



# Alcatel Omni PCX 4400 Release 6.0 using SIP Trunk to Cisco Unified CallManager Release 5.0.

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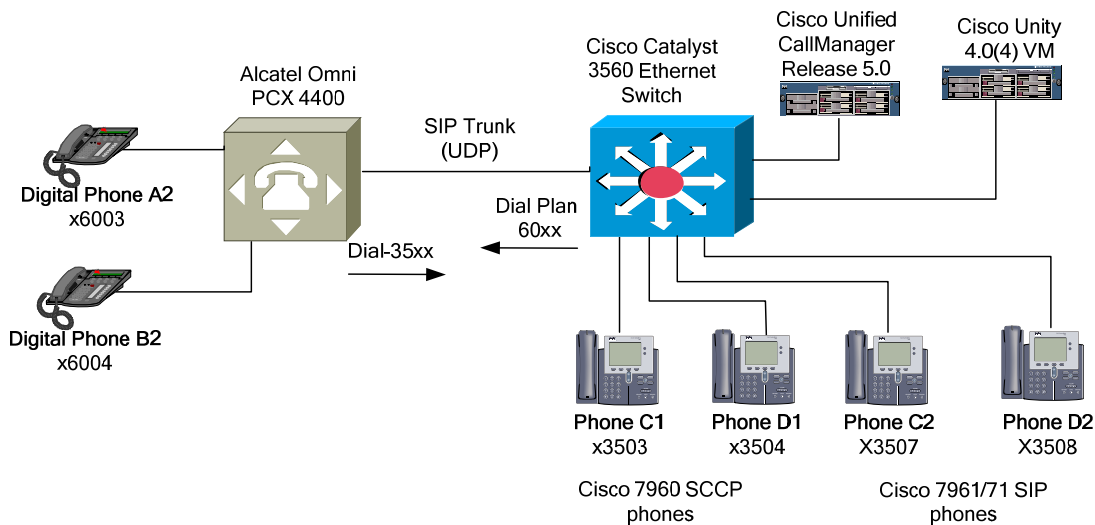
## Introduction

- 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

Diagram of the network topology setup

**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 enabled 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 Omni 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. CUCM 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 works 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) calls CUCM phone (3503) and 3503 performs a CFNA back to another Alcatel phone (6004), one-way audio occurred. Audio path works 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 encounters 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 performs 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 passes 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 interpreted 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.

- No support for Callback feature via the SIP trunk.

## System Components

### Hardware Requirements

- Cisco Unified CallManager MCS -7835H server,
- Unity server MCS-7835H
- Catalyst switch 3560
- Cisco 7970, 7971 and 7960 IP phones
- 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 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

## 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

## Configuration Screen Menu and Commands

### Alcatel Omni PCX 4400 Configuration

Alcatel Omni PCX 4400 Software Version and Hardware Configuration List:

Configuration

File Applications Security Preferences Configuration Windows Help

Search: OmniPCX 4400 In Lab

Where: Name Contains

compidea f1.602	
PCX Release	f1.602
PCX Patch ID	3
MIB Delivery	f1.602
MIB Patch ID	3

PCX Organization Connectivity Data Collection Version

[10:48:47 AM] > show nodes displayed.  
[10:48:48 AM] > show nodes displayed.



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”

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia:1	
Network Number	12
Rank of First Digit to be Sent	1
Incoming identification prefix	
Protocol Type	ABC_F
Numbering Plan Descriptor ID	11
ARS Route list	0
Schedule number	-1
ATM Address ID	-1
Network call prefix	
City/Town Name	
Send City/Town Name	<input type="checkbox"/>
Associated Ext SIP gateway	1

All

1

[1:35:11 PM] > Request 20 sent to compedia.  
[1:35:12 PM] > Request 20 completed on compedia: 1 instance(s) received.

Configuration: compedia





Configure SIP Trunk group:

Trunk Group ID: Enter the trunk group number

Trunk Group Type: T2

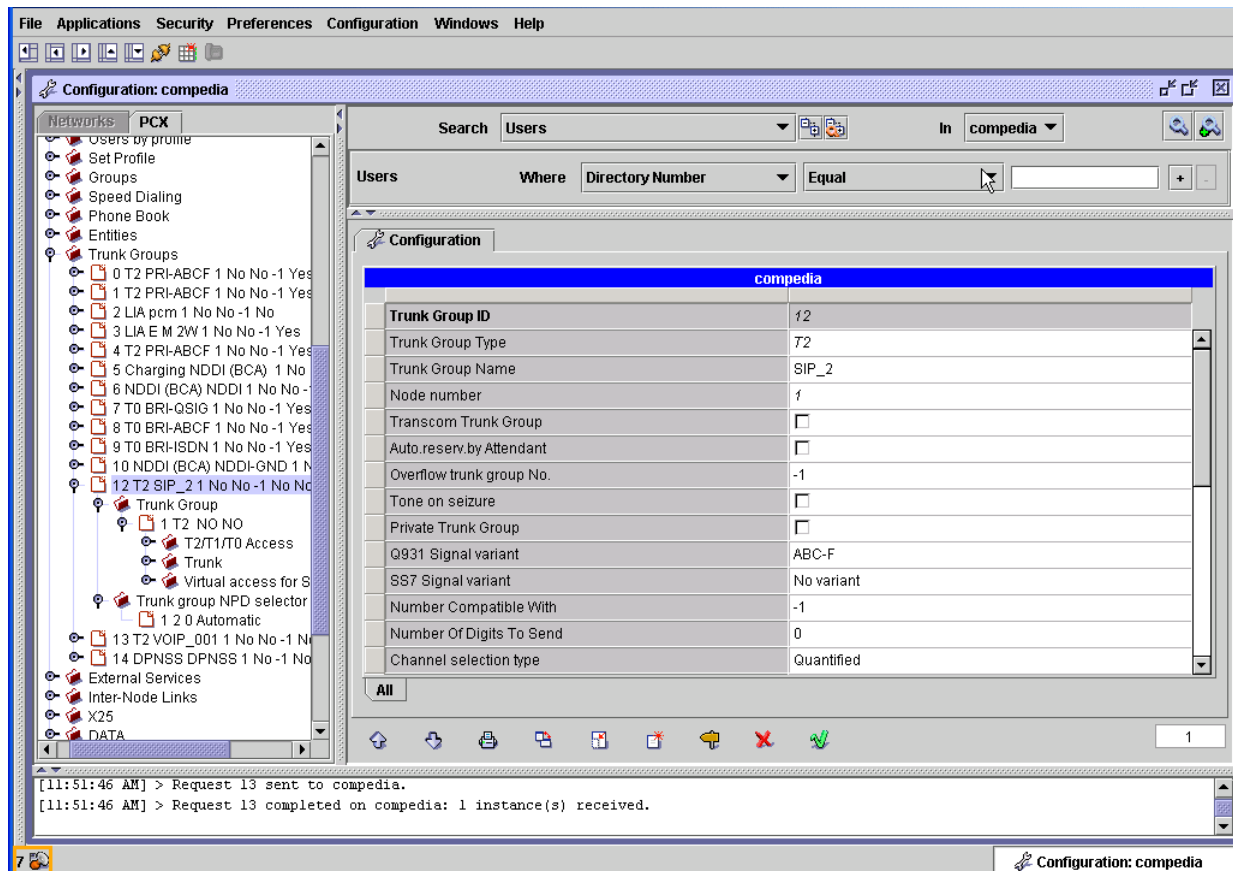
Remote Network: Enter the sub-network number associated with the trunk group.

Node number: Enter the node number

Q931 signal variant: Select ABC-F for the main SIP trunk group

T2 Specification: SIP

Overlap dialing: No





File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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	<input type="checkbox"/>
Auto.DTMF dialing on outgoing call	YES
T2 Specification	SIP
Public Network COS	18
DID transcoding	<input type="checkbox"/>
Special Services	Nothing
Can support UUS in SETUP	<input checked="" type="checkbox"/>
Implicit Priority	
Activation mode	0
Priority Level	0

All

1

[1:35:11 PM] > Request 20 sent to compedia.  
[1:35:12 PM] > Request 20 completed on compedia: 1 instance(s) received.

7 Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia	
Trunk Group ID	12
T2 Specification	SIP
Public Network COS	18
DID transcoding	<input type="checkbox"/>
Special Services	Nothing
Can support UUS in SETUP	<input checked="" type="checkbox"/>
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

All

[12:02:28 PM] > Request 17 sent to compedia.  
[12:02:29 PM] > Request 17 completed on compedia: 1 instance(s) received.

Configuration: compedia



Configure T2 Trunk Group Type:

IP Compression Type: G.711

File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia:12	
Instance (reserved)	1
Trunk Group Type	T2
Public Network Ref.	
End-to-end dialing	NO
DTMF end-to-end signal.	NO
Trunk group used in DISA	NO
DISA Secret Code	
VG for non-existent No.	YES
Routing To Manager	NO
Trunk COS	18
Sending of Progress message	YES
No. of digits unused (ISDN)	4
B Channel Choice	NO
Channels: Attendant Control (Rsvd)	0

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.

7 Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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.

7 Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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.

Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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.

7 Configuration: compedia



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.

The screenshot displays the Cisco Unified Communications Manager (CUCM) configuration interface. The left pane shows the configuration tree with the following structure:

- 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

The right pane shows the configuration for the selected trunk group, titled "compedia:12:1". The configuration table is as follows:

compedia:12:1	
Instance (reserved)	1
Number of SIP Access	8

The bottom status bar indicates the configuration is for "compedia".





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

The screenshot shows the Cisco Configuration Manager interface. The left pane displays the configuration tree with the following structure:

- Networks
  - 1 12 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 -1 Yes No
  - 6 NDDI (BCA) NDDI 1 No No -1 Yes No
  - 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 No -1 No No
  - 12 T2 SIP\_2 1 No No -1 No No
  - 13 T2 VOIP\_001 1 No No -1 No No
  - 14 DPNSS DPNSS 1 No -1 No
- External Services
- Inter-Node Links
- X25
- DATA
- Applications
- Specific Telephone Services
- ATM
- Events Routing Discriminator
- Security and Access Control
- IP
- SIP
  - 1
    - SIP Gateway
      - 1 11 12 172.20.9.250 Compidea 6060
    - SIP Proxy
    - SIP Registrar
    - SIP Dictionary
    - SIP Authentication
    - SIP Ext Gateway
    - Quarantined IP Addresses
    - Trusted IP Addresses
  - DHCP Configuration

The right pane shows the configuration for 'compedia:1' with the following table:

compedia:1	
Instance (reserved)	1
SIP Subnetwork	11
SIP Trunk Group	12
IP Address	172.20.9.250
Machine name - Host	Compidea
SIP Port Number	6060
SIP Proxy Port Number	5060
SIP Subscribe Min Duration	1800
SIP Subscribe Max Duration	86400
DNS local domain name	
SIP DNS1 IP Address	
SIP DNS2 IP Address	

Configure Alcatel SIP Proxy setting:

The screenshot shows the "Configuration: compedia" application window. The left pane displays a tree view of the configuration hierarchy under "Networks". The right pane shows the configuration details for the selected item, "SIP Gateway".

**Left Pane (Tree View):**

- Networks
  - PXC
    - 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 -1 Yes No
    - 6 NDDI (BCA) NDDI 1 No No -1 Yes No
    - 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 No -1 No No
    - 12 T2 SIP\_2 1 No No -1 No No
    - 13 T2 VOIP\_001 1 No No -1 No No
    - 14 DPNSS DPNSS 1 No -1 No
  - External Services
  - Inter-Node Links
  - X25
  - DATA
  - Applications
  - Specific Telephone Services
  - ATM
  - Events Routing Discriminator
  - Security and Access Control
  - IP
  - SIP
    - 1
      - SIP Gateway
        - 1 11 12 172.20.9.250 Compidea 6060
        - SIP Proxy
          - 1 500 4000 180000 No SIP None No
        - SIP Registrar
        - SIP Dictionary
        - SIP Authentication
        - SIP Ext Gateway
        - Quarantined IP Addresses
        - Trusted IP Addresses

**Right Pane (Configuration Details):**

Search Users Where Directory Number Equal

Configuration

compedia:1	
Instance (reserved)	1
SIP initial time-out	500
SIP timer T2	4000
SIP connection duration	180000
Recursive search	<input type="checkbox"/>
Minimal authentication method	SIP None
Authentication realm	
Only authenticated incoming calls	<input type="checkbox"/>

All

Navigation icons at the bottom: back, forward, print, save, delete, refresh, etc.



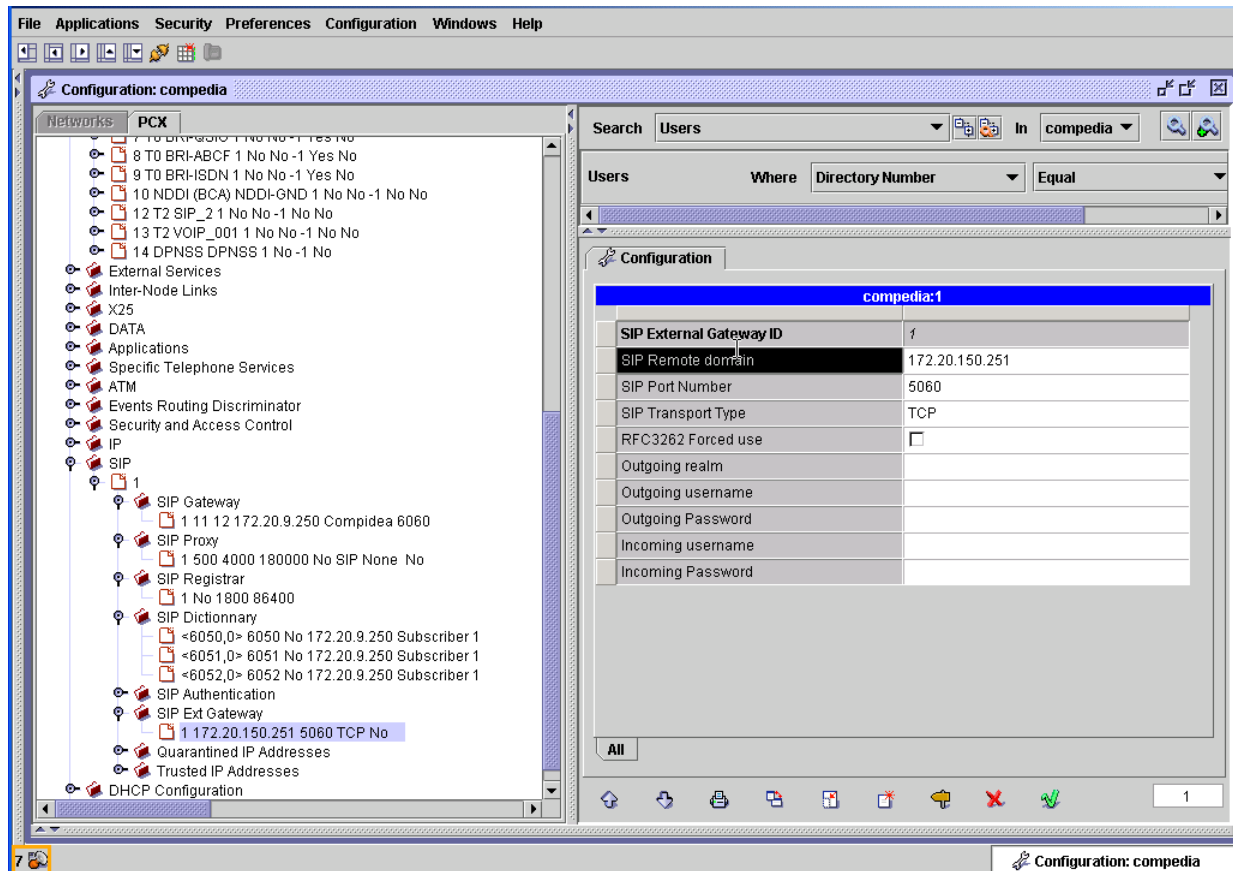
Configure SIP External Gateway:

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

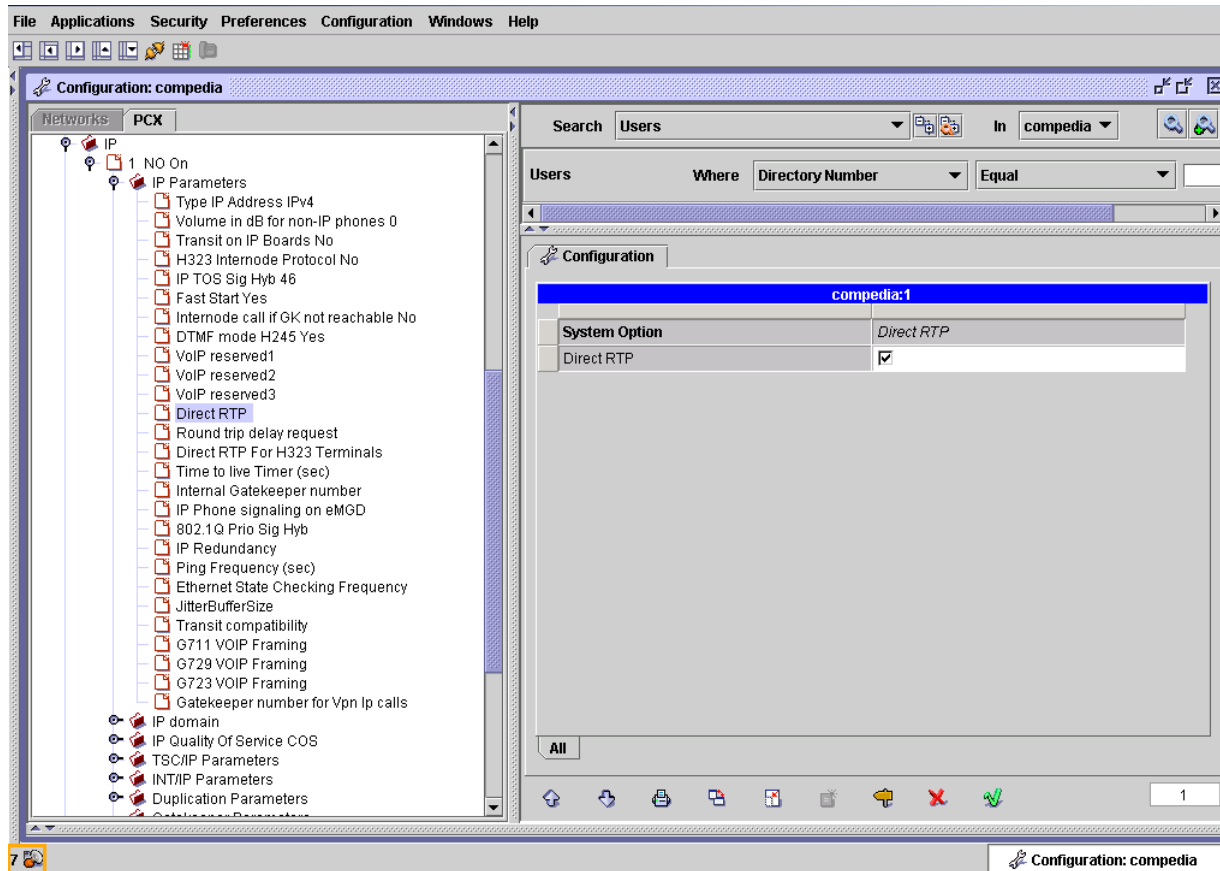
SIP Transport type: Enter TCP or UDP as the transport protocol use for SIP signaling.





Configure IP Parameters:

Direct RTP: enable the checkbox for “Direct RTP”

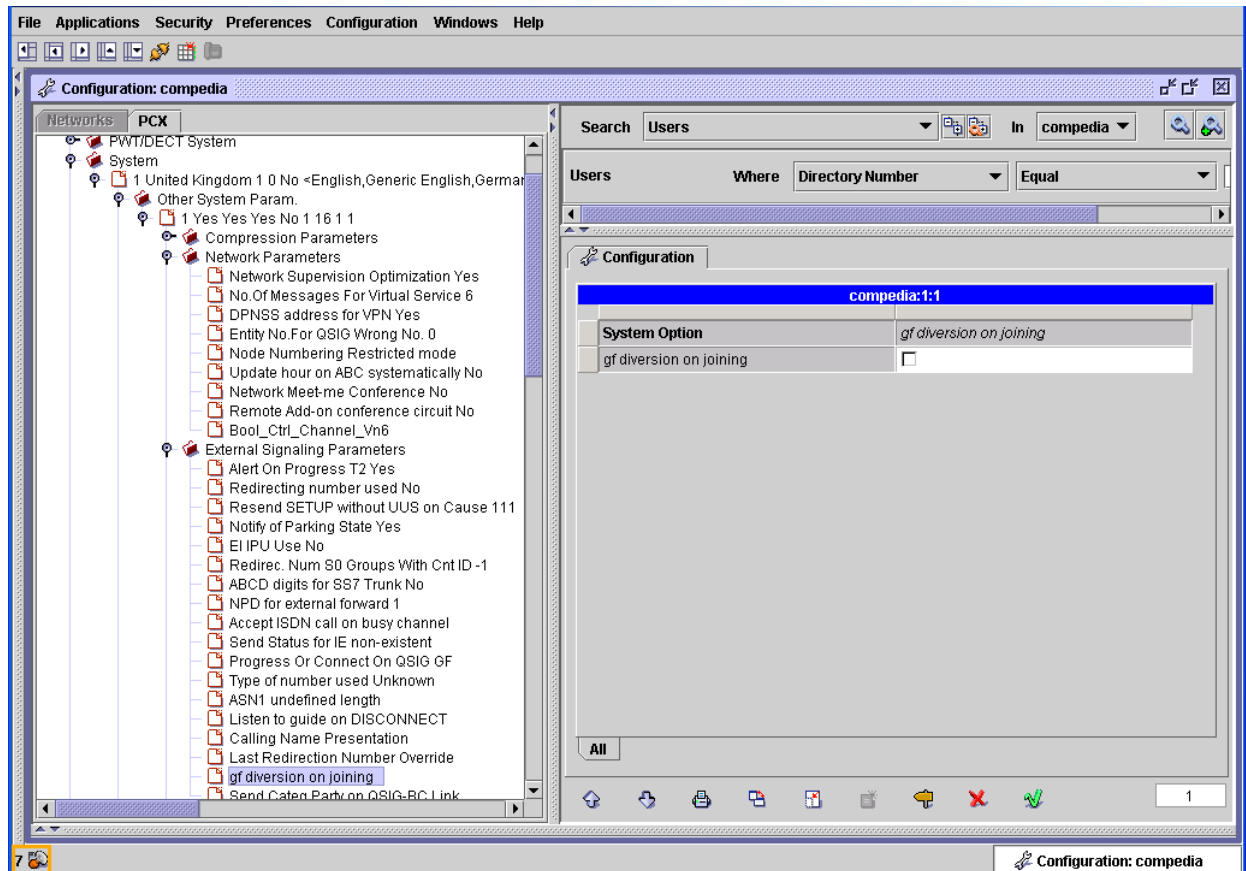




Configure GF diversion on joining:

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

The screenshot shows the Cisco CallManager configuration interface. The left pane displays a list of configuration items under the 'PCX' tab, with '3 Routing No. 0 12 12 4' selected. The right pane shows the configuration for 'compedia:1'.

compedia:1	
Number	3
Prefix Meaning	Routing No.
Domain Identifier	0
Network Number	12
Node Number/ABC-F Trunk Group	12
Number of Digits	4
Number With Subaddress (ISDN)	NO
Default X25 ID.pref.	NO



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](mailto: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)

The screenshot shows the Cisco Configuration Manager interface. On the left, a tree view displays the configuration hierarchy under 'compedia', including 'Users' and '6003 Alcatel DT3 1 0 2 3 4035T 1'. The main pane shows the configuration for the selected user. The 'Directory Number' is set to '6003'. Other fields include 'Add On Module 3' (None), 'External Alphanumeric Keyboard' (None), 'Internal Alphanumeric Keyboard' (English), 'V24 Extension' (unchecked), 'S0 Extension' (unchecked), 'MAC/PC' (NO), 'Z Adapter' (unchecked), 'Language ID' (1), 'Secret Code' (\*\*\*\*), 'Associated Set No.' (6003), 'Cost Center ID' (255), 'Cost Center Name' (Justified), 'Charging COS' (2), and 'Public Network COS' (2). The bottom of the interface shows tabs for 'General Characteristics', 'PIN', 'Assoc.Sets', 'Rights', 'Profile', 'VoiceMail', 'Facilities', 'Set Characteristics', 'Hotel', 'SIP\_Attributes', 'Miscellaneous', 'All', and 'Action'.

Field	Value
Directory Number	6003
Add On Module 3	None
External Alphanumeric Keyboard	None
Internal Alphanumeric Keyboard	English
V24 Extension	<input type="checkbox"/>
S0 Extension	<input type="checkbox"/>
MAC/PC	NO
Z Adapter	<input type="checkbox"/>
Language ID	1
Secret Code	****
Associated Set No.	6003
Cost Center ID	255
Cost Center Name	Justified
Charging COS	2
Public Network COS	2



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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	<input type="checkbox"/>
Voice Mail Dir.No.	
Voice Mail Type	No Voice Mail
Paging Trunk Group	255
Paging Beeper	
Called Associated DECT set	
Tele-Marketing Agent	<input type="checkbox"/>

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia

7





File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia

Directory Number	6003
Tele-Marketing Agent	<input type="checkbox"/>
ISDN User	
External	<input checked="" type="checkbox"/>
Internal	<input type="checkbox"/>
Display ext. calling number	<input checked="" type="checkbox"/>
ISDN Teleservice	Phone
Hotel-Set Operation	Administrative
Use Type Of Dir. No.	Normal
Number Of Set Users	1
Dial by name and text msg.	NO
Multi-line station	NO
Multi-Line Properties	
Automatic Incoming Seizure	<input type="checkbox"/>
Automatic Outgoing Seizure	<input type="checkbox"/>
Relative Filtering	<input type="checkbox"/>

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia

7



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia

Directory Number	6003
Selective Filtering	<input type="checkbox"/>
Overflow on no answer	<input type="checkbox"/>
Overflow on busy	<input type="checkbox"/>
Supervision at off-hook	<input type="checkbox"/>
Automatic Outgoing Seizure for MLA	<input type="checkbox"/>
Dialed number masked	NO
Access Code to UUS messages	NO
Routing Table	0
Associated Videophone	<input type="checkbox"/>
VIP (Very Important Pers.)	<input type="checkbox"/>
Assistant Directory Number	6003
Calls Priority	0
PCBT Associated	NO
Urgent Call	NO
PIN (Personal Ident Nn)	

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia

7



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 Alcatel DT2 1 0 2 2 4035T 1
    - 6003 Alcatel 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
    - lth Profile
    - 6004 Alcatel 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 Alcatel Analog2 1 0 7 1 ANALOG 1
    - 6034 Alcatel Analog3 1 0 7 2 ANALOG 1
    - 6035 Alcatel 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	<input checked="" type="checkbox"/>
Type of control	By COS
PIN group number	1
Can be Called/Dialed By Name	YES
Phone book Name (Dial by name)	Alcatel
Phone book First Name	DT3
Displayed Name	Alcatel DT3
Remote UA	<input type="checkbox"/>
Errors on Secret Code Counter	0
ACD station	NO
NS Right (Notification server)	NO
Incidents Teleservice	NO
CSTA routing	<input type="checkbox"/>

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia

Directory Number	6003
CSTA routing	<input type="checkbox"/>
Voice Guide listening Class	7
Caller COS	4
VSI Transparency	<input type="checkbox"/>
Type of Keyboard	Default keyboard
Errors on Business Code Counter	0
STAP	Off-hook
Tandem	
Tandem Directory Number	
Main set in the tandem	<input type="checkbox"/>
Partial busy	<input type="checkbox"/>
Ringing in partial busy	Long Ring
Specific supervision	<input type="checkbox"/>
Use Personal Calling Number	<input type="checkbox"/>

UA 3G features

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

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 Alcatel DT2 1 0 2 2 4035T 1
    - 6003 Alcatel 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
    - lth Profile
    - 6004 Alcatel 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 Alcatel Analog2 1 0 7 1 ANALOG 1
    - 6034 Alcatel Analog3 1 0 7 2 ANALOG 1
    - 6035 Alcatel 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



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 Alcatel DT2 1 0 2 2 4035T 1
    - 6003 Alcatel 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
    - lth Profile
    - 6004 Alcatel 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 Alcatel Analog2 1 0 7 1 ANALOG 1
    - 6034 Alcatel Analog3 1 0 7 2 ANALOG 1
    - 6035 Alcatel 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	<input type="checkbox"/>
A4980	No 4980
Z IVR	<input type="checkbox"/>
NOMADIC	<input type="checkbox"/>
TAPI premium server	NO
Conference group	-1
Announcement group	-1
Call Restriction COS	0
Applicable Restriction COS	0
Implicit Priority	
Activation mode	0
Priority level	0

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia

Directory Number	6003
Priority Level	0
Explicit Priority	
Activation mode	0
Priority Level	0
Pre-emptable Primary Inc. Line	NO
Pre-emptable Secondary Inc. Line	NO
Priority Presentation	NO
lth Service type	Not Valid
CUG List Number	-1
Preferential CUG	-1
CUG Outgoing Access	<input type="checkbox"/>
CUG Incoming Access	<input type="checkbox"/>
Automatic reconfiguration	CTQ Forbidden - Connection TO
URL UserName	

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

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
lth Service type	Not Valid
CUG List Number	-1
Preferential CUG	-1
CUG Outgoing Access	<input type="checkbox"/>
CUG Incoming Access	<input type="checkbox"/>
Automatic reconfiguration	CTQ Forbidden - Connection TO
URL UserName	
URL Domain	
Advanced configuration	<input type="checkbox"/>

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail Facilities

Set Characteristics Hotel SIP\_Attributes Miscellaneous All Action

1

Configuration: compedia



File Applications Security Preferences Configuration Windows Help

Configuration: compedia

Search Users In compedia

Users Where Directory Number Equal

Configuration

compedia:6003

Key No.	
1	Programmed
Function	Programmed
Content	*40
Mnemo(Pocket,Mobile,4040,IPTouch)	
Locked	NO

All

1

7 Configuration: compedia

Networks PCX

- Users
  - 6001 Alcatel DT1 1 0 2 1 4035T 1
  - 6002 Alcatel DT2 1 0 2 2 4035T 1
  - 6003 Alcatel DT3 1 0 2 3 4035T 1
- Progr.Keys
  - 1 Programmed \*40
  - 2 Three-Party Conference
  - 3 Transfer
  - 4 DTMF End-to-end Dialing
  - 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
  - 6





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.

The screenshot displays the Cisco Configuration Manager (CCM) interface. On the left, a tree view shows the hierarchy: **compedia** > **Users** > **6004 Alcatel DT4 1 0 2 4 4035T 1**. The main pane on the right shows the configuration details for this user. The **Configuration** tab is active, displaying a table of parameters and their values.

compedia	
Directory Number	6004
Location Node	1
Shelf Address	0
Board Address	2
Equipment Address	4
Set Type	4035T
Entity Number	1
Set Function	Default
Domain Identifier	0
Language ID	1
Secret Code	****
Can be Called/Dialed By Name	YES
Phone book Name (Dial by name)	Alcatel
Phone book First Name	DT4
Displayed Name	Alcatel DT4

Below the table, there are tabs for **General Characteristics**, **PIN**, **Assoc.Sets**, **Rights**, **Profile**, **VoiceMail**, and **Facilities**. The **General Characteristics** tab is selected, showing sub-tabs: **Set Characteristics**, **Hotel**, **SIP\_Attributes**, **Miscellaneous**, **All**, and **Action**.



## Cisco Unified CallManager Configuration

### Cisco Unified CallManager Software Version

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



## Cisco Unified CallManager Administration

System version: 5.0.4.2106-1  
Administration version: 1.1.0.0-1

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If you require further assistance please contact us by sending email to [export@cisco.com](mailto:export@cisco.com).



## SIP Trunk Security Profile

Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Log Off

SIP Trunk Security Profile Configuration

Related Links: Back To Find/List ▾ Go

Status

Status: Ready

SIP Trunk Security Profile Information

Name\*

Alcatel\_5060

Description

Alcatel Compidea

Device Security Mode

Non Secure ▾

Incoming Transport Type\*

TCP+UDP ▾

Outgoing Transport Type

TCP ▾

☐ Enable Digest Authentication

Nonce Validity Time (mins)\*

600

X.509 Subject Name

Incoming Port\*

5060

☐ Enable Application Level Authorization

☐ Accept Presence Subscription

☒ Accept Out-of-Dialog REFER

☒ Accept Unsolicited Notification

☒ Accept Replaces Header

Save

Delete

Copy

Reset

Add New

i

\*- indicates required item.



## SIP Phone Security Profile

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

Phone Security Profile Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Phone Security Profile Information

Product Type:

Cisco 7970

Device Protocol:

SIP

Name\*

Cisco 7970 - Standard SIP Non-Secure Profile

Description

Cisco 7970 - Standard SIP Non-Secure Profile

Nonce Validity Time\*

600

Device Security Mode

Non Secure

Transport Type\*

TCP+UDP

☐ Enable Digest Authentication

☐ TFTP Encrypted Config

☐ Exclude Digest Credentials in Configuration File

Phone Security Profile CAPE Information

Authentication Mode\*

By Null String

Key Size (Bits)\*

1024

Parameters used in Phone

SIP Phone Port\*

5060

Copy

Reset

Add New

\*

\*- indicates required item.





## SCCP Phone Security Profile

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

Phone Security Profile Configuration Related Links: Back To Find/List Go



Status

 Status: Ready

Phone Security Profile Information

Product Type: Cisco 7961

Device Protocol: SCCP

Name\* Cisco 7961 - Standard SCCP Non-Secure Profile

Description Cisco 7961 - Standard SCCP Non-Secure Profile

Device Security Mode Non Secure


☐ TFTP Encrypted Config

Phone Security Profile CAPE Information

Authentication Mode\* By Null String

Key Size (Bits)\* 1024

Copy Reset Add New

 \*- indicates required item.



## SIP Profile for SIP Trunk to Alcatel Call Server

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

SIP Profile Configuration Related Links: Back To Find/List ▾ Go

Status

Status: Ready

SIP Profile Information

Name\*

Alcatel\_SIP

Description

Alcatel SIP Profile

Default MTP Telephony Event Payload Type\*

96

☒ Redirect by Application

☐ Disable Early Media on 180

Parameters used in Phone

Timer Invite Expires (seconds)\*

180

Timer Register Delta (seconds)\*

5

Timer Register Expires (seconds)\*

3600

Timer T1 (msec)\*

500

Timer T2 (msec)\*

4000

Retry INVITE\*

6

Retry Non-INVITE\*

10

Start Media Port\*

16384

Stop Media Port\*

32766

Call Pickup URI\*

x-cisco-serviceuri-pickup

Call Pickup Group Other URI\*

x-cisco-serviceuri-opickup

Call Pickup Group URI\*

x-cisco-serviceuri-gpickup

Meet Me Service URI\*

x-cisco-serviceuri-meetme

User Info\*

None ▾

DTMF DB Level\*

Nominal ▾

Call Hold Ring Back\*

Off ▾

Anonymous Call Block\*

Off ▾

Caller ID Blocking\*

Off ▾

Do Not Disturb Control\*

User ▾

Telnet Level for 7940 and 7960\*

Disabled ▾

Timer Keep Alive Expires (seconds)\*

120



Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32768"/>
Call Pickup URI*	<input type="text" value="x-cisco-serviceuri-pickup"/>
Call Pickup Group Other URI*	<input type="text" value="x-cisco-serviceuri-opickup"/>
Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (microseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Abbreviated Dial URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> Call Stats	

\*- indicates required item.



## Standard SIP Profile

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

SIP Profile Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

SIP Profile Information

Name\*

Standard SIP Profile

Description

Default SIP Profile

Default MTP Telephony Event Payload Type\*

101

☐ Redirect by Application

☐ Disable Early Media on 180

Parameters used in Phone

Timer Invite Expires (seconds)\*

180

Timer Register Delta (seconds)\*

5

Timer Register Expires (seconds)\*

3600

Timer T1 (msec)\*

500

Timer T2 (msec)\*

4000

Retry INVITE\*

6

Retry Non-INVITE\*

10

Start Media Port\*

16384

Stop Media Port\*

32766

Call Pickup URI\*

x-cisco-serviceuri-pickup

Call Pickup Group Other URI\*

x-cisco-serviceuri-opickup

Call Pickup Group URI\*

x-cisco-serviceuri-gpickup

Meet Me Service URI\*

x-cisco-serviceuri-meetme

User Info\*

None

DTMF DB Level\*

Nominal

Call Hold Ring Back\*

Off

Anonymous Call Block\*

Off

Caller ID Blocking\*

Off

Do Not Disturb Control\*

User

Telnet Level for 3040 and 3060\*





Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="6"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
Call Pickup URI*	<input type="text" value="x-cisco-serviceuri-pickup"/>
Call Pickup Group Other URI*	<input type="text" value="x-cisco-serviceuri-opickup"/>
Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
TotNet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (microseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Abbreviated Dial URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> Call Stats	

\*- indicates required item.



## Media Resource Group

Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Log Off

Media Resource Group Configuration

Related Links: Back To Find/List Go

Status

Status: Ready

Media Resource Group Status

Media Resource Group: SW\_MRG (used by 41 devices)

Media Resource Group Information

Name\* SW\_MRG

Description SW\_MRG

Devices for this Group

Available Media Resources\*\*

CFB000E3879AEF9

Selected Media Resources\*

ANN\_2 (ANN)  
CFB\_2 (CFB)  
MOH\_2 (MOH)  
MTP\_2 (MTP)

☐ Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

Save Delete Copy Reset Add New

\*- indicates required item.

\*\*Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)



## Media Resource Group List

Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Log Off

Media Resource Group List Configuration

Related Links: Back To Find/List

Go

**Status**  
 Status: Ready

**Media Resource Group List Status**  
Media Resource Group List: SW\_MRGL (used by 37 devices)

**Media Resource Group List Information**  
Name\*

**Media Resource Groups for this List**  
Available Media Resource Groups   
  

Selected Media Resource Groups

\*- indicates required item.



## Assigned MRGL to Default Device Pool

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

Device Pool Configuration Related Links: Back To Find/List Go

Status

Status: Ready

Device Pool: Default (37 members\*\*)

Device Pool Settings

Device Pool Name\*

Default

Cisco Unified CallManager Group\*

Default

Date/Time Group\*

CMLocal

Region\*

Default

Softkey Template\*

Standard User

SRST Reference\*

Disable

Calling Search Space for Auto-registration

< None >

Media Resource Group List

SW\_MRGL

Network Hold MOH Audio Source

< None >

User Hold MOH Audio Source

< None >

Network Locale

< None >

User Locale

< None >

Connection Monitor Duration

Multilevel Precedence and Preemption (MLPP) Information

MLPP Indication\*

Default

MLPP Preemption\*

Default

MLPP Domain

< None >

Save Delete Copy Reset Add New

\*- indicates required item.

\*\* Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.



## SIP Trunk to Alcatel SIP Call Server/Proxy Server

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

Trunk Configuration

Related Links: Back To Find/List Go

Status

Status: Ready

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Device Name\*

Alcatel\_SIP

Description

SIP Trunk to Alcatel

Device Pool\*

Default

Call Classification\*

Use System Default

Media Resource Group List

SW\_MRGL

Location\*

Hub\_None

AAR Group

< None >

Packet Capture Mode\*

None

Packet Capture Duration

0

☒ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Transmit UTF-8 for Calling Party Name

☐ Unattended Port

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



**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.



## SCCP Phone Ext. 3503 Device Level Configuration

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

Phone Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Association Information

Modify Button Items

1 796 Line [1] - 3503 (no partition)

2 796 Line [2] - 3506 (no partition)

3 Add a new SD

4 Add a new SD

5 Add a new SD

6 Add a new SD

7 Unassigned Associated Items -----

796 Line [3] - Add a new DN

8 Add a new SD

9 Add a new SURL

10 Add a new BLF SD

11 Privacy

12 None

Phone Type

Product Type: Cisco 7960

Device Protocol: SCCP

Device Information

Registration Registered with Cisco Unified CallManager cm-kings

IP Address 172.20.150.17

MAC Address\* 00146A3C1BB9

Description SCCP 3503

Device Pool\* Default

Phone Button Template\* Standard 7960 SCCP

Softkey Template Standard User with CallBack

Common Phone Profile\* Standard Common Phone Profile

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List SW\_MRGL

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Location\* Hub\_None

User Locale < None >

Network Locale < None >

Built In Bridge\* Default

Privacy\* Default

Owner User ID < None >

Phone Load Name

☒ Retry Video Call as Audio

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

Protocol Specific Information



<b>Protocol Specific Information</b>	
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 >
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	
<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="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.	
<b>Expansion Module Information</b>	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
<b>External Data Locations Information (Leave blank to use default)</b>	
Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Not Selected --
Login User ID & Pass	









Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

**Extension Information**  
☐ Enable Extension Mobility  
Log Out Profile   
Login in User ID   
Log in Time   
Log out Time

**MLPP Information**  
MLPP Domain   
MLPP Indication\*   
MLPP Preemption\*

**Product Specific Configuration Layout**   
☐ Disable Speakerphone  
☐ Disable Speakerphone and Headset  
PC Port \*   
Settings Access\*   
Gratuitous ARP\*   
PC Voice VLAN Access\*   
Video Capabilities\*   
Auto Line Select\*   
Web Access\*

 \*- 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.



## SCCP Phone Ext. 3503 Directory Number Level Configuration

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

Directory Number Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Directory Number Information

Directory Number\* 3503

Route Partition < None >

Description SCCP 3503

Alerting Name SCCP\_3503A

ASCII Alerting Name SCCP\_3503A

☒ Allow Control of Device from CTI

Associated Devices SEP00146A3C1BB9

Edit Device

Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile < None >

Calling Search Space < None >

Presence Group\* Standard Presence group

AAR Group < None >

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer\* Auto Answer Off

Call Forward and Call Pickup Settings

Forward All ☒ or

Secondary Calling Search Space for Forward All < None >

Forward Busy Internal ☐ or 6004

Forward Busy External ☐ or



Secondary Calling Search Space for Forward All

Forward Busy Internal	<input type="checkbox"/> or	6004	< None >
Forward Busy External	<input type="checkbox"/> or	6004	< None >
Forward No Answer Internal	<input type="checkbox"/> or	6004	< None >
Forward No Answer External	<input type="checkbox"/> or	6004	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	6004	< None >
Forward No Coverage External	<input type="checkbox"/> or	6004	< None >
Forward on CTI Failure	<input type="checkbox"/> 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



## SCCP Phone Ext. 3504 Device Level Configuration

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

Phone Configuration Related Links: Back To Find/List Go

Status

Status: Ready

Association Information

Modify Button Items

1 Line [1] - 3504 (no partition)

2 Line [2] - 3506 (no partition)

3 Add a new SD

4 Add a new SD

5 Add a new SD

6 Add a new SD

Unassigned Associated Items -----

7 Line [3] - Add a new DN

8 Add a new SD

9 Add a new SURL

10 Add a new BLF SD

11 Privacy

12 None

Phone Type

Product Type: Cisco 7960

Device Protocol: SCCP

Device Information

Registration Registered with Cisco Unified CallManager cm-kings

IP Address 172.20.150.18

MAC Address\* 00146A4D3BF5

Description SCCP 3504

Device Pool\* Default

Phone Button Template\* Standard 7960 SCCP

Softkey Template Standard User with CallBack

Common Phone Profile\* Standard Common Phone Profile

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List SW\_MRGL

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Location\* Hub\_None

User Locale < None >

Network Locale < None >

Built In Bridge\* Default

Privacy\* Default

Owner User ID < None >

Phone Load Name

☒ Retry Video Call as Audio

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

Protocol Specific Information




<b>Protocol Specific Information</b>	
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 >
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<input type="checkbox"/> RFC2833 Disabled	
<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="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.	
<b>Expansion Module Information</b>	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
<b>External Data Locations Information (Leave blank to use default)</b>	
Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Not Selected --
Login User ID & Pass	





Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>


**Extension Information**  
☐ Enable Extension Mobility  
Log Out Profile   
Login in User ID   
Log in Time   
Log out Time

**MLPP Information**  
MLPP Domain   
MLPP Indication\*   
MLPP Preemption\*

**Product Specific Configuration Layout**  
  
☐ Disable Speakerphone  
☐ Disable Speakerphone and Headset  
PC Port \*   
Settings Access\*   
Gratuitous ARP\*   
PC Voice VLAN Access\*   
Video Capabilities\*   
Auto Line Select\*   
Web Access\*

 \*- 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.



## SCCP Phone Ext. 3504 Directory Number Level Configuration

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

Directory Number Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Directory Number Information

Directory Number\* 3504

Route Partition < None >

Description SCCP 3504

Alerting Name SCCP4

ASCII Alerting Name SCCP4A

☒ Allow Control of Device from CTI

Associated Devices SEP00146A4D3BF5

Edit Device

Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile < None >

Calling Search Space < None >

Presence Group\* Standard Presence group

AAR Group < None >

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Auto Answer\* Auto Answer Off

Call Forward and Call Pickup Settings

Forward All ☐ or ☐

Secondary Calling Search Space for Forward All < None >

Forward Busy Internal ☐ or ☐

Forward Busy External ☐ or ☐



Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group			< None >

  
**MLPP Alternate Party Settings**  
Target (Destination)   
MLPP Calling Search Space   
MLPP No Answer Ring Duration (seconds)   
  
**Line 1 on Device SEP00146A4D3BF5**  
Display (Internal Caller ID)  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\*   
Ring Setting (Phone Idle)\*   
Ring Setting (Phone Active)\*  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\*   
Busy Trigger\*  (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





## SIP Phone Ext. 3507 Device Level Configuration

NavigationCisco Unified CallManager AdministrationGo

Cisco Unified CallManager AdministrationFor Cisco Unified Communications SolutionsLogged in as: CCMAdministrator

SystemCall RoutingMedia ResourcesVoice MailDeviceApplicationUser ManagementBulk AdministrationHelpLog Off

Phone ConfigurationRelated Links: Back To Find/ListGo

Status

Status: Ready

Association Information

Modify Button Items

1792 Line [1] - 3507 (no partition)

2792 Line [2] - 3505 (no partition)

3Add a new SD

4Add a new SD

5Add a new SD

6Add a new SD

7Add a new SD

8Add a new SD

----- Unassigned Associated Items -----

9792 Line [3] - Add a new DN

10Add a new SD

11Add a new SURF

12Add a new BLF SD

13 Privacy

14 None

Phone Type

Product Type: Cisco 7970

Device Protocol: SIP

Device Information

RegistrationRegistered with Cisco Unified CallManager cm-kins

IP Address172.20.150.16

MAC Address\*000E84F600D6

DescriptionSIP7970 3507

Device Pool\*Default

Phone Button Template\*Standard 7970 SIP

Softkey TemplateStandard User with CallBack

Common Phone Profile\*Standard Common Phone Profile

Calling Search Space< None >

AAR Calling Search Space< None >

Media Resource Group ListSW\_MRGL

User Hold MOH Audio Source1-SampleAudioSource

Network Hold MOH Audio Source1-SampleAudioSource

Location\*Hub\_None

User Locale< None >

Network Locale< None >

Built In Bridge\*Default

Privacy\*Default

Owner User ID< None >

Phone Load Name

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

Protocol Specific Information

Packet Capture Mode\*None



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 >
<input checked="" type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="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	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Not Selected --
Login in User ID < None >	



<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Not Selected --
Login in User ID	< None >
Log in Time	< None >
Log out Time	< None >
<b>MLPP Information</b>	
MLPP Domain	000000
<b>Secure Shell Information</b>	
Secure Shell User	
Secure Shell Password	
<b>Product Specific Configuration Layout</b>	
<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> 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

**i** \*- indicates required item.

**i** \*\*- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

**i** \*\*\*Note: Security Profile Contains Addition CAPF Settings.



## SIP Phone Ext. 3507 Directory Number Level Configuration

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

Directory Number Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Directory Number Information

Directory Number\* 3507

Route Partition < None >

Description 7970 SIP Shareline 2

Alerting Name SIP\_3507A

ASCII Alerting Name SIP\_3507A

☒ Allow Control of Device from CTI

Associated Devices SEP000E84F600D6

Edit Device

Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile Default (Choose <None> to use system default)

Calling Search Space < None >

Presence Group\* Standard Presence group

AAR Group < None >

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer\* Auto Answer Off

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Forward All	<input checked="" type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or	6004	< None >
Forward Busy External	<input type="checkbox"/> or		< None >



Forward Busy Internal	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward Busy External	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Answer External	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward No Coverage External	<input type="checkbox"/> or	<input type="text" value="6004"/>	<input type="text" value=" &lt; None &gt;"/>
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" &lt; None &gt;"/>
No Answer Ring Duration (seconds)	<input type="text"/>		
Call Pickup Group	<input type="text" value=" &lt; None &gt;"/>		

**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)  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\*   
Ring Setting (Phone Idle)\*   
Ring Setting (Phone Active)  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\*   
Busy Trigger\*  (Less than or equal to Max. Calls)

**Forwarded Call Information Display on Device SEP000E84F600D6**  
☒ Caller Name  
☒ Caller Number  
☒ Redirected Number  
☒ Dialed Number



## SIP Phone Ext. 3508 Device Level Configuration

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

Phone Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Association Information

Modify Button Items

1 Line [1] - 3508 (no partition)

2 Line [2] - 3505 (no partition)

3 Add a new SD

4 Add a new SD

5 Add a new SD

6 Add a new SD

7 Add a new SD

8 Add a new SD

Unassigned Associated Items

9 Line [3] - Add a new DN

10 Add a new SD

11 Add a new SURF

12 Add a new BLF SD

13 Privacy

14 None

Phone Type

Product Type: Cisco 7970

Device Protocol: SIP

Device Information

Registration Registered with Cisco Unified CallManager cm-kings

IP Address 172.20.150.11

MAC Address\* 000E839C1229

Description SIP 7970 3508 3505

Device Pool\* Default

Phone Button Template\* Standard 7970 SIP

Softkey Template Standard User with CallBack

Common Phone Profile\* Standard Common Phone Profile

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List SW\_MRGL

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Location\* Hub\_None

User Locale < None >

Network Locale < None >

Built In Bridge\* Default

Privacy\* Default

Owner User ID < None >

Phone Load Name

☐ Ignore Presentation Indicators (internal calls only)

☒ Allow Control of Device from CTI

Protocol Specific Information

Packet Capture Mode\* None



<b>Protocol Specific Information</b>	
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 >
<input checked="" type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	
<b>Certificate Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="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.	
<b>External Data Locations Information (Leave blank to use default)</b>	
Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Not Selected --
Login in User ID < None >	



<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile <span>-- Not Selected --</span>	
Login in User ID <span>&lt; None &gt;</span>	
Log in Time <span>&lt; None &gt;</span>	
Log out Time <span>&lt; None &gt;</span>	
<b>MLPP Information</b>	
MLPP Domain <span>000000</span>	
<b>Secure Shell Information</b>	
Secure Shell User <input type="text"/>	
Secure Shell Password <input type="password"/>	
<b>Product Specific Configuration Layout</b>	
<input type="checkbox"/> Disable Speakerphone	
<input type="checkbox"/> Disable Speakerphone and Headset	
PC Port *	<span>Enabled</span>
Settings Access*	<span>Enabled</span>
Gratuitous ARP*	<span>Enabled</span>
PC Voice VLAN Access*	<span>Enabled</span>
Video Capabilities*	<span>Disabled</span>
Auto Line Select*	<span>Disabled</span>
Web Access*	<span>Enabled</span>
Days Display Not Active	<span>Sunday</span>
	<span>Monday</span>
	<span>Tuesday</span>
Display On Time	<span>07:30</span>
Display On Duration	<span>10:30</span>
Display Idle Timeout	<span>01:00</span>
Span to PC Port*	<span>Disabled</span>
Logging Display*	<span>PC Controlled</span>
Load Server	<input type="text"/>

**i** \*- indicates required item.

**i** \*\*- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

**i** \*\*\*Note: Security Profile Contains Addition CAPF Settings.





## SIP Phone Ext. 3508 Directory Number Level Configuration

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

Directory Number Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Directory Number Information

Directory Number\* 3508

Route Partition < None >

Description 7970 3508

Alerting Name SIP\_3508A

ASCII Alerting Name SIP\_3508A

☒ Allow Control of Device from CTI

Associated Devices SEP000E839C1229

Edit Device

Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile Default (Choose <None> to use system default)

Calling Search Space < None >

Presence Group\* Standard Presence group

AAR Group < None >

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer\* Auto Answer Off

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		..



Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)	<input type="text"/>		
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



## Route Pattern to Alcatel PBX digital phone extensions Configuration

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

Route Pattern Configuration

Related Links: Back To Find/List ▾ Go

Status

Status: Ready

Pattern Definition

Route Pattern\* 600X

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



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	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	
Call Classification*	OnNet	
<input type="checkbox"/> Allow Device Override <input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		
<b>Calling Party Transformations</b>		
<input type="checkbox"/> 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	
<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
<b>ISDN Network-Specific Facilities Information Element</b>		
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.		



## Voice Mail Ports to Unity Voice Mail system

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

Find and List Voice Mail Ports

Status

8 records found

Search Options

Find Voice Mail Port where Device Name begins with Find Search Within Results

(device.name begins with any)

Search Results

Device Name	Description	Device Pool	Device Security Mode	Status	IP Address	Copy
<a href="#">CiscoUM1-V11</a>	Unity VM Ports	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.150.252	
<a href="#">CiscoUM1-V12</a>	Unity VM Ports	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.150.252	
<a href="#">CiscoUM1-V13</a>	Unity VM Ports	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.150.252	
<a href="#">CiscoUM1-V14</a>	Unity VM Ports	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.150.252	
<a href="#">UnityP10-V11</a>	UnityP10 - Standalone PIMG/TIMG	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.17.254	
<a href="#">UnityP10-V12</a>	UnityP10 - Standalone PIMG/TIMG	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.17.254	
<a href="#">UnityP10-V13</a>	UnityP10 - Standalone PIMG/TIMG	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.17.254	
<a href="#">UnityP10-V14</a>	UnityP10 - Standalone PIMG/TIMG	<a href="#">Default</a>	Non Secure Voice Mail Port	Registered with cm-kings	172.20.17.254	

Add New Select All Clear All Delete Selected Reset Selected Rows per Page 50



Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

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-V11

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.



## Voice Mail Pilot for Unity Voice Mail system

Navigation Cisco Unified CallManager Administration Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged in as: CCMAAdministrator

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help Log Off

Voice Mail Pilot Configuration Related Links: Back To Find/List Go

**Status**  
 Status: Ready

**Voice Mail Pilot Information**  
Voice Mail Pilot Number   
Calling Search Space   
Description   
☒ Make this the default Voice Mail Pilot for the system

\*- indicates required item.

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## Voice Mail Profile for Unity Voice Mail system

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 Profile Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Voice Mail Profile Information

Voice Mail Profile Default (used by 12 devices)

Voice Mail Profile Name\* Default

Description Default voice messaging profile

Voice Mail Pilot\*\* 3590/< None >

Voice Mail Box Mask

☒ Make this the default Voice Mail Profile for the System

Save Delete Copy Reset Add New

\*- indicates required item.

\*\*-. The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (< Voice Mail Pilot Number >/< Calling Search Space >).





## Voice Mail MWI ON and OFF for Unity Voice Mail system

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\* 3598

Partition < None >

Description MWI ON

Message Waiting Indicator\* ☒ On ☐ Off

Calling Search Space < None >

Save

Delete

Copy

Add New

\*- indicates required item.

Larry Whitfill (lwhitfill)

CCM 5.0 RFP Responses

We are responding to an RFP for the Health and Human Services Commission in Texas. They have asked for



Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

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.



## Voice Mail Line Group

**Line Group Configuration**Related Links: [Back To Find/List](#) [Go](#)

**Line Group Information**  
Line Group Name\*   
RNA Reversion Timeout\*   
Distribution Algorithm\*

**Hunt Options**  
No Answer\*   
Busy\*\*   
Not Available\*\*

**Line Group Member Information**  
**Find Directory Numbers to Add to Line Group**  
Partition   
Directory Number Contains    
Available DN/Route Partition 

1000

3501

3502

3503

3504





**Current Line Group Members**  
**Reverse Order of Selected DN/Route Partitions**  
Selected DN/Route Partition 

3591


3592


3593


3594


   
Removed DN/Route Partition  

**Directory Numbers**  

 3591 (no partition)

 3592 (no partition)

 3593 (no partition)

 3594 (no partition)



Navigation

Cisco Unified CallManager Administration

Go

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

Logged in as: CCMAAdministrator

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Log Off

Directory Number Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Directory Number Information

Directory Number\*3591

Route Partition< None >

Description

Alerting NameVoiceMail

ASCII Alerting Name

Line GroupCiscoUM1

Associated DevicesCiscoUM1-V11

▼▲

Dissociate Devices

Edit Line Group

Edit Device

Edit Line Appearance

Directory Number Settings

Voice Mail ProfileNoVoiceMail

Calling Search Space< None >

Presence Group\*Standard Presence group

AAR Group< None >

User Hold MOH Audio Source< None >

Network Hold MOH Audio Source< None >

(Choose <None> to use system default)

Call Forward and Call Pickup Settings

Voice MailDestinationCalling Search Space

Forward Allor

Secondary Calling Search Space for Forward All

Forward Busy Internalor

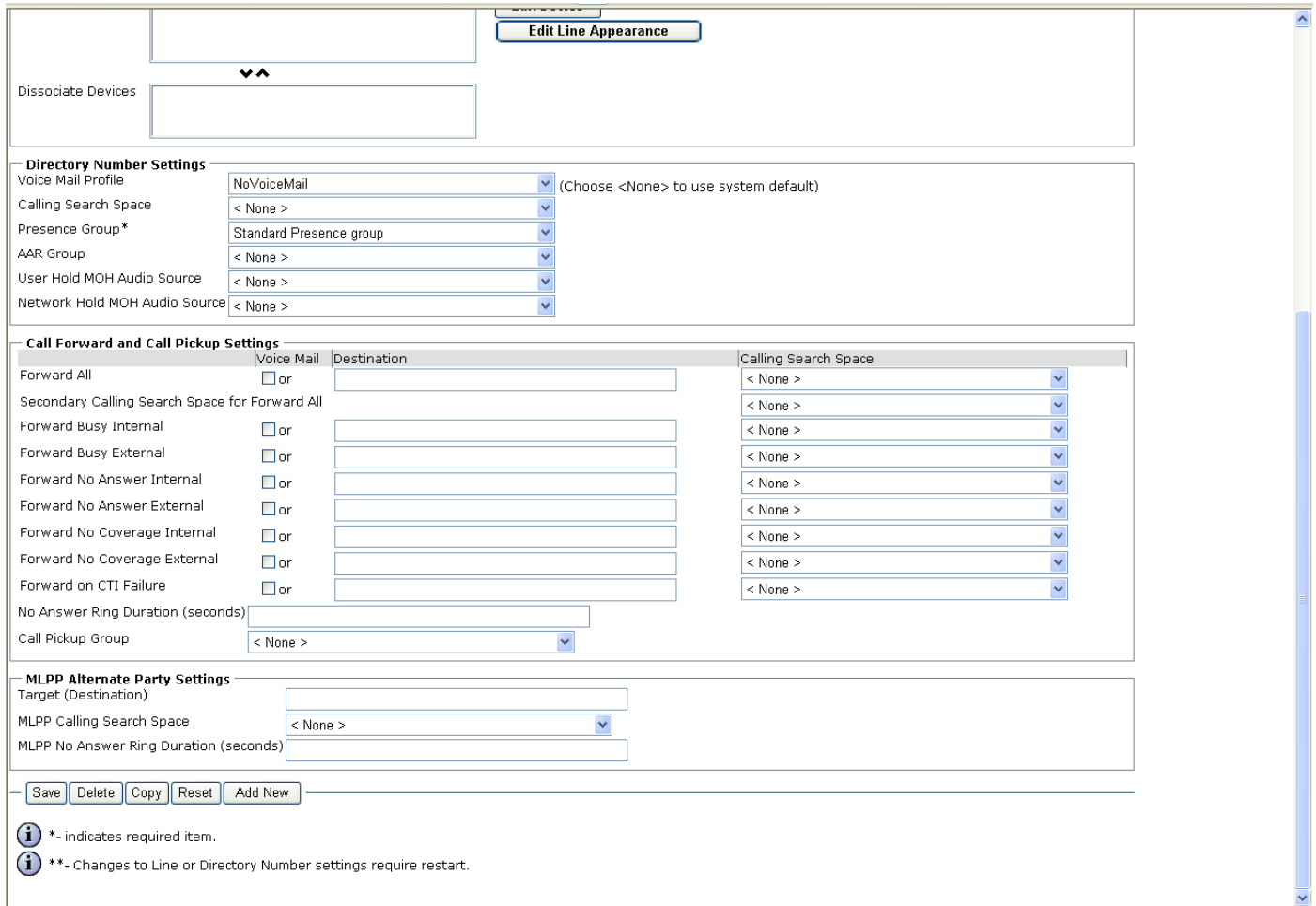
Forward Busy Externalor

< None >

< None >

< None >

< None >





## Voice Mail Hunt List

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

Hunt List Configuration

Related Links: Back To Find/List

Go

Status

Status: Ready

Hunt List Information

Name\*

VM\_HuntList

Description

Unity VM HuntList

Cisco Unified CallManager Group\*

Default

☒ Enable this Hunt List (change effective on Save; no reset required)

Hunt List Member Information

Add Line Group

Selected Groups\*\*

CiscoUM1

Removed Groups\*\*\*

Hunt List Details

CiscoUM1

Save

Delete

Copy

Reset

Add New

\*- indicates required item.

\*\*ordered by highest priority

\*\*\*will be removed from Hunt List when you click Save



## Voice Mail Hunt Pilot

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

Hunt Pilot Configuration

Related Links: Back To Find/List ▾ Go

Status

Status: Ready

Pattern Definition

Hunt Pilot\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Hunt List\*

Route Option

3590

< None >

VM\_Pilot

< None >

< None >

Default

VM\_HuntList

Route this pattern

Block this pattern

No Error

☐ Provide Outside Dial Tone

☐ Urgent Priority

(Edit)

Hunt Forward Settings

Use Personal Preferences

Destination

Calling Search Space

Forward Hunt No Answer

Forward Hunt Busy

Maximum Hunt Timer

☐ or

☐ or

< None >

< None >

< None >

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Default

Default

Connected Party Transformations

Connected Line ID Presentation\*

Connected Name Presentation\*

Default

Default

Called Party Transformations

Discard Digits

< None >



Route Partition	< None >
Description	VM_Pilot
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Hunt List*	VM_HuntList (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
<input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Urgent Priority	

<b>Hunt Forward Settings</b>			
Forward Hunt No Answer	<input type="checkbox"/> or	Destination	Calling Search Space
Forward Hunt Busy	<input type="checkbox"/> or		< None >
Maximum Hunt Timer			< None >

<b>Calling Party Transformations</b>	
<input type="checkbox"/> 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

<b>Connected Party Transformations</b>	
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

<b>Called Party Transformations</b>	
Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	

<b>AAR Group Settings</b>	
AAR Group	< None >
External Number Mask	

[Save](#) [Delete](#) [Copy](#) [Add New](#)

\*- indicates required item.





## Cisco Unity Configuration

### Cisco Unity Software Version

Configuration

[Settings](#)  
[Software Versions](#)  
[Recordings](#)  
[Contacts](#)  
[Phone Languages](#)  
[GUI Languages](#)

Configuration

Software Versions

Cisco Unity Version	4.1
Build Number	4.1(1)
Windows Server Version	Microsoft Windows 2000 build 2195 (Service Pack 4)
System Administrator DLL	4.1.0.237
AVLOGMGRSVR	4.1.0.111
AVRESLOADERSVR	4.0.4.53
DOH	4.1.0.259
AvResMgr	4.0.3.46
AvMiuSvr	4.1.0.209
AVVIRTUALQUEUESVR	4.0.3.21
AVSASCHEDULERSVR	4.0.4.39
AvRulerSvr	4.0.3.86
AVARBITERSVR	4.1.0.220
AVCONVENGSVR	4.1.0.146
AvPhraseServerSvr	4.1.0.89
AVPAGERCONVSVR	4.1.0.118
AVFAILURECONVSVR	4.0.3.34
AVCONVMGRSVR	4.1.0.146
AVDOHMMSVR	4.0.4.2
AvStatMonSvr	4.1.0.105
AVTrapSVR	4.1.0.84
AVRSASVR	4.0.4.21

Cisco  
**Unity**

[Log off](#)

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## Cisco Unity Integration

Integration

[Cisco CallManager](#)

Cisco

Unity

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[Log off](#)

Integrations

Cisco CallManager

Integration Type

Cisco CallManager

Switch File

cisco0002.ini

CM-KINGS

Primary Server

172.20.150.251:2000

Device Name Prefix

CiscoUM1-VI

MWI On Extension

3598

MWI Off Extension

3599

Reconnect After CallManager Failback

Yes

CCM41

Primary Server

172.20.150.253:2000

Device Name Prefix

CCM41-VI

MWI On Extension

3698

MWI Off Extension

3699

Reconnect After CallManager Failback

Yes

CM-MOON

Primary Server

172.20.201.254:2000

Device Name Prefix

MoonUM1-VI

MWI On Extension

4198

MWI Off Extension

4199

Reconnect After CallManager Failback

Yes

CM-LAKERS

Primary Server

172.20.152.253:2000

Device Name Prefix

Lakers-VI

MWI On Extension

6198

MWI Off Extension

6199

Reconnect After CallManager Failback

Yes



## Cisco Unity Voice Mail Ports

Ports

Ports

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

Cisco  
Unity

Log off

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## 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



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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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