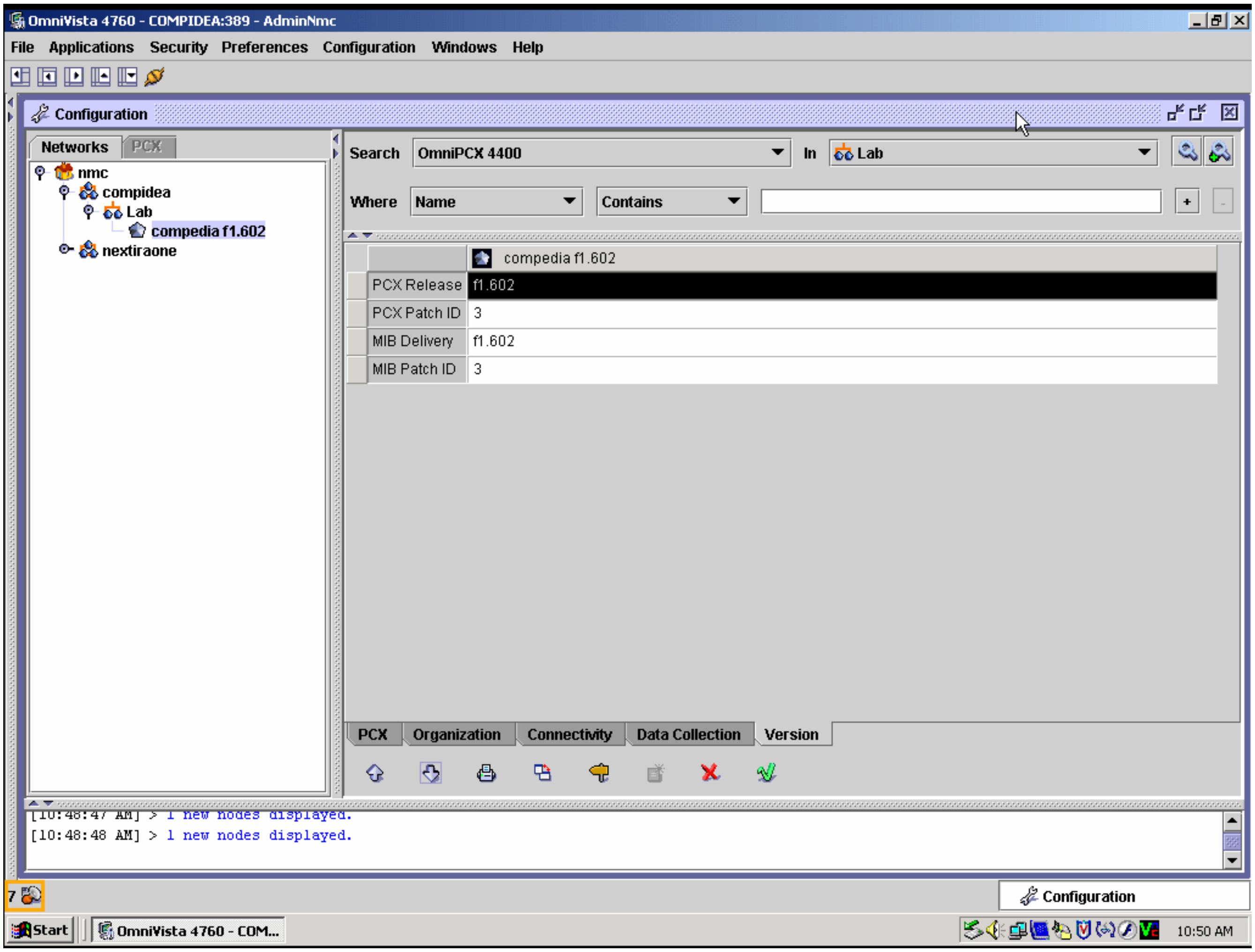
1. Cisco Unified CallManager Software Version
2. Enterprise Parameter Top Level Domain Setting
3. SIP Trunk Security Profile
4. SIP Phone Security Profile
5. SCCP Phone Security Profile
6. SIP Profile for SIP Trunk to Alcatel Call Server/Proxy Server
7. Standard SIP Profile
8. Media Resource Group
9. Media Resource Group List
10. Assigned MGRL in the Default Device Pool
11. SIP Trunk to Alcatel Call Server/Proxy Server
12. SIP and SCCP Device Level and DN Level configuration
13. Route Pattern to Alcatel phone extensions
14. Voice Mail Ports for Unity Voice Mail system
15. Voice Mail Pilot for Unity Voice Mail system
16. Voice Mail Profile for Unity Voice Mail system
17. Voice Mail MWI ON and OFF for Unity Voice Mail system
18. Voice Mail Line Group
19. Voice Mail Hunt List
20. Voice Mail Hunt Pilot

Cisco Unity:

1. Cisco Unity software version
2. Cisco Unity Integration with Cisco Unified CallManager
3. Cisco Unity Voice Mail ports

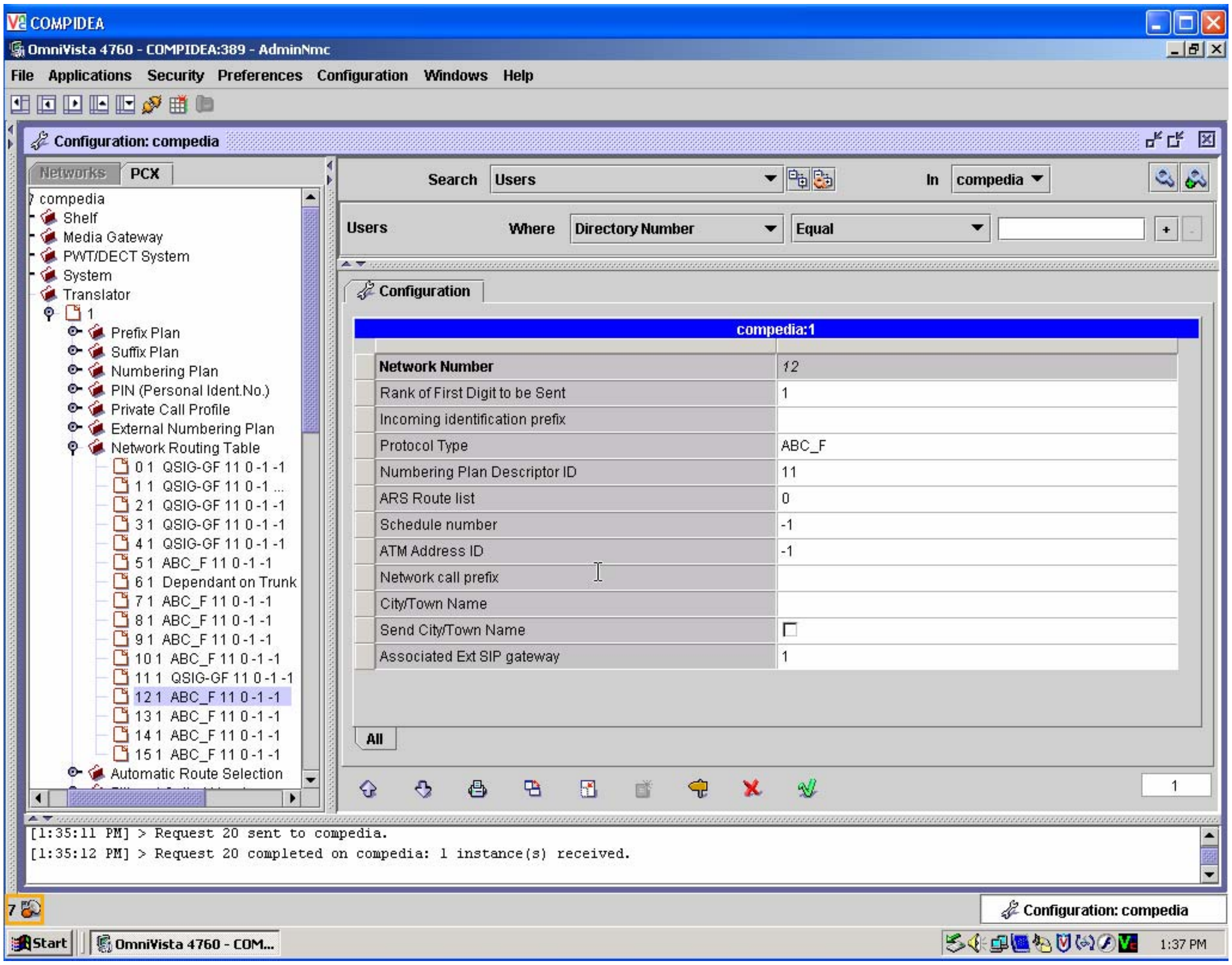
Alcatel Omni PCX 4400 Configuration

Alcatel Omni PCX 4400 Software Version and Hardware Configuration List:



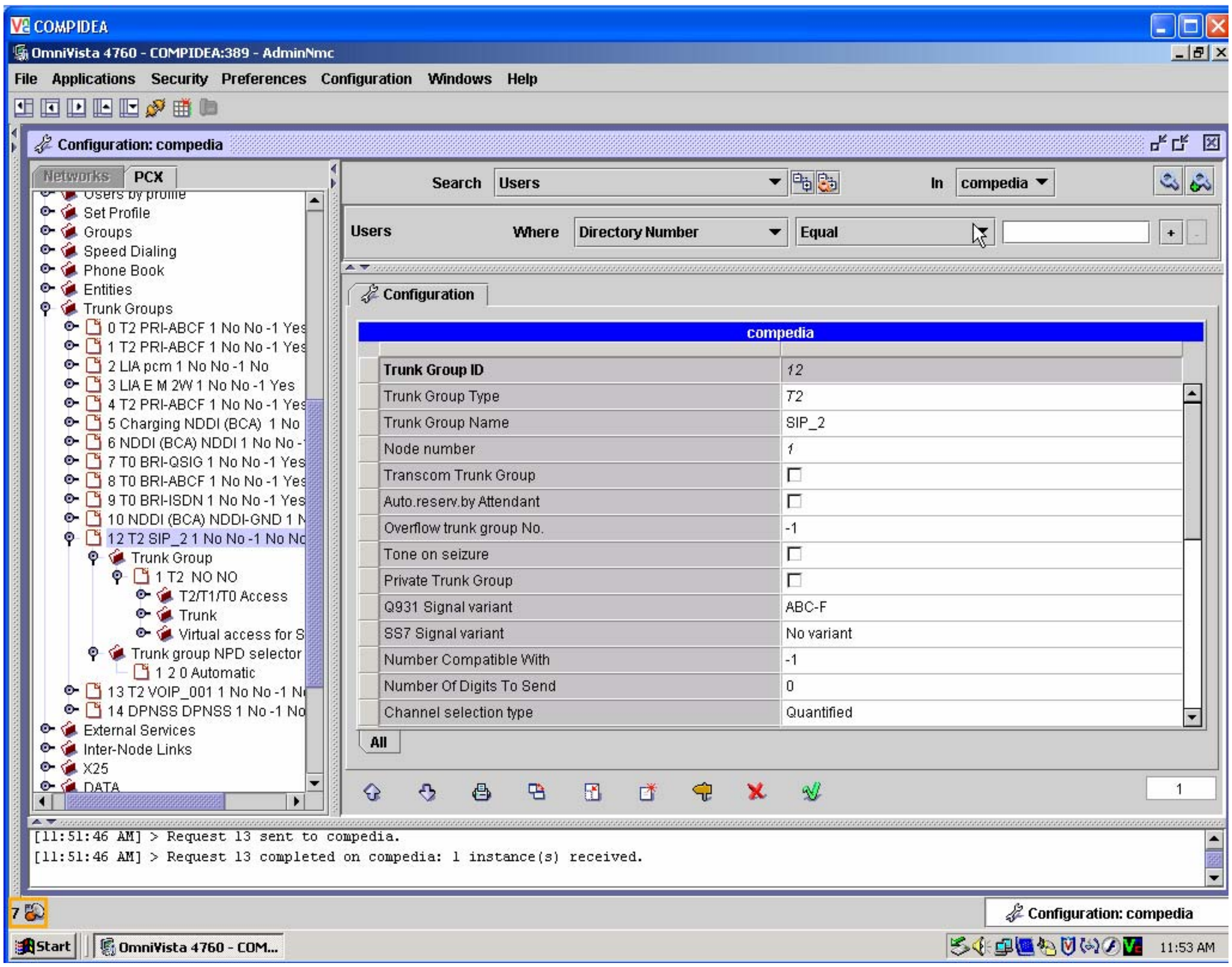
Configure SIP Network: Translator → Network Routing Table:

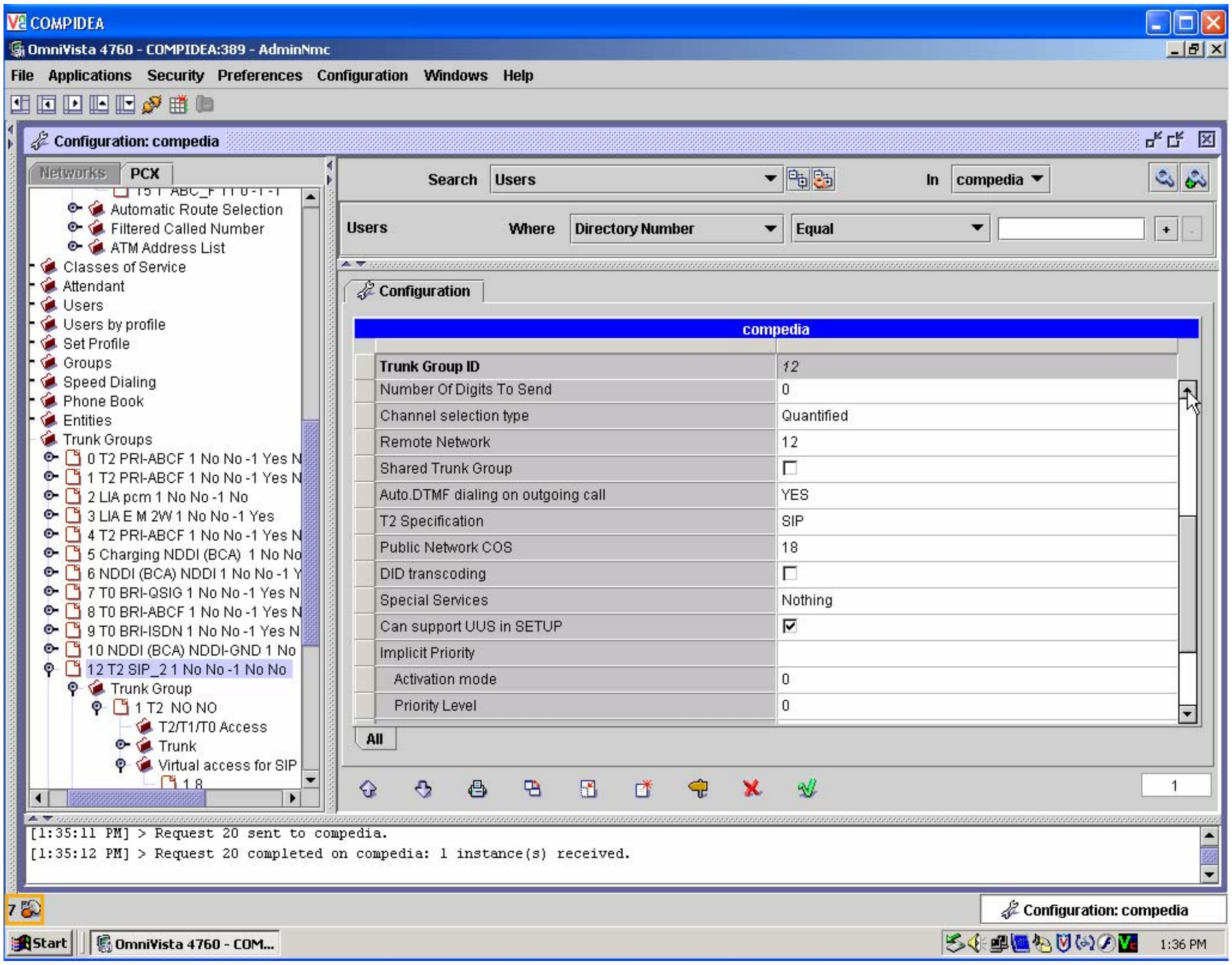
- Ensure the sub-network number used by SIP sets and SIP trunk group have the "Protocol Type = ABC_F"

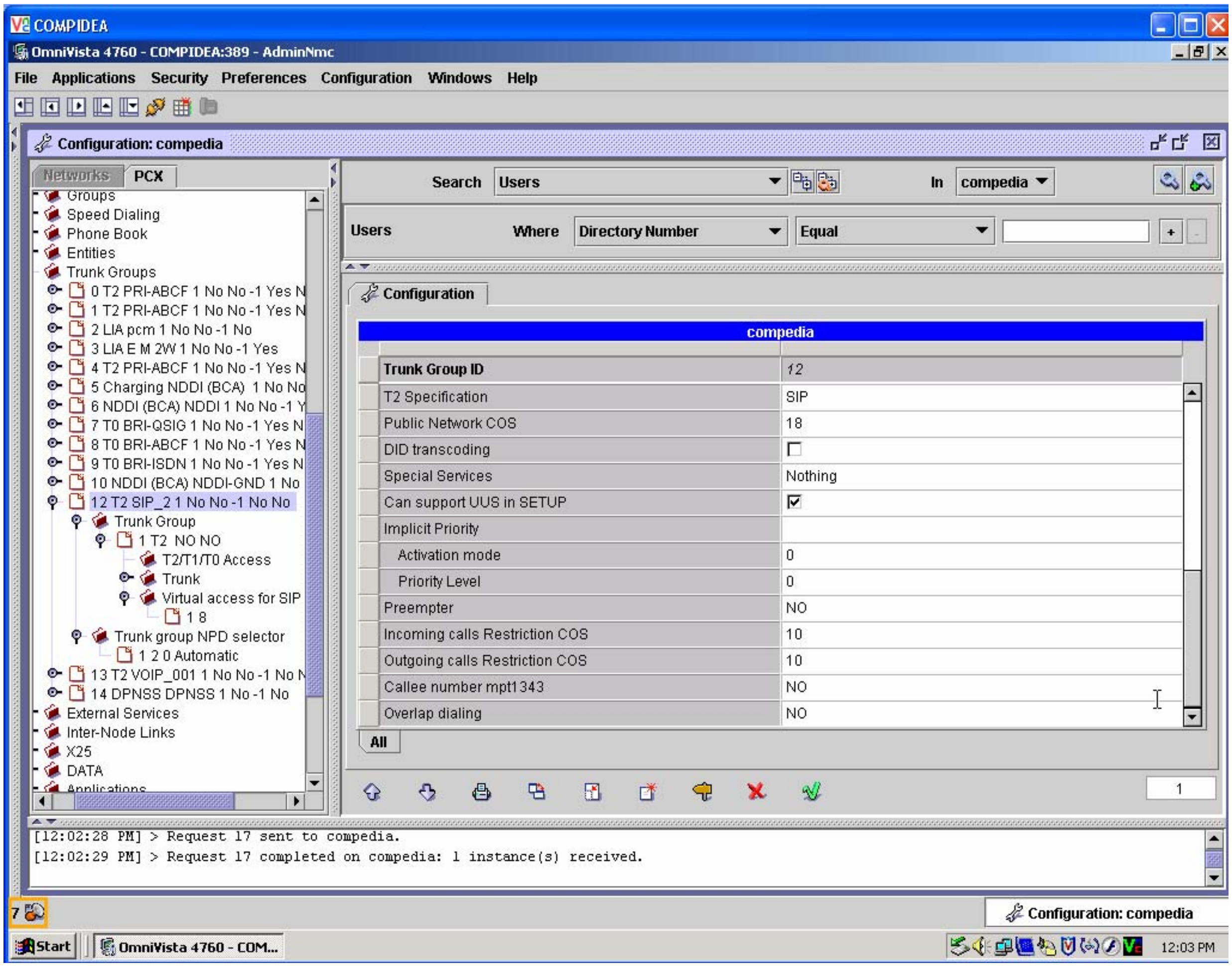


Configure SIP Trunk group:

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

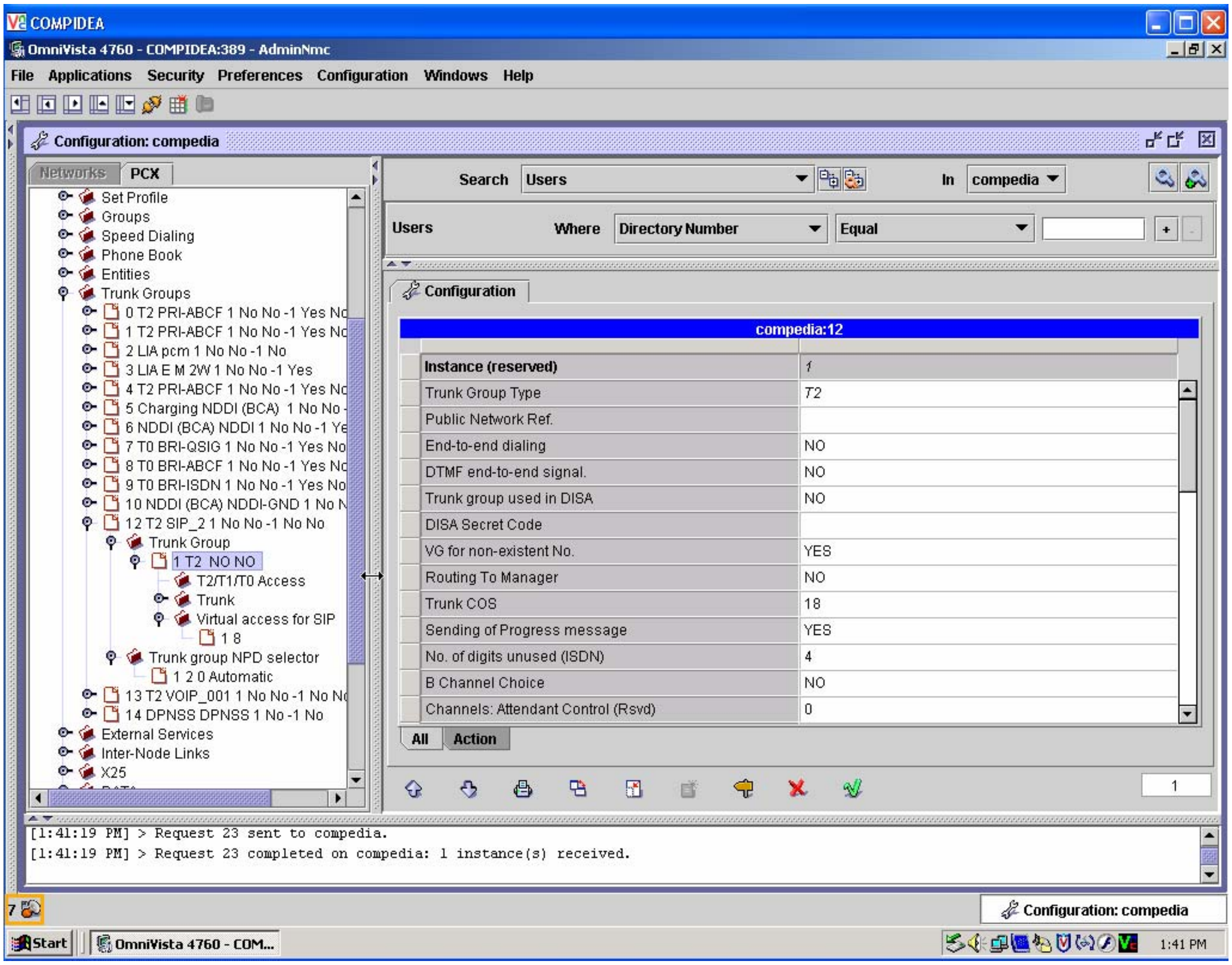


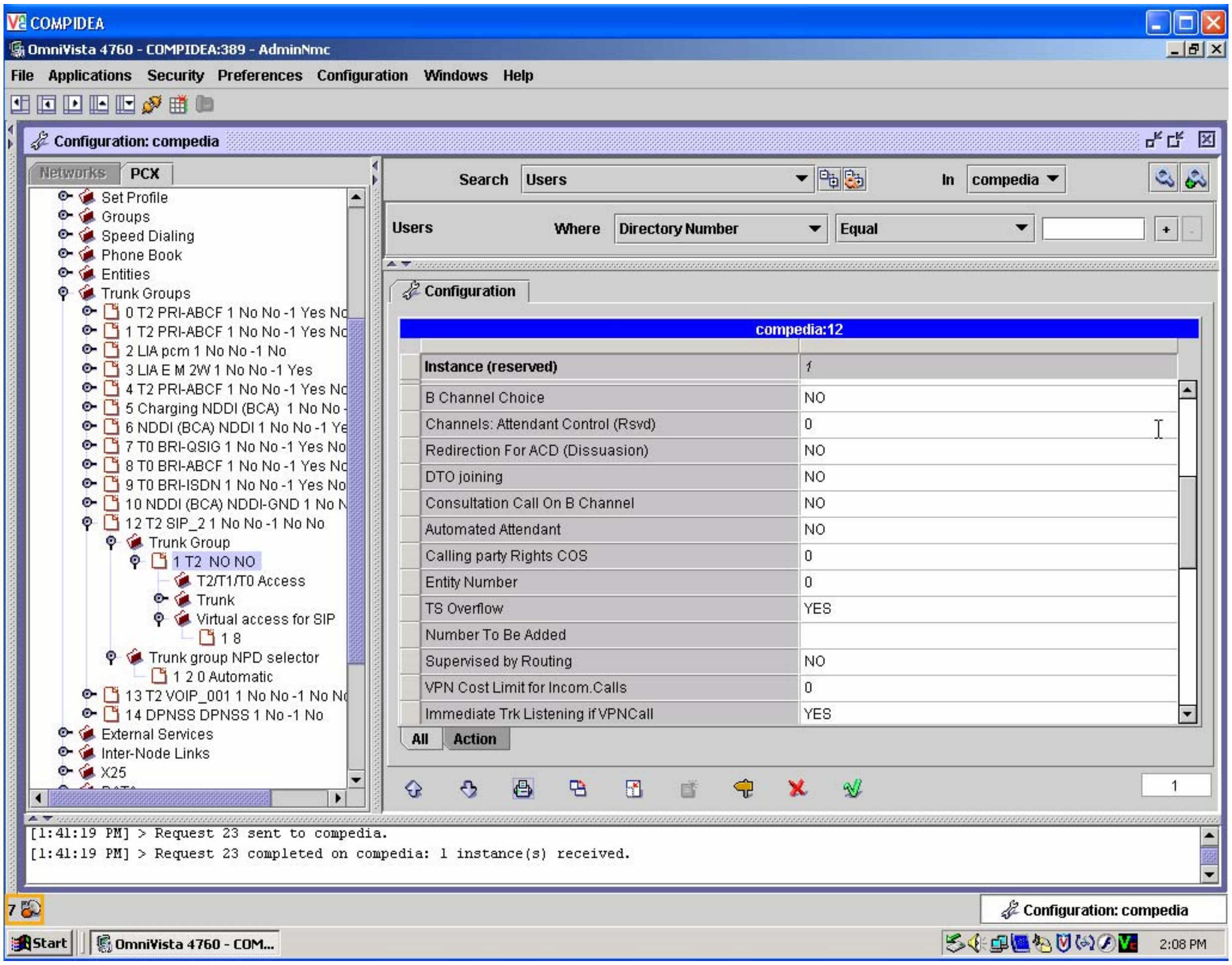

COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Automatic Route Selection
Filtered Called Number
ATM Address List
Classes of Service
Attendant
Users
Users by profile
Set Profile
Groups
Speed Dialing
Phone Book
Entities
Trunk Groups
12 T2 SIP_2 1 No No -1 No No
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual access for SIP
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Trunk Group ID 12
Number Of Digits To Send 0
Channel selection type Quantified
Remote Network 12
Shared Trunk Group
Auto.DTMF dialing on outgoing call YES
T2 Specification SIP
Public Network COS 18
DID transcoding
Special Services Nothing
Can support UUS in SETUP
Implicit Priority
Activation mode 0
Priority Level 0
All
[1:35:11 PM] > Request 20 sent to compedia.
[1:35:12 PM] > Request 20 completed on compedia: 1 instance(s) received.
Configuration: compedia
Start OmniVista 4760 - COM...
1:36 PM

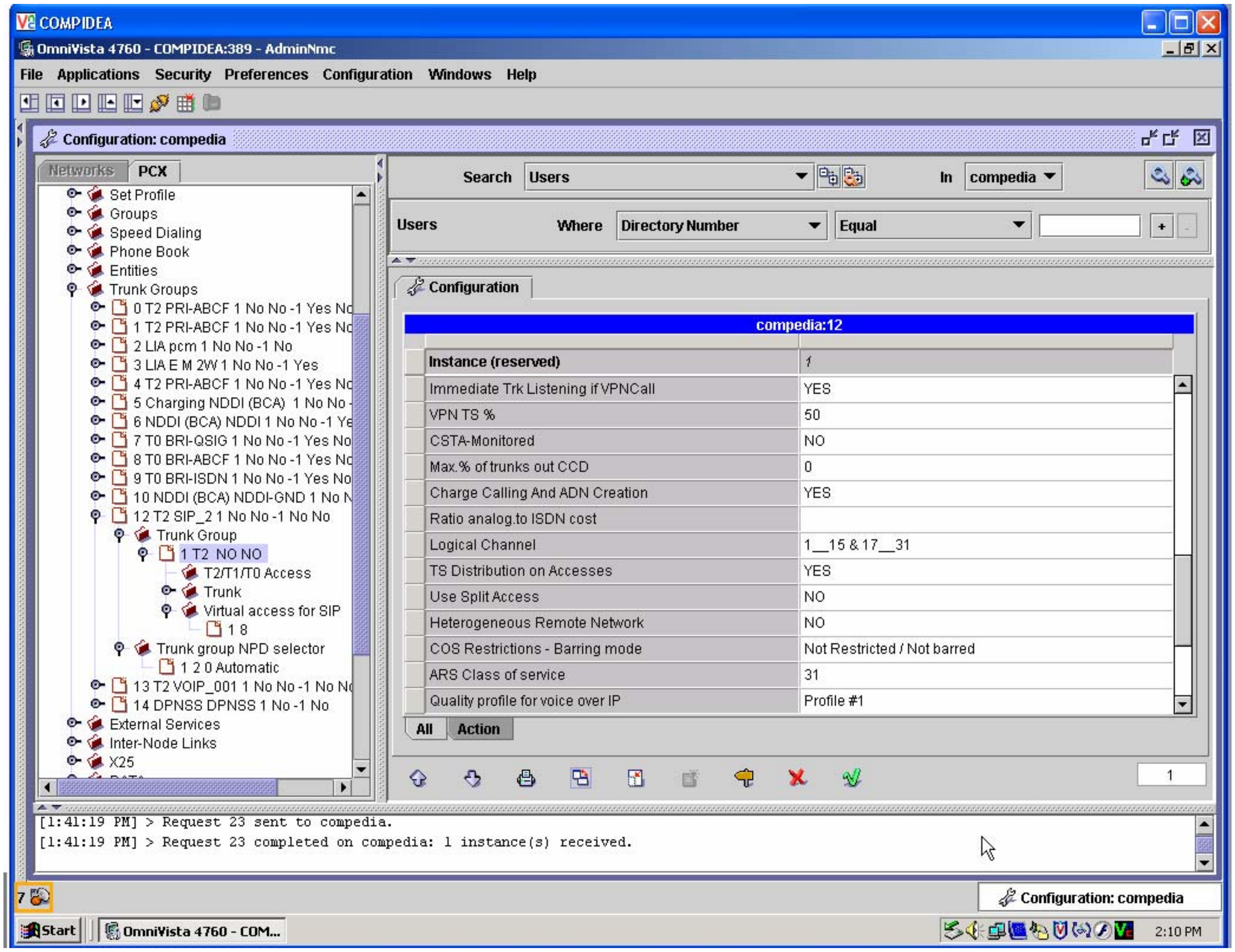

COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Search Users In compedia
Users Where Directory Number Equal
Trunk Groups
12 T2 SIP_2 1 No No -1 No No
Configuration
compedia
Trunk Group ID 12
T2 Specification SIP
Public Network COS 18
DID transcoding
Special Services Nothing
Can support UUS in SETUP
Implicit Priority
Activation mode 0
Priority Level 0
Preempter NO
Incoming calls Restriction COS 10
Outgoing calls Restriction COS 10
Callee number mpt1343 NO
Overlap dialing NO
[12:02:28 PM] > Request 17 sent to compedia.
[12:02:29 PM] > Request 17 completed on compedia: 1 instance(s) received.

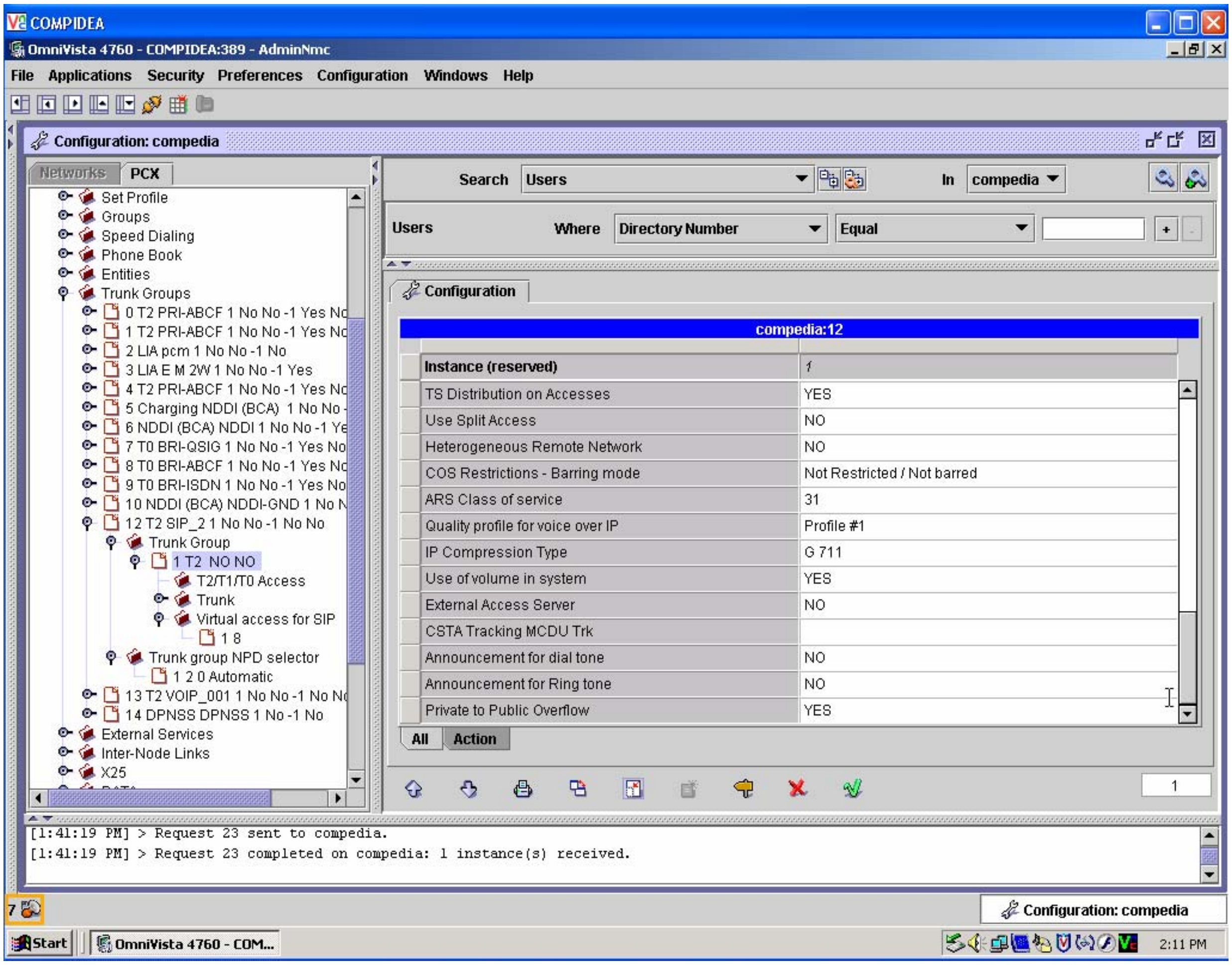
Configure T2 Trunk Group Type:

IP Compression Type: G.711




COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Set Profile
Groups
Speed Dialing
Phone Book
Entities
Trunk Groups
0 T2 PRI-ABCF 1 No No -1 Yes No
1 T2 PRI-ABCF 1 No No -1 Yes No
2 LIA pcm 1 No No -1 No
3 LIA E M 2W 1 No No -1 Yes
4 T2 PRI-ABCF 1 No No -1 Yes No
5 Charging NDDI (BCA) 1 No No
6 NDDI (BCA) NDDI 1 No No -1 Ye
7 T0 BRI-QSIG 1 No No -1 Yes No
8 T0 BRI-ABCF 1 No No -1 Yes No
9 T0 BRI-ISDN 1 No No -1 Yes No
10 NDDI (BCA) NDDI-GND 1 No N
12 T2 SIP_2 1 No No -1 No No
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual access for SIP
1 8
Trunk group NPD selector
1 2 0 Automatic
13 T2 VOIP_001 1 No No -1 No No
14 DPNSS DPNSS 1 No -1 No
External Services
Inter-Node Links
X25
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia:12
Instance (reserved) 1
B Channel Choice NO
Channels: Attendant Control (Rsvd) 0
Redirection For ACD (Dissuasion) NO
DTO joining NO
Consultation Call On B Channel NO
Automated Attendant NO
Calling party Rights COS 0
Entity Number 0
TS Overflow YES
Number To Be Added
Supervised by Routing NO
VPN Cost Limit for Incom.Calls 0
Immediate Trk Listening if VPNCall YES
All Action
1
[1:41:19 PM] > Request 23 sent to compedia.
[1:41:19 PM] > Request 23 completed on compedia: 1 instance(s) received.
Configuration: compedia
Start OmniVista 4760 - COM...
2:08 PM

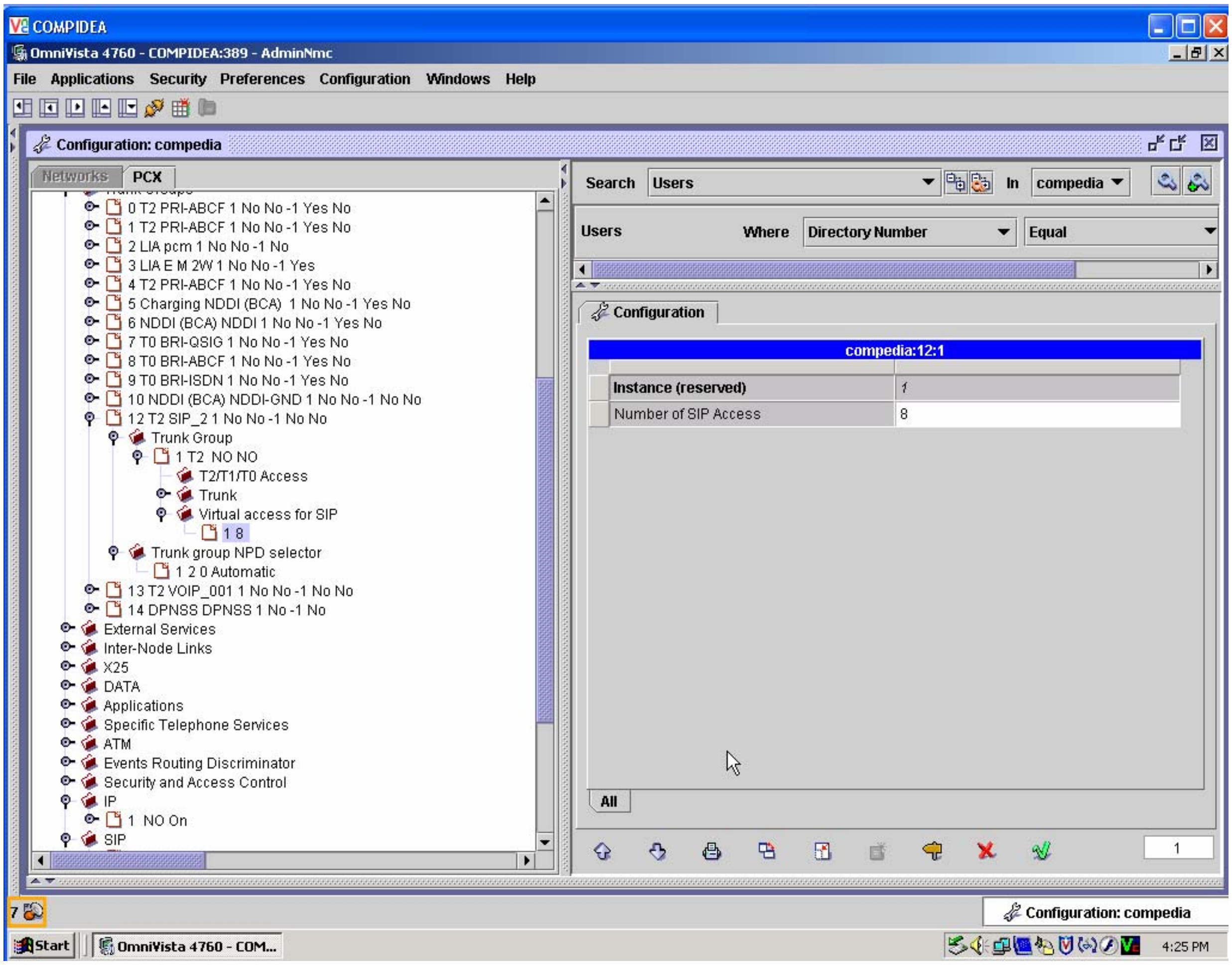

COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Set Profile
Groups
Speed Dialing
Phone Book
Entities
Trunk Groups
12 T2 SIP_2 1 No No -1 No No
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual access for SIP
Trunk group NPD selector
External Services
Inter-Node Links
X25
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia:12
Instance (reserved) 1
Immediate Trk Listening if VPNCall YES
VPN TS % 50
CSTA-Monitored NO
Max.% of trunks out CCD 0
Charge Calling And ADN Creation YES
Ratio analog.to ISDN cost
Logical Channel 1__15 & 17__31
TS Distribution on Accesses YES
Use Split Access NO
Heterogeneous Remote Network NO
COS Restrictions - Barring mode Not Restricted / Not barred
ARS Class of service 31
Quality profile for voice over IP Profile #1
All Action
[1:41:19 PM] > Request 23 sent to compedia.
[1:41:19 PM] > Request 23 completed on compedia: 1 instance(s) received.
Start OmniVista 4760 - COM...
2:10 PM


COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Set Profile
Groups
Speed Dialing
Phone Book
Entities
Trunk Groups
0 T2 PRI-ABCF 1 No No -1 Yes No
1 T2 PRI-ABCF 1 No No -1 Yes No
2 LIA pcm 1 No No -1 No
3 LIA E M 2W 1 No No -1 Yes
4 T2 PRI-ABCF 1 No No -1 Yes No
5 Charging NDDI (BCA) 1 No No
6 NDDI (BCA) NDDI 1 No No -1 Ye
7 T0 BRI-QSIG 1 No No -1 Yes No
8 T0 BRI-ABCF 1 No No -1 Yes No
9 T0 BRI-ISDN 1 No No -1 Yes No
10 NDDI (BCA) NDDI-GND 1 No N
12 T2 SIP_2 1 No No -1 No No
Trunk Group
1 T2 NO NO
T2/T1/T0 Access
Trunk
Virtual access for SIP
1 8
Trunk group NPD selector
1 2 0 Automatic
13 T2 VOIP_001 1 No No -1 No No
14 DPNSS DPNSS 1 No -1 No
External Services
Inter-Node Links
X25
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia:12
Instance (reserved) 1
TS Distribution on Accesses YES
Use Split Access NO
Heterogeneous Remote Network NO
COS Restrictions - Barring mode Not Restricted / Not barred
ARS Class of service 31
Quality profile for voice over IP Profile #1
IP Compression Type G 711
Use of volume in system YES
External Access Server NO
CSTA Tracking MCDU Trk
Announcement for dial tone NO
Announcement for Ring tone NO
Private to Public Overflow YES
All Action
1
[1:41:19 PM] > Request 23 sent to compedia.
[1:41:19 PM] > Request 23 completed on compedia: 1 instance(s) received.
Configuration: compedia
Start OmniVista 4760 - COM...
2:11 PM

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:

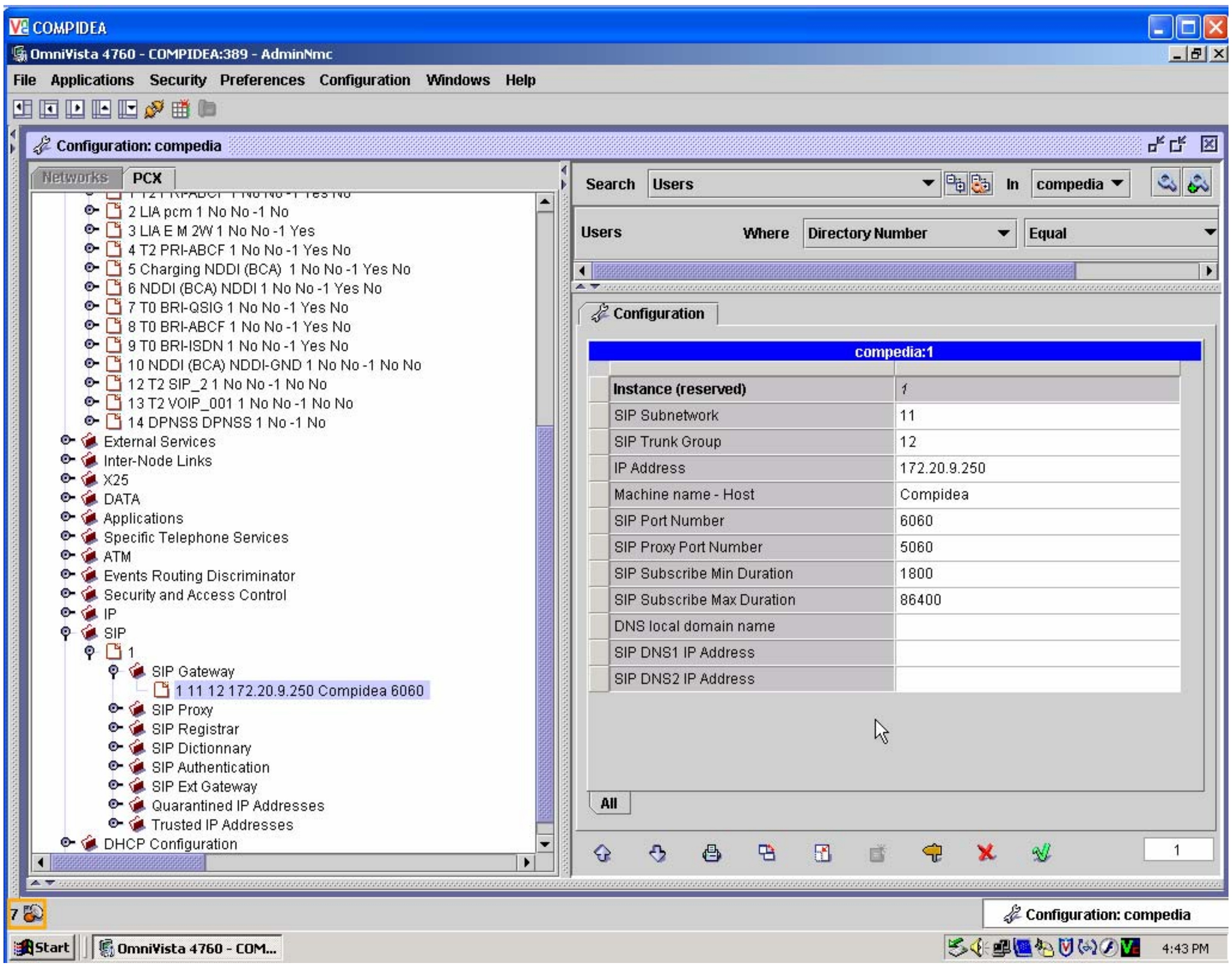
SIP Subnetwork: Enter the sub-network number used by SIP sets and SIP trunk group

SIP Trunk Group: Enter the SIP trunk group number

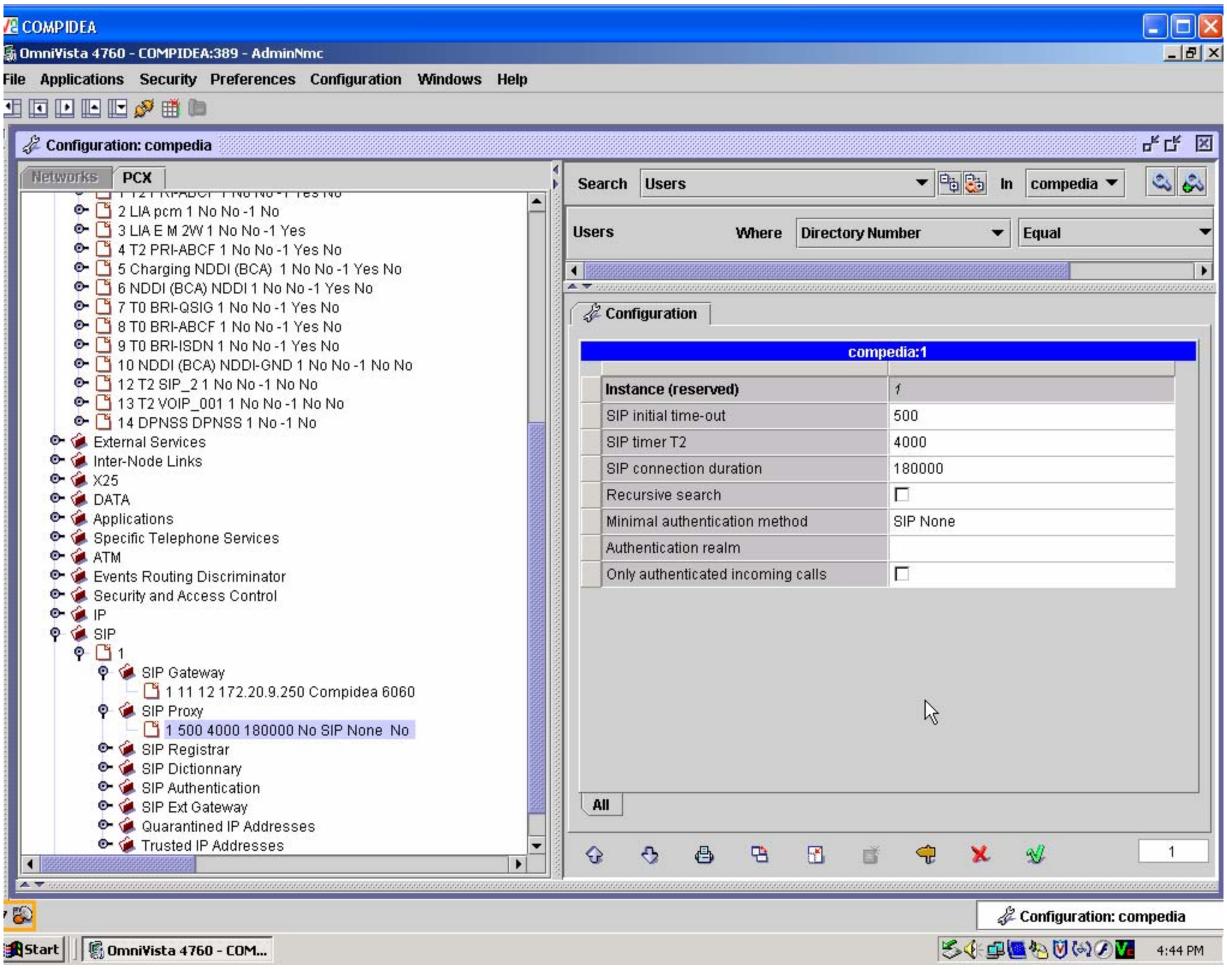
IP Address: Enter the IP Address of the Alcatel Call Server

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:



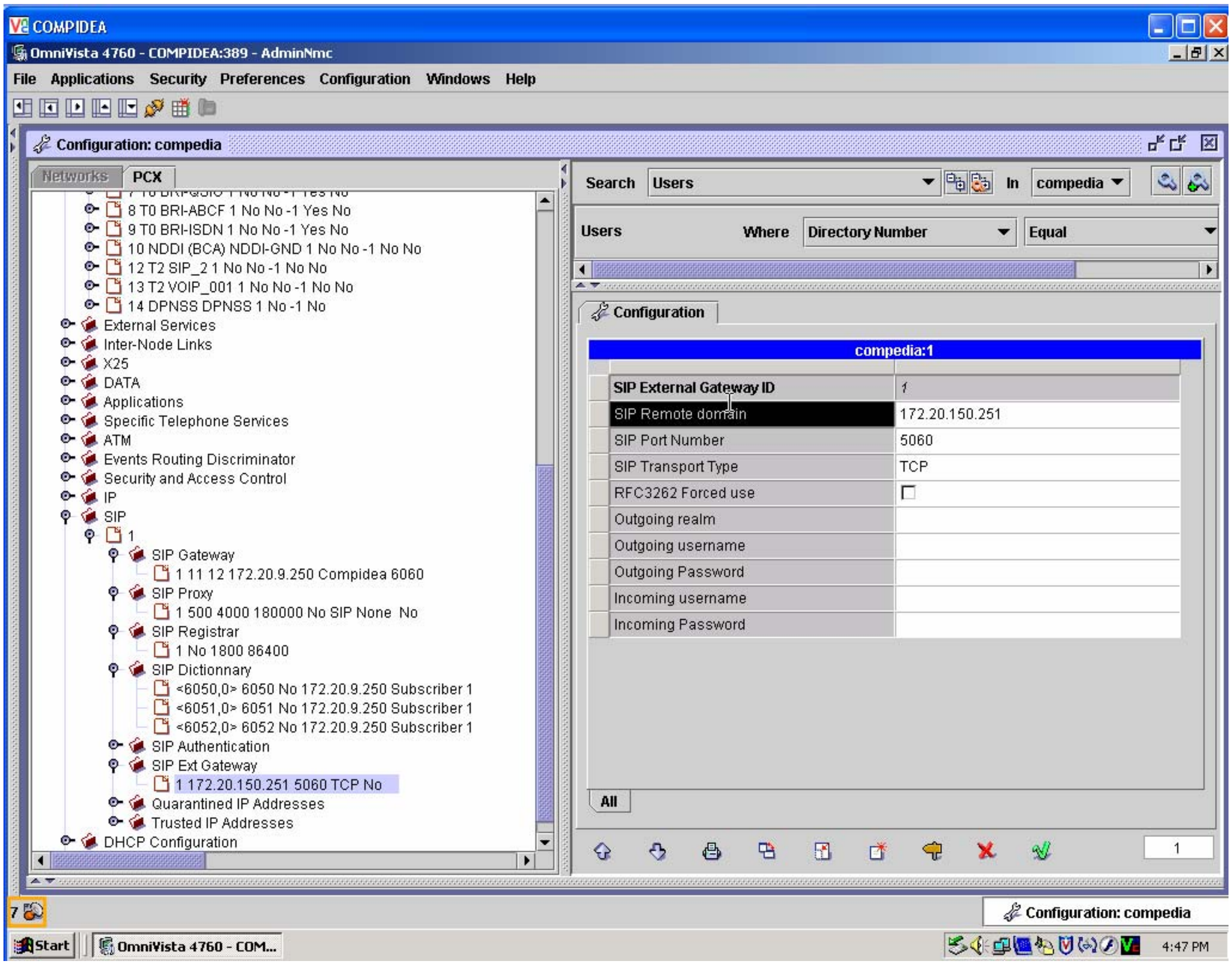
Configure SIP External Gateway:

This is Cisco CallManager server configuration

SIP Remote Domain: Enter the IP address or FQDN of Cisco CallManager server

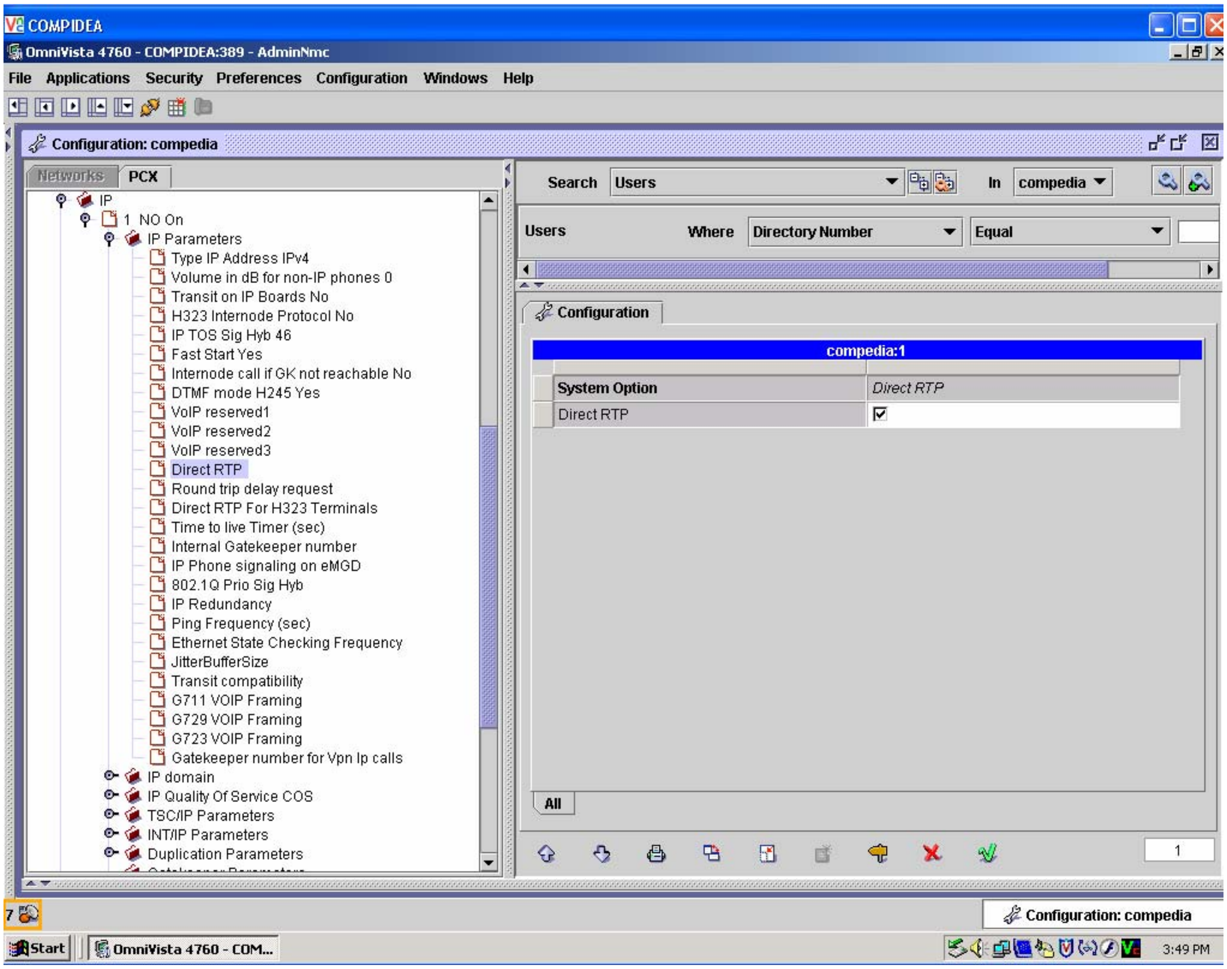
SIP Port Number: Enter the TCP or UDP port number use by Cisco CallManager for SIP signaling message

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



Configure IP Parameters:

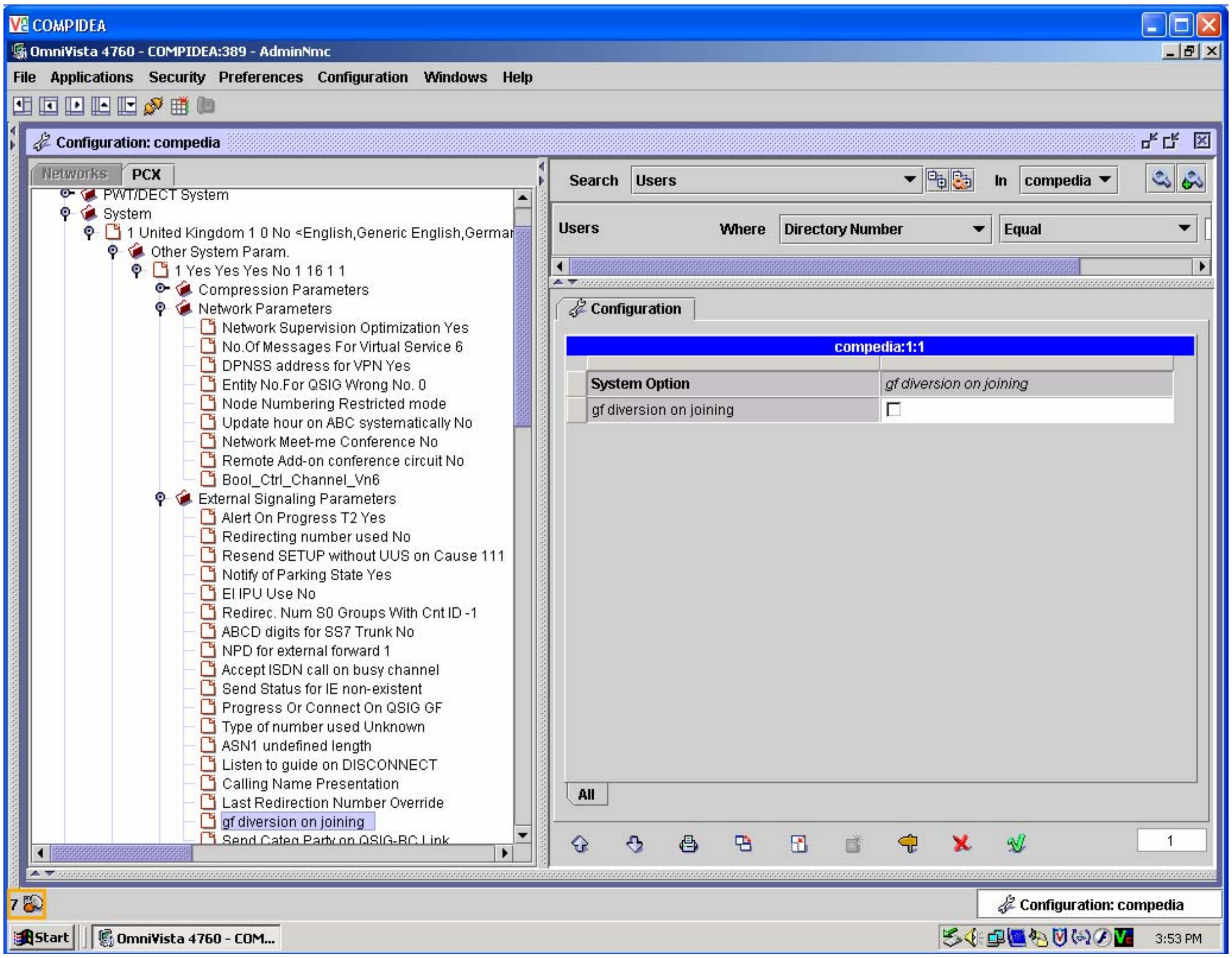
Direct RTP: enable the checkbox for “Direct RTP”



Configure GF diversion on joining:

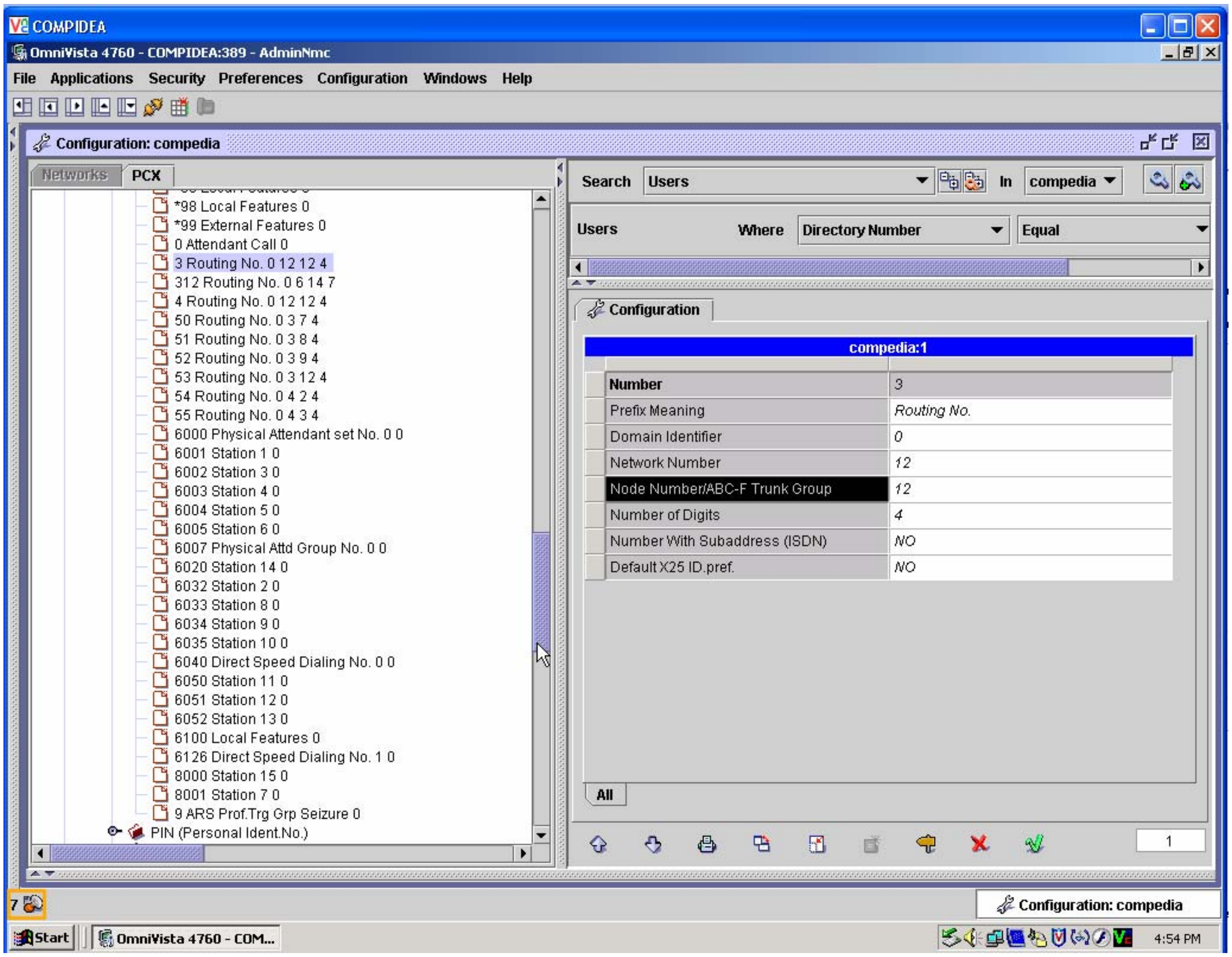
Disable the gf diversion on joining parameter by uncheck the box under the following:

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



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

Select Translator → Prefix Plan → Create a new prefix for 3xxx to use SIP trunk group 12

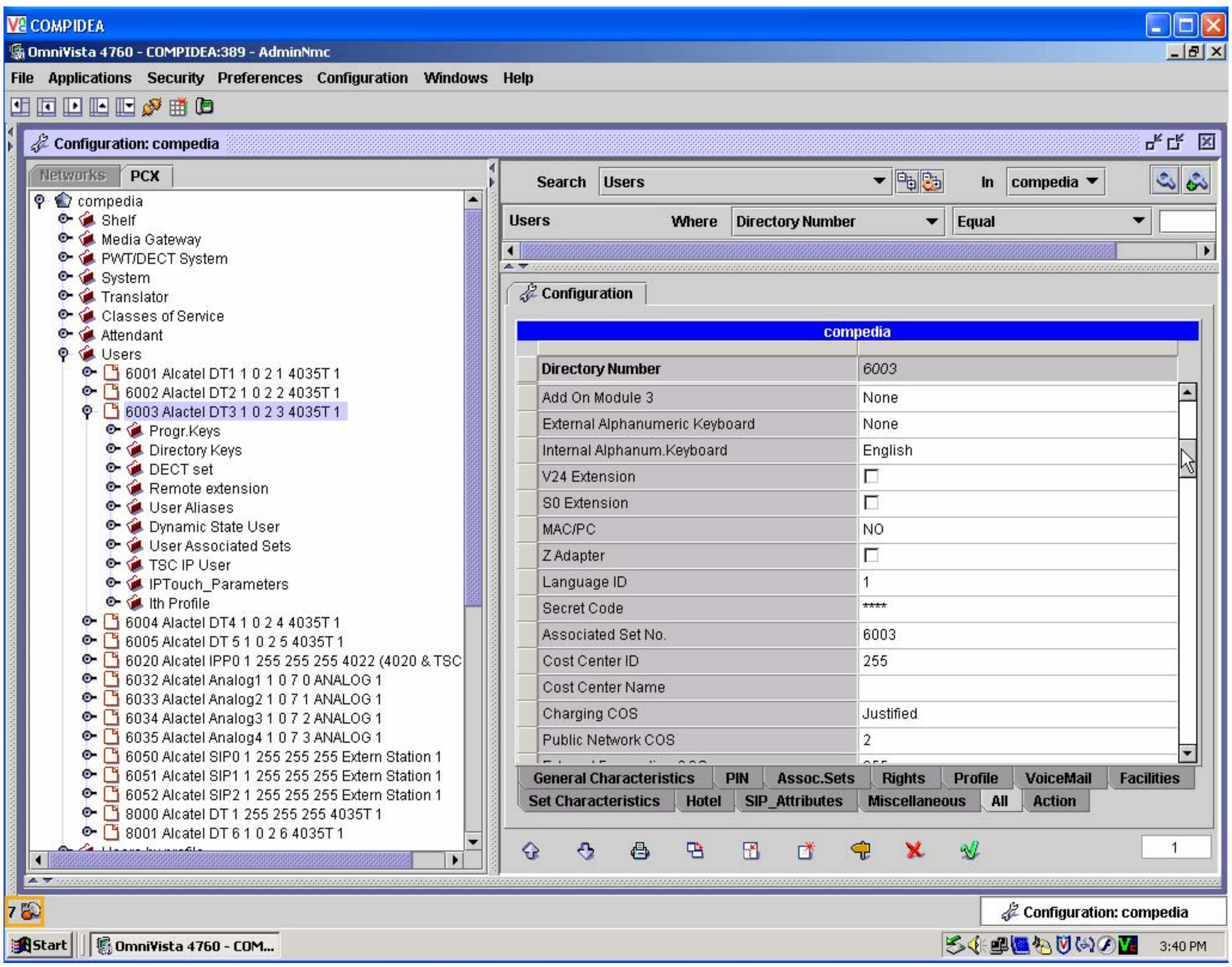


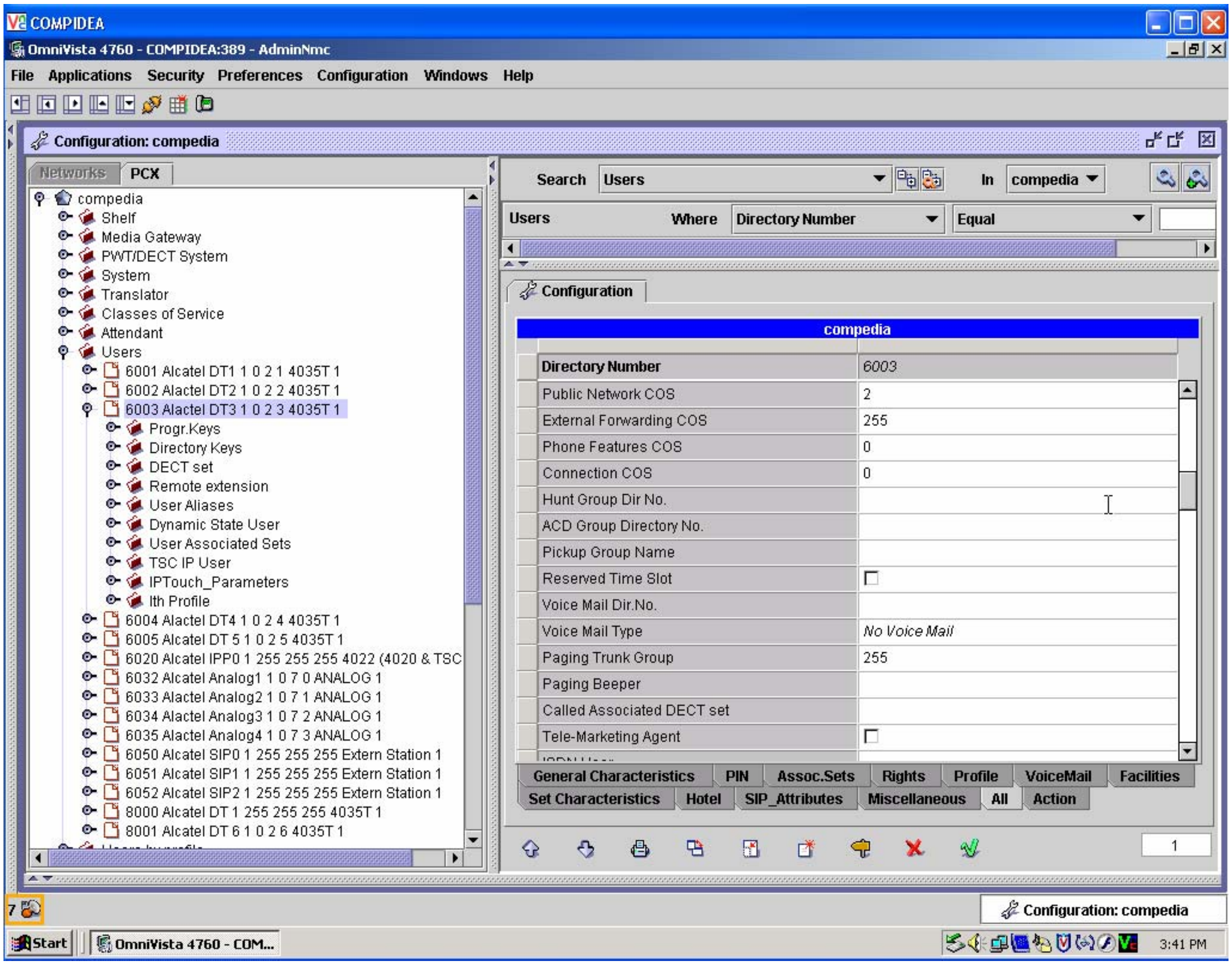
Configure Alcatel standard users (digital stations):

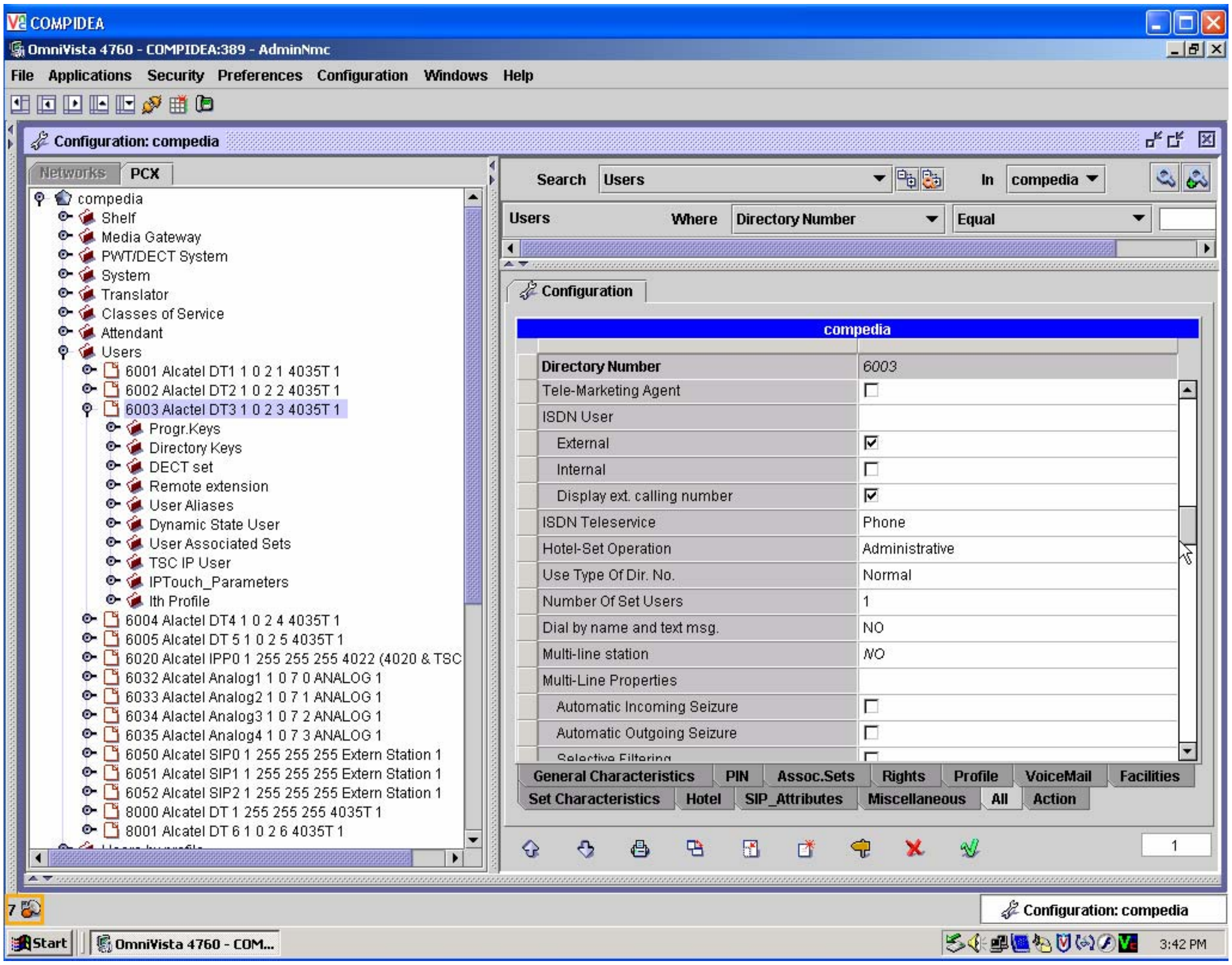
Select User → Create → Create a new user for the digital phone

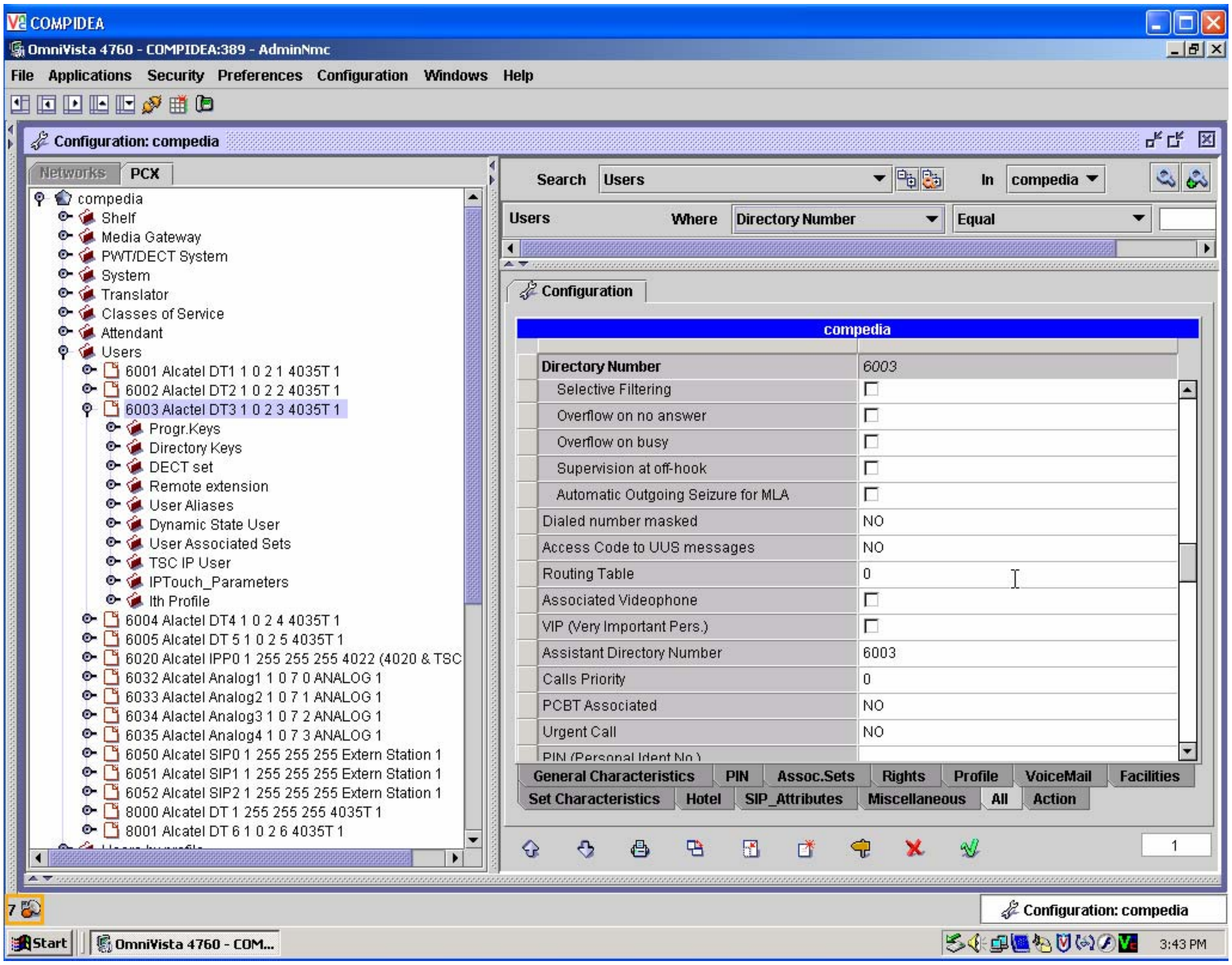
For a standard user, the URL <username> and <domain> attributes are optional. They can be completed to make the set accessible to the SIP world by a specific SIP URL in form of username@domain type. If they are not configured, the URL is automatically constructed by the system from MAO system configuration 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 = 6003 will have SIP URL = 6003@172.20.9.250 where 172.20.9.250 is the IP Address of the Alcatel SIP media gateway.

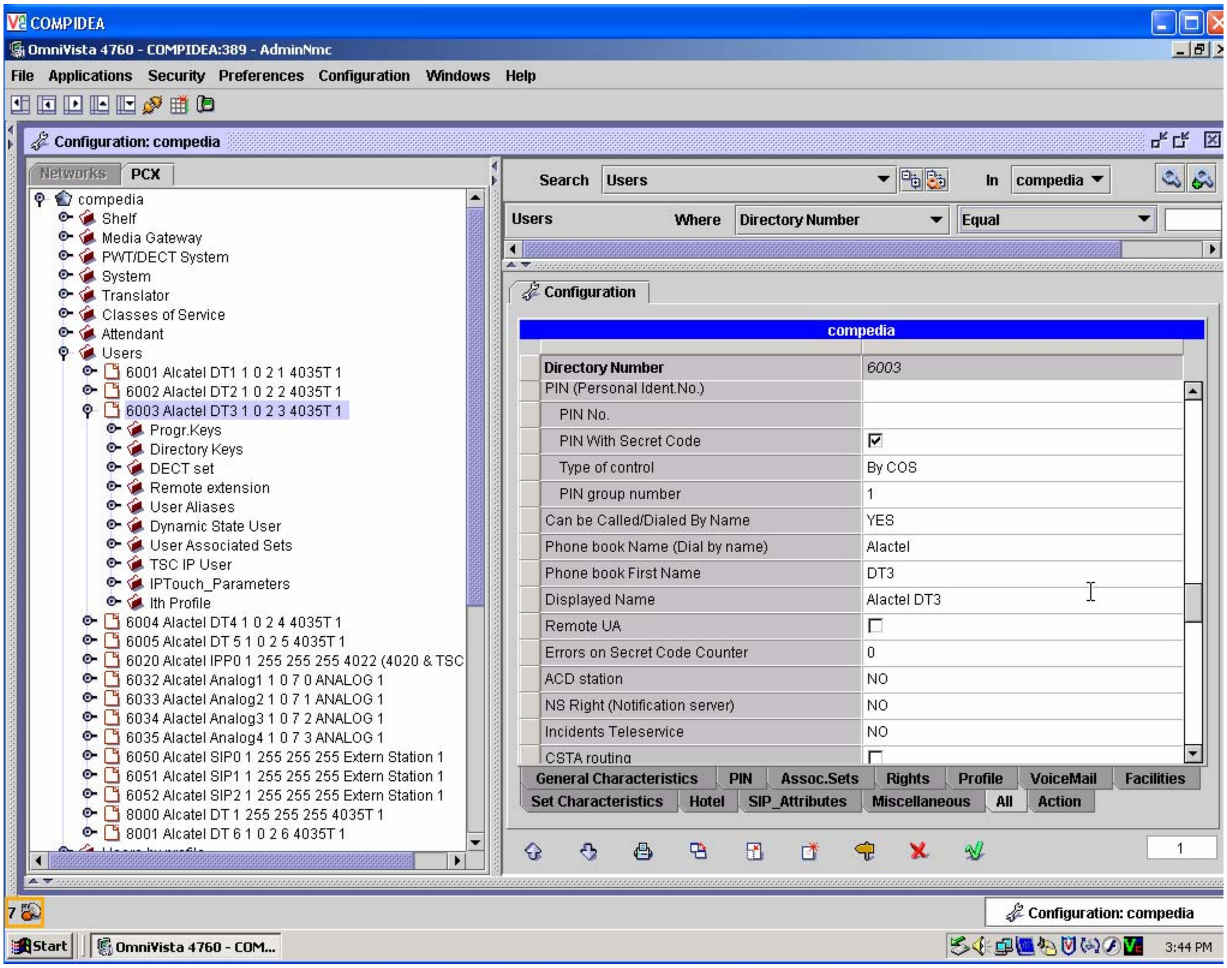
Alcatel digital type 4035 phone (phone A2 with extension 6003)



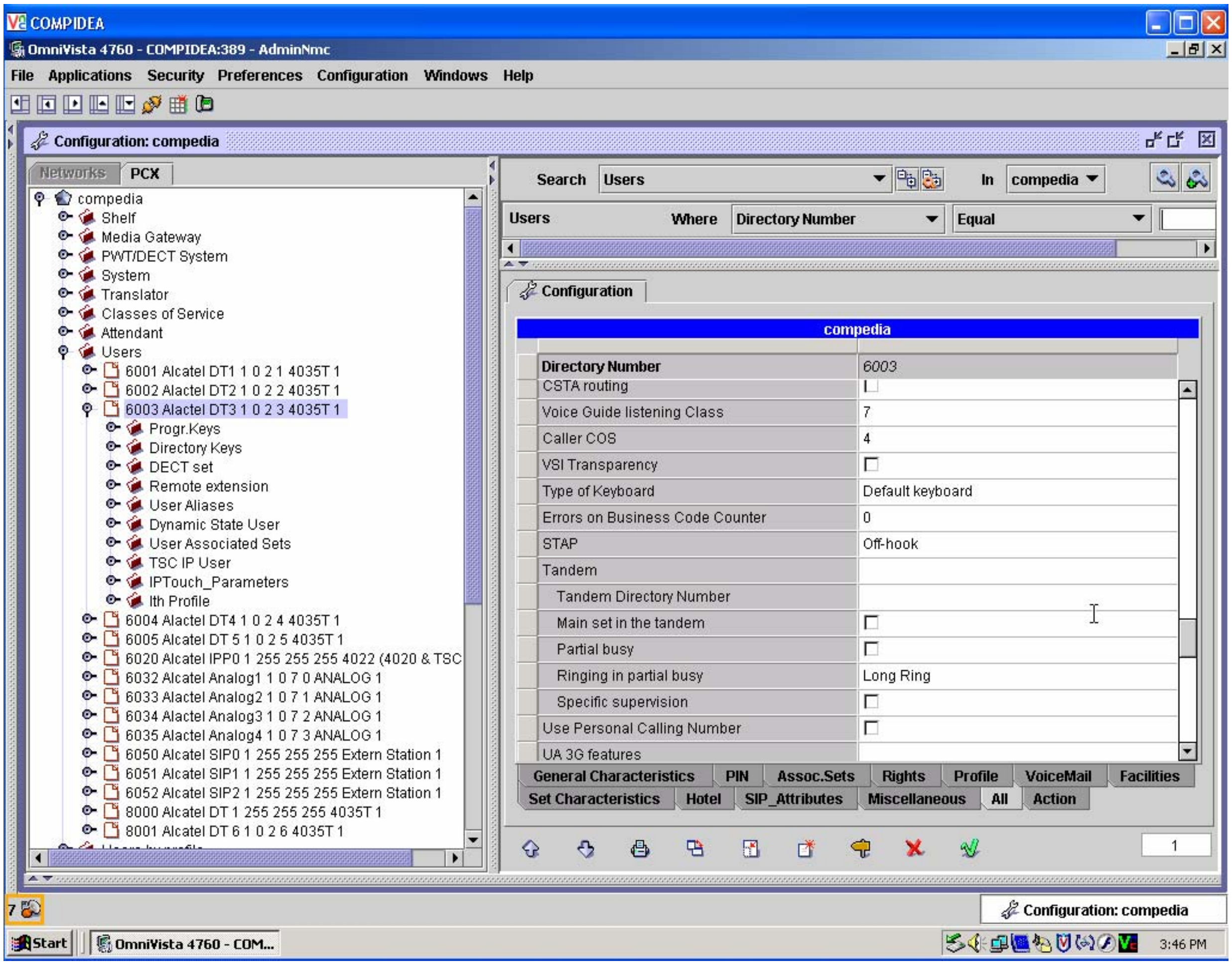

COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
compedia
Shelf
Media Gateway
PWT/DECT System
System
Translator
Classes of Service
Attendant
Users
6001 Alcatel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
Directory Keys
DECT set
Remote extension
User Aliases
Dynamic State User
User Associated Sets
TSC IP User
IPTouch_Parameters
Ith Profile
6004 Alactel DT4 1 0 2 4 4035T 1
6005 Alcatel DT 5 1 0 2 5 4035T 1
6020 Alcatel IPP0 1 255 255 255 4022 (4020 & TSC
6032 Alcatel Analog1 1 0 7 0 ANALOG 1
6033 Alactel Analog2 1 0 7 1 ANALOG 1
6034 Alactel Analog3 1 0 7 2 ANALOG 1
6035 Alactel Analog4 1 0 7 3 ANALOG 1
6050 Alcatel SIP0 1 255 255 255 Extern Station 1
6051 Alcatel SIP1 1 255 255 255 Extern Station 1
6052 Alcatel SIP2 1 255 255 255 Extern Station 1
8000 Alcatel DT 1 255 255 255 4035T 1
8001 Alcatel DT 6 1 0 2 6 4035T 1
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Directory Number 6003
Public Network COS 2
External Forwarding COS 255
Phone Features COS 0
Connection COS 0
Hunt Group Dir No.
ACD Group Directory No.
Pickup Group Name
Reserved Time Slot
Voice Mail Dir.No.
Voice Mail Type No Voice Mail
Paging Trunk Group 255
Paging Beeper
Called Associated DECT set
Tele-Marketing Agent
General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities
Set Characteristics Hotel SIP_Attributes Miscellaneous All Action
1
Configuration: compedia
Start
OmniVista 4760 - COM...
3:41 PM

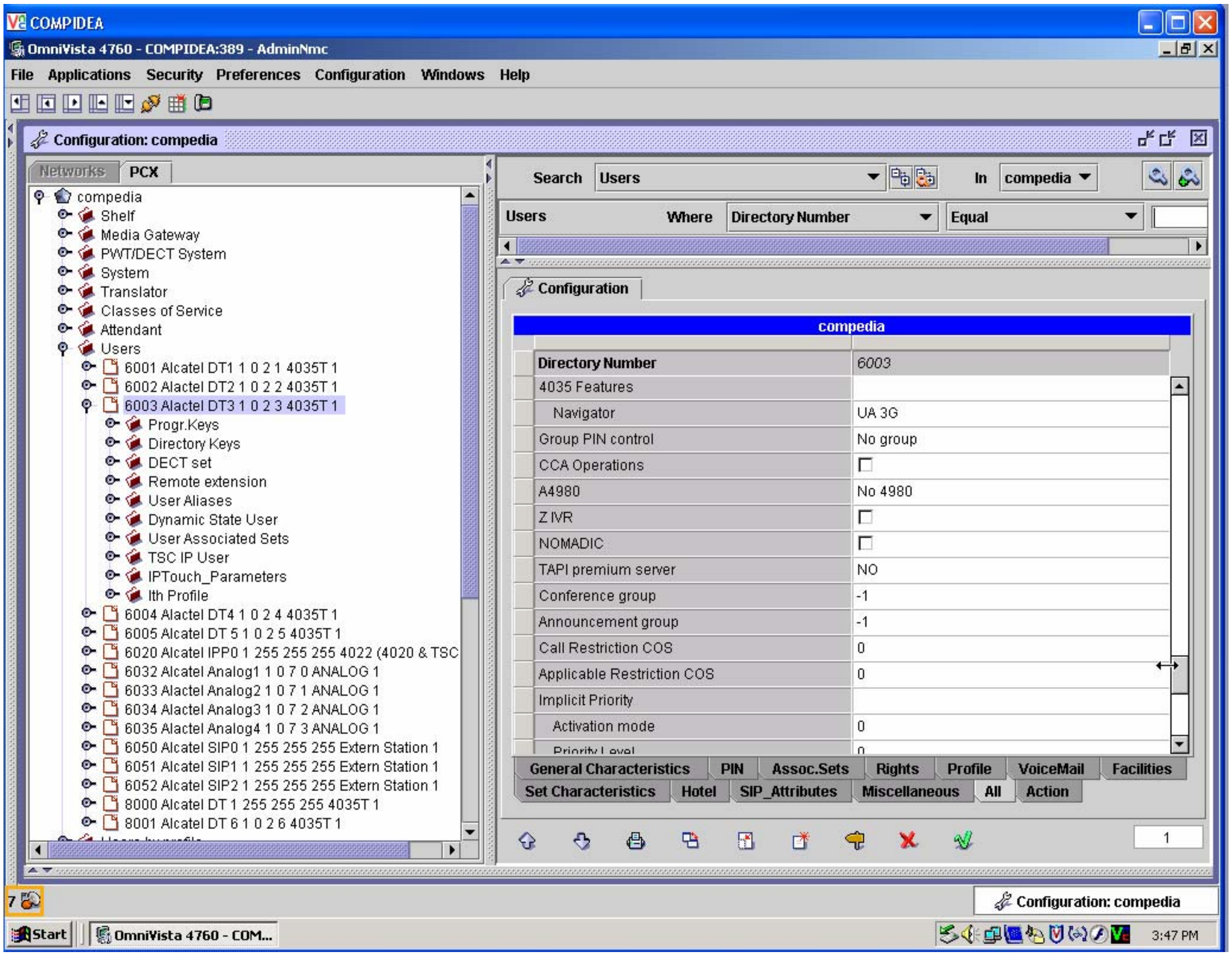


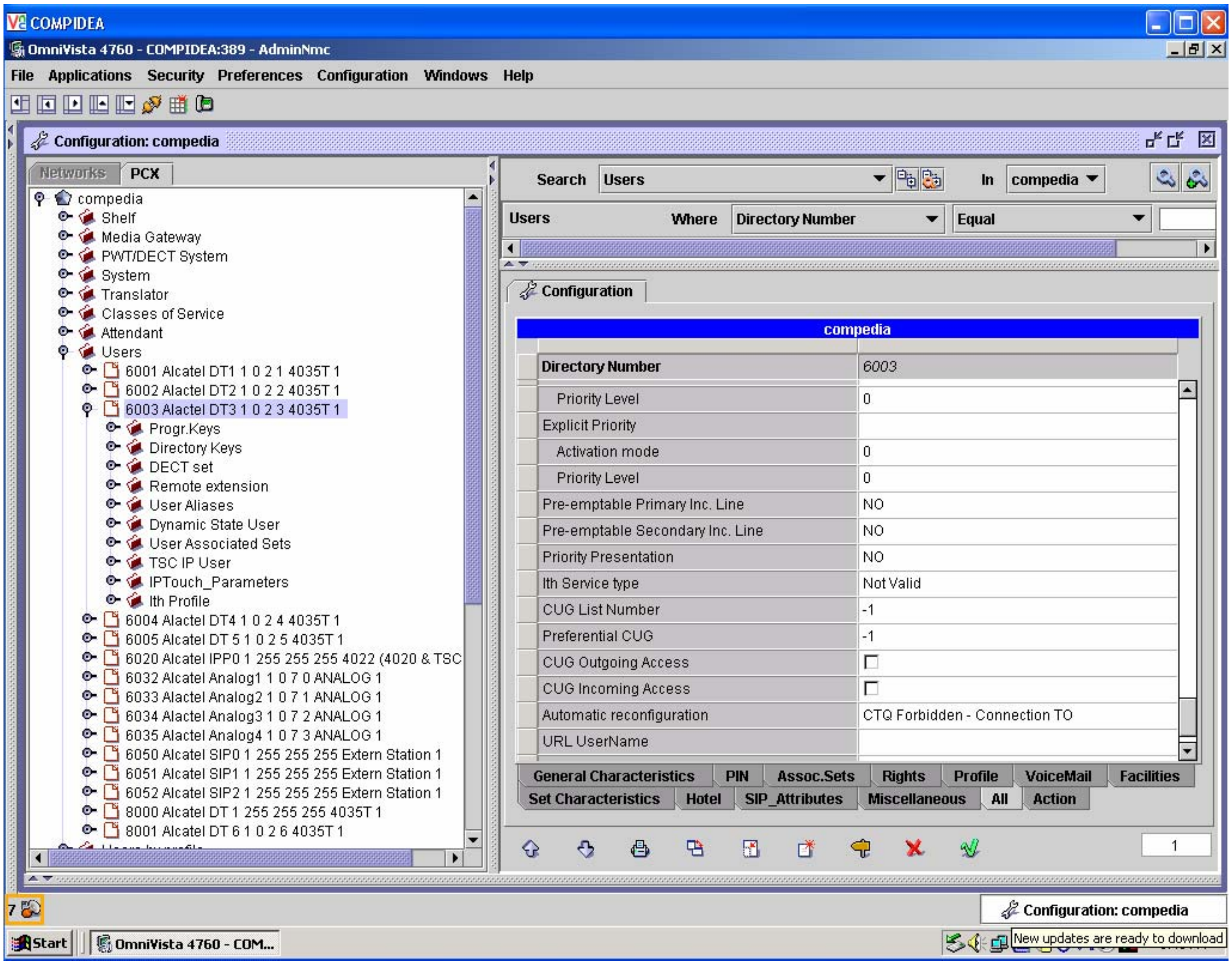

COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
compedia
Shelf
Media Gateway
PWT/DECT System
System
Translator
Classes of Service
Attendant
Users
6001 Alactel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
Directory Keys
DECT set
Remote extension
User Aliases
Dynamic State User
User Associated Sets
TSC IP User
IPTouch_Parameters
Ith Profile
6004 Alactel DT4 1 0 2 4 4035T 1
6005 Alcatel DT 5 1 0 2 5 4035T 1
6020 Alcatel IPP0 1 255 255 255 4022 (4020 & TSC
6032 Alcatel Analog1 1 0 7 0 ANALOG 1
6033 Alactel Analog2 1 0 7 1 ANALOG 1
6034 Alactel Analog3 1 0 7 2 ANALOG 1
6035 Alactel Analog4 1 0 7 3 ANALOG 1
6050 Alcatel SIP0 1 255 255 255 Extern Station 1
6051 Alcatel SIP1 1 255 255 255 Extern Station 1
6052 Alcatel SIP2 1 255 255 255 Extern Station 1
8000 Alcatel DT 1 255 255 255 4035T 1
8001 Alcatel DT 6 1 0 2 6 4035T 1
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Directory Number 6003
Selective Filtering
Overflow on no answer
Overflow on busy
Supervision at off-hook
Automatic Outgoing Seizure for MLA
Dialed number masked NO
Access Code to UUS messages NO
Routing Table 0
Associated Videophone
VIP (Very Important Pers.)
Assistant Directory Number 6003
Calls Priority 0
PCBT Associated NO
Urgent Call NO
General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities
Set Characteristics Hotel SIP_Attributes Miscellaneous All Action
1
Configuration: compedia
Start OmniVista 4760 - COM...
3:43 PM

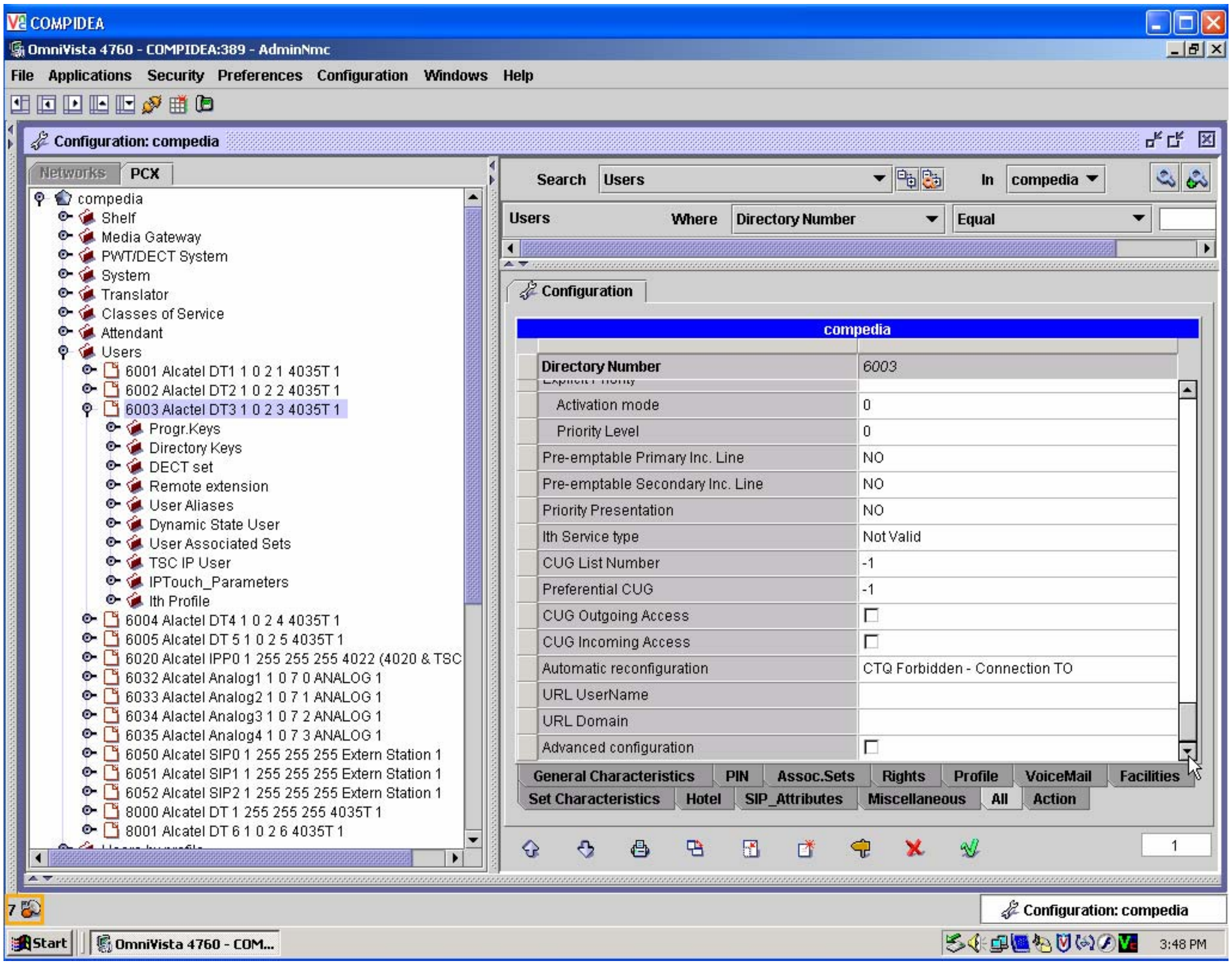


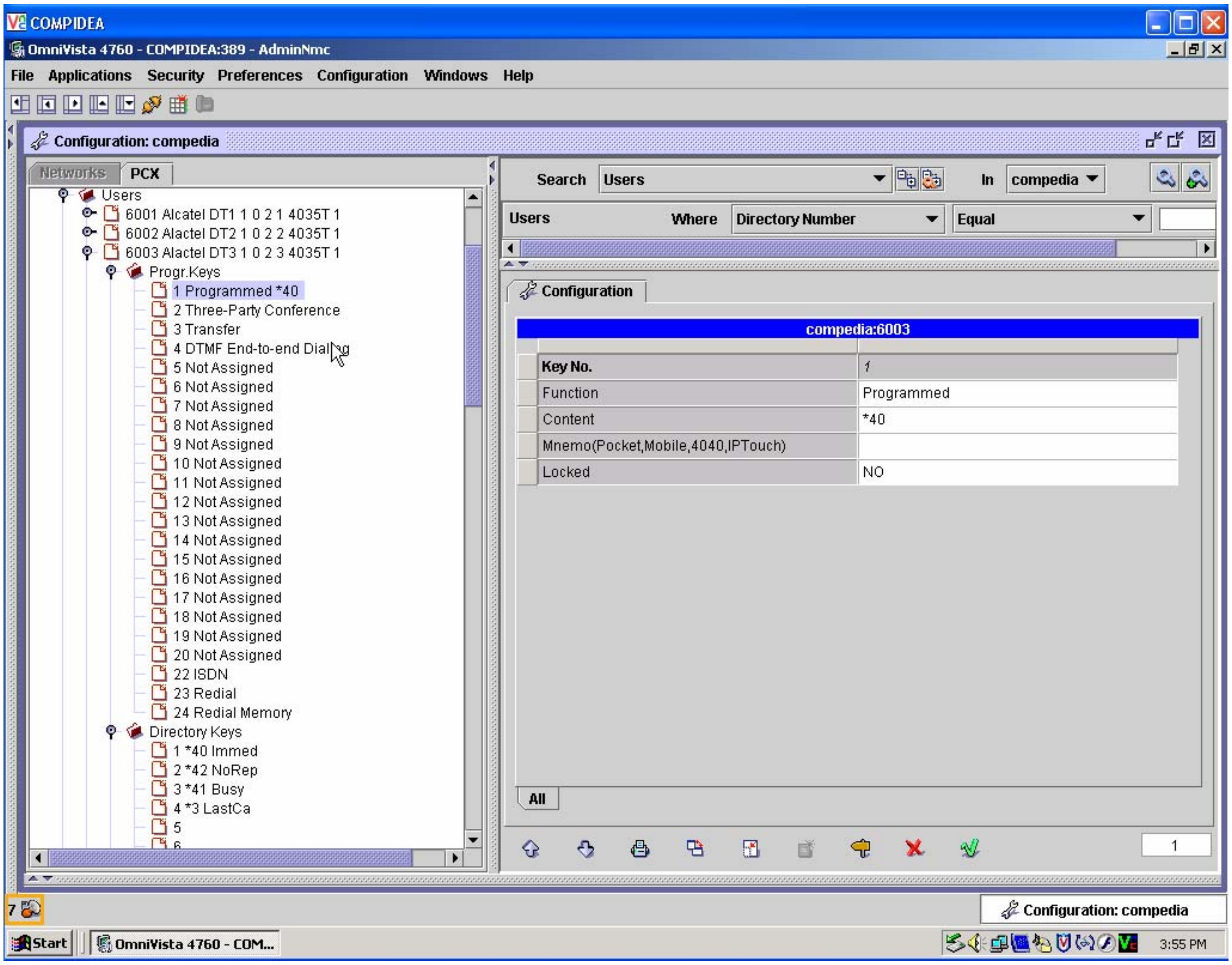
COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
compedia
Shelf
Media Gateway
PWT/DECT System
System
Translator
Classes of Service
Attendant
Users
6001 Alactel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
Directory Keys
DECT set
Remote extension
User Aliases
Dynamic State User
User Associated Sets
TSC IP User
IPTouch_Parameters
Ith Profile
6004 Alactel DT4 1 0 2 4 4035T 1
6005 Alactel DT 5 1 0 2 5 4035T 1
6020 Alcatel IPP0 1 255 255 255 4022 (4020 & TSC
6032 Alcatel Analog1 1 0 7 0 ANALOG 1
6033 Alactel Analog2 1 0 7 1 ANALOG 1
6034 Alactel Analog3 1 0 7 2 ANALOG 1
6035 Alactel Analog4 1 0 7 3 ANALOG 1
6050 Alcatel SIP0 1 255 255 255 Extern Station 1
6051 Alcatel SIP1 1 255 255 255 Extern Station 1
6052 Alcatel SIP2 1 255 255 255 Extern Station 1
8000 Alcatel DT 1 255 255 255 4035T 1
8001 Alcatel DT 6 1 0 2 6 4035T 1
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Directory Number 6003
PIN (Personal Ident.No.)
PIN No.
PIN With Secret Code
Type of control By COS
PIN group number 1
Can be Called/Dialed By Name YES
Phone book Name (Dial by name) Alactel
Phone book First Name DT3
Displayed Name Alactel DT3
Remote UA
Errors on Secret Code Counter 0
ACD station NO
NS Right (Notification server) NO
Incidents Teleservice NO
CSTA routing
General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities
Set Characteristics Hotel SIP_Attributes Miscellaneous All Action
1
Configuration: compedia
Start
OmniVista 4760 - COM...
3:44 PM




COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
compedia
Shelf
Media Gateway
PWT/DECT System
System
Translator
Classes of Service
Attendant
Users
6001 Alcatel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
Directory Keys
DECT set
Remote extension
User Aliases
Dynamic State User
User Associated Sets
TSC IP User
IPTouch_Parameters
Ith Profile
6004 Alactel DT4 1 0 2 4 4035T 1
6005 Alcatel DT 5 1 0 2 5 4035T 1
6020 Alcatel IPP0 1 255 255 255 4022 (4020 & TSC
6032 Alcatel Analog1 1 0 7 0 ANALOG 1
6033 Alactel Analog2 1 0 7 1 ANALOG 1
6034 Alactel Analog3 1 0 7 2 ANALOG 1
6035 Alactel Analog4 1 0 7 3 ANALOG 1
6050 Alcatel SIP0 1 255 255 255 Extern Station 1
6051 Alcatel SIP1 1 255 255 255 Extern Station 1
6052 Alcatel SIP2 1 255 255 255 Extern Station 1
8000 Alcatel DT 1 255 255 255 4035T 1
8001 Alcatel DT 6 1 0 2 6 4035T 1
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Directory Number 6003
4035 Features
Navigator UA 3G
Group PIN control No group
CCA Operations
A4980 No 4980
Z IVR
NOMADIC
TAPI premium server NO
Conference group -1
Announcement group -1
Call Restriction COS 0
Applicable Restriction COS 0
Implicit Priority
Activation mode 0
General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities
Set Characteristics Hotel SIP_Attributes Miscellaneous All Action
1
Configuration: compedia
Start
OmniVista 4760 - COM...
3:47 PM

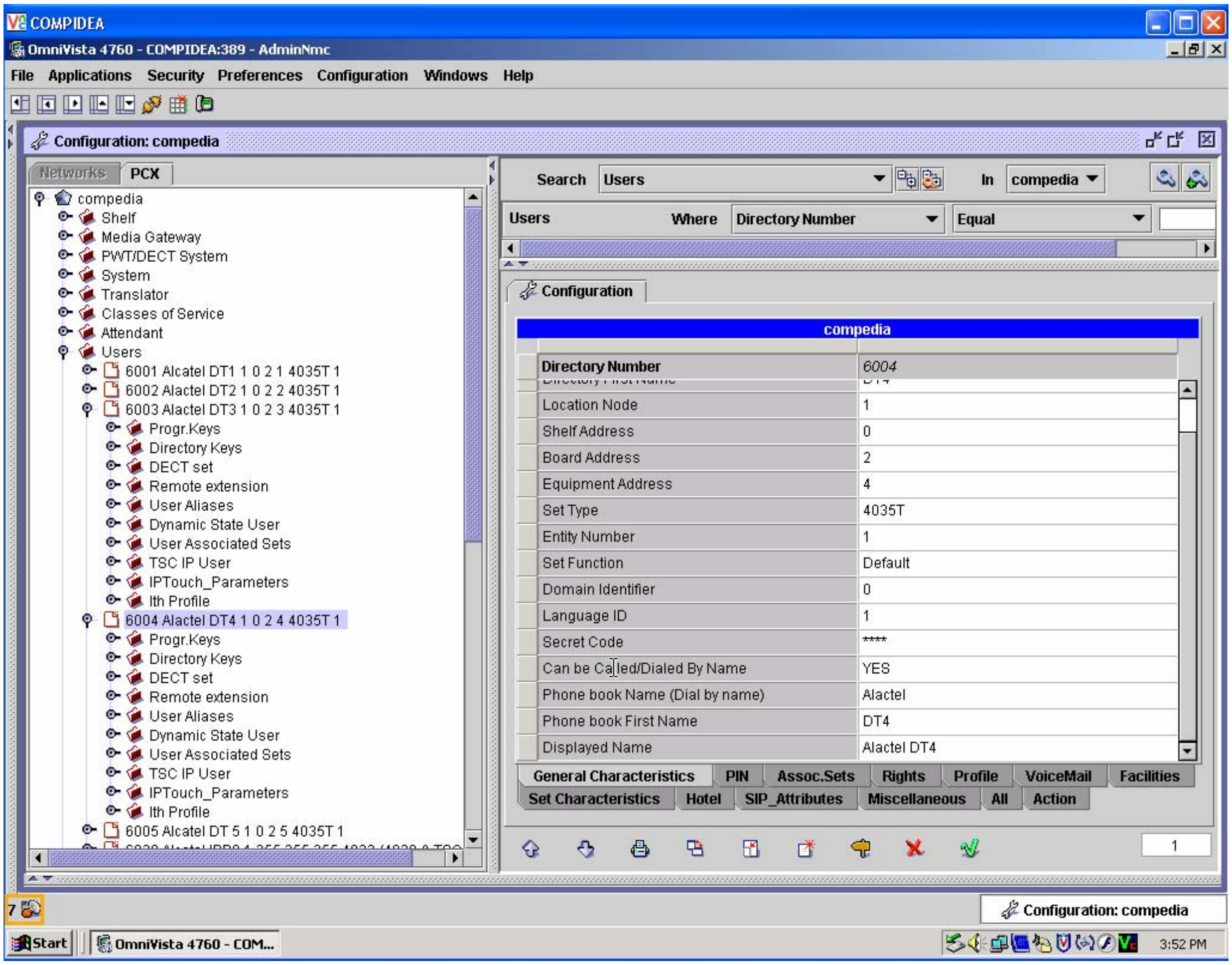



COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
compedia
Shelf
Media Gateway
PWT/DECT System
System
Translator
Classes of Service
Attendant
Users
6001 Alcatel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
Directory Keys
DECT set
Remote extension
User Aliases
Dynamic State User
User Associated Sets
TSC IP User
IPTouch_Parameters
Ith Profile
6004 Alactel DT4 1 0 2 4 4035T 1
6005 Alcatel DT 5 1 0 2 5 4035T 1
6020 Alcatel IPP0 1 255 255 255 4022 (4020 & TSC
6032 Alcatel Analog1 1 0 7 0 ANALOG 1
6033 Alactel Analog2 1 0 7 1 ANALOG 1
6034 Alactel Analog3 1 0 7 2 ANALOG 1
6035 Alactel Analog4 1 0 7 3 ANALOG 1
6050 Alcatel SIP0 1 255 255 255 Extern Station 1
6051 Alcatel SIP1 1 255 255 255 Extern Station 1
6052 Alcatel SIP2 1 255 255 255 Extern Station 1
8000 Alcatel DT 1 255 255 255 4035T 1
8001 Alcatel DT 6 1 0 2 6 4035T 1
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia
Directory Number 6003
Activation mode 0
Priority Level 0
Pre-emptable Primary Inc. Line NO
Pre-emptable Secondary Inc. Line NO
Priority Presentation NO
Ith Service type Not Valid
CUG List Number -1
Preferential CUG -1
CUG Outgoing Access
CUG Incoming Access
Automatic reconfiguration CTQ Forbidden - Connection TO
URL UserName
URL Domain
Advanced configuration
General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities
Set Characteristics Hotel SIP_Attributes Miscellaneous All Action
Configuration: compedia
Start
OmniVista 4760 - COM...
3:48 PM


COMPIDEA
OmniVista 4760 - COMPIDEA:389 - AdminNmc
File Applications Security Preferences Configuration Windows Help
Configuration: compedia
Networks PCX
Users
6001 Alcatel DT1 1 0 2 1 4035T 1
6002 Alactel DT2 1 0 2 2 4035T 1
6003 Alactel DT3 1 0 2 3 4035T 1
Progr.Keys
1 Programmed *40
2 Three-Party Conference
3 Transfer
4 DTMF End-to-end Dialling
5 Not Assigned
6 Not Assigned
7 Not Assigned
8 Not Assigned
9 Not Assigned
10 Not Assigned
11 Not Assigned
12 Not Assigned
13 Not Assigned
14 Not Assigned
15 Not Assigned
16 Not Assigned
17 Not Assigned
18 Not Assigned
19 Not Assigned
20 Not Assigned
22 ISDN
23 Redial
24 Redial Memory
Directory Keys
1 *40 Immed
2 *42 NoRep
3 *41 Busy
4 *3 LastCa
5
Search Users In compedia
Users Where Directory Number Equal
Configuration
compedia:6003
Key No. 1
Function Programmed
Content *40
Mnemo(Pocket,Mobile,4040,IPTouch)
Locked NO
All
1
Configuration: compedia
Start
OmniVista 4760 - COM...
3:55 PM

Alcatel digital type 4035 phone (phone B2 with extension 6004)

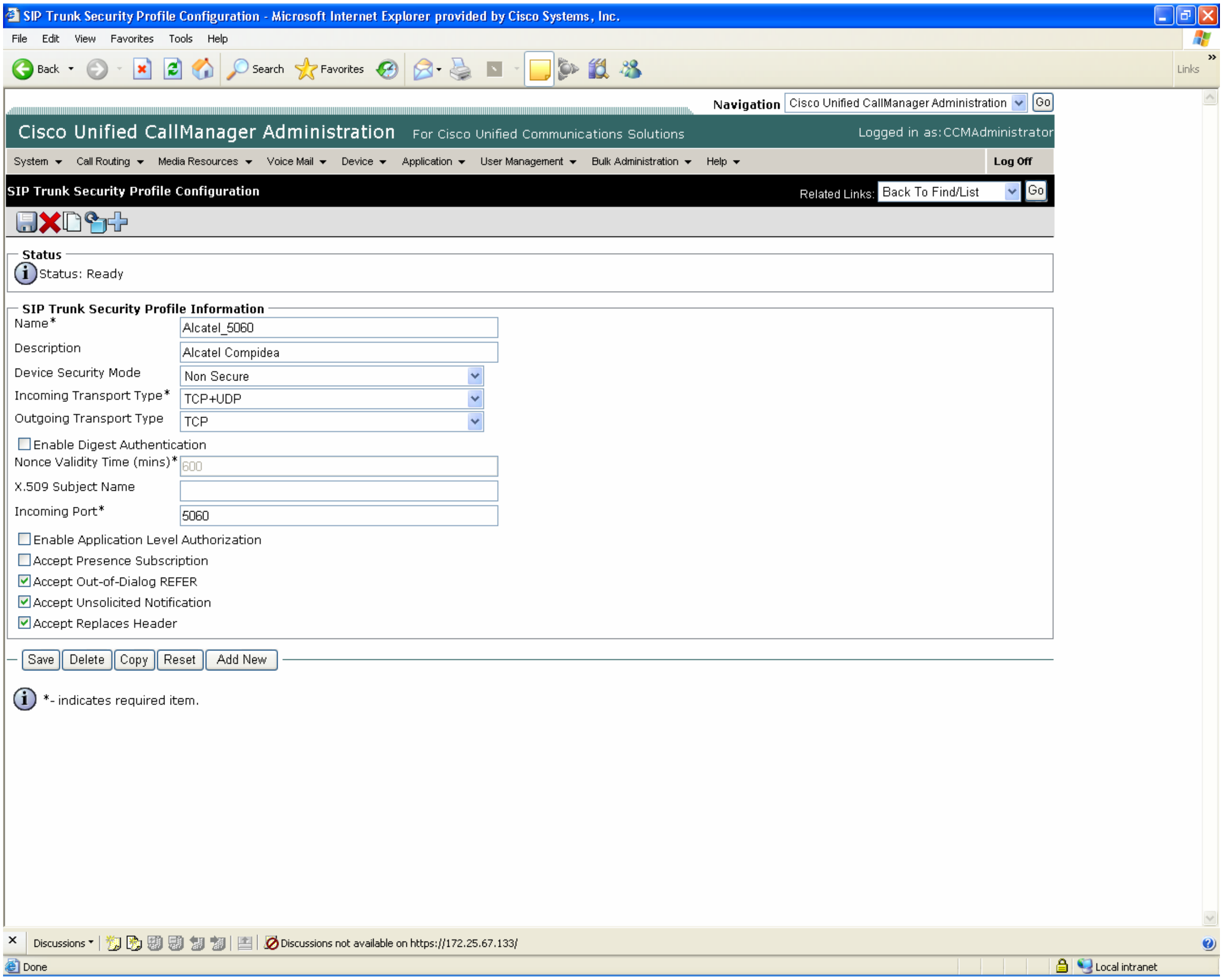
- Most of the parameters are the same as extension 6003 with exception to the DN and displayed name.



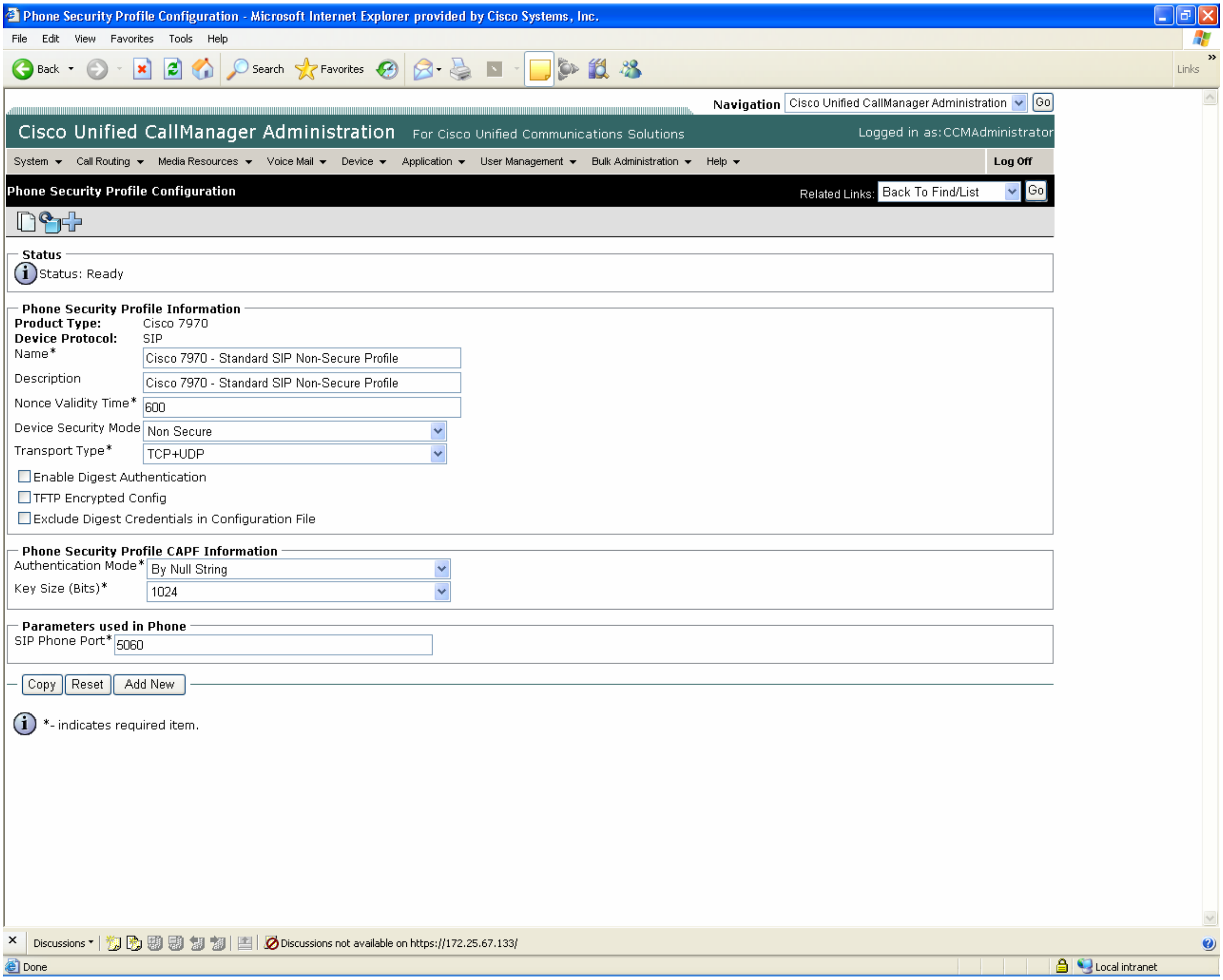
Cisco Unified CallManager Configuration

Cisco Unified CallManager Software Version

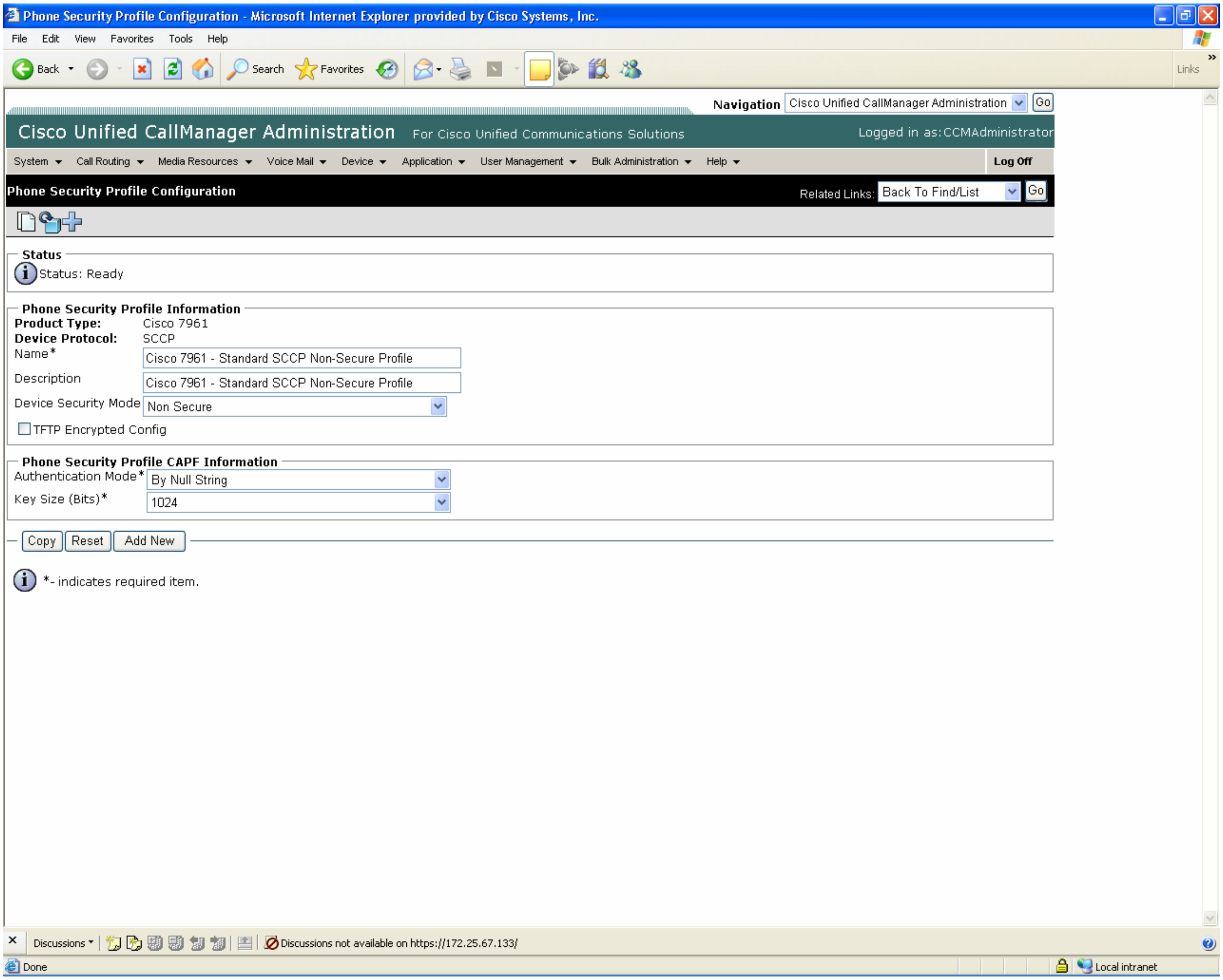
SIP Trunk Security Profile



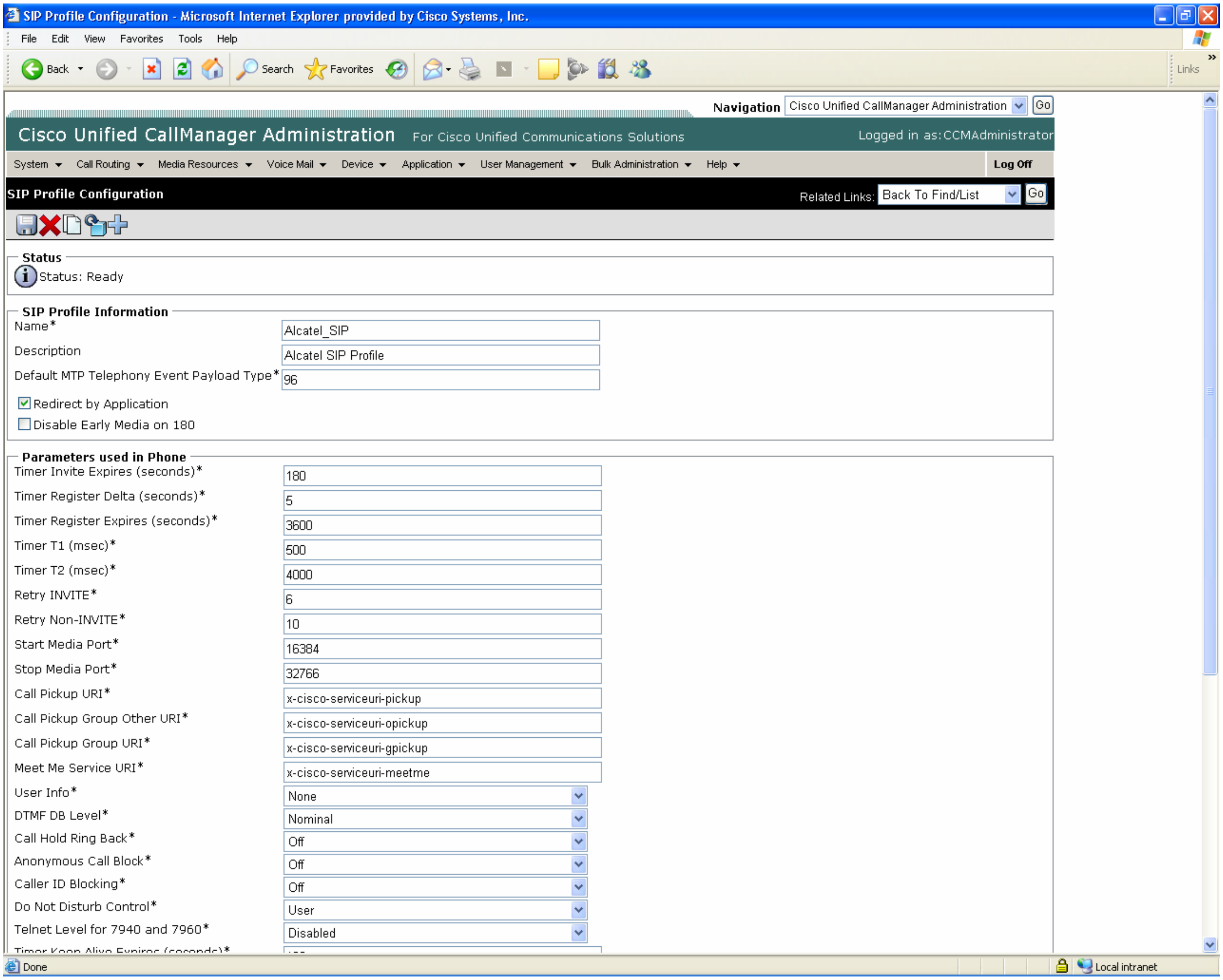
SIP Phone Security Profile

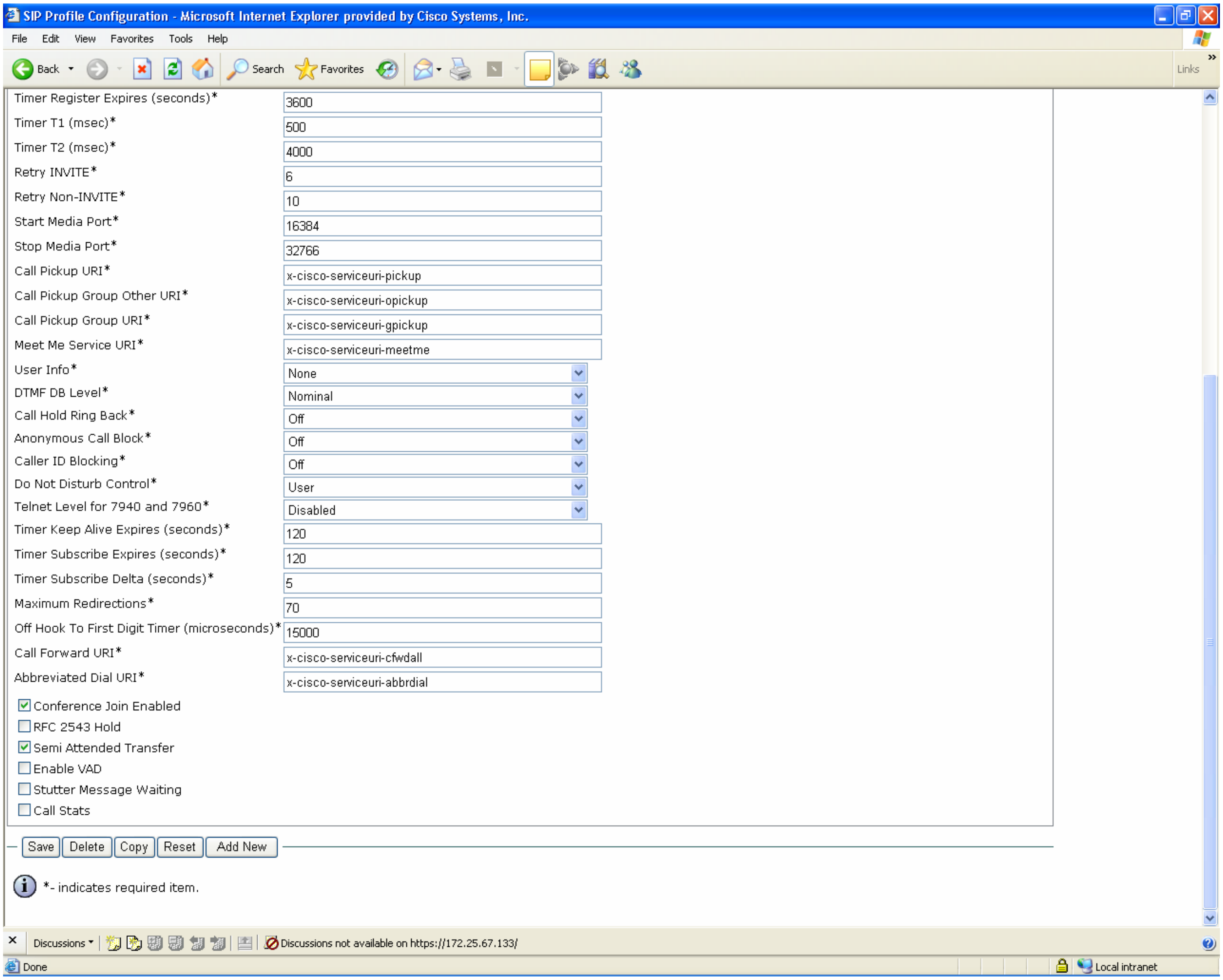


SCCP Phone Security Profile

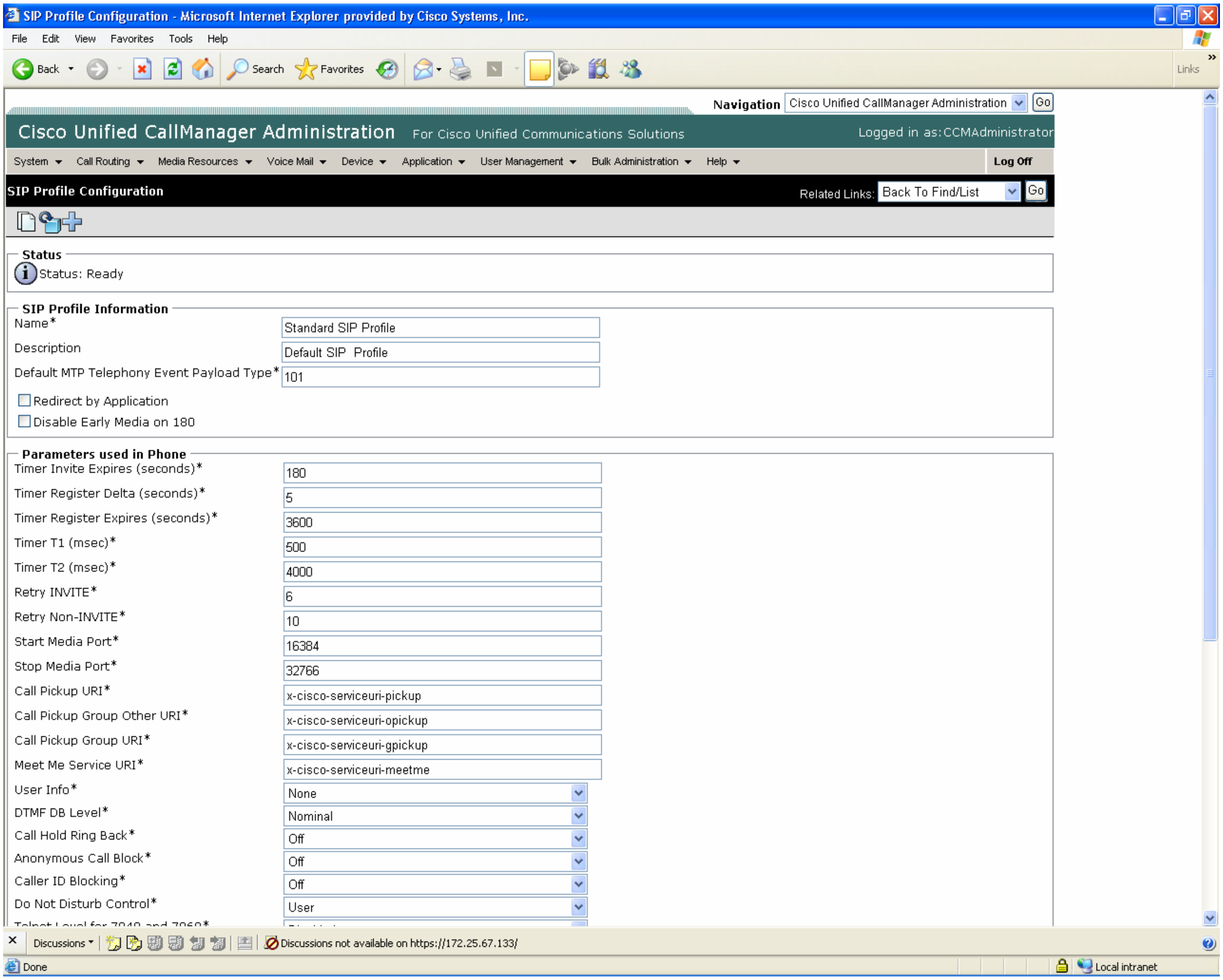


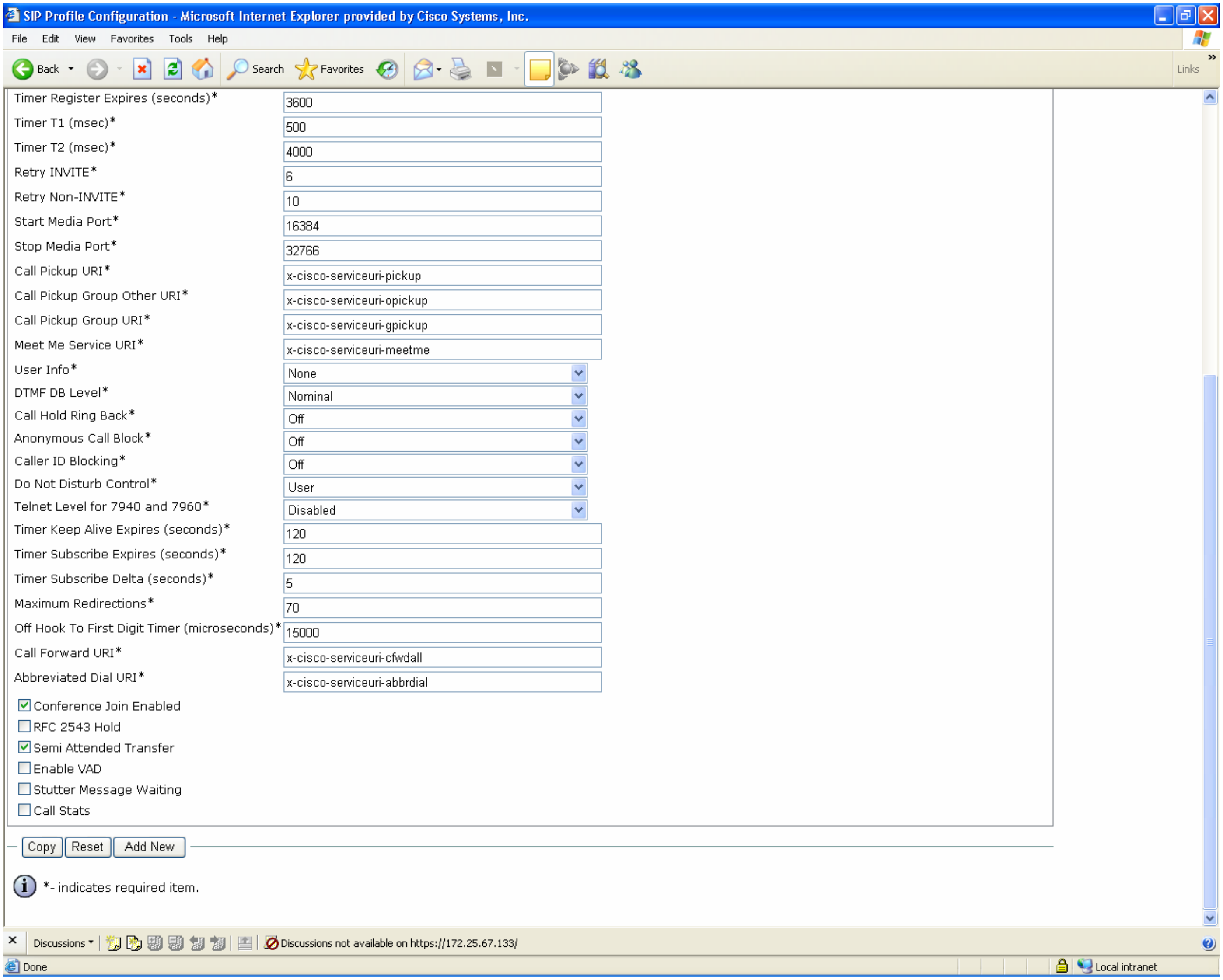
SIP Profile for SIP Trunk to Alcatel Call Server



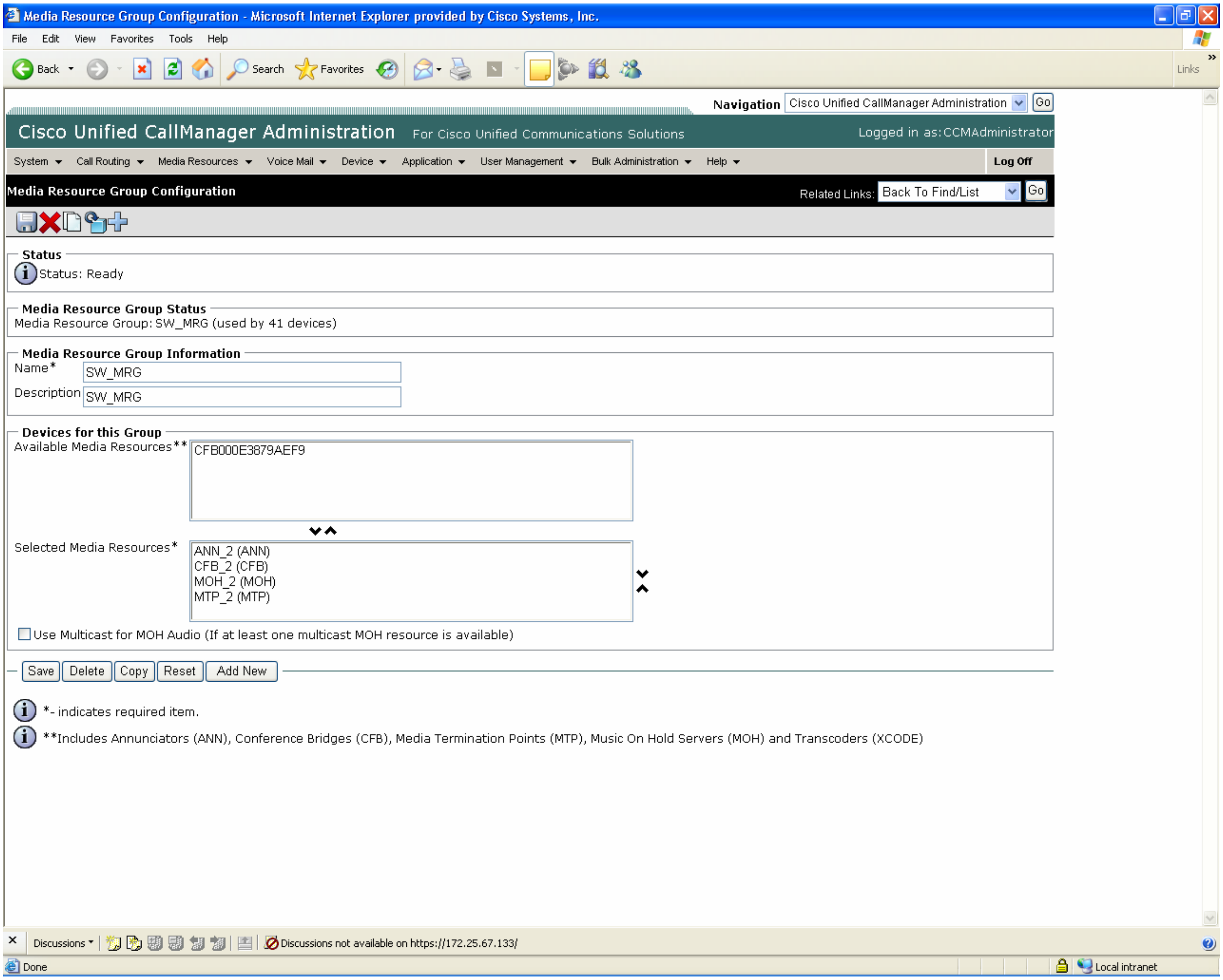

SIP Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Timer Register Expires (seconds)* 3600
Timer T1 (msec)* 500
Timer T2 (msec)* 4000
Retry INVITE* 6
Retry Non-INVITE* 10
Start Media Port* 16384
Stop Media 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 (microseconds)* 15000
Call Forward URI* x-cisco-serviceuri-cfwdall
Abbreviated Dial URI* x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
Stutter Message Waiting
Call Stats
Save Delete Copy Reset Add New
*- indicates required item.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

Standard SIP Profile

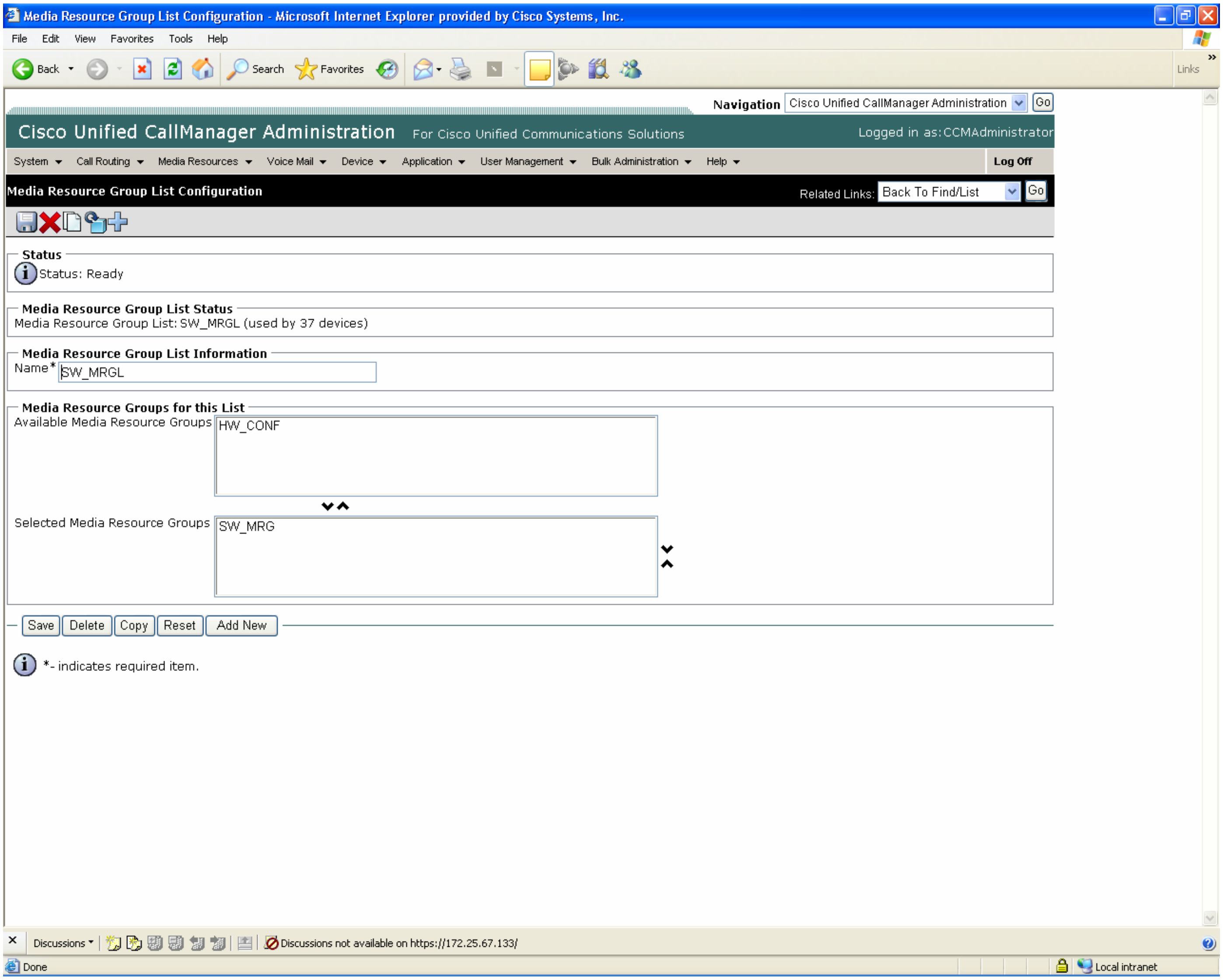



SIP Profile Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Links
Timer Register Expires (seconds)* 3600
Timer T1 (msec)* 500
Timer T2 (msec)* 4000
Retry INVITE* 6
Retry Non-INVITE* 10
Start Media Port* 16384
Stop Media 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 (microseconds)* 15000
Call Forward URI* x-cisco-serviceuri-cfwdall
Abbreviated Dial URI* x-cisco-serviceuri-abbrdial
Conference Join Enabled
RFC 2543 Hold
Semi Attended Transfer
Enable VAD
Stutter Message Waiting
Call Stats
Copy Reset Add New
*- indicates required item.
Discussions Discussions not available on https://172.25.67.133/
Done Local intranet

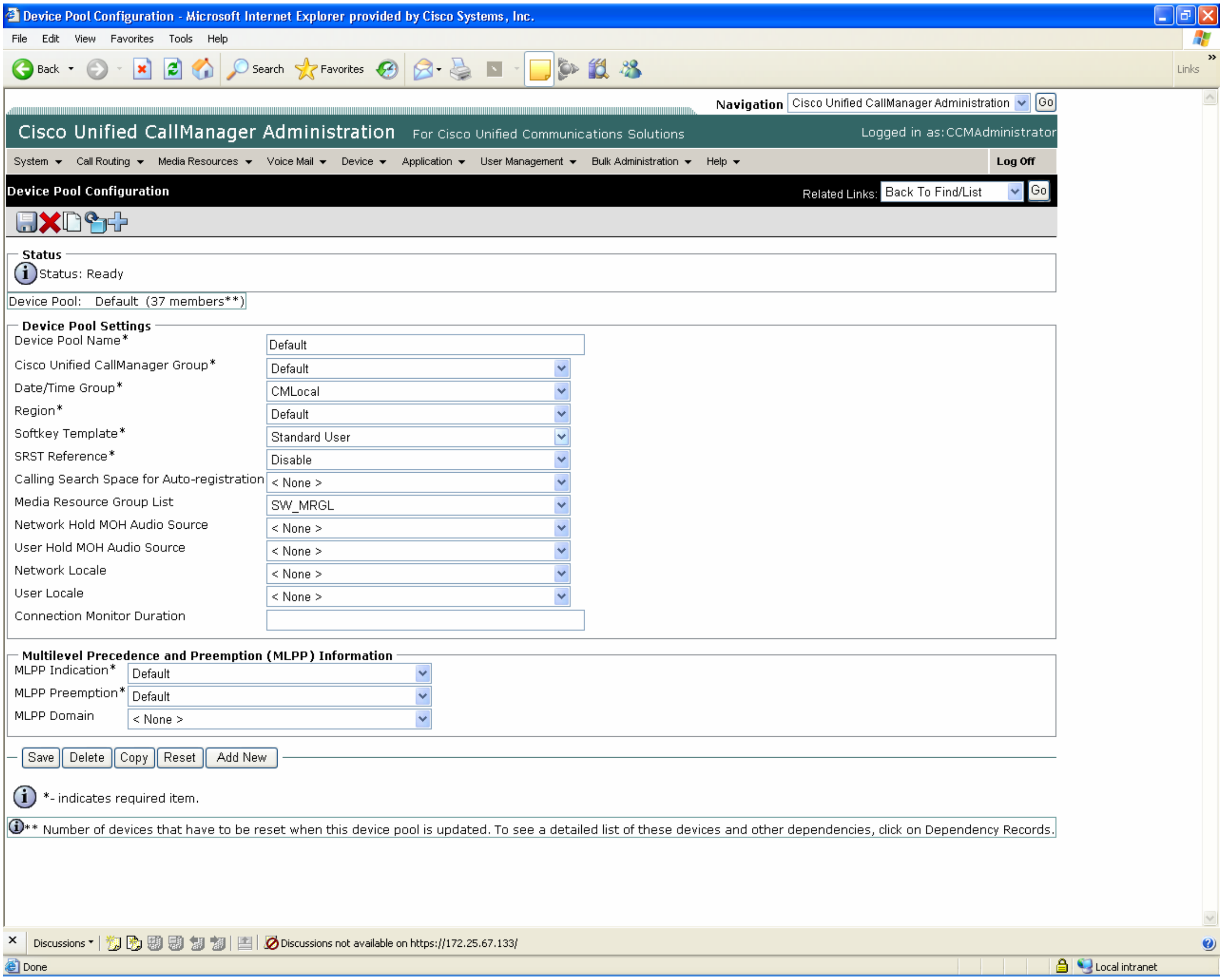
Media Resource Group



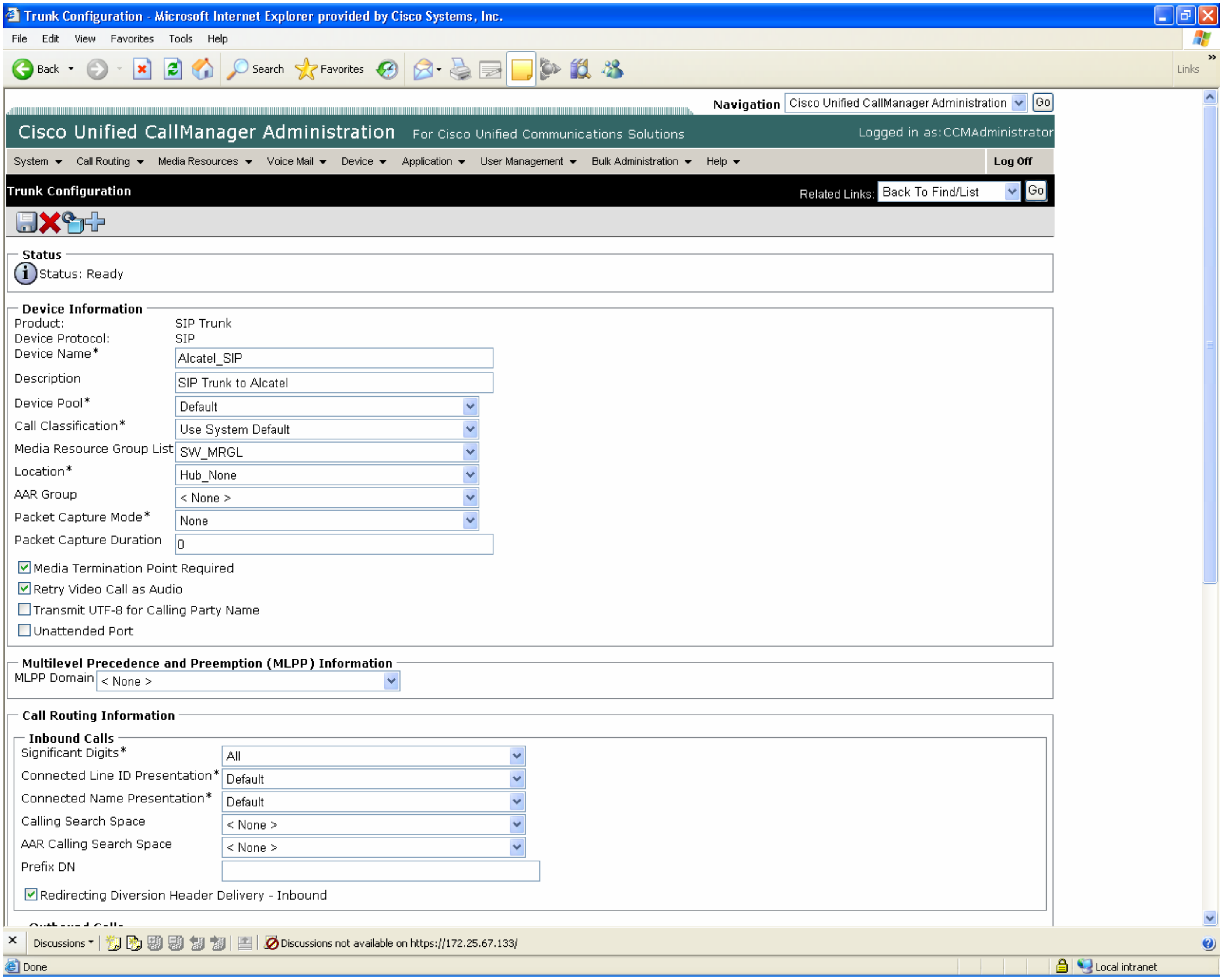
Media Resource Group List

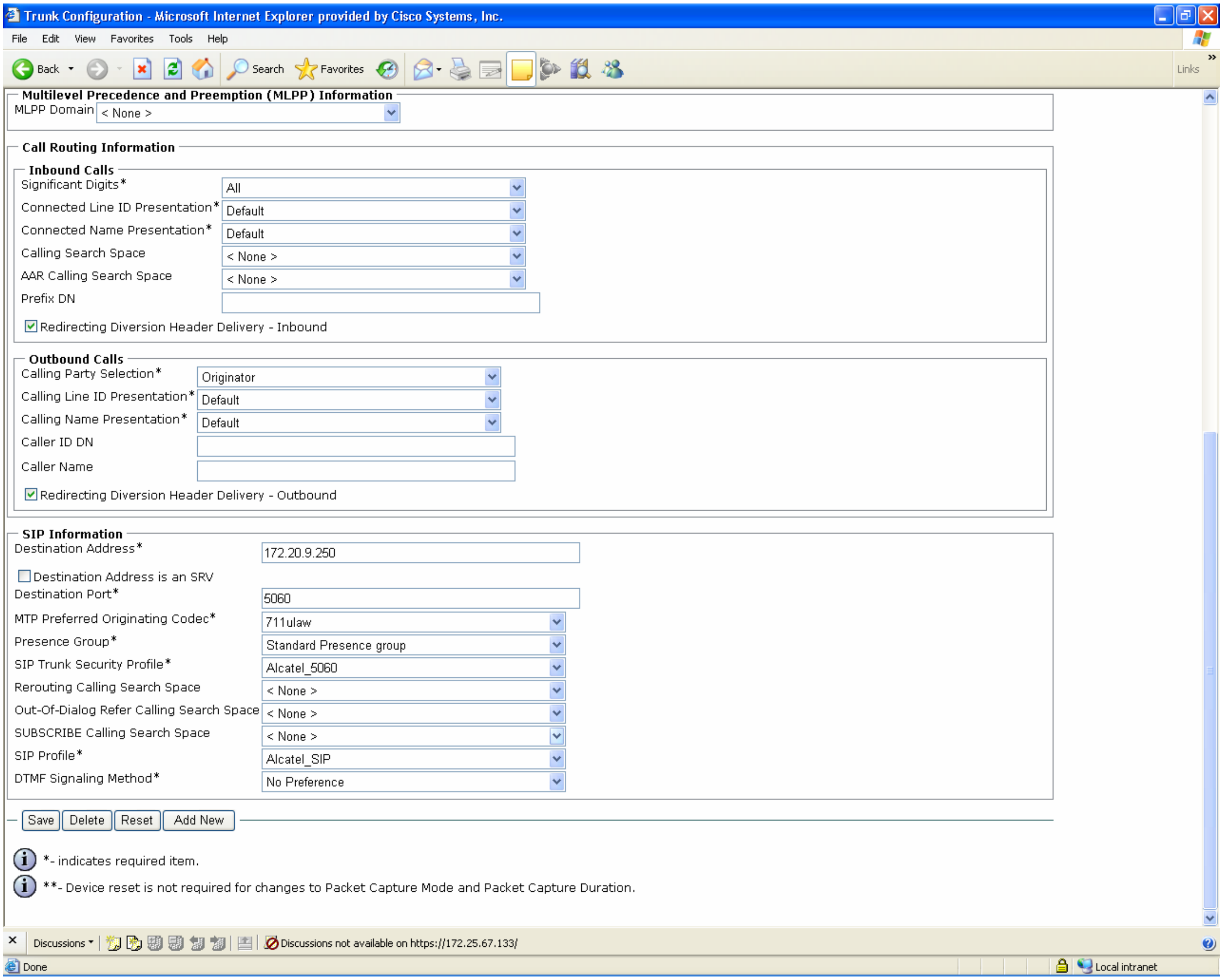


Assigned MRGL to Default Device Pool

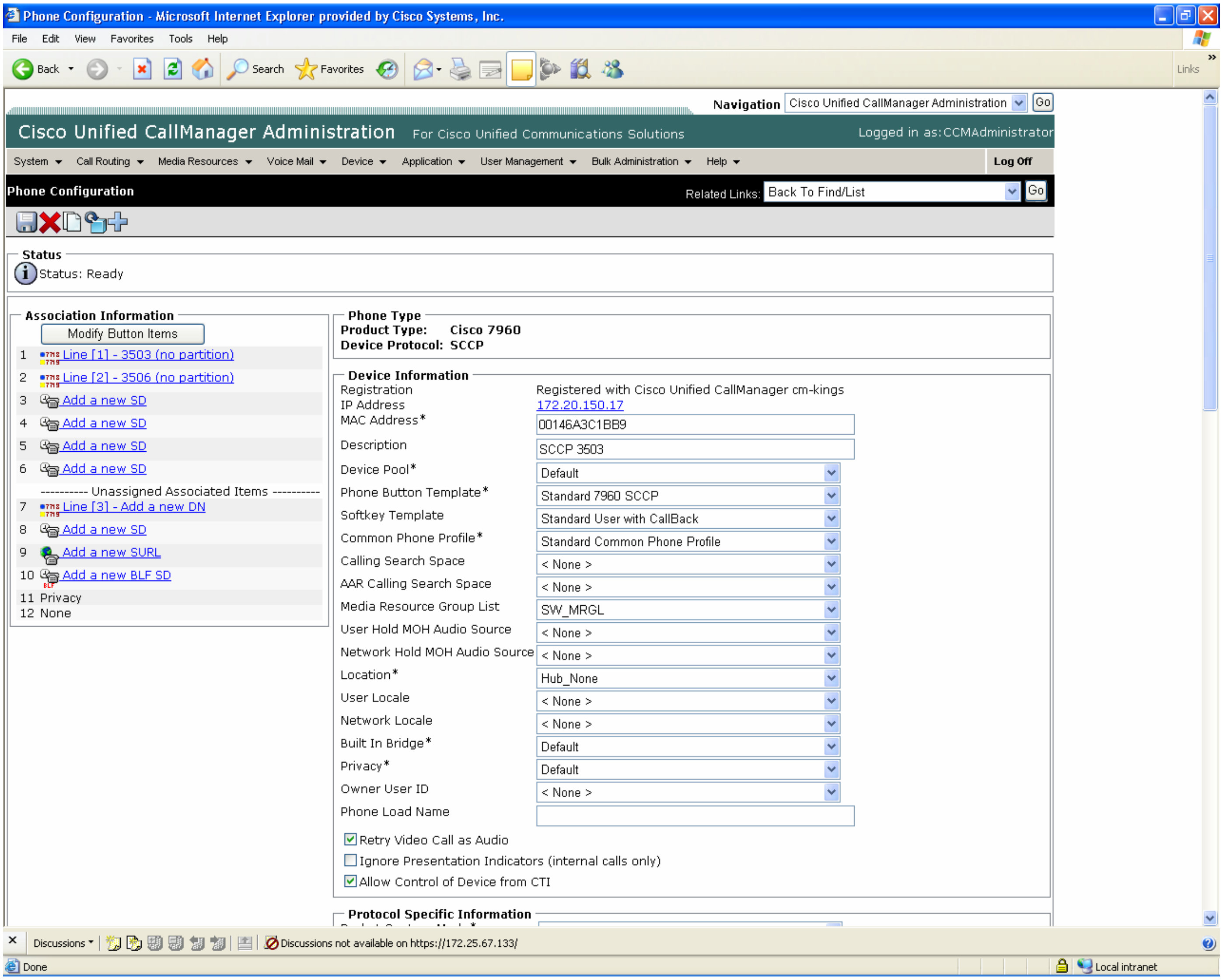


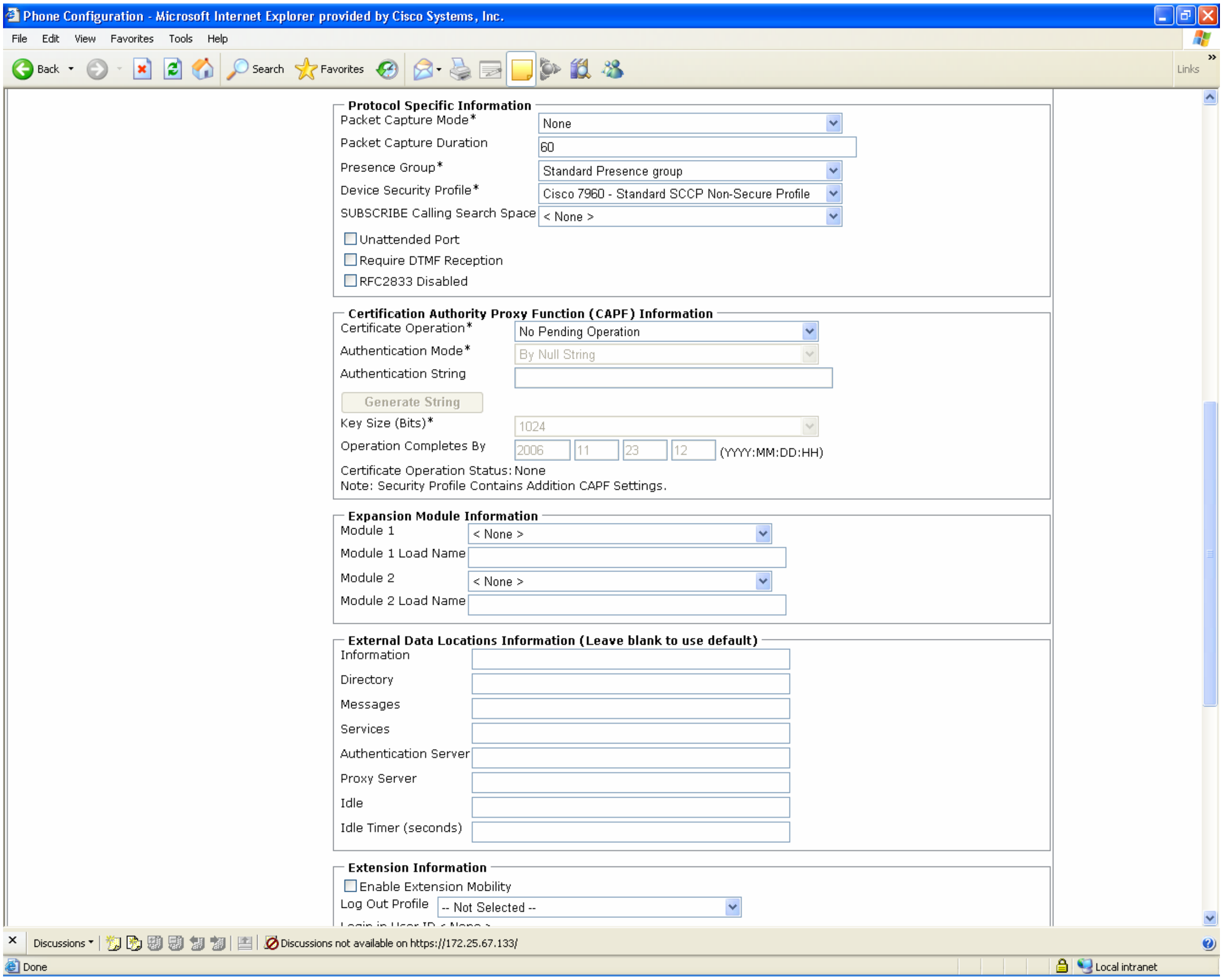
SIP Trunk to Alcatel SIP Call Server/Proxy Server

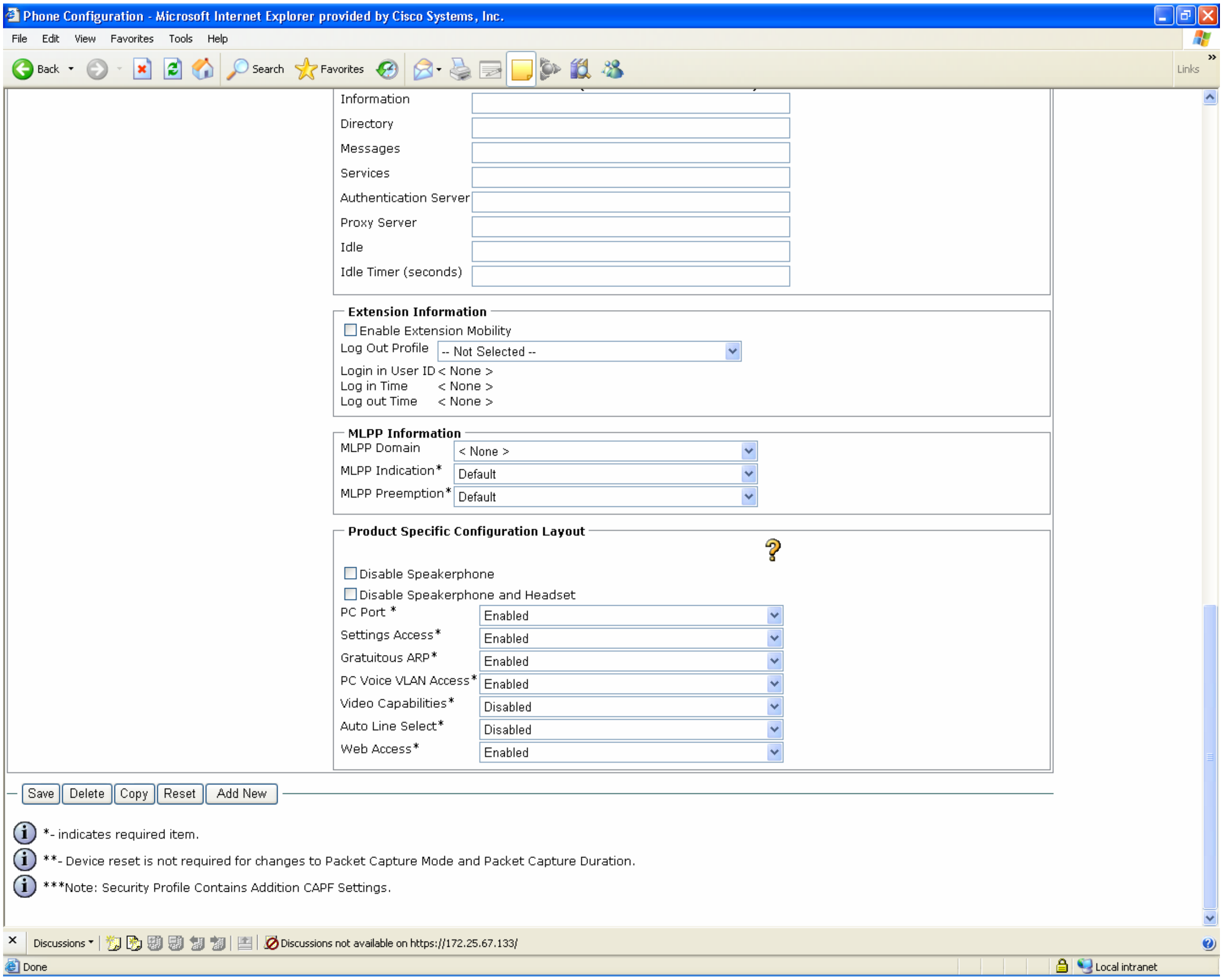



Trunk Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Links
Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >
Call Routing Information
Inbound Calls
Significant Digits* All
Connected Line ID Presentation* Default
Connected Name Presentation* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Prefix DN
Redirecting Diversion Header Delivery - Inbound
Outbound Calls
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 Address* 172.20.9.250
Destination Address is an SRV
Destination Port* 5060
MTP Preferred Originating Codec* 711ulaw
Presence Group* Standard Presence group
SIP Trunk Security Profile* Alcatel_5060
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Alcatel_SIP
DTMF Signaling Method* No Preference
Save Delete Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Discussions Discussions not available on https://172.25.67.133/
Done
Local intranet

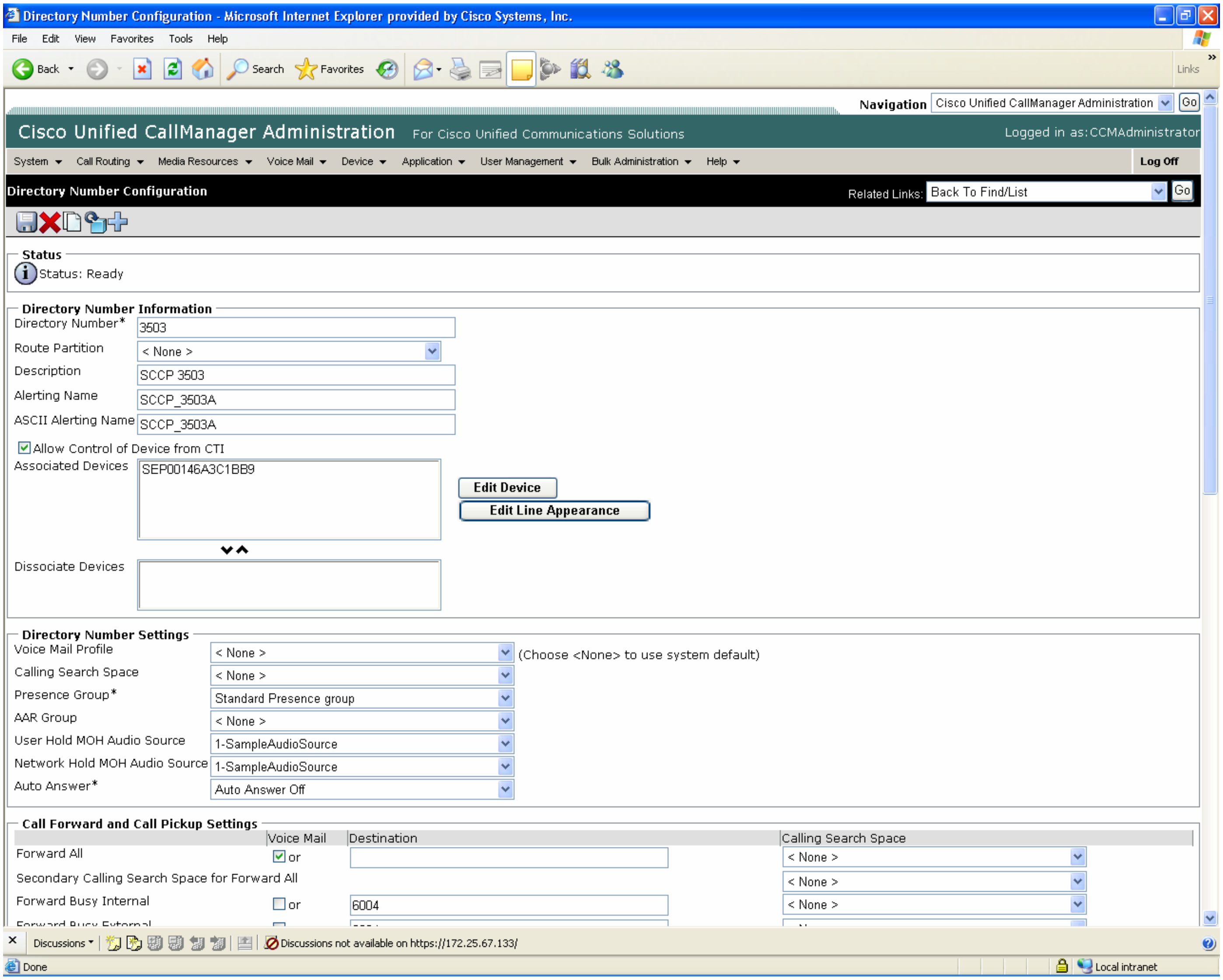
SCCP Phone Ext. 3503 Device Level Configuration

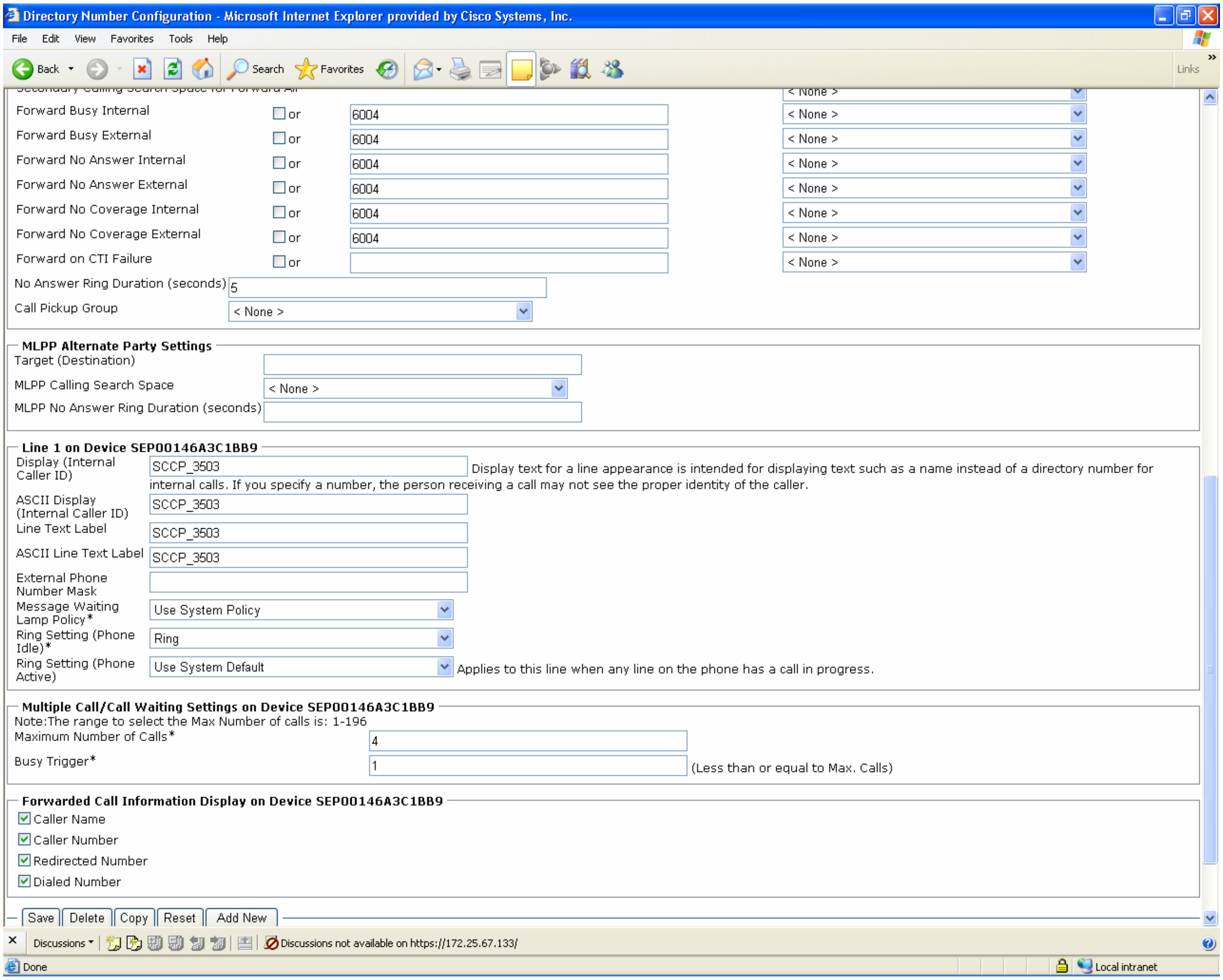



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 60
Presence Group* Standard Presence group
Device Security Profile* Cisco 7960 - Standard SCCP Non-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 2006 11 23 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 -- Not Selected --
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet


Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Extension Information
Enable Extension Mobility
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain < None >
MLPP Indication* Default
MLPP Preemption* Default
Product Specific Configuration Layout
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Enabled
PC Voice VLAN Access* Enabled
Video Capabilities* Disabled
Auto Line Select* Disabled
Web Access* Enabled
Save Delete Copy Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
***Note: Security Profile Contains Addition CAPF Settings.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

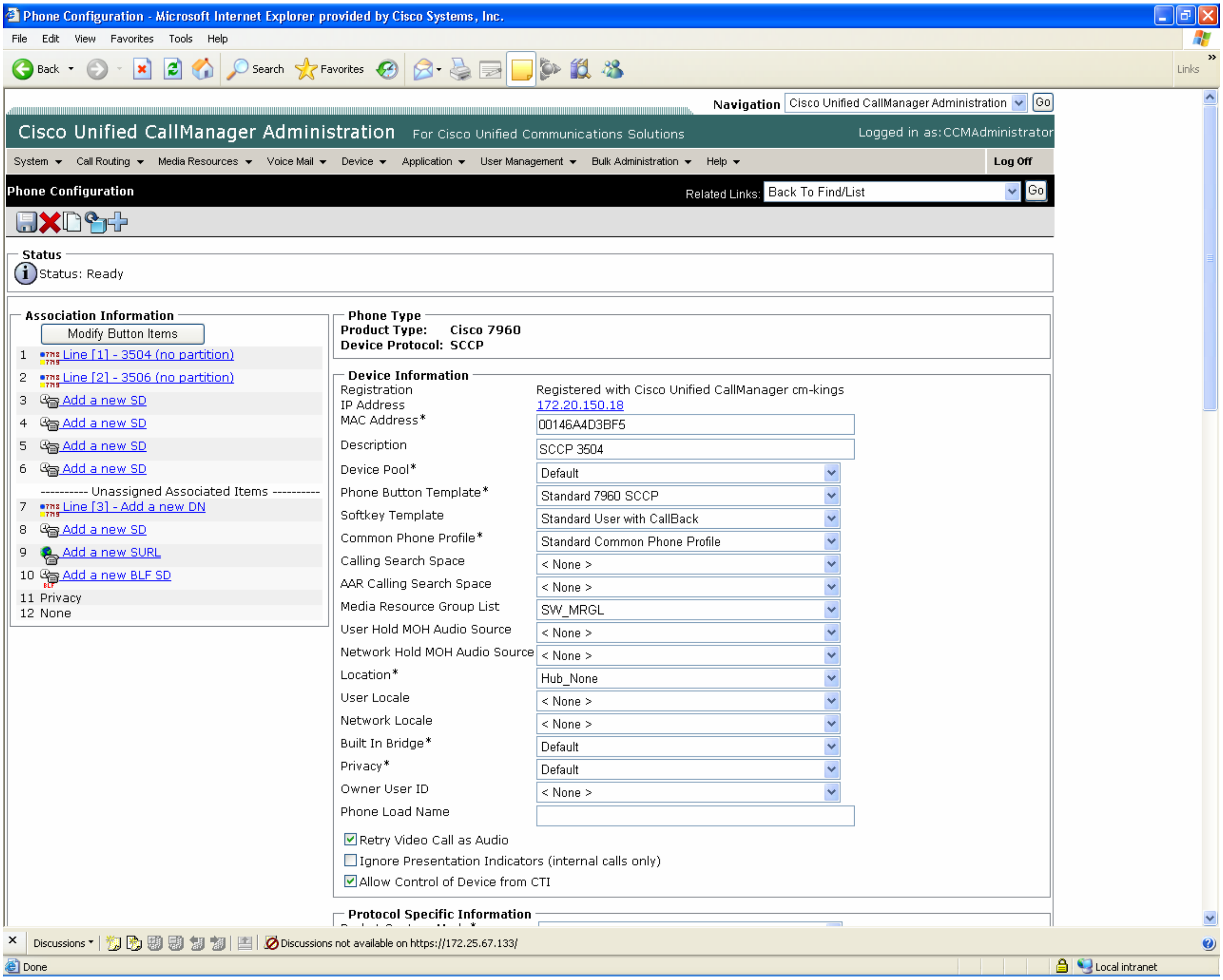
SCCP Phone Ext. 3503 Directory Number Level Configuration

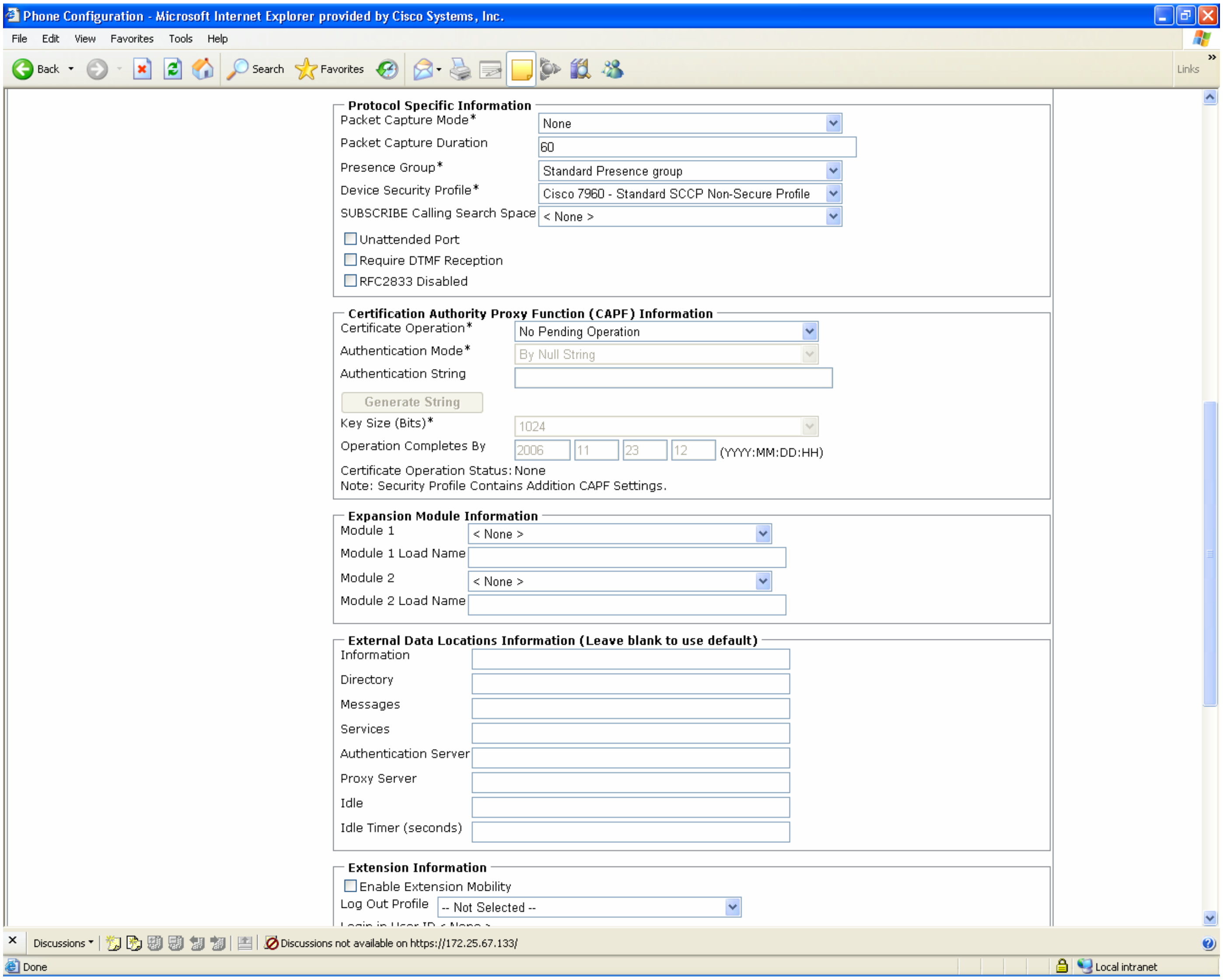


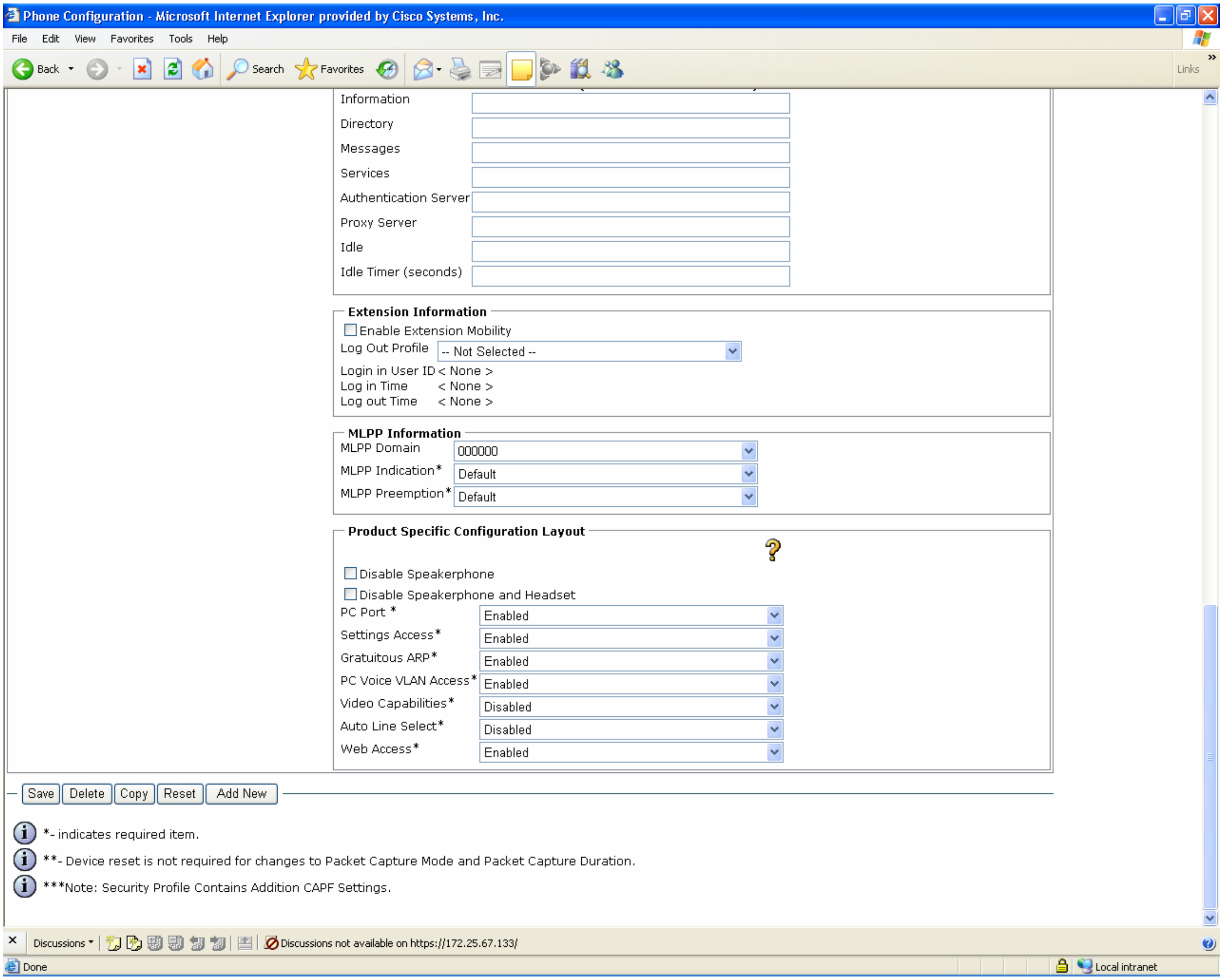


Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Forward Busy Internal or 6004 < None >
Forward Busy External or 6004 < None >
Forward No Answer Internal or 6004 < None >
Forward No Answer External or 6004 < None >
Forward No Coverage Internal or 6004 < None >
Forward No Coverage External or 6004 < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds) 5
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP00146A3C1BB9
Display (Internal Caller ID) SCCP_3503
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) SCCP_3503
Line Text Label SCCP_3503
ASCII Line Text Label SCCP_3503
External Phone Number Mask
Message Waiting Lamp Policy* Use System Policy
Ring Setting (Phone Idle)* Ring
Ring Setting (Phone Active) Use System Default
Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP00146A3C1BB9
Note:The range to select the Max Number of calls is: 1-196
Maximum Number of Calls* 4
Busy Trigger* 1
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP00146A3C1BB9
Caller Name
Caller Number
Redirected Number
Dialed Number
Save Delete Copy Reset Add New
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

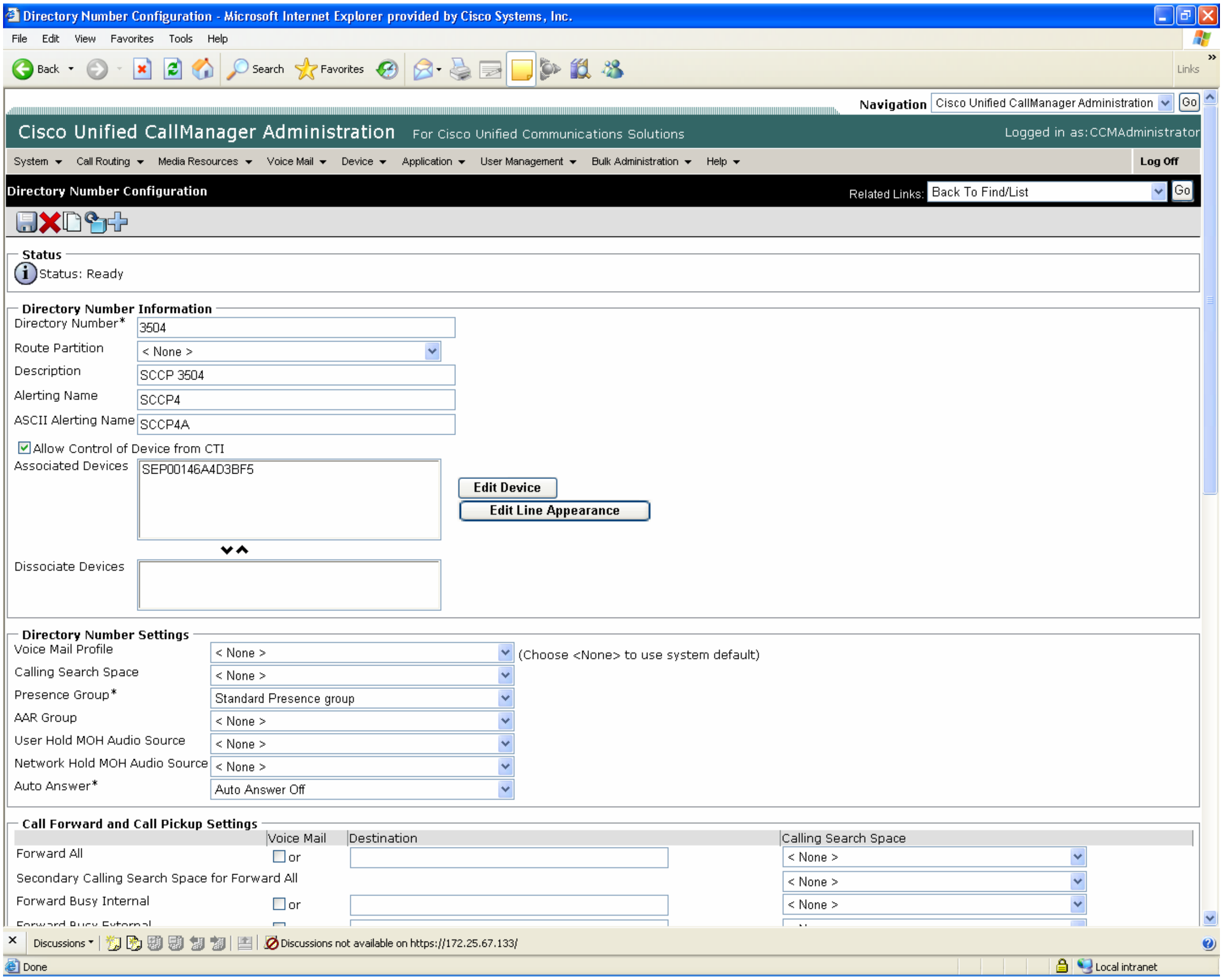
SCCP Phone Ext. 3504 Device Level Configuration

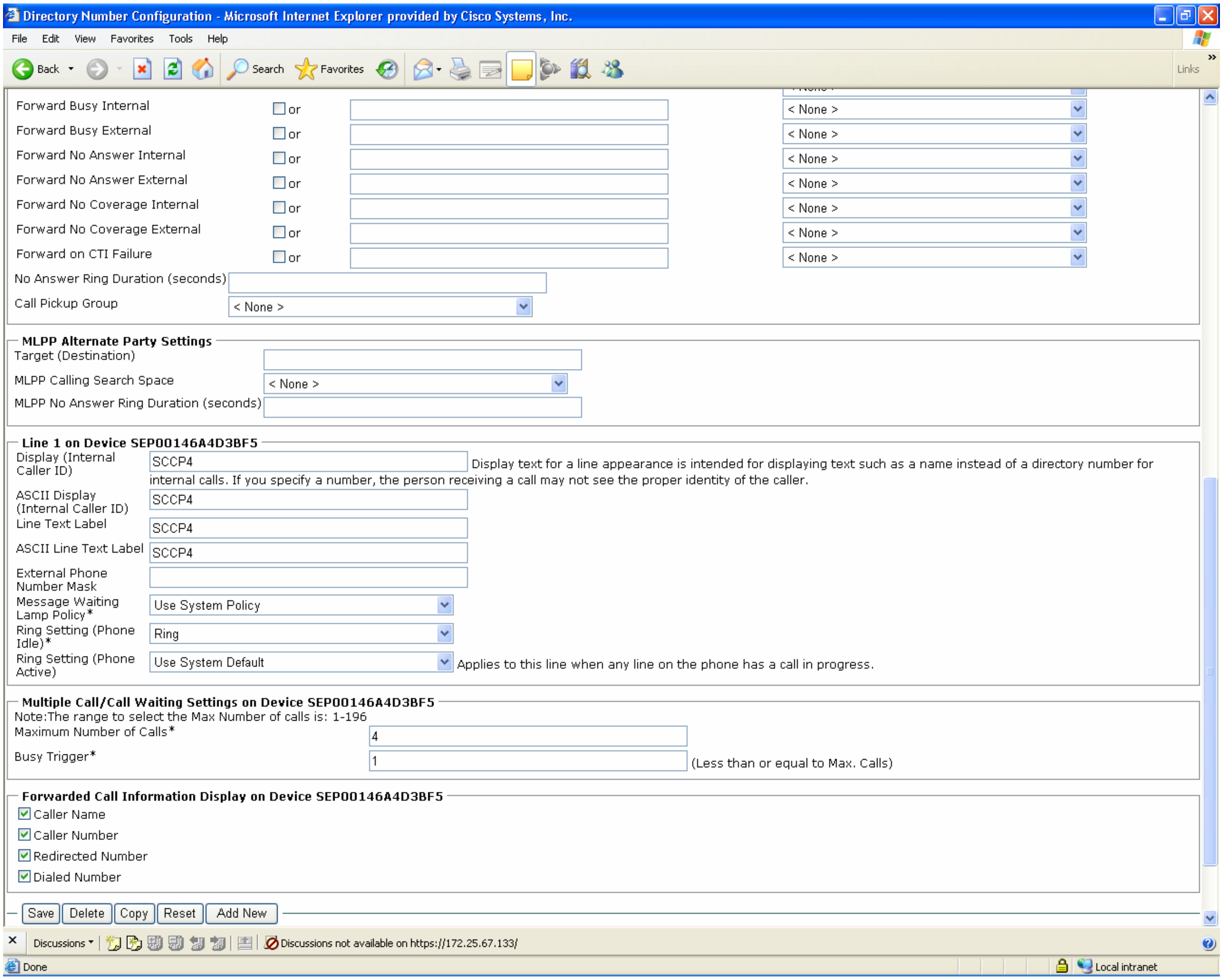



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 60
Presence Group* Standard Presence group
Device Security Profile* Cisco 7960 - Standard SCCP Non-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 2006 11 23 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 -- Not Selected --
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet


Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Information
Directory
Messages
Services
Authentication Server
Proxy Server
Idle
Idle Timer (seconds)
Extension Information
Enable Extension Mobility
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain 000000
MLPP Indication* Default
MLPP Preemption* Default
Product Specific Configuration Layout
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Enabled
PC Voice VLAN Access* Enabled
Video Capabilities* Disabled
Auto Line Select* Disabled
Web Access* Enabled
Save Delete Copy Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
***Note: Security Profile Contains Addition CAPF Settings.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

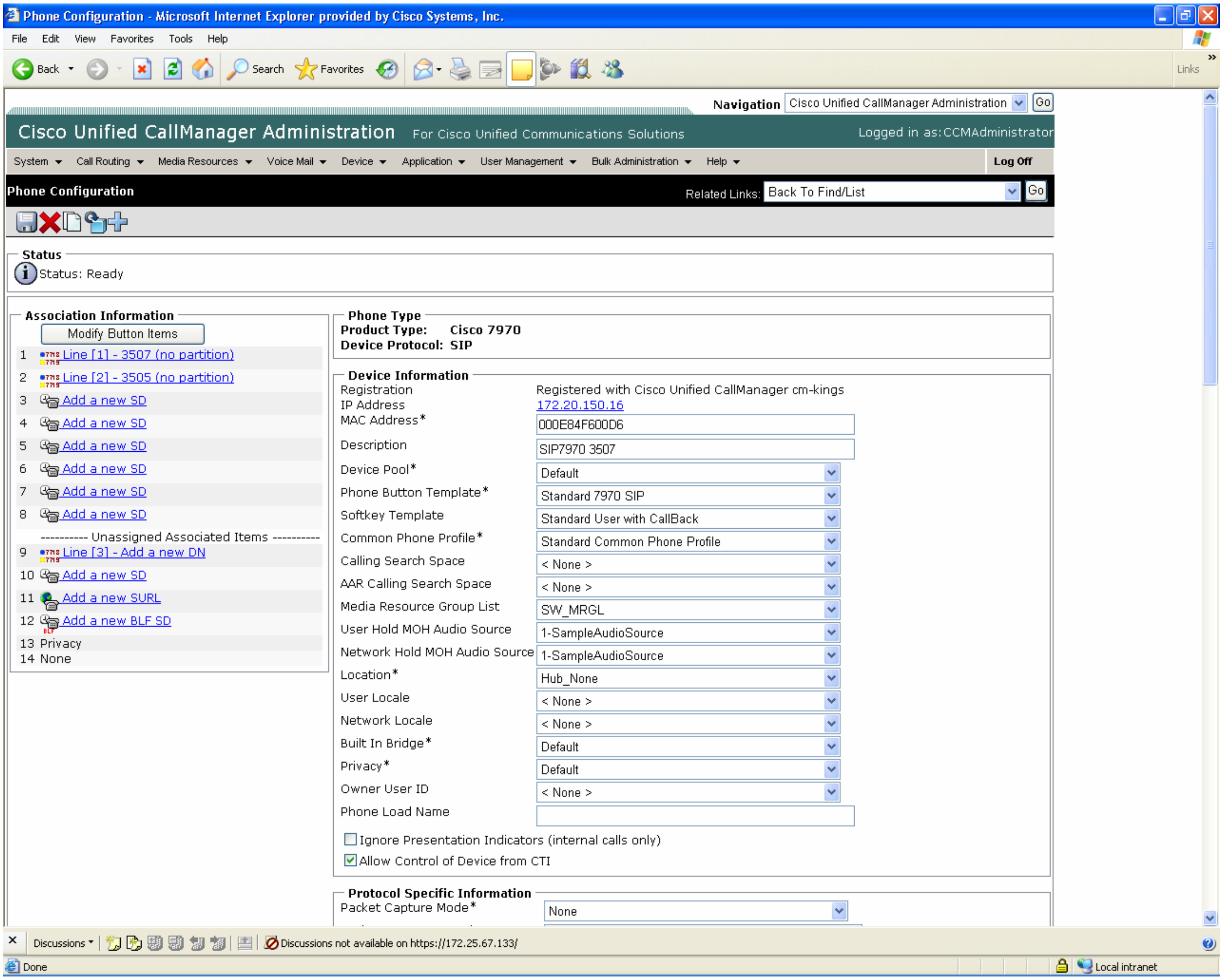
SCCP Phone Ext. 3504 Directory Number Level Configuration

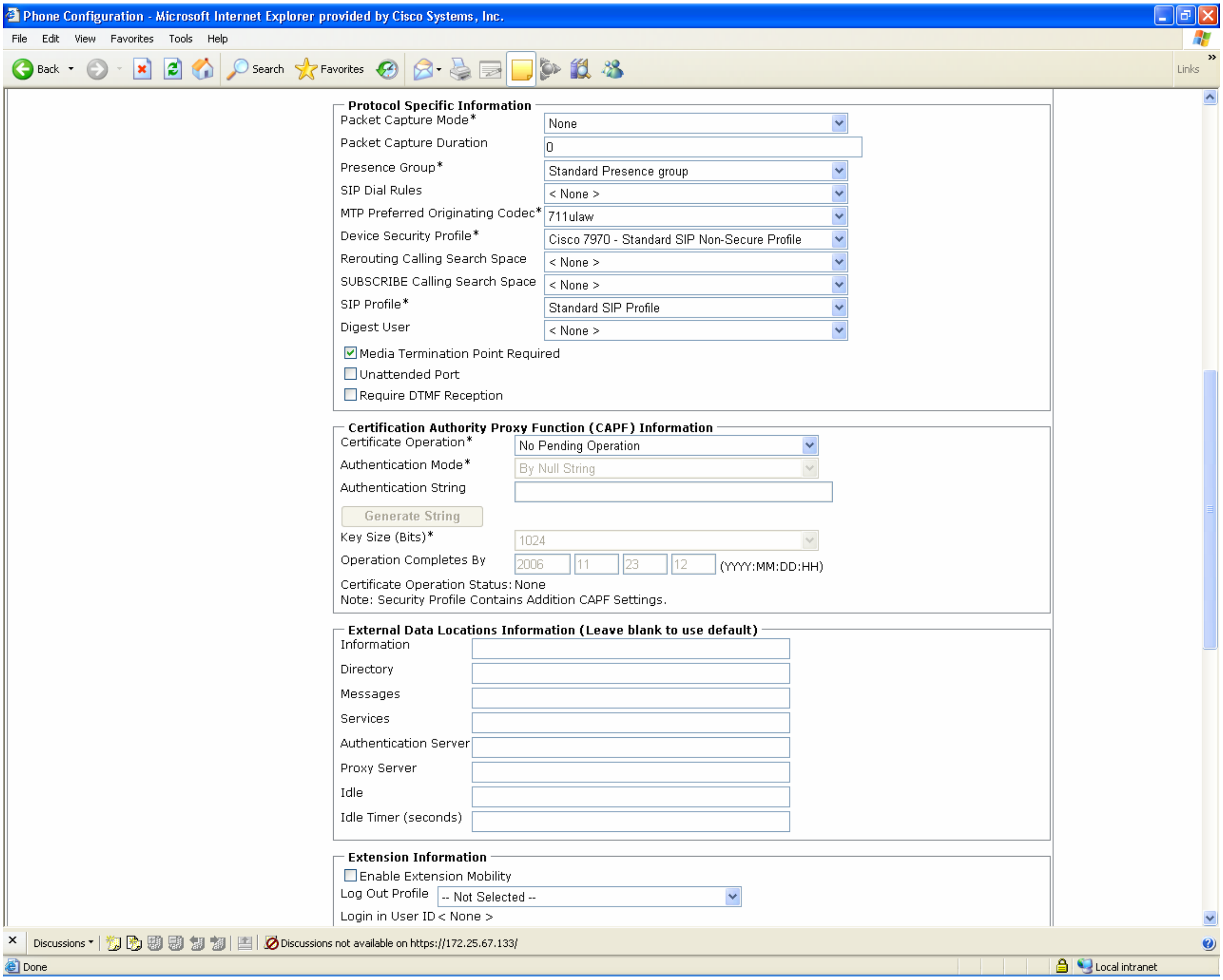


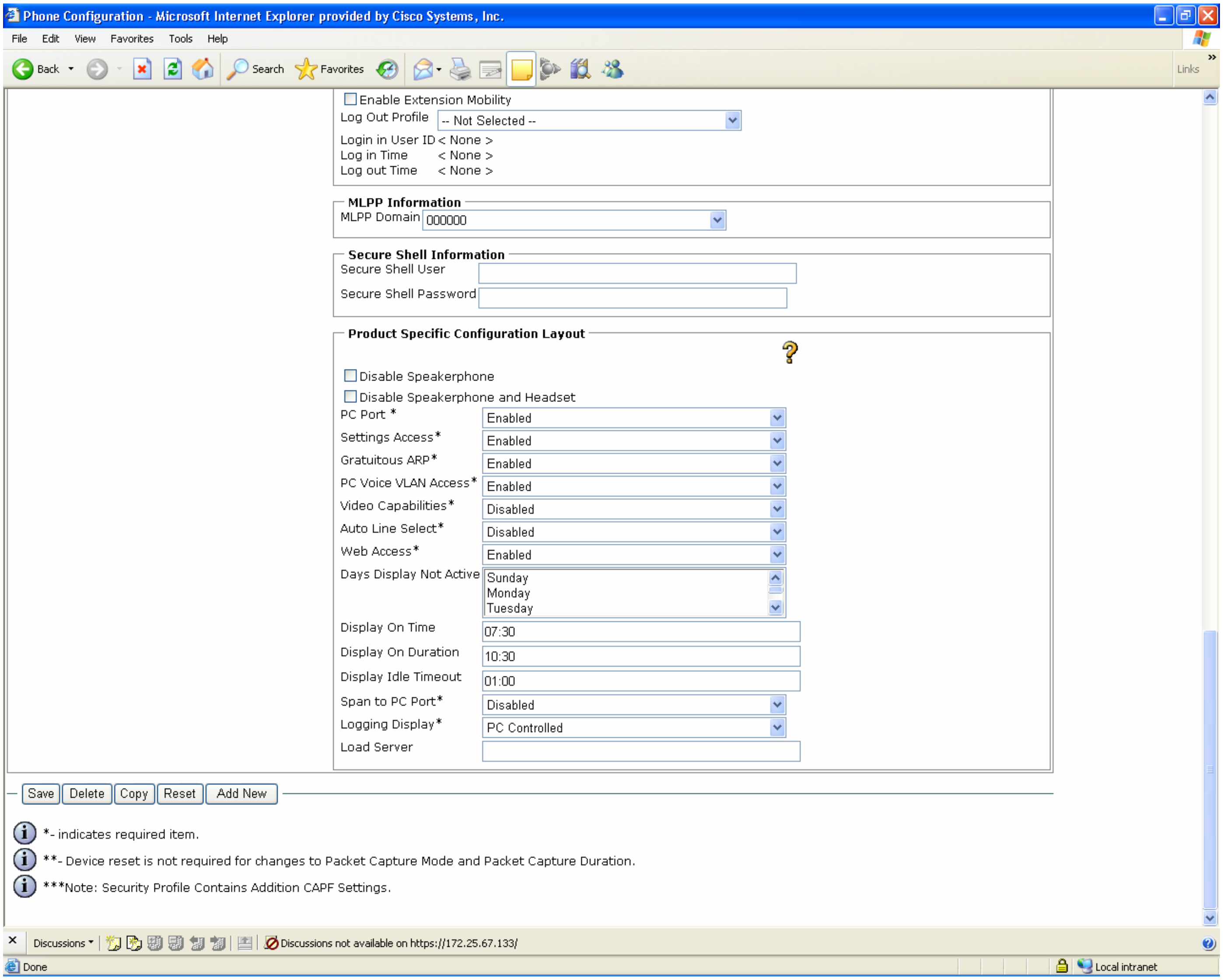


Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Forward Busy Internal or < None >
Forward Busy External or < None >
Forward No Answer Internal or < None >
Forward No Answer External or < None >
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP00146A4D3BF5
Display (Internal Caller ID) SCCP4
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) SCCP4
Line Text Label SCCP4
ASCII Line Text Label SCCP4
External Phone Number Mask
Message Waiting Lamp Policy* Use System Policy
Ring Setting (Phone Idle)* Ring
Ring Setting (Phone Active) Use System Default
Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP00146A4D3BF5
Note:The range to select the Max Number of calls is: 1-196
Maximum Number of Calls* 4
Busy Trigger* 1
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP00146A4D3BF5
Caller Name
Caller Number
Redirected Number
Dialed Number
Save Delete Copy Reset Add New
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

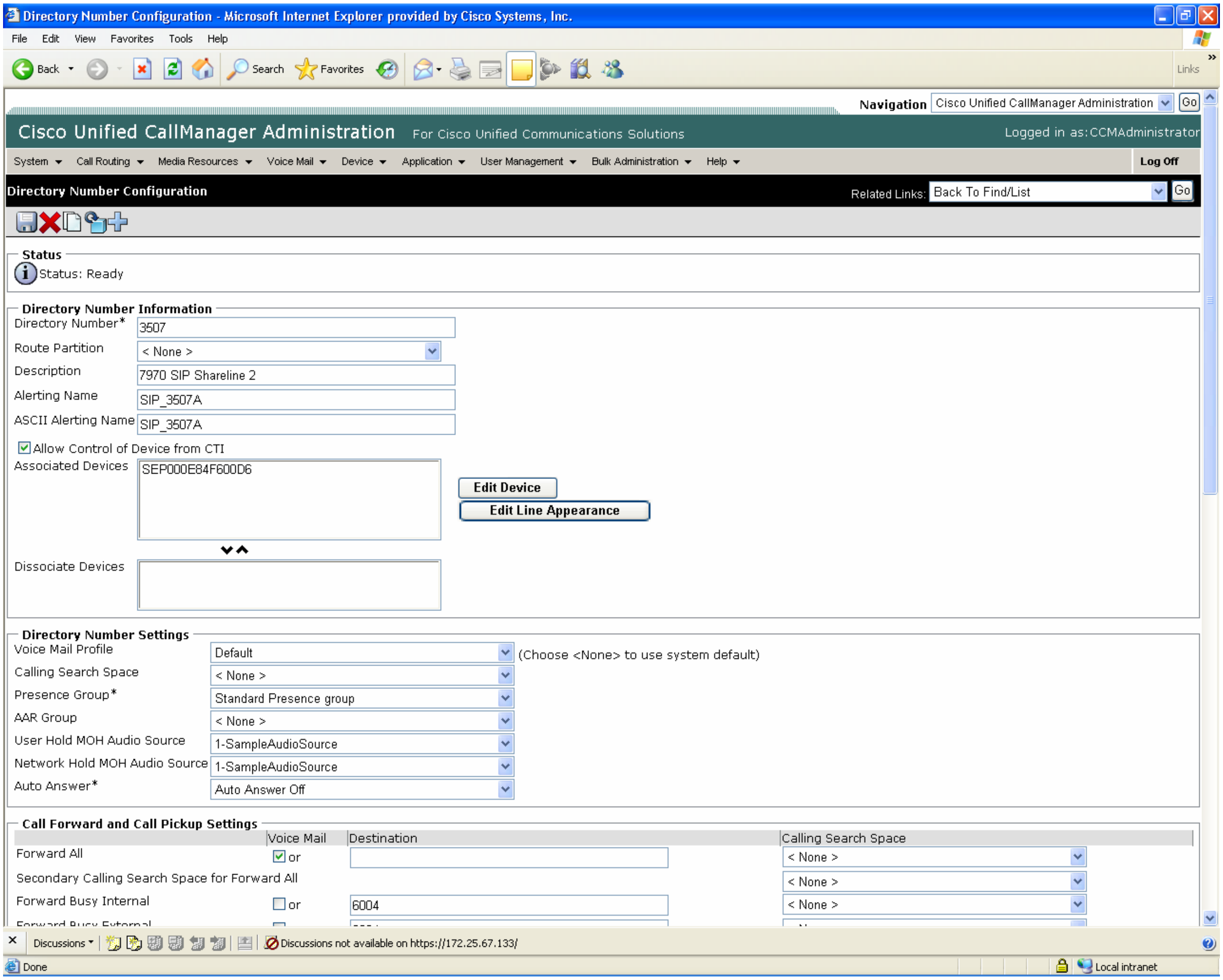
SIP Phone Ext. 3507 Device Level Configuration

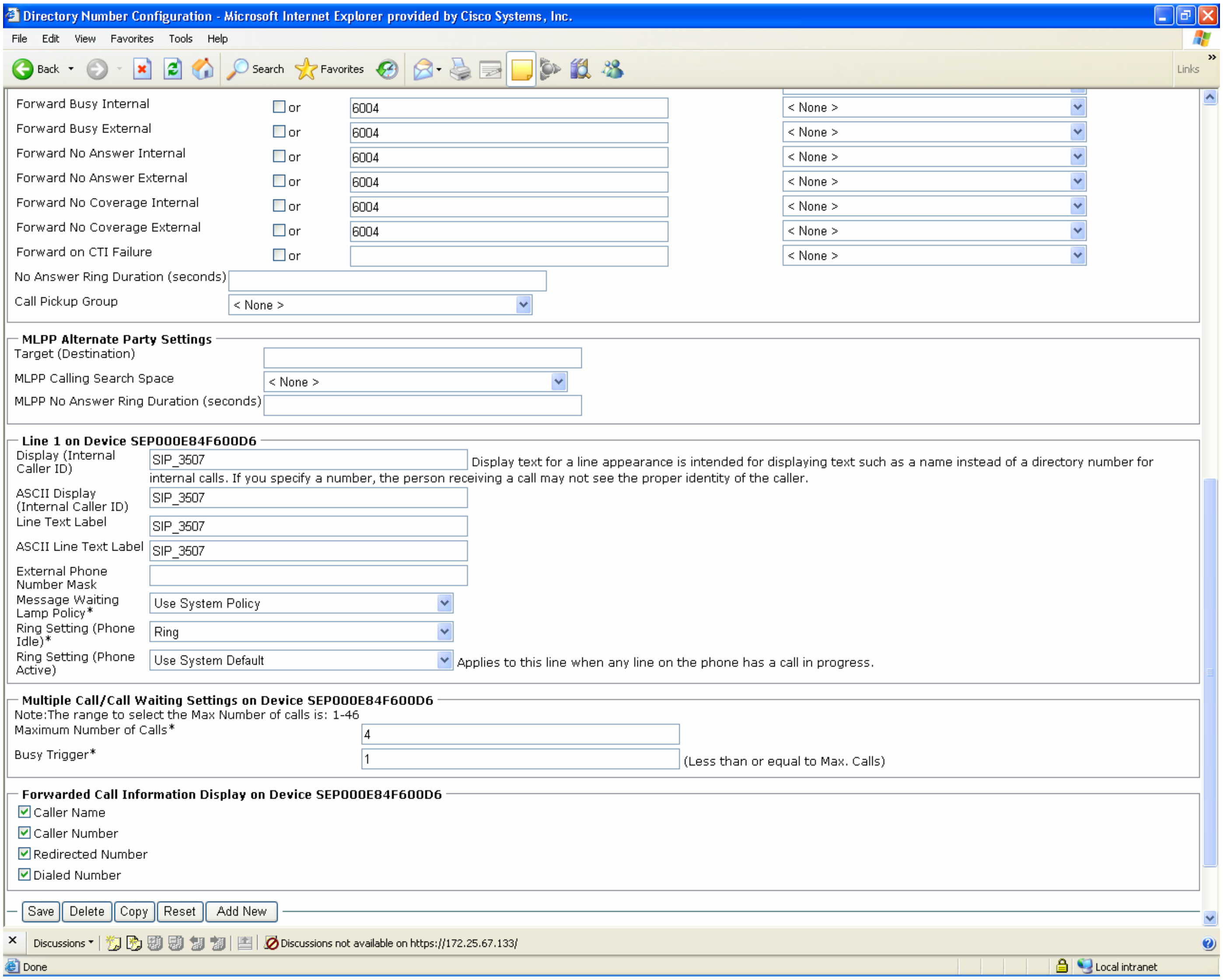



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites Links
Protocol Specific Information
Packet Capture Mode* None
Packet Capture Duration 0
Presence Group* Standard Presence group
SIP Dial Rules < None >
MTP Preferred Originating Codec* 711ulaw
Device Security Profile* Cisco 7970 - 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 2006 11 23 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.
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 -- Not Selected --
Login in User ID < None >
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

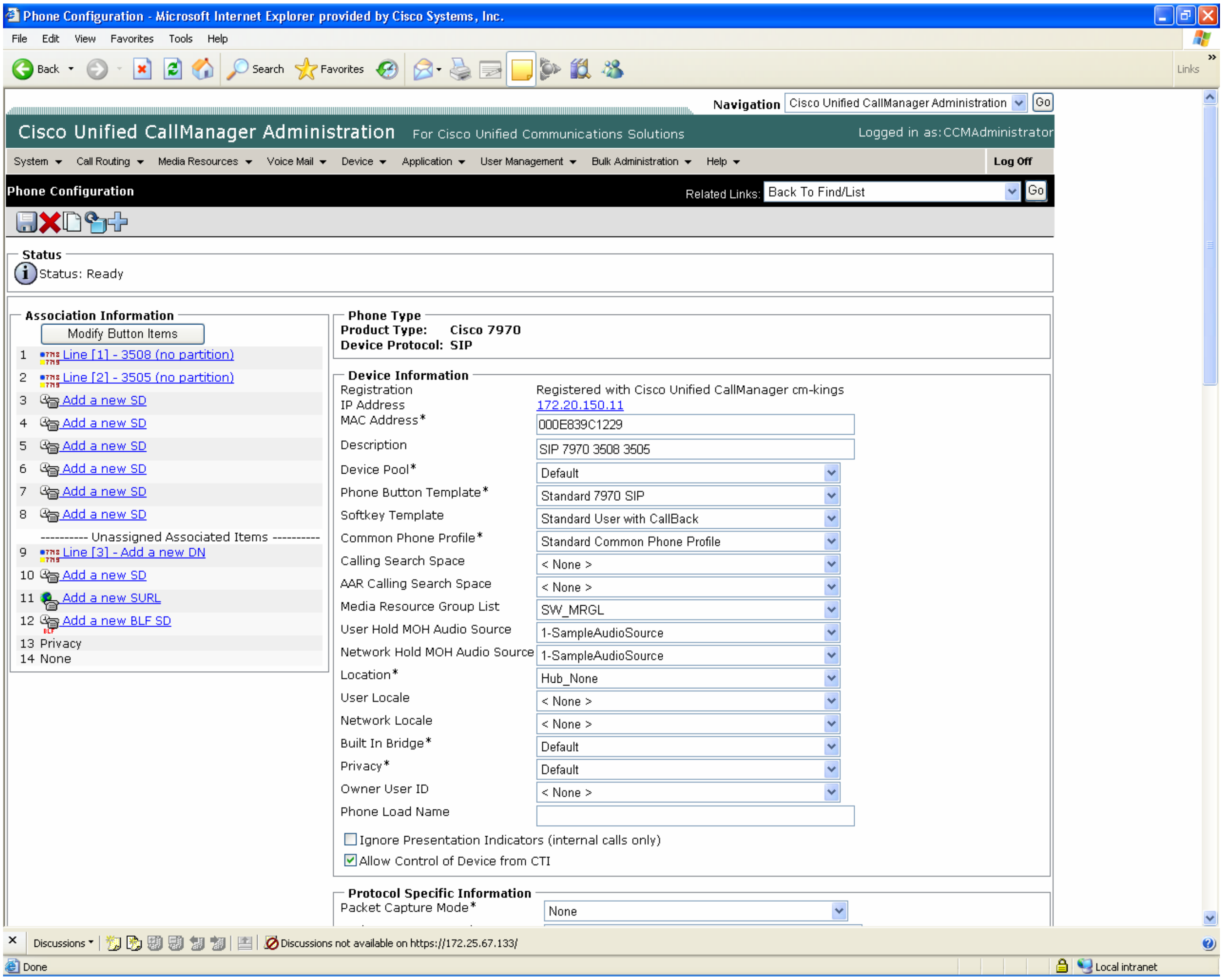

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Enable Extension Mobility
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain 000000
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration Layout
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Enabled
PC Voice VLAN Access* Enabled
Video Capabilities* Disabled
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* Disabled
Logging Display* PC Controlled
Load Server
Save Delete Copy Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
***Note: Security Profile Contains Addition CAPF Settings.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

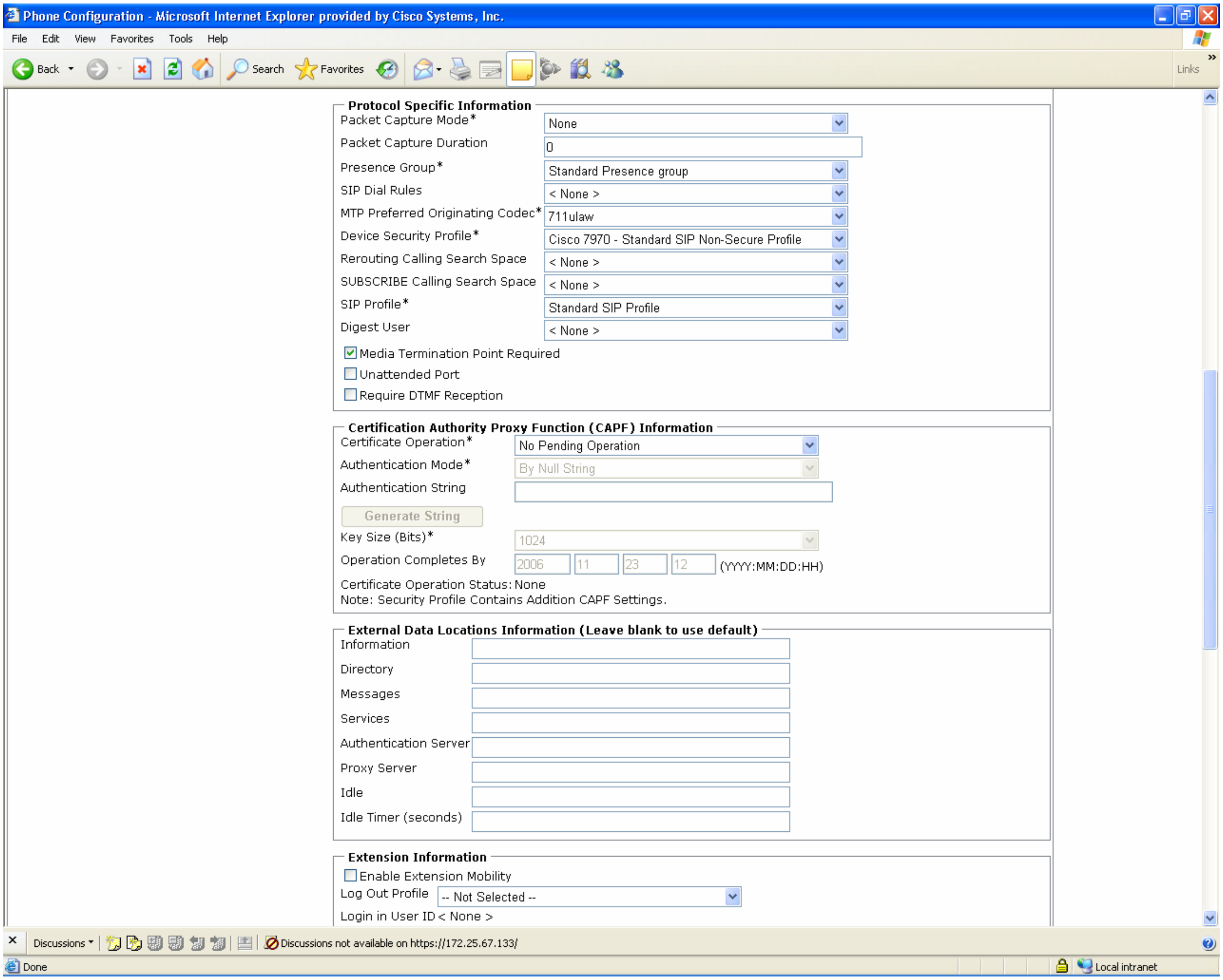
SIP Phone Ext. 3507 Directory Number Level Configuration

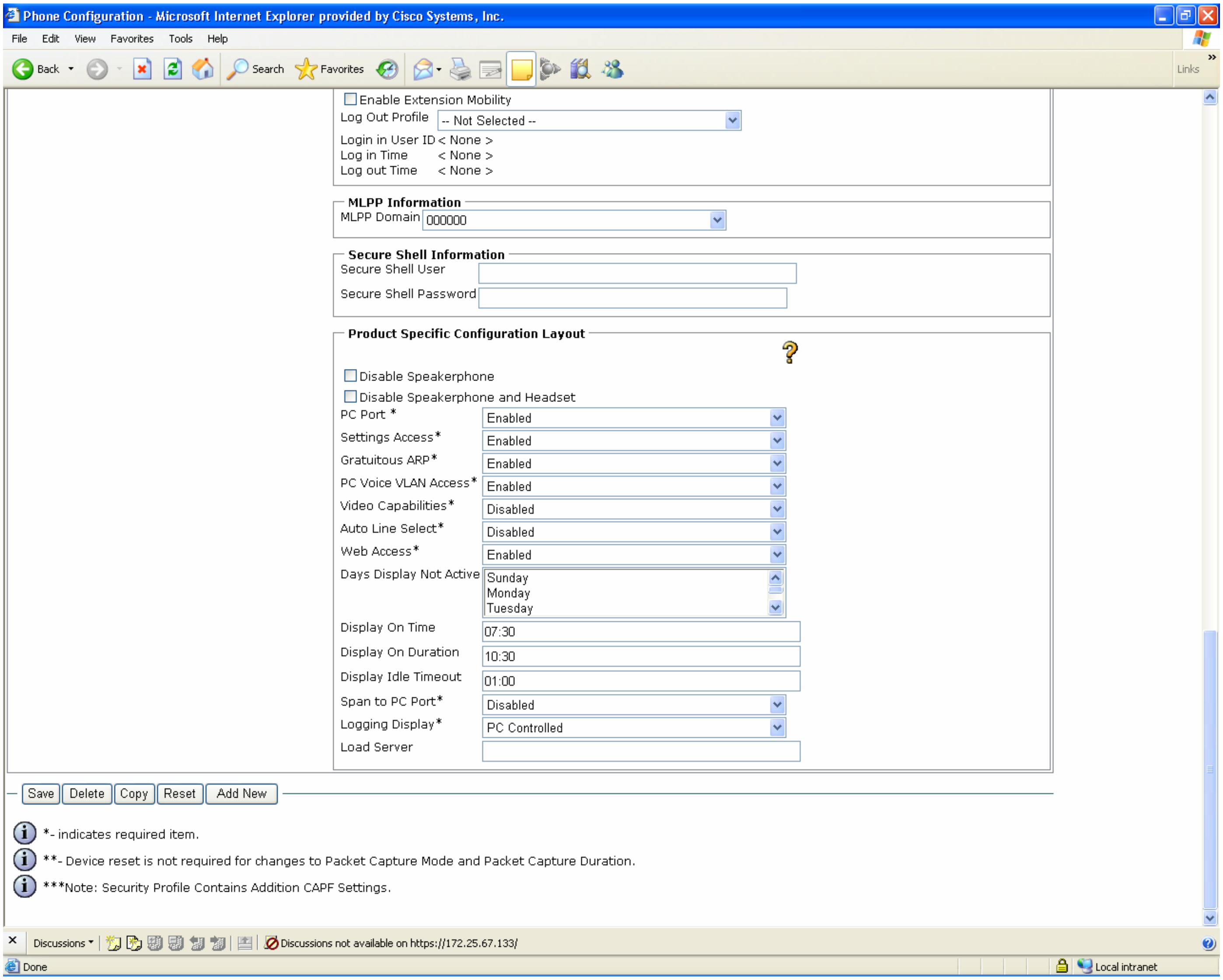



Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
or
6004
< None >
No Answer Ring Duration (seconds)
Call Pickup Group
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000E84F600D6
Display (Internal Caller ID)
SIP_3507
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
Line Text Label
ASCII Line Text Label
External Phone Number Mask
Message Waiting Lamp Policy*
Use System Policy
Ring Setting (Phone Idle)*
Ring
Ring Setting (Phone Active)
Use System Default
Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP000E84F600D6
Note:The range to select the Max Number of calls is: 1-46
Maximum Number of Calls*
4
Busy Trigger*
1
(Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP000E84F600D6
Caller Name
Caller Number
Redirected Number
Dialed Number
Save
Delete
Copy
Reset
Add New
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

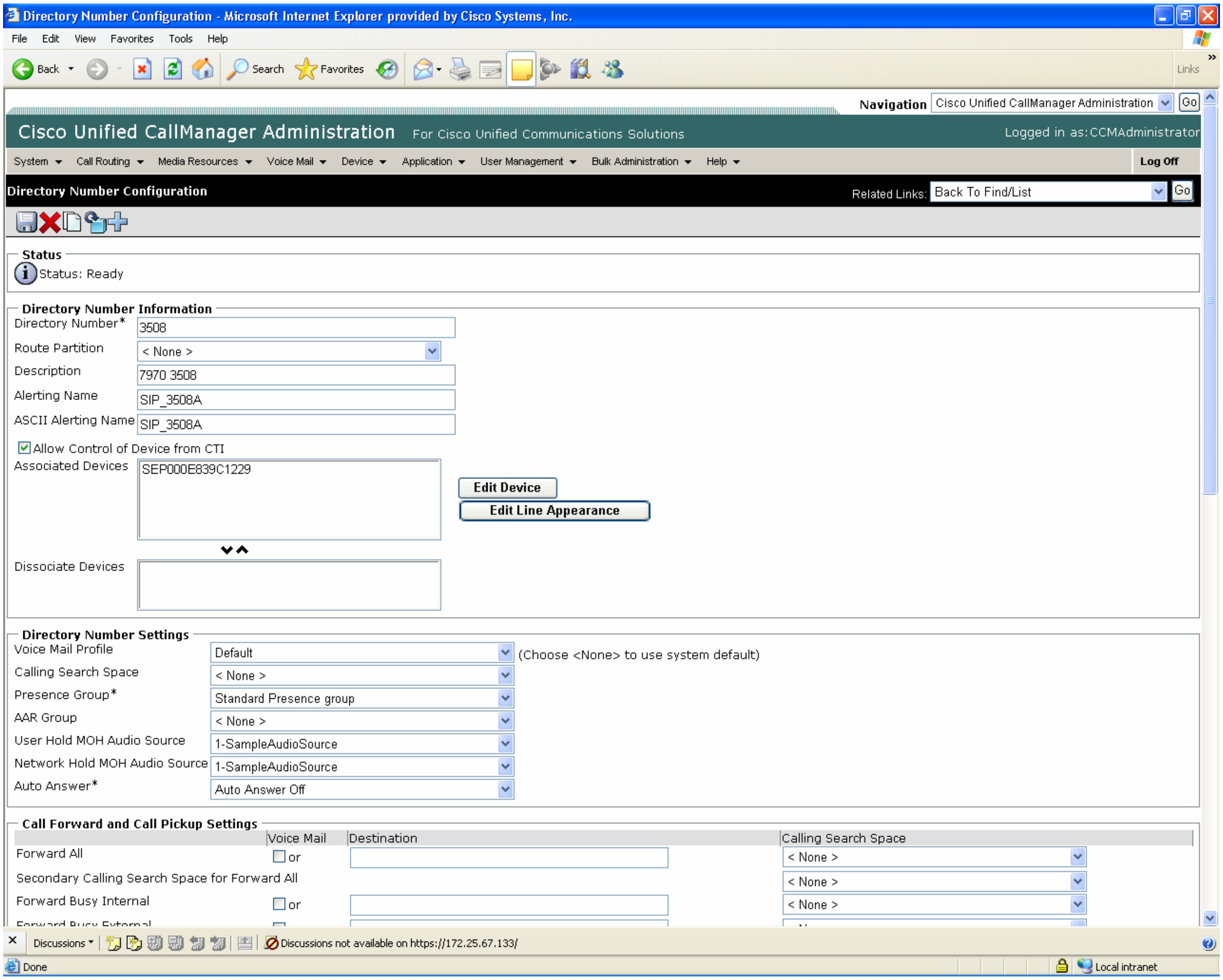
SIP Phone Ext. 3508 Device Level Configuration

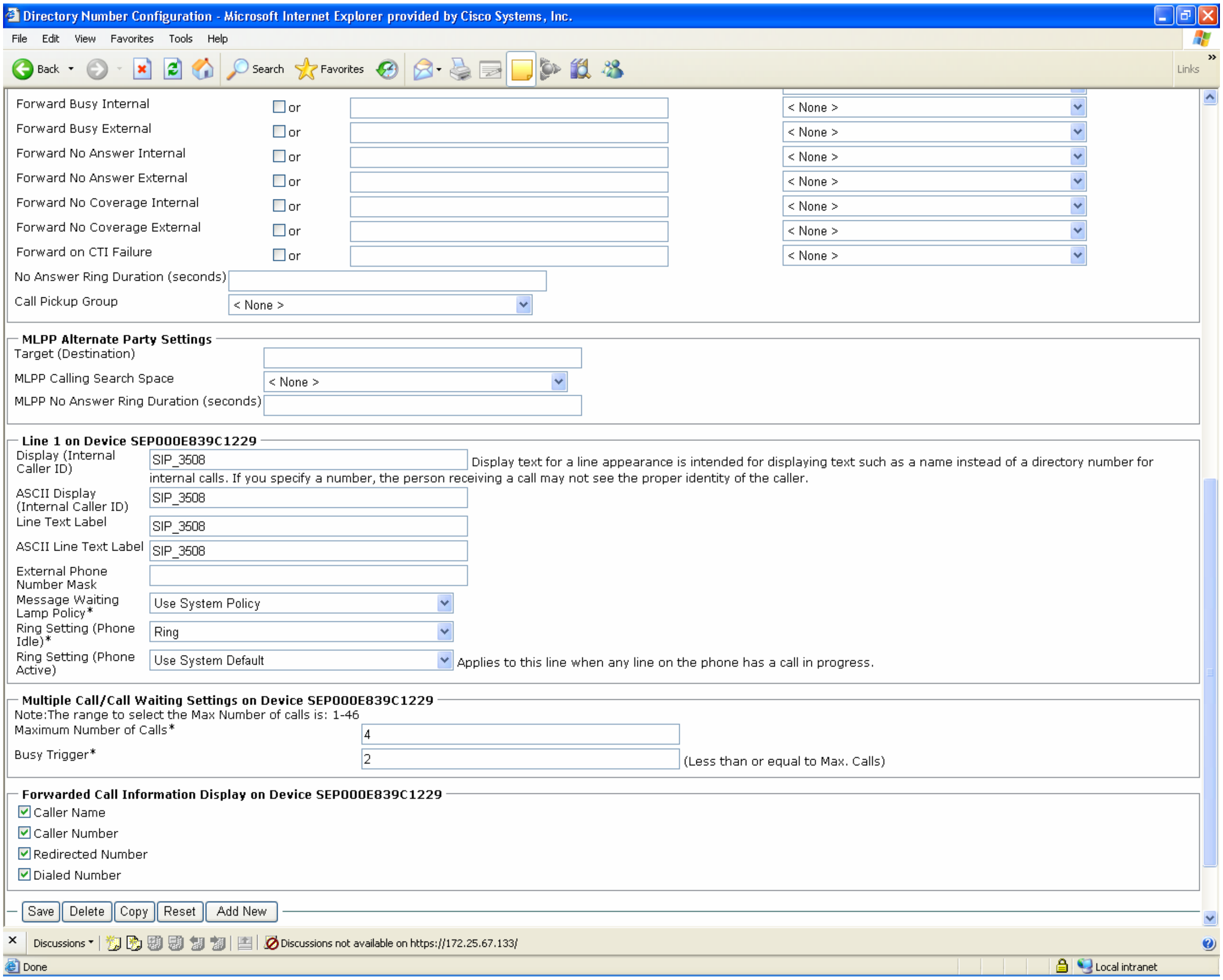



Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Protocol Specific Information
Packet Capture Mode*
None
Packet Capture Duration
0
Presence Group*
Standard Presence group
SIP Dial Rules
< None >
MTP Preferred Originating Codec*
711ulaw
Device Security Profile*
Cisco 7970 - 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
2006
11
23
12
(YYYY:MM:DD:HH)
Certificate Operation Status: None
Note: Security Profile Contains Addition CAPF Settings.
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
-- Not Selected --
Login in User ID < None >
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

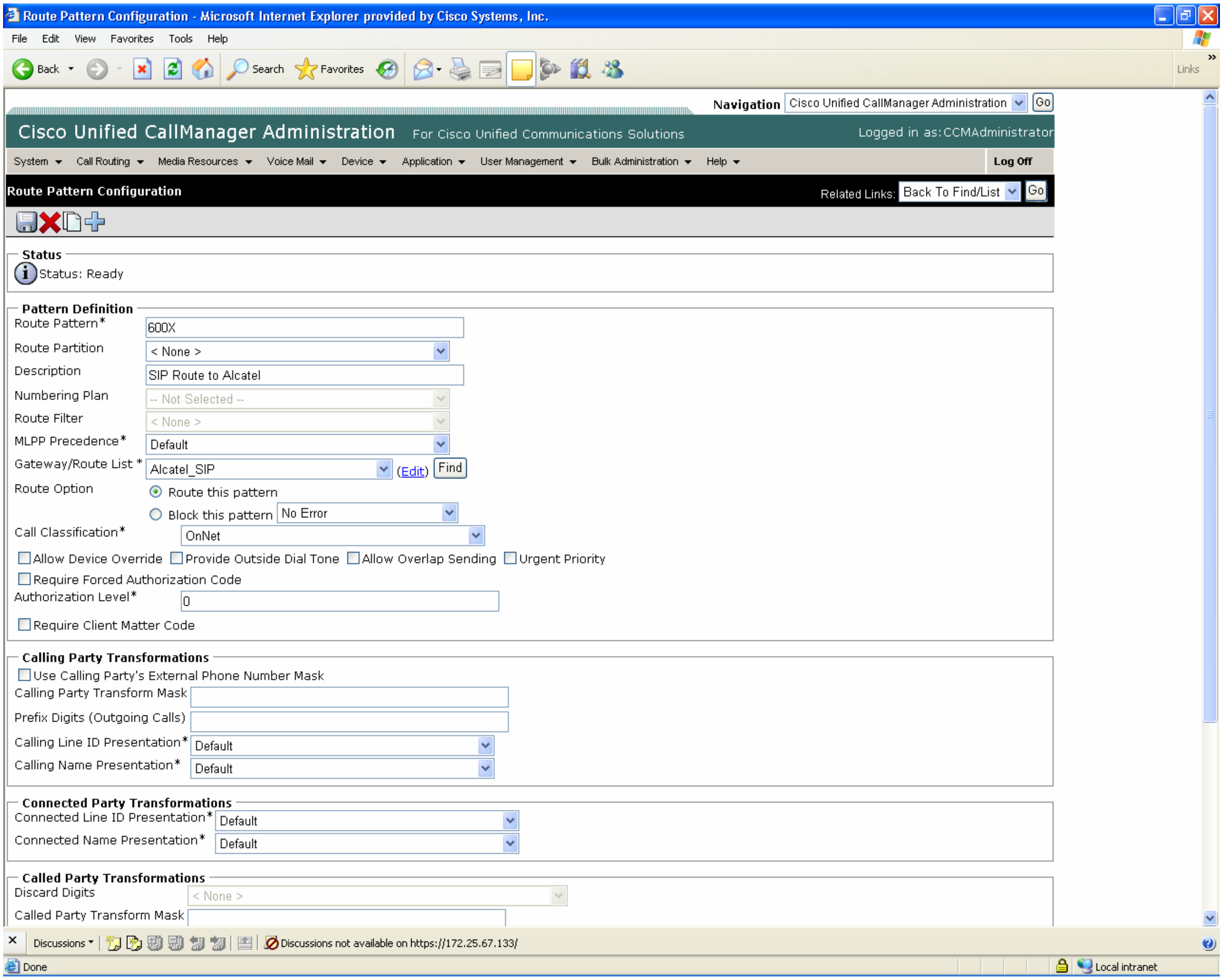

Phone Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Enable Extension Mobility
Log Out Profile -- Not Selected --
Login in User ID < None >
Log in Time < None >
Log out Time < None >
MLPP Information
MLPP Domain 000000
Secure Shell Information
Secure Shell User
Secure Shell Password
Product Specific Configuration Layout
Disable Speakerphone
Disable Speakerphone and Headset
PC Port * Enabled
Settings Access* Enabled
Gratuitous ARP* Enabled
PC Voice VLAN Access* Enabled
Video Capabilities* Disabled
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* Disabled
Logging Display* PC Controlled
Load Server
Save Delete Copy Reset Add New
*- indicates required item.
**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
***Note: Security Profile Contains Addition CAPF Settings.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

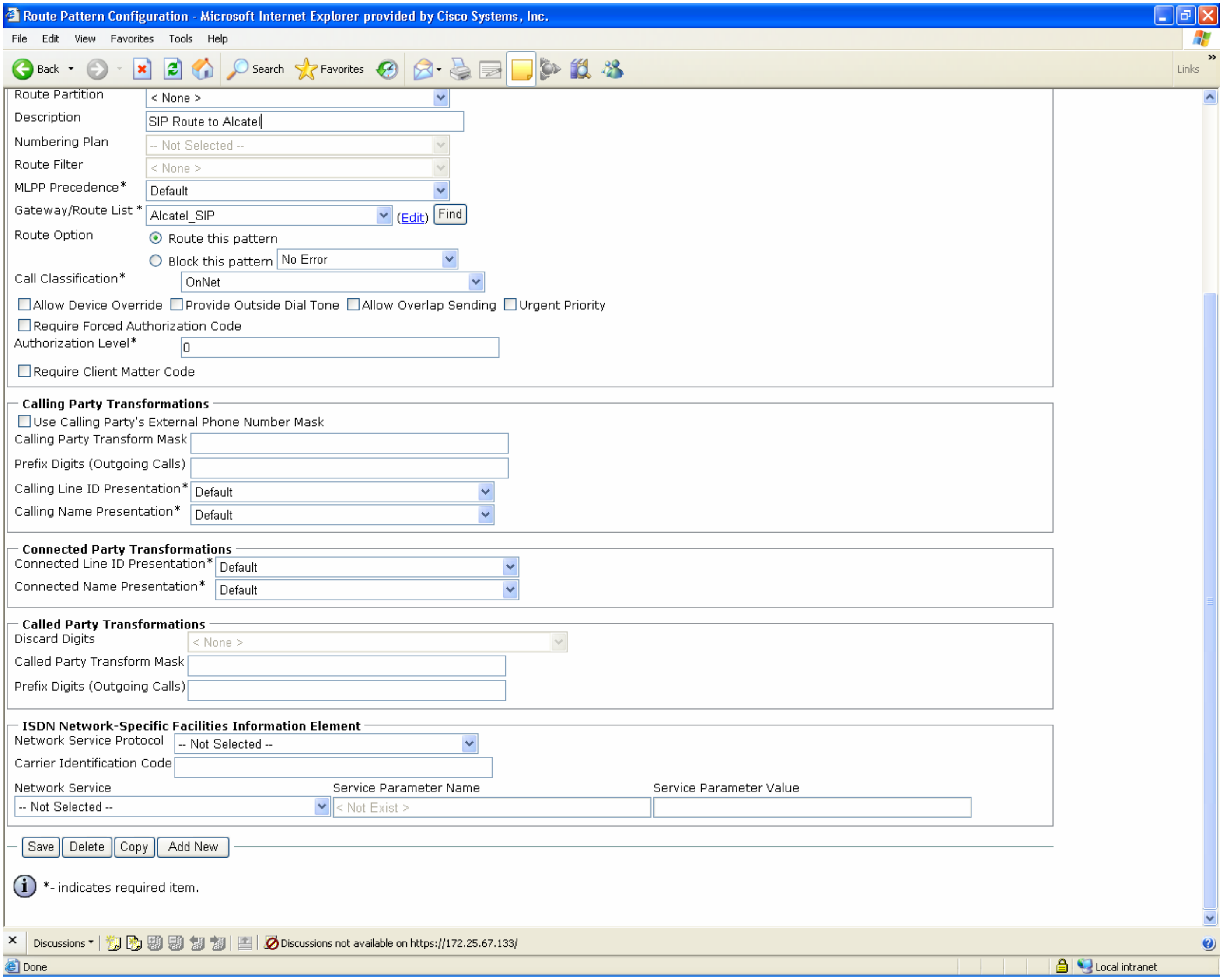
SIP Phone Ext. 3508 Directory Number Level Configuration



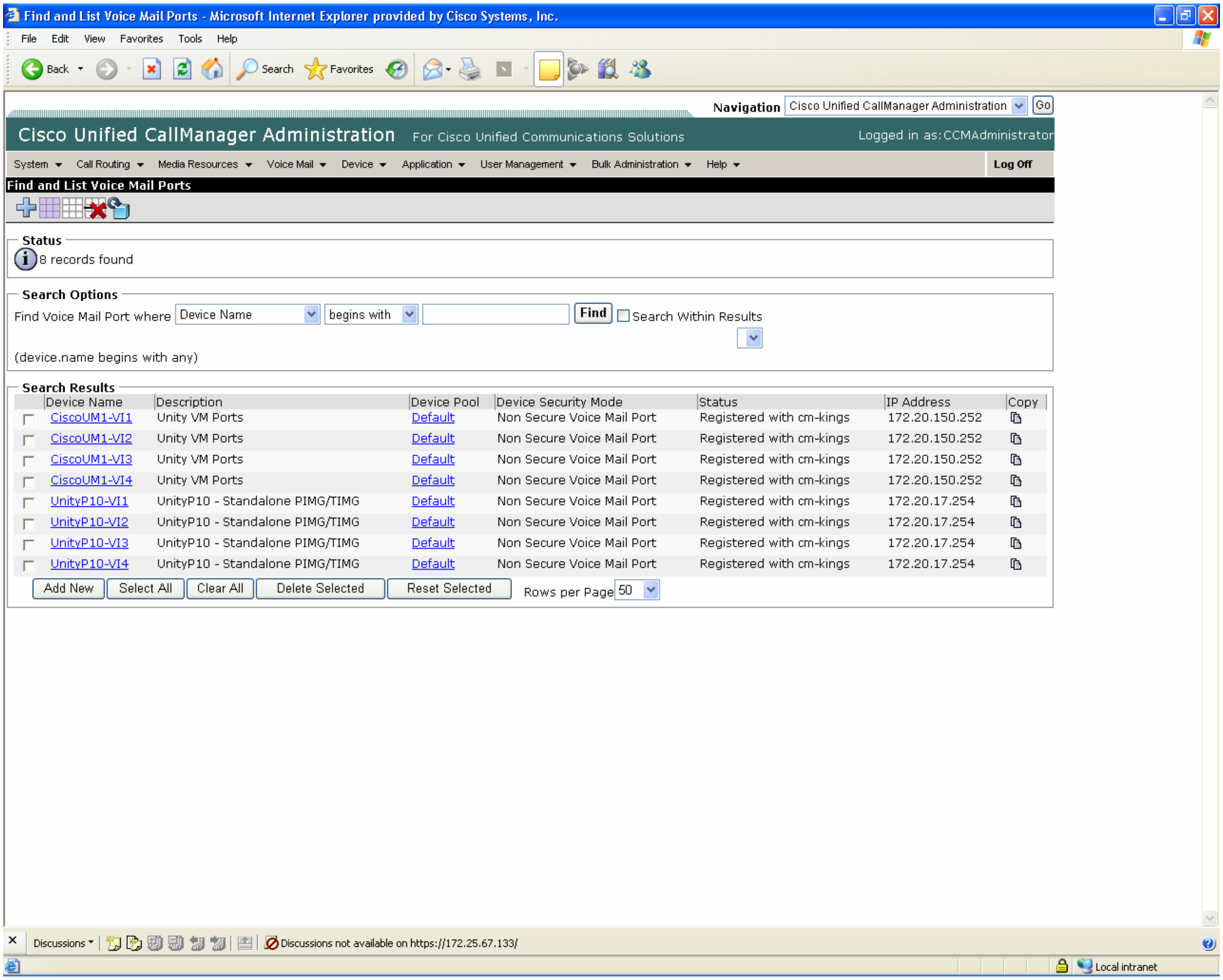

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Forward Busy Internal or < None >
Forward Busy External or < None >
Forward No Answer Internal or < None >
Forward No Answer External or < None >
Forward No Coverage Internal or < None >
Forward No Coverage External or < None >
Forward on CTI Failure or < None >
No Answer Ring Duration (seconds)
Call Pickup Group < None >
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)
Line 1 on Device SEP000E839C1229
Display (Internal Caller ID) SIP_3508 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID) SIP_3508
Line Text Label SIP_3508
ASCII Line Text Label SIP_3508
External Phone Number Mask
Message Waiting Lamp Policy* Use System Policy
Ring Setting (Phone Idle)* Ring
Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.
Multiple Call/Call Waiting Settings on Device SEP000E839C1229
Note:The range to select the Max Number of calls is: 1-46
Maximum Number of Calls* 4
Busy Trigger* 2 (Less than or equal to Max. Calls)
Forwarded Call Information Display on Device SEP000E839C1229
Caller Name
Caller Number
Redirected Number
Dialed Number
Save Delete Copy Reset Add New
Discussions Discussions not available on https://172.25.67.133/
Done Local intranet

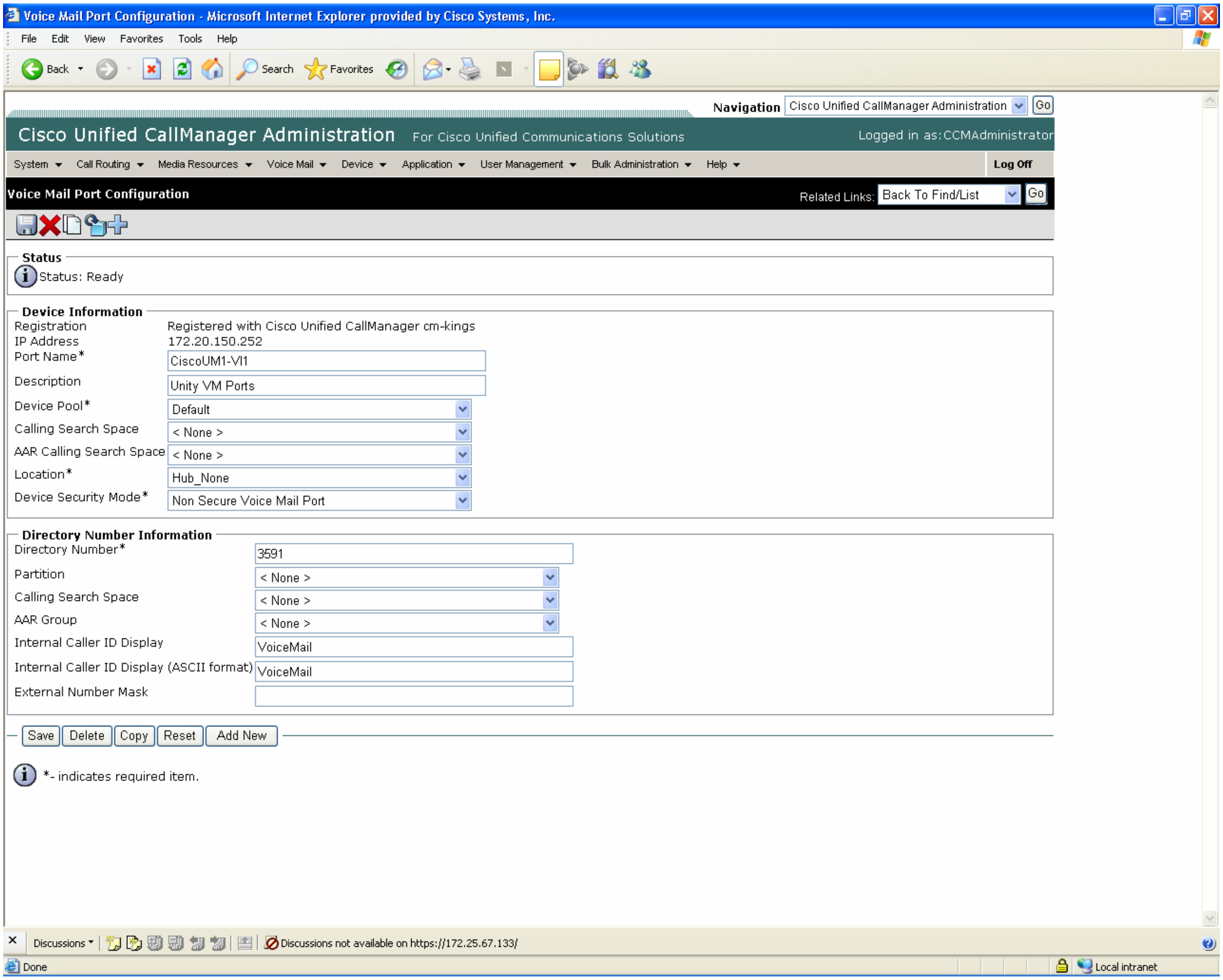
Route Pattern to Alcatel PBX digital phone extensions Configuration



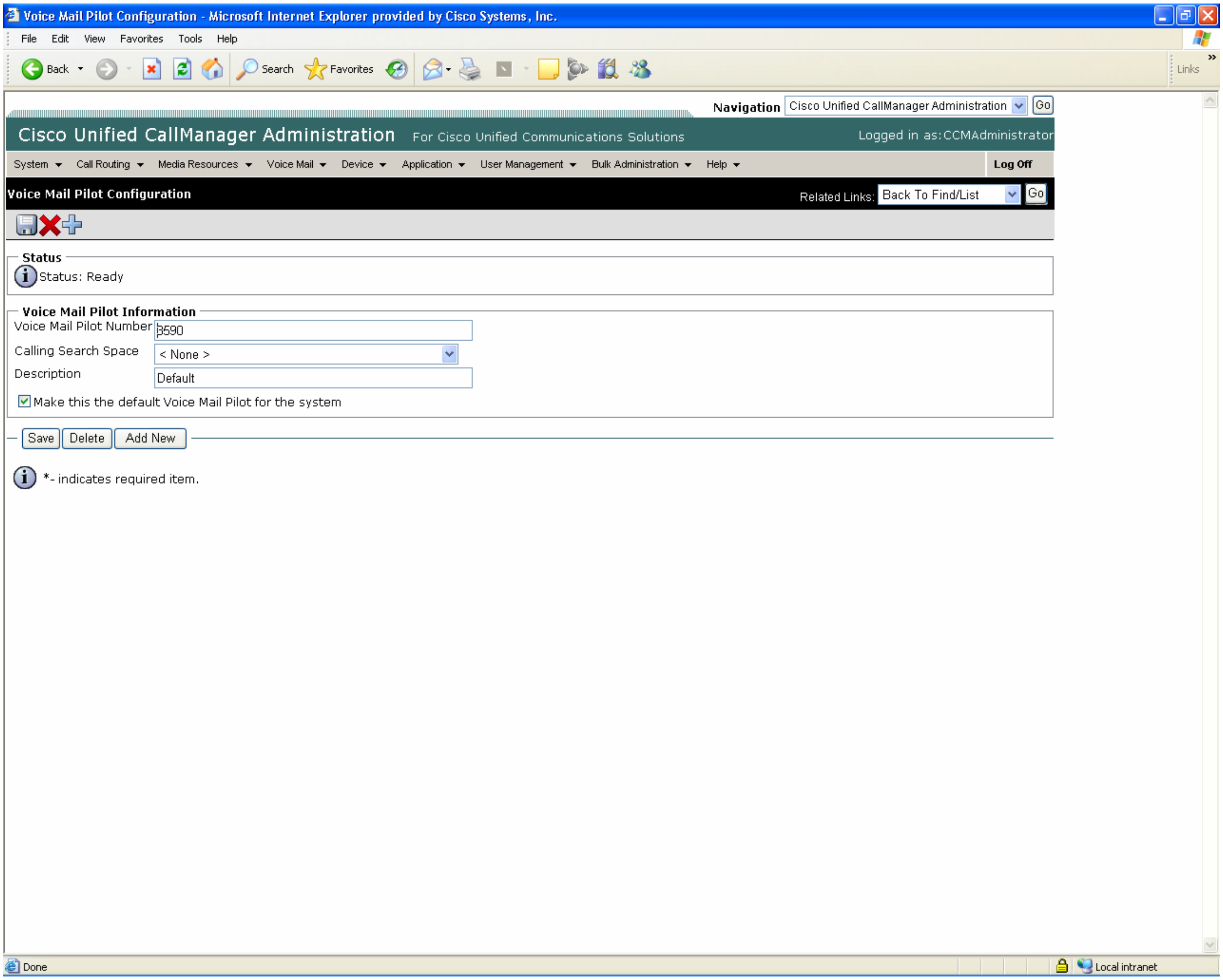

Route Pattern Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Links
Route Partition
< None >
Description
SIP Route to Alcatel
Numbering Plan
-- Not Selected --
Route Filter
< None >
MLPP Precedence*
Default
Gateway/Route List*
Alcatel_SIP
(Edit)
Find
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
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)
ISDN Network-Specific Facilities Information Element
Network Service Protocol
-- Not Selected --
Carrier Identification Code
Network Service
Service Parameter Name
Service Parameter Value
-- Not Selected --
< Not Exist >
Save
Delete
Copy
Add New
*- indicates required item.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

Voice Mail Ports to Unity Voice Mail system

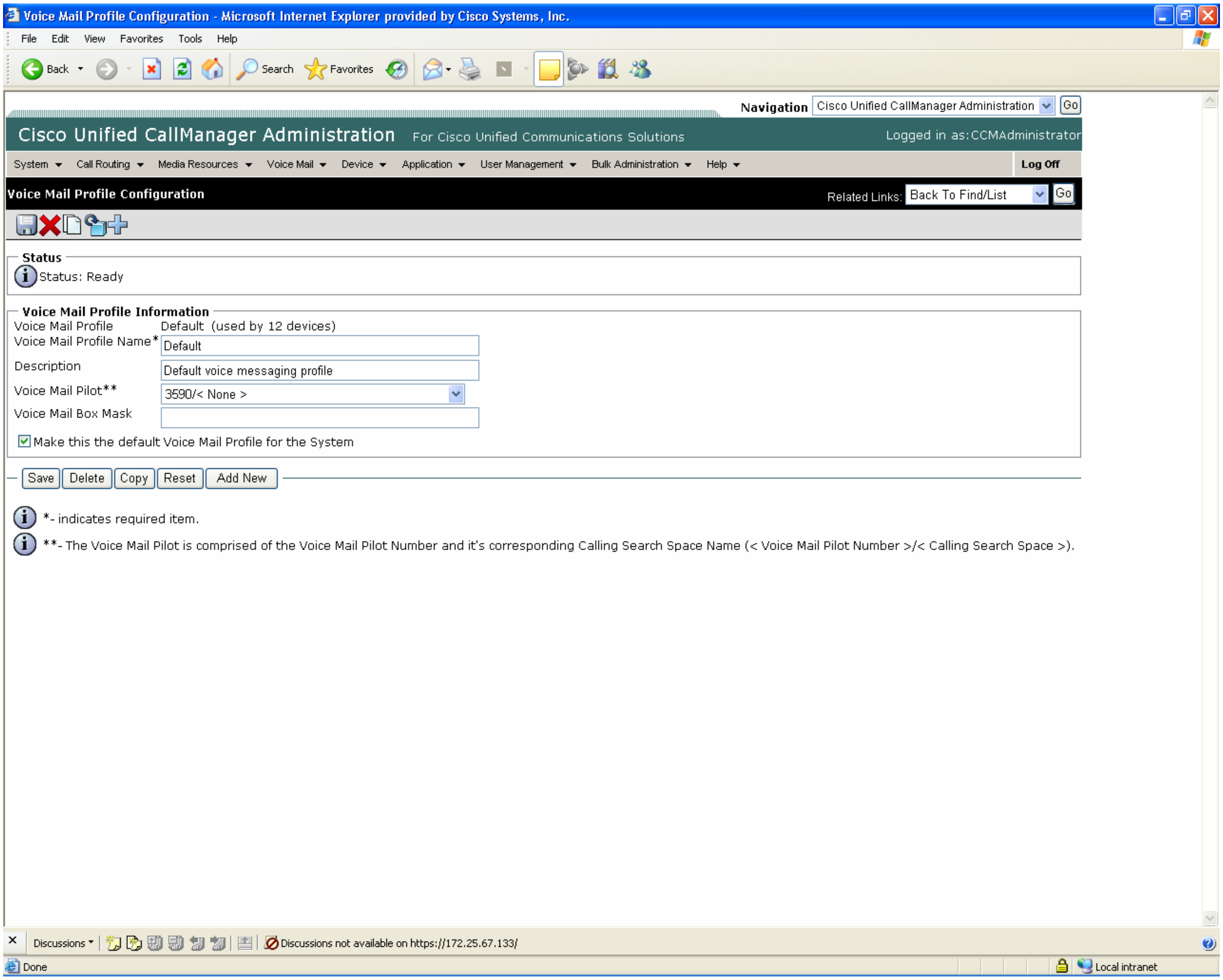



Voice Mail Port Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
Navigation Cisco Unified CallManager Administration Go
Cisco Unified CallManager Administration For Cisco Unified Communications Solutions
Logged in as:CCMAdministrator
System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help
Log Off
Voice Mail Port Configuration
Related Links: Back To Find/List Go
Status
Status: Ready
Device Information
Registration Registered with Cisco Unified CallManager cm-kings
IP Address 172.20.150.252
Port Name* CiscoUM1-VI1
Description Unity VM Ports
Device Pool* Default
Calling Search Space < None >
AAR Calling Search Space < None >
Location* Hub_None
Device Security Mode* Non Secure Voice Mail Port
Directory Number Information
Directory Number* 3591
Partition < None >
Calling Search Space < None >
AAR Group < None >
Internal Caller ID Display VoiceMail
Internal Caller ID Display (ASCII format) VoiceMail
External Number Mask
Save Delete Copy Reset Add New
*- indicates required item.
Discussions not available on https://172.25.67.133/
Done
Local intranet

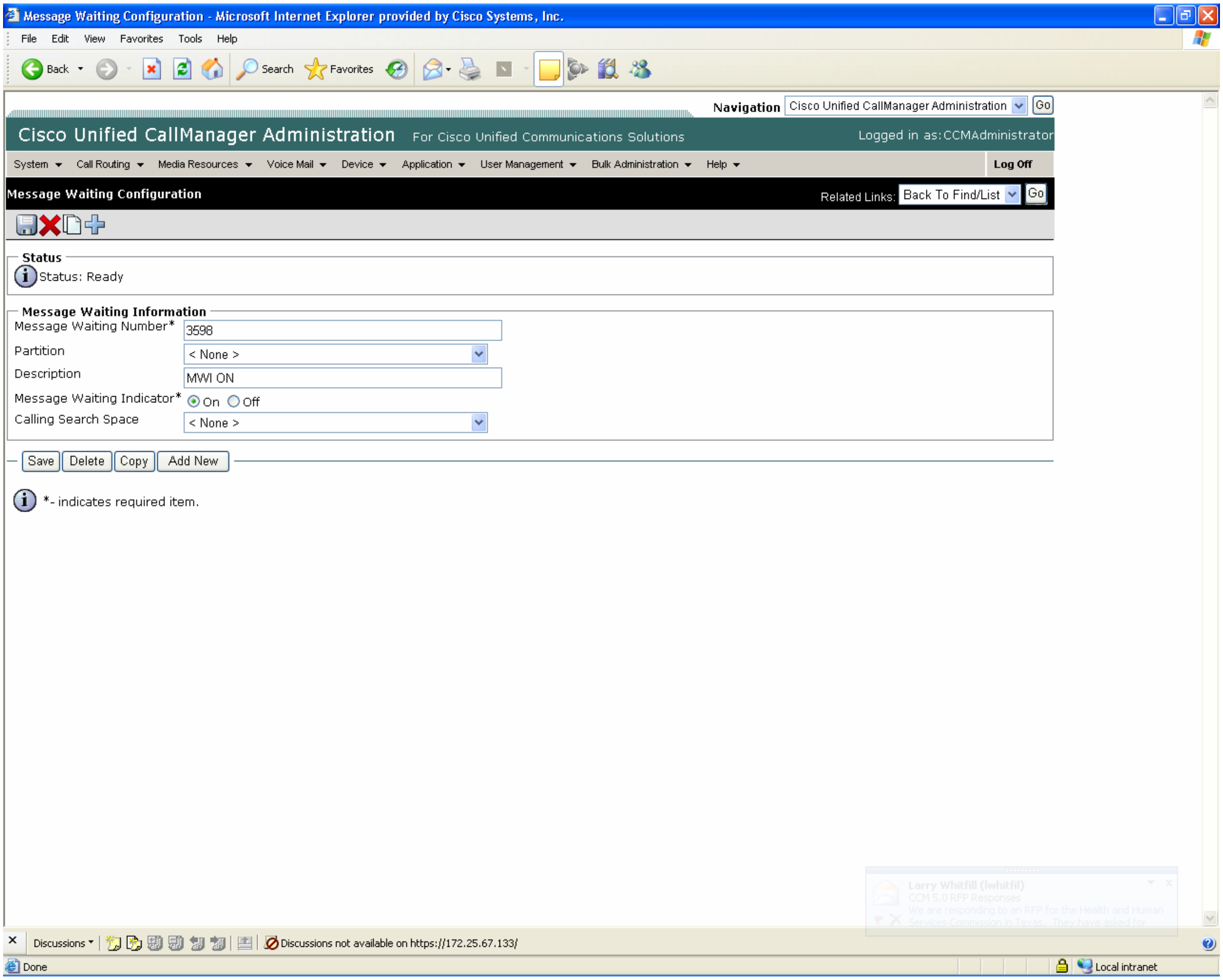
Voice Mail Pilot for Unity Voice Mail system

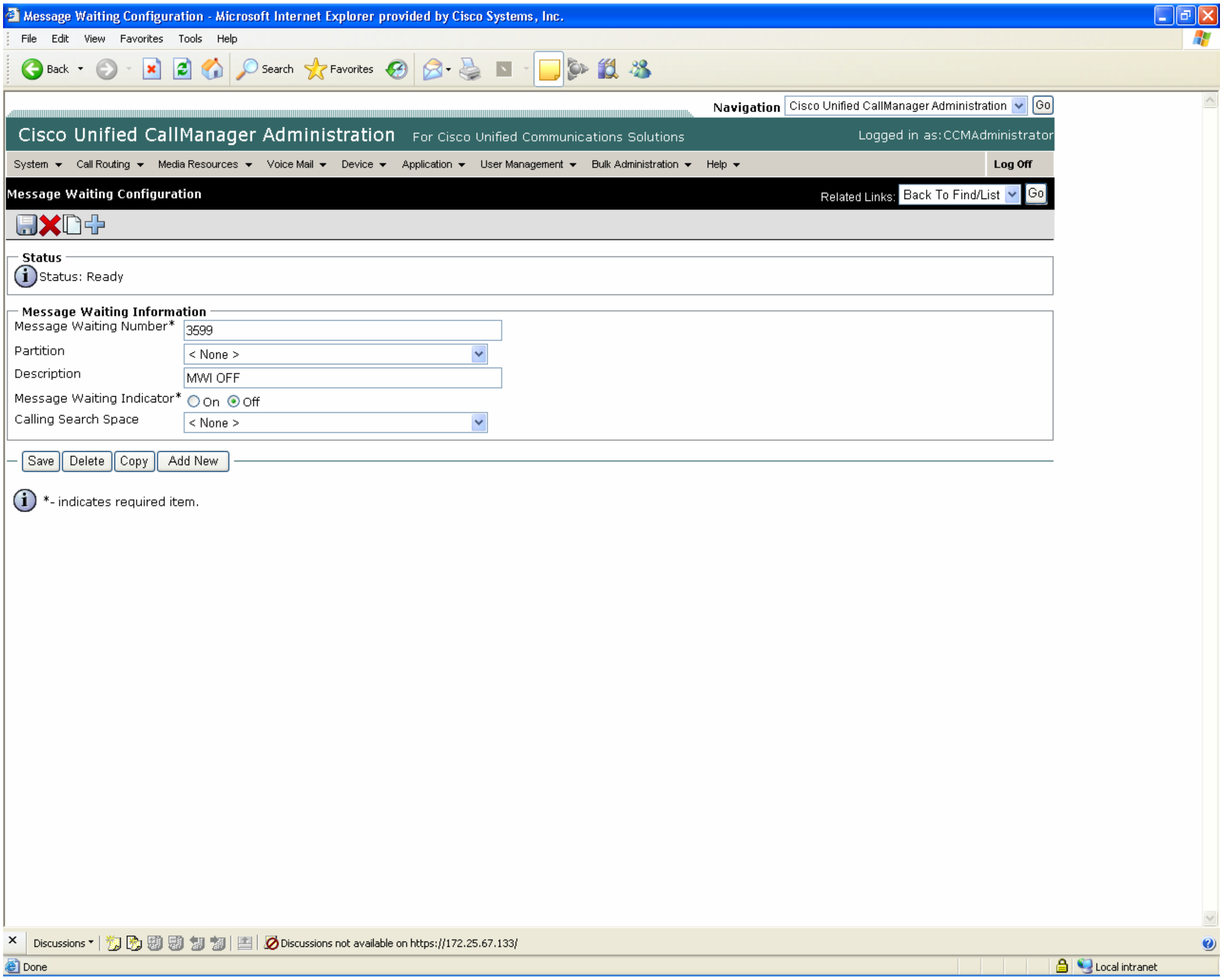


Voice Mail Profile for Unity Voice Mail system

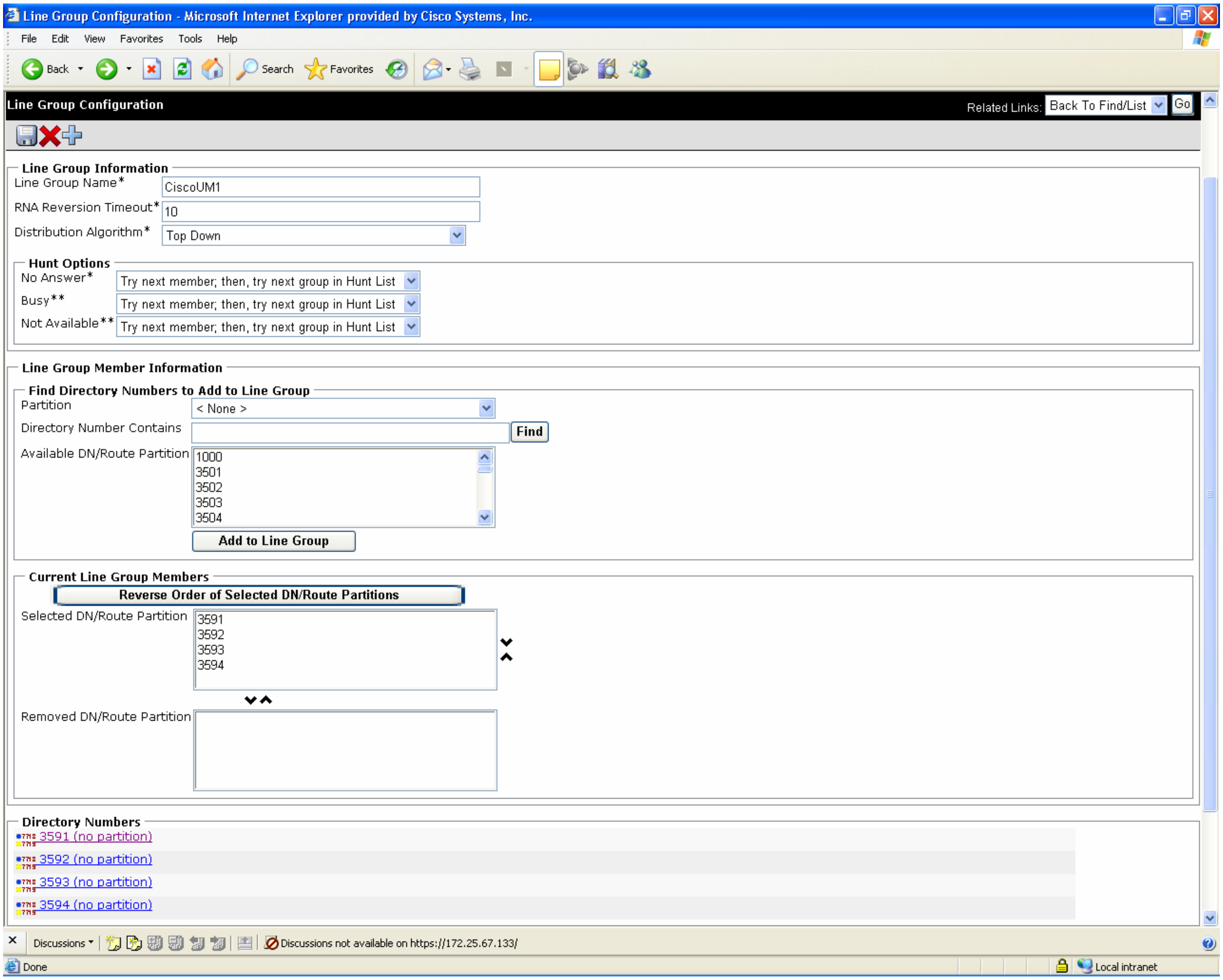


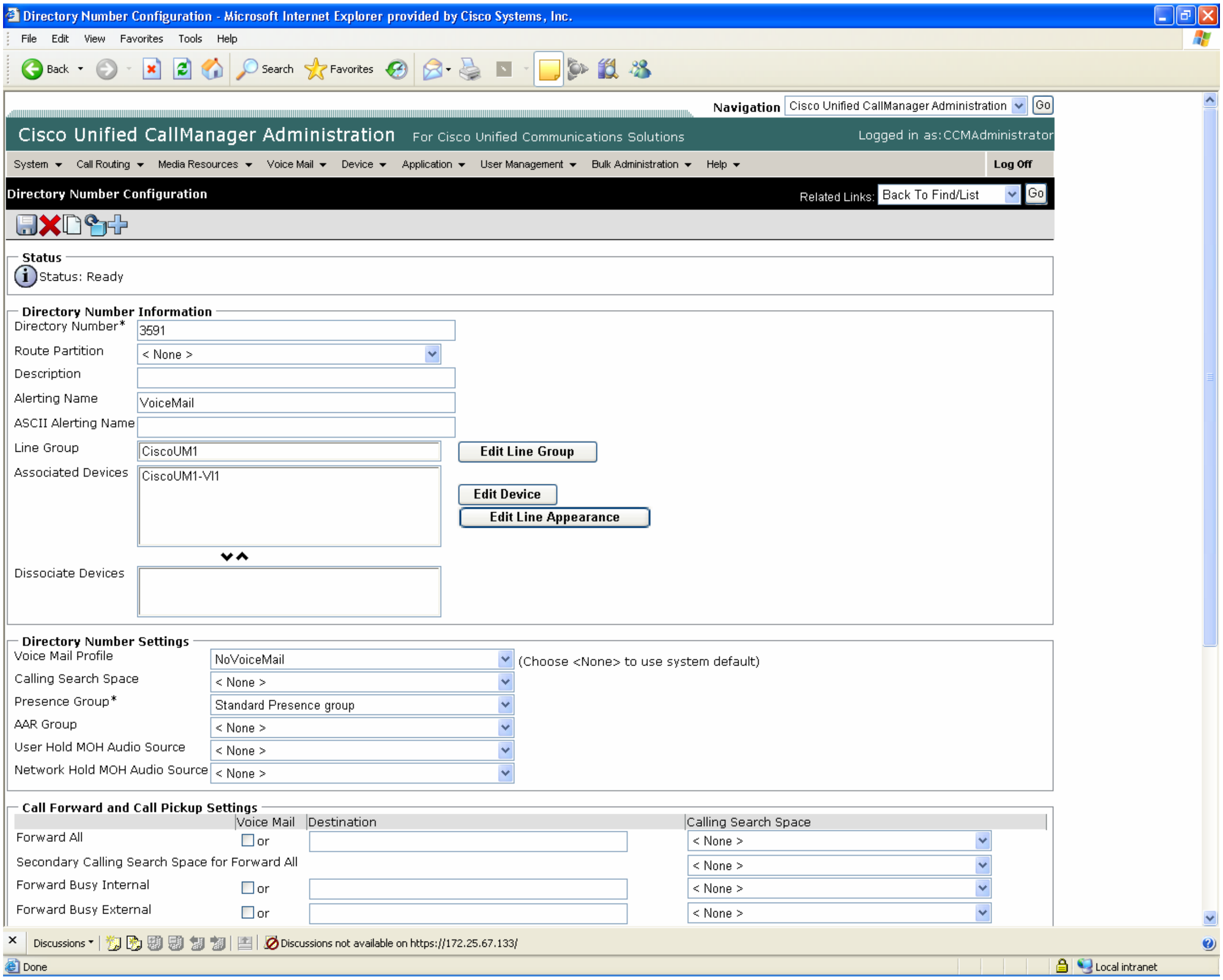
Voice Mail MWI ON and OFF for Unity Voice Mail system

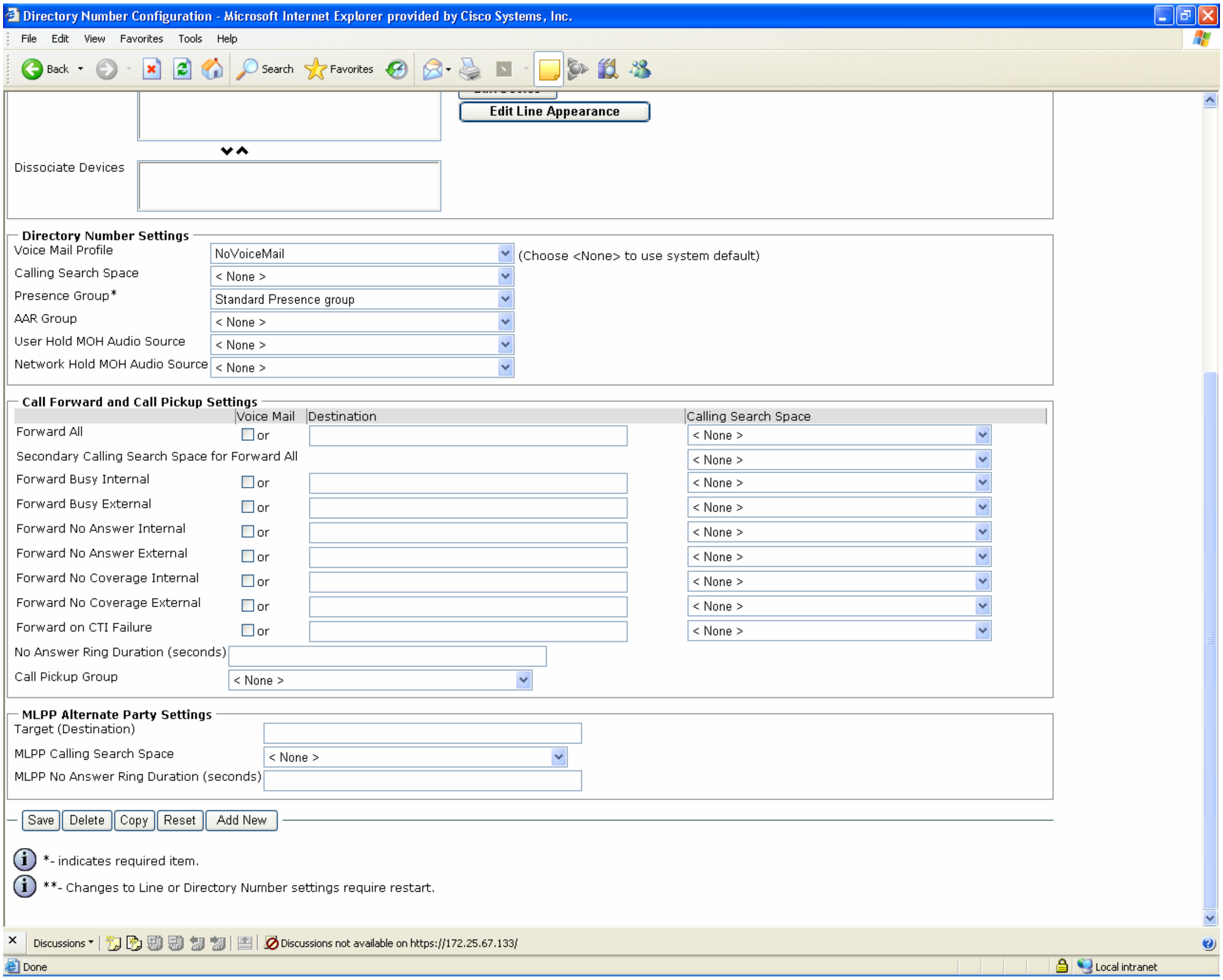



Message Waiting Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Navigation Cisco Unified CallManager Administration Go
Cisco Unified CallManager Administration For Cisco Unified Communications Solutions
Logged in as:CCMAdministrator
System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help
Log Off
Message Waiting Configuration
Related Links: Back To Find/List Go
Status
Status: Ready
Message Waiting Information
Message Waiting Number* 3599
Partition < None >
Description MWI OFF
Message Waiting Indicator* On Off
Calling Search Space < None >
Save Delete Copy Add New
*- indicates required item.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

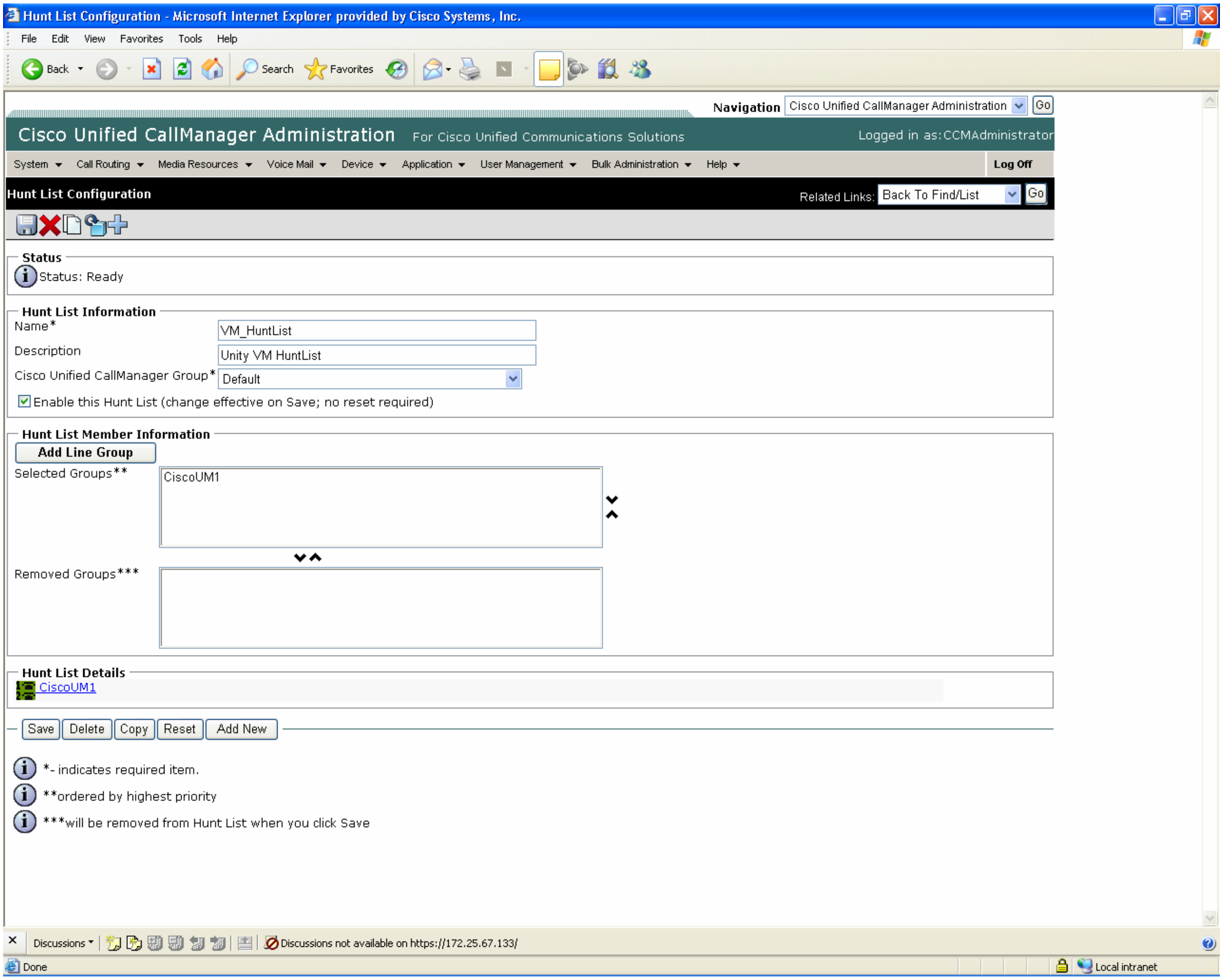
Voice Mail Line Group



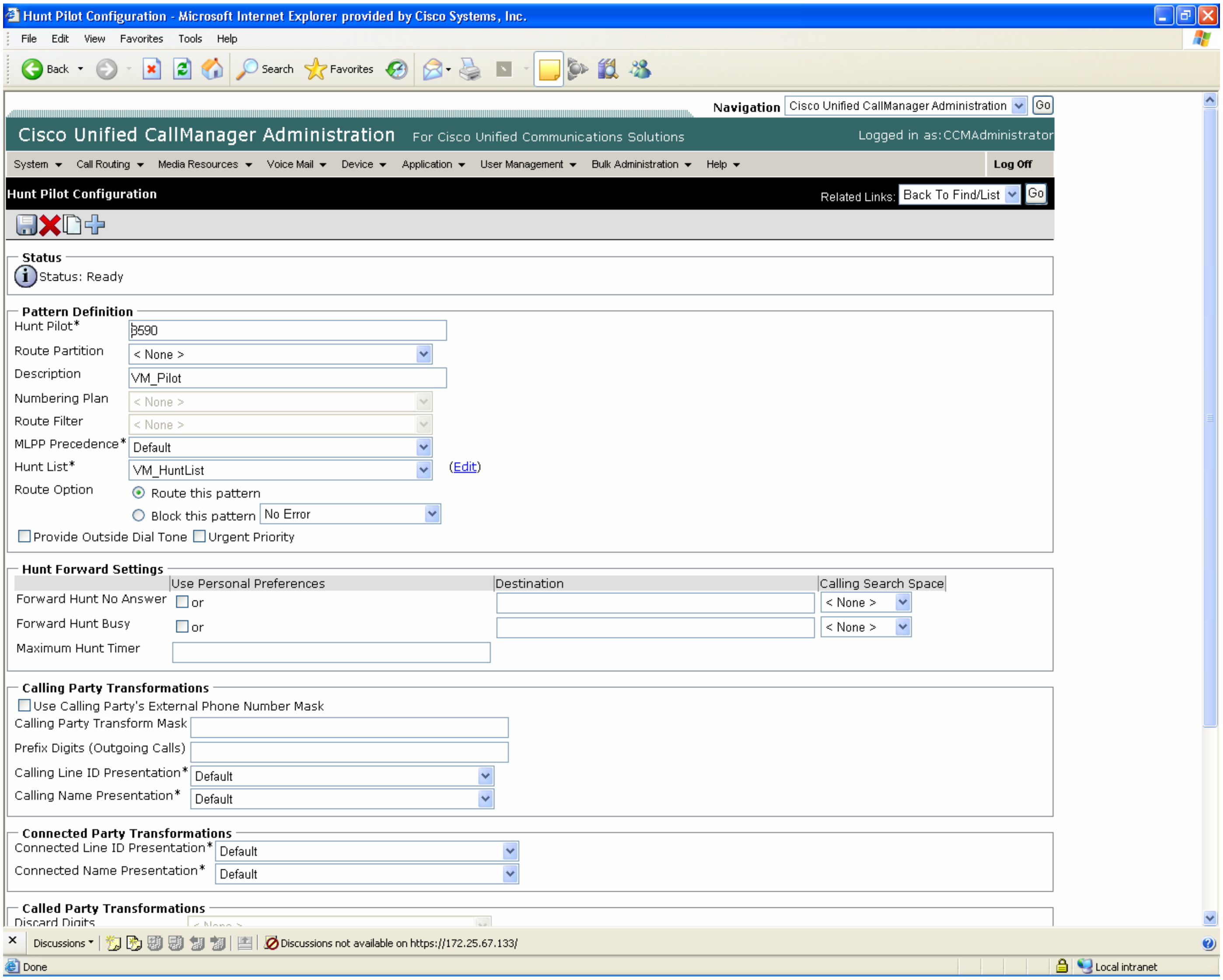


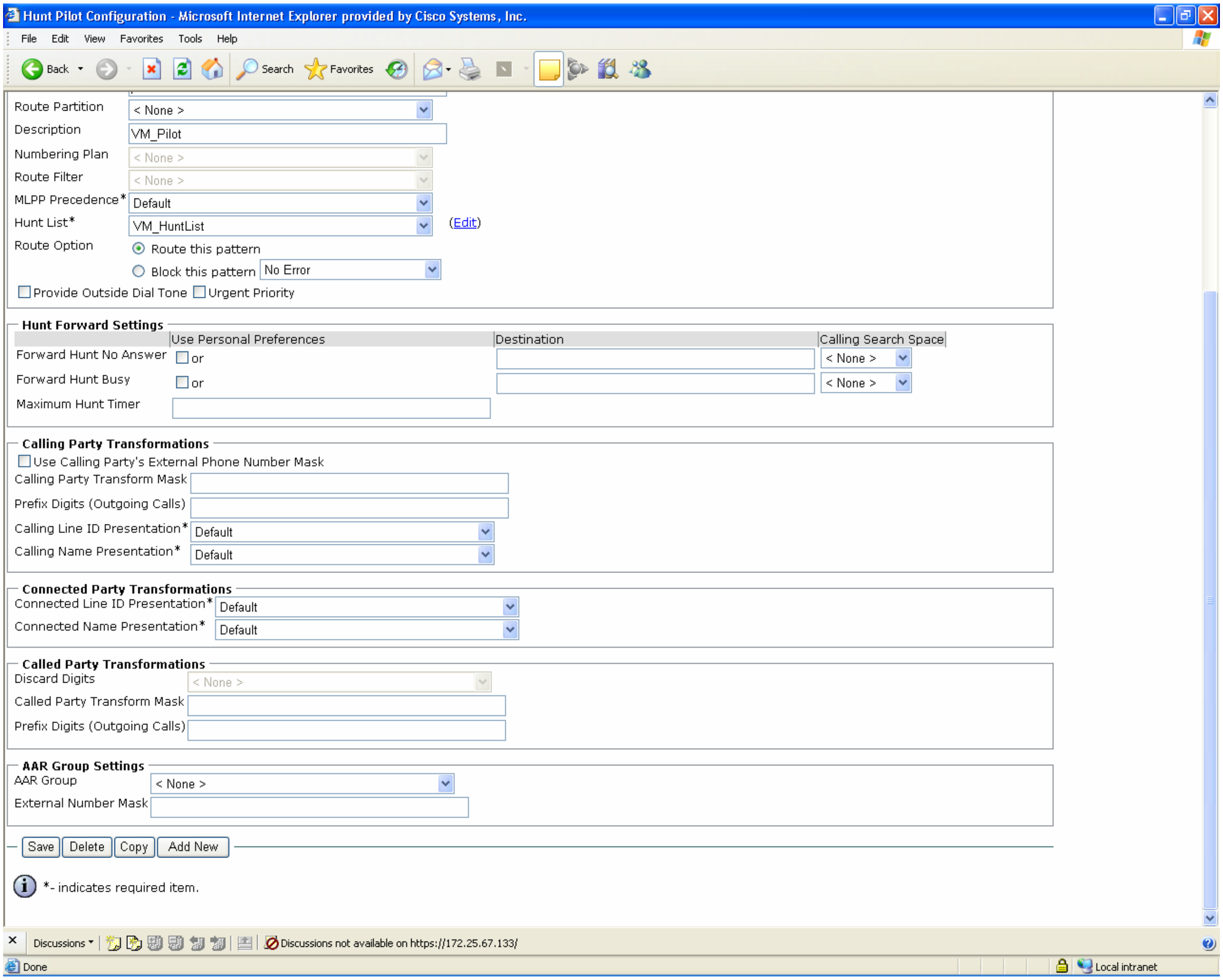

Directory Number Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back Search Favorites
Edit Line Appearance
Dissociate Devices
Directory Number Settings
Voice Mail Profile
NoVoiceMail
(Choose <None> to use system default)
Calling Search Space
< None >
Presence Group*
Standard Presence group
AAR Group
User Hold MOH Audio Source
Network Hold MOH Audio Source
Call Forward and Call Pickup Settings
Voice Mail
Destination
Calling Search Space
Forward All
Secondary Calling Search Space for Forward All
Forward Busy Internal
Forward Busy External
Forward No Answer Internal
Forward No Answer External
Forward No Coverage Internal
Forward No Coverage External
Forward on CTI Failure
or
No Answer Ring Duration (seconds)
Call Pickup Group
MLPP Alternate Party Settings
Target (Destination)
MLPP Calling Search Space
MLPP No Answer Ring Duration (seconds)
Save
Delete
Copy
Reset
Add New
*- indicates required item.
**- Changes to Line or Directory Number settings require restart.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

Voice Mail Hunt List



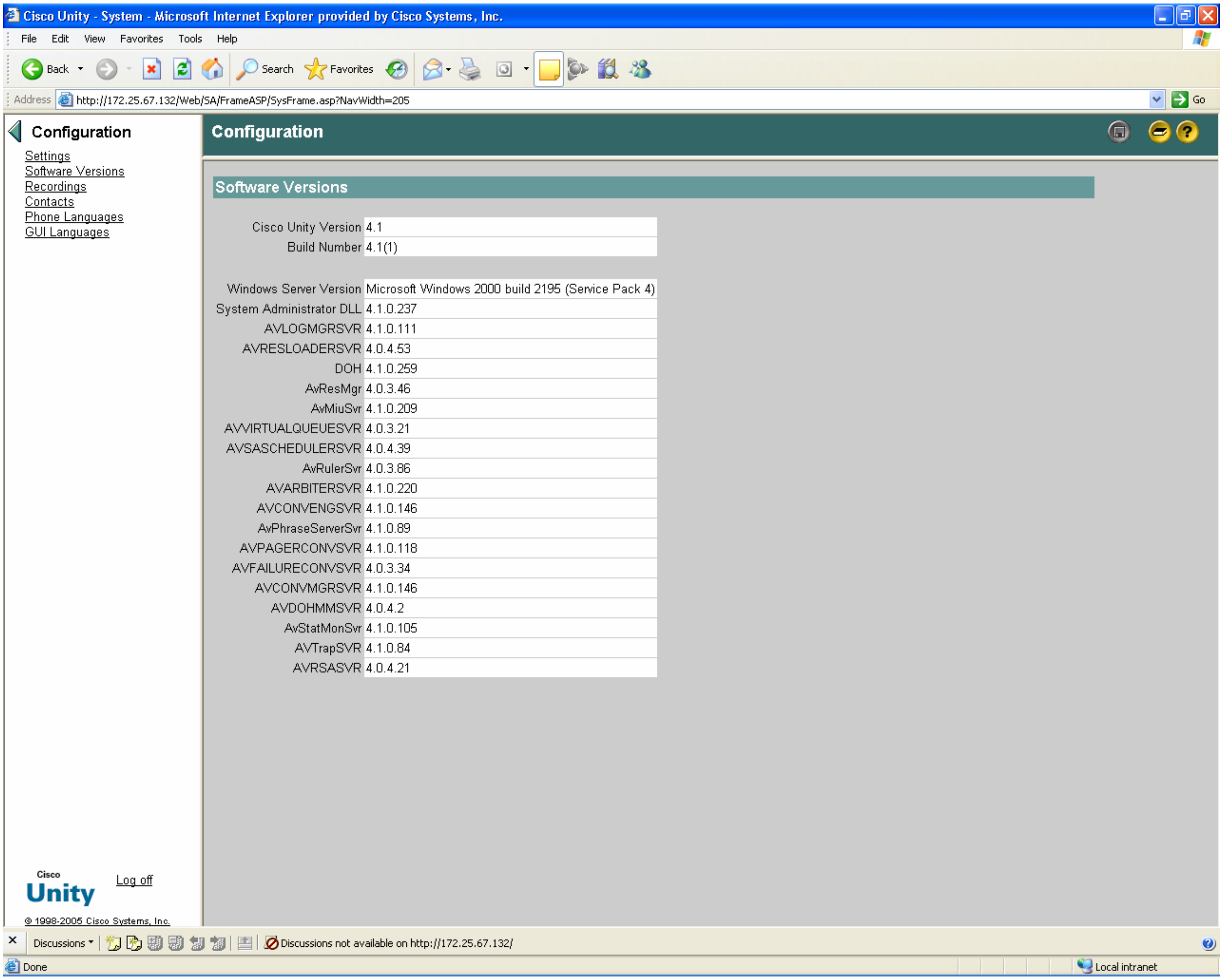
Voice Mail Hunt Pilot



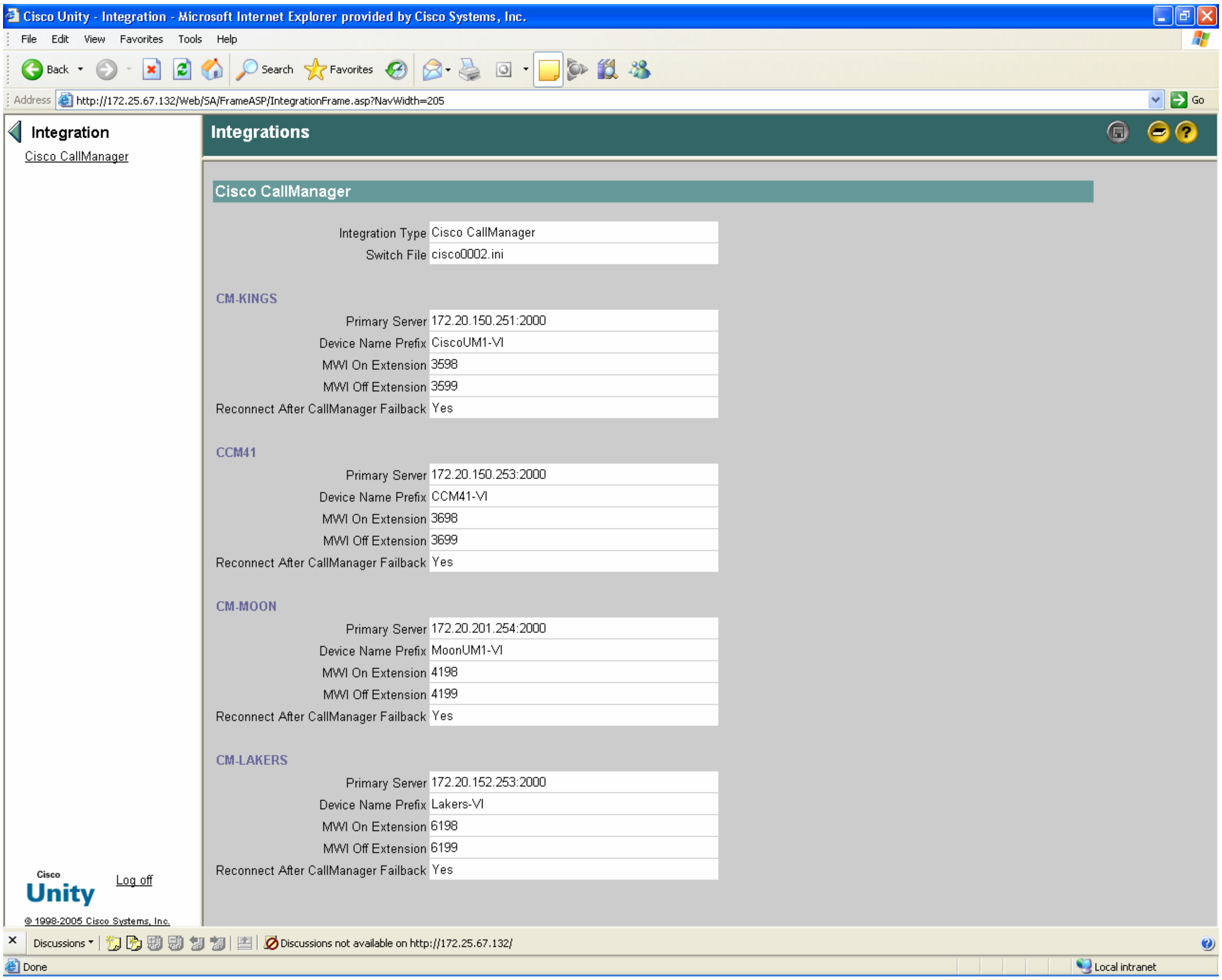

Hunt Pilot Configuration - Microsoft Internet Explorer provided by Cisco Systems, Inc.
File Edit View Favorites Tools Help
Back
Search
Favorites
Route Partition
< None >
Description
VM_Pilot
Numbering Plan
< None >
Route Filter
< None >
MLPP Precedence*
Default
Hunt List*
VM_HuntList
(Edit)
Route Option
Route this pattern
Block this pattern
No Error
Provide Outside Dial Tone
Urgent Priority
Hunt Forward Settings
Use Personal Preferences
Destination
Calling Search Space
Forward Hunt No Answer
or
< None >
Forward Hunt Busy
or
< None >
Maximum Hunt Timer
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
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)
AAR Group Settings
AAR Group
< None >
External Number Mask
Save
Delete
Copy
Add New
*- indicates required item.
Discussions
Discussions not available on https://172.25.67.133/
Done
Local intranet

Cisco Unity Configuration

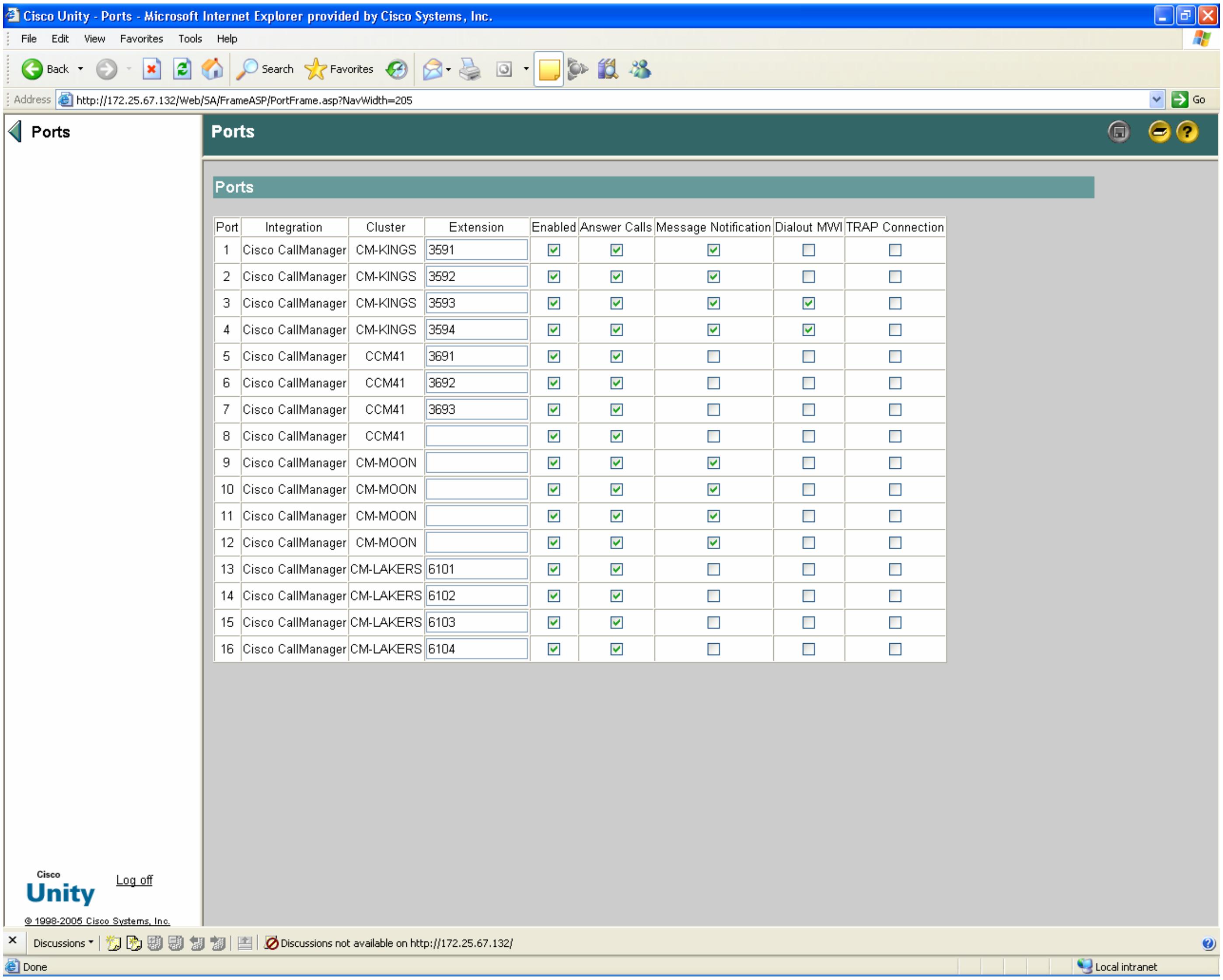
Cisco Unity Software Version



Cisco Unity Integration



Cisco Unity Voice Mail Ports



Port	Integration	Cluster	Extension	Enabled	Answer Calls	Message Notification	Dialout MWI	TRAP Connection
1	Cisco CallManager	CM-KINGS	3591	☑	☑	☑	☐	☐
2	Cisco CallManager	CM-KINGS	3592	☑	☑	☑	☐	☐
3	Cisco CallManager	CM-KINGS	3593	☑	☑	☑	☑	☐
4	Cisco CallManager	CM-KINGS	3594	☑	☑	☑	☑	☐
5	Cisco CallManager	CCM41	3691	☑	☑	☐	☐	☐
6	Cisco CallManager	CCM41	3692	☑	☑	☐	☐	☐
7	Cisco CallManager	CCM41	3693	☑	☑	☐	☐	☐
8	Cisco CallManager	CCM41		☑	☑	☐	☐	☐
9	Cisco CallManager	CM-MOON		☑	☑	☑	☐	☐
10	Cisco CallManager	CM-MOON		☑	☑	☑	☐	☐
11	Cisco CallManager	CM-MOON		☑	☑	☑	☐	☐
12	Cisco CallManager	CM-MOON		☑	☑	☑	☐	☐
13	Cisco CallManager	CM-LAKERS	6101	☑	☑	☐	☐	☐
14	Cisco CallManager	CM-LAKERS	6102	☑	☑	☐	☐	☐
15	Cisco CallManager	CM-LAKERS	6103	☑	☑	☐	☐	☐
16	Cisco CallManager	CM-LAKERS	6104	☑	☑	☐	☐	☐

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
CUCM	Cisco Unified CallManager
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

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

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

Printed in the USA