



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

April 6, 2011 – Rev. 4

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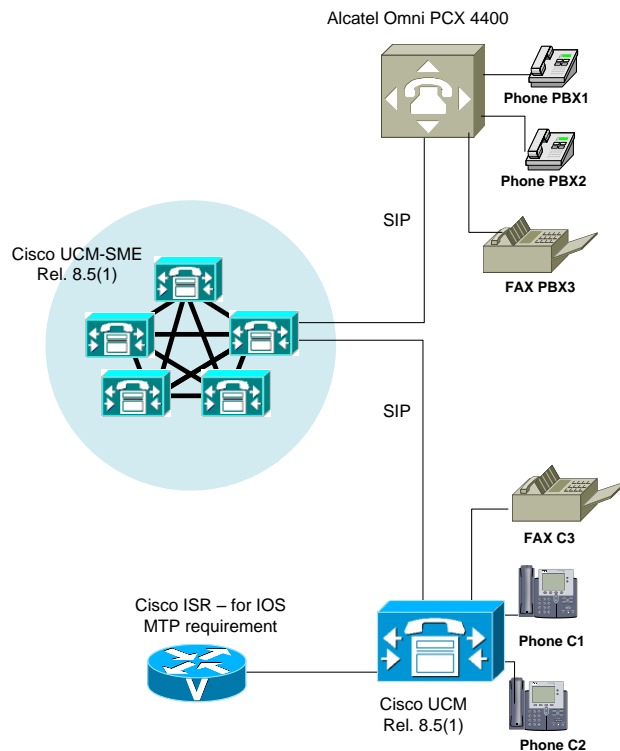
## Introduction

- This application note describes the necessary steps and configurations for connectivity between Alcatel OmniPCX 4400 Rel. 8.0, and a Cisco Unified Communications Manager (Cisco UCM) version 8.5(1) with Cisco Unified Communications Manager-Session Management Edition (Cisco UCM-SME) Version 8.5(1).
- The network topology diagram (Figures 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager (Cisco UCM) Release 8.5(1) connected to the Alcatel PBX via a Cisco UCM-SME using SIP trunks (between Cisco UCM-SME and Alcatel PBX) and SIP trunks (between the Cisco UCM-SME and Cisco UCM). Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, fax transmission, and DTMF interworking.

## Network Topology

Diagram of the network topology setup

**Figure 1.** Network Topology



## Capabilities

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



## Limitations

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



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

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

## System Components

### Hardware Requirements

- Cisco MCS 7800 Unified Communications Manager Appliance
- 2 Cisco Unified IP phone 7960 configured as SCCP phones
- 2 Cisco Unified IP phone 7970 configured as SIP phones
- Alcatel Omni PCX 4400 PBX with INT-IP2 card
- Alcatel digital and IP phones

### Software Requirements

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

### Features Supported

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



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

### **Features Not Supported**

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

### **Configuration**

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

#### **Configuring the Alcatel Omni PCX 4400**

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



Alcatel Omni PCX 4400 Software Version and Hardware Configuration List:

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration

Search: OmniPCX 4400 / Enterprise In: Nextiraone g1.503

Where: Name Contains

Nextiraone g1.503

PCX Release	g1.503
PCX Patch ID	12
MIB Delivery	g1.503
MIB Patch ID	12

PCX Organization Customer data

Connectivity Data Collection Version Miscellaneous Software download

Configuration



Configure System → Other System Parameters → Compression Type

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

The screenshot shows the Cisco AdminMmc interface for configuring the Nextiraone system. The left pane displays a tree view of the configuration hierarchy, with 'Compression type G 729' selected. The right pane shows a table with the following data:

Nextiraone:1:1	
System_Option	Compression type
Compression type	G 729



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

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

The screenshot shows the Cisco AdminMmc interface for configuring Nextiraone. The left pane displays a tree view of the network configuration, with 'IP Domain' selected under the 'IP' category. The right pane shows the configuration table for 'Nextiraone:1'. The table lists various parameters and their values, including 'IP Domain Number' (0), 'IP Domain Name', 'Intra Domain Coding Algorithm' (Default), 'Extra Domain Coding Algorithm' (Without Compression), 'FAX/MODEM Intra domain call transp' (NO), 'FAX/MODEM Extra domain call transp' (NO), 'Domain Max Voice Connection' (-1), 'IP Quality of service' (0), 'Contact Number', 'Backup IP address', 'Trunk Group Id' (-1), and 'Quality of service IP recording' (0).

Nextiraone:1	
IP Domain Number	0
IP Domain Name	
Intra Domain Coding Algorithm	Default
Extra Domain Coding Algorithm	Without Compression
FAX/MODEM Intra domain call transp	NO
FAX/MODEM Extra domain call transp	NO
Domain Max Voice Connection	-1
IP Quality of service	0
Contact Number	
Backup IP address	
Trunk Group Id	-1
Quality of service IP recording	0





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”

OmniVista 4760 - nextiraone:389 - AdminMmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Search Users In Nextiraone

Users Where Directory Number Equal

Configuration

Nextiraone:1	
Network Number	11
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	
Town Name	
Send Town Name	<input type="checkbox"/>
Associated SIP gateway	1
Enable UTF8 names sending	<input checked="" type="checkbox"/>

All

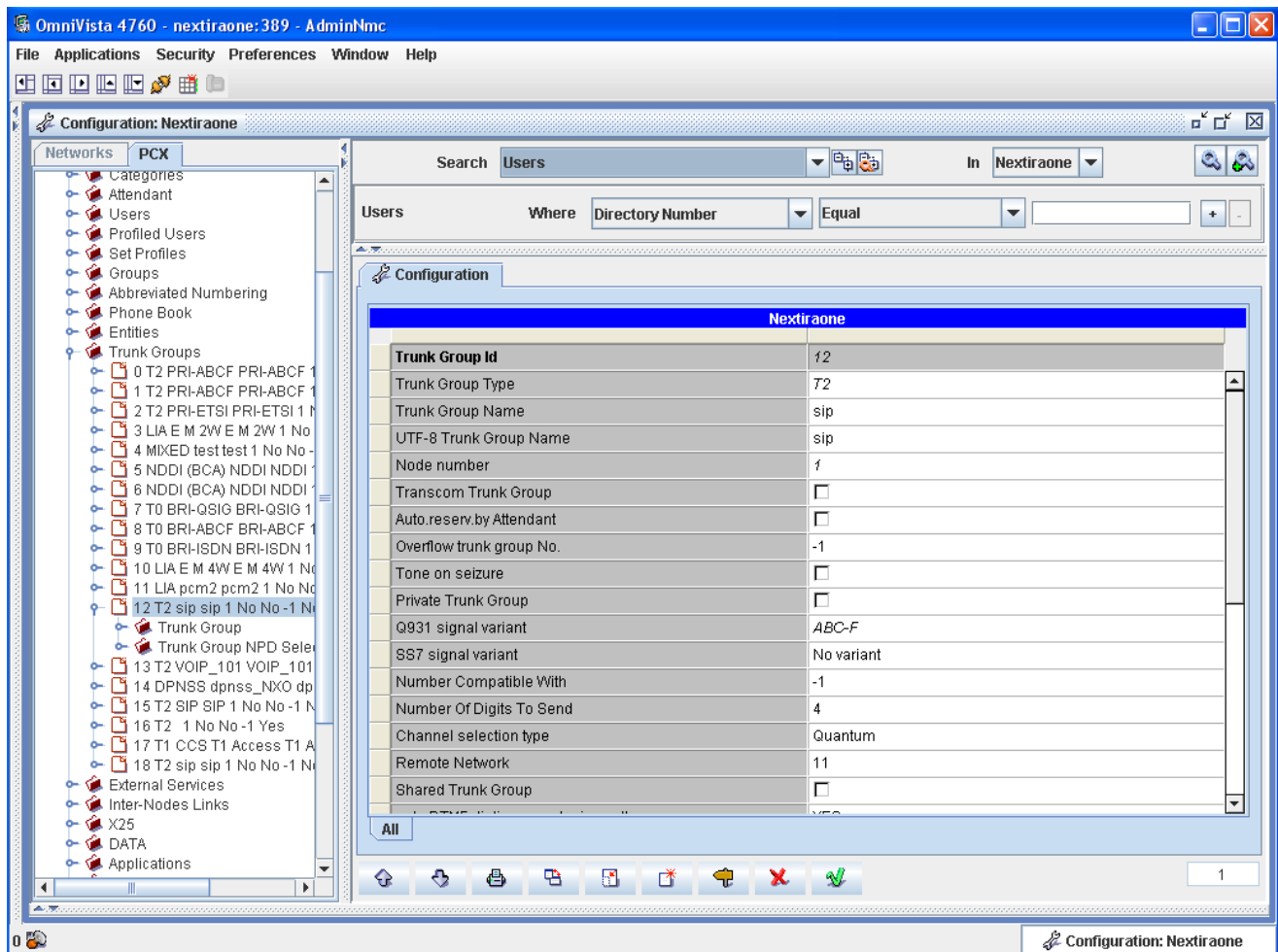
1

Configuration: Nextiraone



Configure SIP Trunk group:

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





OmniVista 4760 - nextiraone: 389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Search Users In Nextiraone

Users Where Directory Number Equal

Configuration

Nextiraone	
Trunk Group Id	12
Shared Trunk Group	<input type="checkbox"/>
auto.DTMF dialing on outgoing call	YES
T2 Specificity	SIP
Public Network Category	18
DDI 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 categ.	10
Outgoing calls Restriction categ.	10
mpt1 343 callee number	NO
Overlap dialing	NO
Call diversion in ISDN	NO

All

1

Configuration: Nextiraone



Configure T2 Trunk Group Type:

IP Compression Type: G.711

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Search Users In Nextiraone

Users Where Directory Number Equal

Configuration

Nextiraone:12

Instance (reserved)	1
Trunk Group Type	T2
Public Network Ref.	
Dialling end to end	NO
DTMF end to end signal.	NO
Trunk group used in DISA	NO
DISA Secret Code	
VG for non-existent No.	YES
Routing To Executive	NO
Trunk Category Id	18
Sending of Progress message	YES
Nb of digits unused (ISDN)	4
B Channel Choice	NO
Channels Reserved By Attend.	0
Dissuasion For ACD	NO
DTO joining	NO
Enquiry Call On B Channel	NO

All Action

Configuration: Nextiraone



OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Search Users In Nextiraone

Users Where Directory Number Equal

Configuration

Nextiraone:12

Instance (reserved)	1
Automated Attendant	NO
Calling party Rights category	0
Entity Number	0
TS Overflow	YES
Number To Be Added	
Supervised by Routing	NO
VPN Cost Limit for Incom.Calls	0
Immediat Trk Listening For VPN Call	YES
VPN TS %	50
Csta Monitored	NO
Max.% of trunks	0
Charge Calling And ADN Creation	YES
Ratio analog.to ISDN tax	
LogicalChannel	1__15 & 17__31
TS Distribution on Accesses	YES
Use Split Acces	NO

All Action

Configuration: Nextiraone



OmniVista 4760 - nextiraone: 389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Search Users In Nextiraone

Users Where Directory Number Equal

Configuration

Nextiraone:12

Instance (reserved)	1
Charge Calling And ADN Creation	YES
Ratio analog.to ISDN tax	
LogicalChannel	1__15 & 17__31
TS Distribution on Accesses	YES
Use Split Acces	NO
Heterogeneous Remote Network	NO
Barring mode	Not barred
ARS class of service	31
Quality Profile for Voice on IP	Profile #1
IP compression type	G 711
Use of volume in system	YES
External Access Server	NO
Modu Trk MonitCsta	
Announcement for Dialtone	NO
Announcement for Ringtone	NO
Private to Public Overflow	YES

All Action

1

Configuration: Nextiraone



Configure Virtual Access for SIP:

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

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

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Networks PCX

Phone Book  
Entities  
Trunk Groups  
0 T2 PRI-ABCF PRI-ABCF 1 No No  
1 T2 PRI-ABCF PRI-ABCF 1 No No  
2 T2 PRI-ETSI PRI-ETSI 1 No No  
3 LIA E M 2W E M 2W 1 No No -1  
4 MIXED test test 1 No No -1 No  
5 NDDI (BCA) NDDI NDDI 1 No No  
6 NDDI (BCA) NDDI NDDI 1 No No  
7 T0 BRI-QSIG BRI-QSIG 1 No No  
8 T0 BRI-ABCF BRI-ABCF 1 No No  
9 T0 BRI-ISDN BRI-ISDN 1 No No  
10 LIA E M 4W E M 4W 1 No No -1  
11 LIA pcm2 pcm2 1 No No -1 No  
12 T2 sip sip 1 No No -1 No  
Trunk Group  
1 T2 NO NO  
T2/T1/T0 Access  
Trunk  
Virtual Access for SIP  
1 2  
Trunk Group NPD Selector  
13 T2 VOIP\_101 VOIP\_101 1 No No  
14 DPNSS dpnss\_NX0 dpnss\_N  
15 T2 SIP SIP 1 No No -1 No  
16 T2 1 No No -1 Yes  
17 T1 CCS T1 Access T1 Access  
18 T2 sip sip 1 No No -1 No  
External Services  
Inter-Nodes Links  
X25  
DATA  
Applications  
Specific Telephone Services  
Atm

Configuration

Nextiraone:12:1

Instance (reserved)	1
Number of SIP Acces	2

All

1

Configuration: Nextiraone



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 Nextiraone Configuration window. The left pane displays a tree view of the configuration hierarchy, with 'SIP Gateway' selected under 'SIP'. The right pane shows the configuration details for 'Nextiraone:1'.

Nextiraone:1	
Instance (reserved)	1
Subnetwork number	11
Trunk Group	12
IP Address	172.20.9.251
Machin name	node001001
Proxy Port Number	5060
sipSubscribeMinDuration	1800
sipSubscribeMaxDuration	86400
Session Timer	1800
sipLocalDomainName	
First DNS IP Address	
Second DNS IP Address	
SDP IN 180	<input checked="" type="checkbox"/>
Cac SIP-SIP	<input type="checkbox"/>
INFO method for remote extension	<input type="checkbox"/>
Dynamic Payload type for dtmf	97





Configure Alcatel SIP Proxy setting:

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Networks PCX

Nextiraone

- Shelf
- Media Gateway
- Dect System
- System
- Translator
- Categories
- Attendant
- Users
- Profiled Users
- Set Profiles
- Groups
- Abbreviated Numbering
- Phone Book
- Entities
- Trunk Groups
- External Services
- Inter-Nodes Links
- X25
- DATA
- Applications
- Specific Telephone Services
- Atm
- Events Routing Discriminator
- Security and Access Control
- IP
- SIP
  - 1
    - SIP Gateway
      - 1 11 12 172.20.9.251
    - Proxy
      - 1 500 4000 180000 N
    - Registrar
    - SIP Dictionary
    - Authentication
    - External Gateways

Configuration

Nextiraone:1

Instance (reserved)	1
Initial timeout	500
timer T2	4000
Dynamic Payload type for dtmf	180000
Recursive search	<input type="checkbox"/>
Minimal authentication method	None
Authentication realm	
Only authenticated incoming calls	<input type="checkbox"/>
Framework Period	3
Framework Nb Message By Period	25
Framework Quarantine Period	1800

All

1

Configuration: Nextiraone



### Configure SIP External Gateway:

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

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

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

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

Dynamic Payload Type for dtmf: Enter 101

The screenshot shows the Cisco Configuration Manager (CCM) interface. The left pane displays the configuration tree with 'SIP' expanded, showing 'SIP Gateway' and 'External Gateways'. The right pane shows the configuration for 'Nextiraone:1'.

Nextiraone:1	
Instance	1
Gateway Name	CUCM-Titan
Remote domain	172.20.236.252
Port number	5060
Transport type	TCP
RFC3262 forced use	<input type="checkbox"/>
Belonging domain	
Registration Id	
Registration Id in P_Asserted	<input type="checkbox"/>
Registration timer	0
SIP Outbound Proxy	
Supervision timer	0
Trunk group number	12
Pool Number	-1
Outgoing realm	
Outgoing username	
Outgoing Password	
Incoming username	
Incoming Password	
RFC 3325 supported by the distant	<input checked="" type="checkbox"/>
SDP IN 180	<input checked="" type="checkbox"/>



OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Networks PCX

- Attendant
- Users
- Profiled Users
- Set Profiles
- Groups
- Abbreviated Numbering
- Phone Book
- Entities
- Trunk Groups
- External Services
- Inter-Nodes Links
- X25
- DATA
- Applications
- Specific Telephone Services
- Atm
- Events Routing Discriminator
- Security and Access Control
- IP
- SIP
  - 1
    - SIP Gateway
      - Proxy
      - Registrar
      - SIP Dictionary
      - Authentication
      - External Gateways
        - 1 CUCM-Titan 172.20.236.252
        - 2 CUCM-Mercury 172.20.215.25
      - Quarantined IP Addresses
      - Trusted IP Addresses
- DHCP Configuration
- Alcatel-Lucent 8&9 Series
- Encryption
- Passive Com. Server
- SNMP Configuration

Configuration

Nextiraone:1

Instance	1
RFC3262 forced use	<input type="checkbox"/>
Belonging domain	
Registration Id	
Registration Id in P_Asserted	<input type="checkbox"/>
Registration timer	0
SIP Outbound Proxy	
Supervision timer	0
Trunk group number	12
Pool Number	-1
Outgoing realm	
Outgoing username	
Outgoing Password	
Incoming username	
Incoming Password	
RFC 3325 supported by the distant	<input checked="" type="checkbox"/>
SDP IN 180	<input checked="" type="checkbox"/>
Minimal authentication method	None
INFO method for remote extension	<input type="checkbox"/>
Send only trunk group algo	<input type="checkbox"/>
Dynamic Payload type for dtmf	101

All

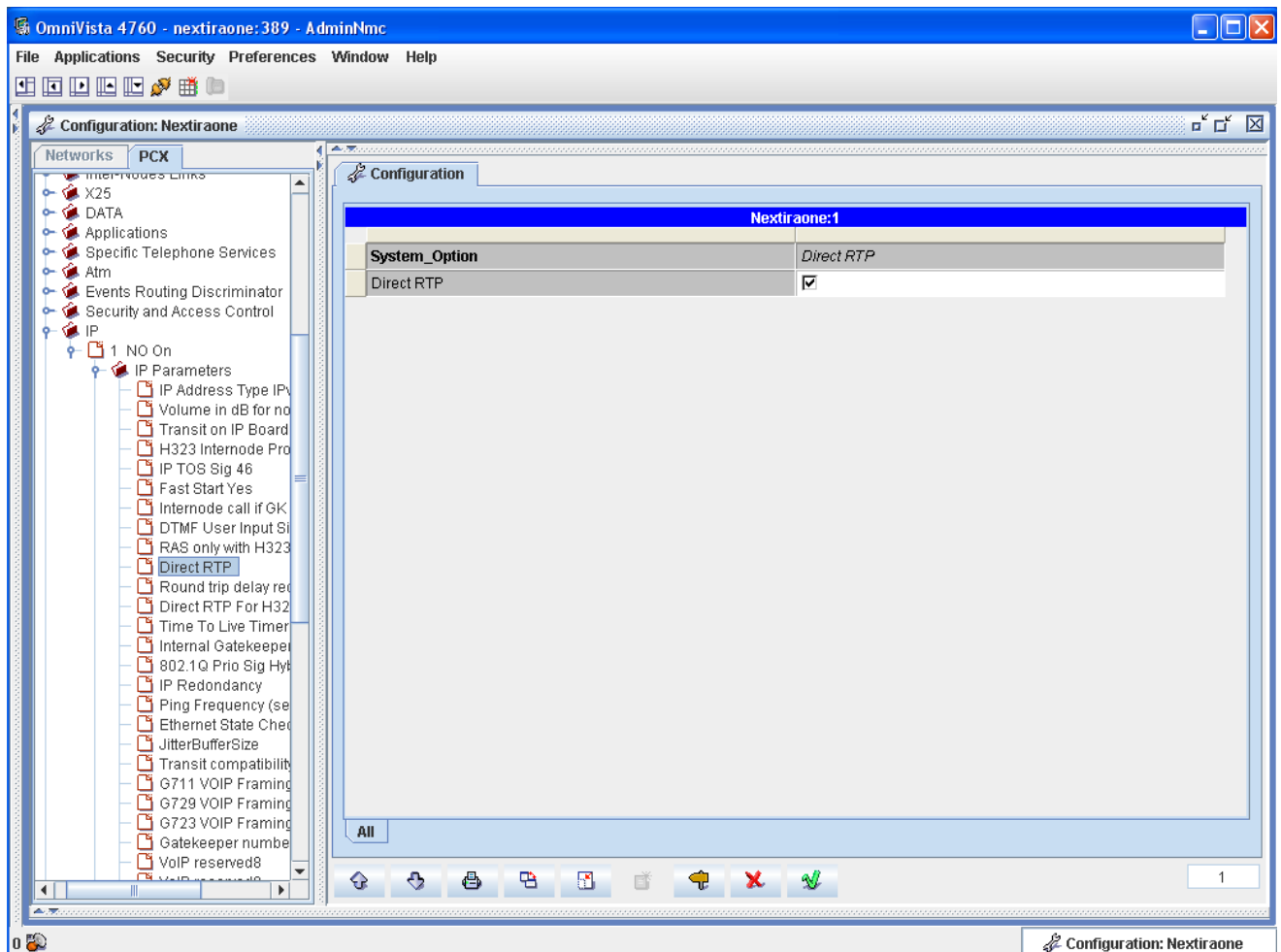
1

Configuration: Nextiraone



Configure IP Parameters:

Direct RTP: enable the checkbox for “Direct RTP”

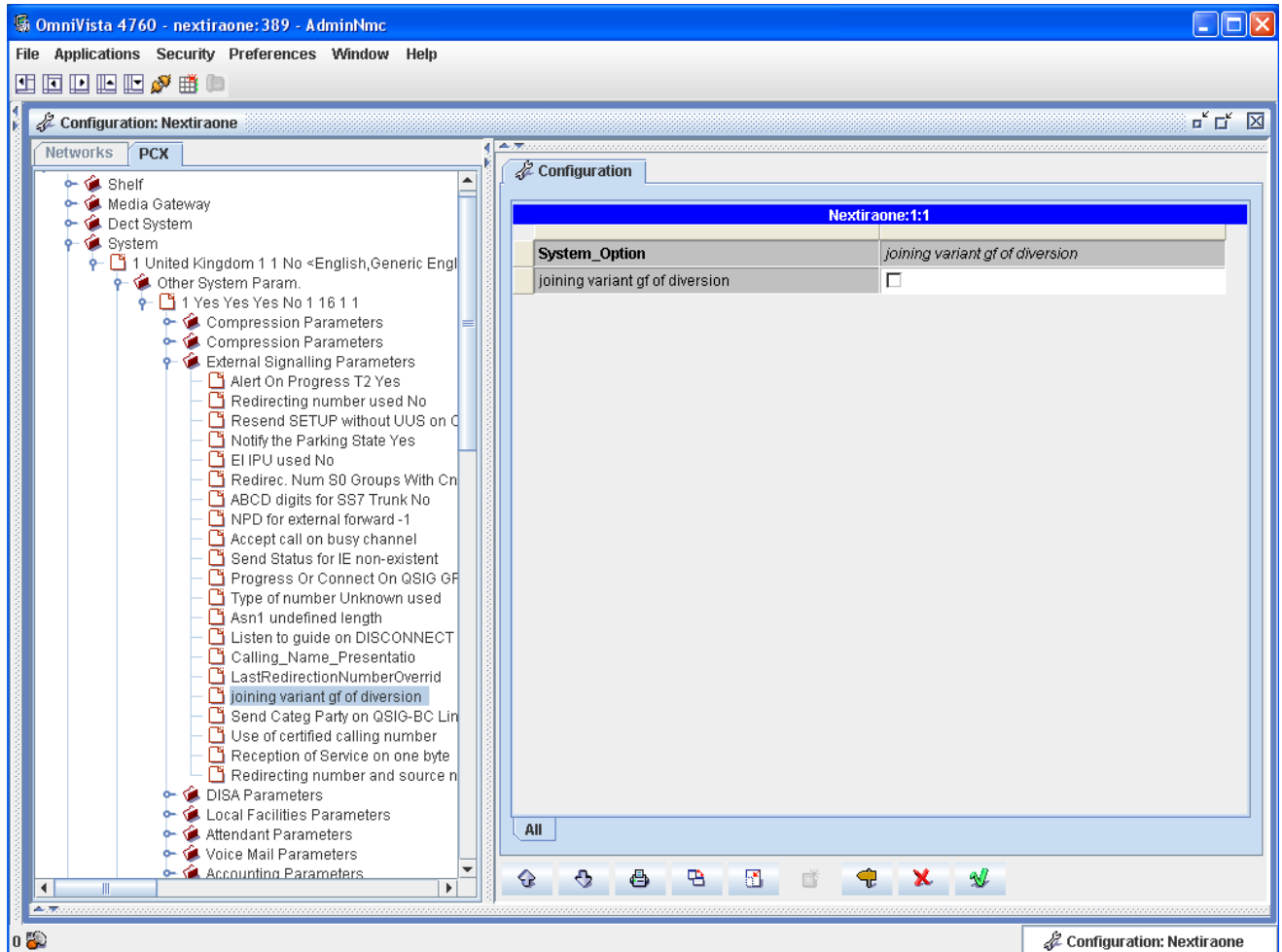




Configure GF diversion on joining:

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

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





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

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

The screenshot shows the Cisco UCM Administration interface. The left pane displays a tree view under 'Configuration: Nextiraone' with 'PCX' selected. The right pane shows the 'Configuration' tab for 'Nextiraone:1' with a table of settings.

Nextiraone:1	
Number	5
Prefix Meaning	Routing No.
Network Number	11
Node Number/ABC-F Trunk Group	12
Number of Digits	4
Number With Sub Address (ISDN)	NO
Default X25 Id.prf.	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 = 3003 will have SIP URL = [3003@172.20.9.251](mailto:3003@172.20.9.251) where 172.20.9.251 is the IP Address of the Alcatel SIP media gateway.

Alcatel 4035 digital phone ext. 3003

The screenshot shows the Cisco AdminMmc interface for configuration. The left pane displays a tree view of the configuration hierarchy, with 'Users' expanded and '3003 NextiraOne DT3 1 0 2 3' selected. The right pane shows the configuration details for this user.

NextiraOne	
Directory Number	3003
Directory name	NextiraOne
Directory First Name	DT3
UTF-8 Directory Name	
UTF-8 Directory First Name	
Location Node	1
Shelf Address	0
Board Address	2
Equipment Address	3
Set Type	4035T
Entity Number	1
Set Function	Default
Identifier of Domain	0
Language Id.	1
Secret Code	****
Can Be Called By Name	YES
Phone book Name (Call by name)	NextiraOne
Phone book First Name	DT3
Displayed Name	NextiraOne DT3

At the bottom of the configuration pane, there are tabs for 'General Characteristics', 'PIN', 'Assoc.Sets', 'Rights', 'Profile', and 'VoiceMail'. The 'General Characteristics' tab is currently selected, showing sub-tabs for 'Facilities', 'Set Characteristics', 'Hotel', 'SIP', 'Miscellaneous', 'All', and 'Action'.



Alcatel IP Touch 4068 phone ext. 3012

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Networks PCX

- Nextiraone
  - Shelf
  - Media Gateway
  - Dect System
  - System
  - Translator
  - Categories
  - Attendant
  - Users
    - 3001 NextiraOne DT1 1 0 2 1
    - 3002 NextiraOne DT2 1 0 2 2
    - 3003 NextiraOne DT3 1 0 2 3
    - 3004 NextiraOne DT4 1 0 2 4
    - 3005 NextiraOne DT5 1 0 2 5
    - 3006 NextiraOne DT6 1 0 2 6
    - 3010 alcatel IP-Phone0 1 255 255
    - 3011 Alcatel IP-Phone1 1 255 255
    - 3012 Alcatel IP-Phone2 1 255 255**
    - 3032 Mummy Doe 1 0 7 0
    - 3033 Daddy Doe 1 0 7 1
    - 3034 Brother Doe 1 0 7 2
    - 3035 Sister Doe 1 0 7 3
    - 3038 uk\_test\_1 1 255 255 255
    - 3039 uk\_test\_2 1 255 255 255
    - 3050 Alcatel SIP0 1 255 255 255
    - 3051 Alcatel SIP1 1 255 255 255
    - 3200 BRI Next DT0 1 0 2 11
    - 3201 BRI Next DT1 1 0 2 10
  - Profiled Users
  - Set Profiles
  - Groups
  - Abbreviated Numbering
  - Phone Book
  - Entities
  - Trunk Groups
  - External Services

Configuration

Nextiraone	
Directory Number	3012
Directory name	Alcatel
Directory First Name	IP-Phone2
UTF-8 Directory Name	
UTF-8 Directory First Name	
Location Node	1
Shelf Address	255
Board Address	255
Equipment Address	255
Set Type	IPTouch 4068
Entity Number	1
Set Function	Default
Identifier of Domain	0
Language Id.	1
Secret Code	****
Can Be Called By Name	YES
Phone book Name (Call by name)	Alcatel
Phone book First Name	IP-Phone
Displayed Name	Alcatel IP-Phone

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail

Facilities Set Characteristics Hotel SIP Miscellaneous All Action

1

Configuration: Nextiraone





Alcatel analog station ext. 3032

OmniVista 4760 - nextiraone:389 - AdminNmc

File Applications Security Preferences Window Help

Configuration: Nextiraone

Networks PCX

Nextiraone

- Shelf
- Media Gateway
- Dect System
- System
- Translator
- Categories
- Attendant
- Users
  - 3001 NextiraOne DT1 1 0 2 1
  - 3002 NextiraOne DT2 1 0 2 2
  - 3003 NextiraOne DT3 1 0 2 3
  - 3004 NextiraOne DT4 1 0 2 4
  - 3005 NextiraOne DT5 1 0 2 5
  - 3006 NextiraOne DT6 1 0 2 6
  - 3010 alcatel IP-Phone0 1 255 255 255
  - 3011 Alcatel IP-Phone1 1 255 255 255
  - 3012 Alcatel IP-Phone2 1 255 255 255
  - 3032 Test Fax Station 1 0 7 1
  - 3033 Daddy Doe 1 0 7 1
  - 3034 Brother Doe 1 0 7 2
  - 3035 Sister Doe 1 0 7 3
  - 3038 uk\_test\_1 1 255 255 255
  - 3039 uk\_test\_2 1 255 255 255
  - 3050 Alcatel SIP0 1 255 255 255
  - 3051 Alcatel SIP1 1 255 255 255
  - 3200 BRI Next DT0 1 0 2 11
  - 3201 BRI Next DT1 1 0 2 10
- Profiled Users
- Set Profiles
- Groups
- Abbreviated Numbering
- Phone Book
- Entities
- Trunk Groups
- External Services

Configuration

Nextiraone	
Directory Number	3032
Directory name	Test Fax Station
Directory First Name	
UTF-8 Directory Name	
UTF-8 Directory First Name	
Location Node	1
Shelf Address	0
Board Address	7
Equipment Address	0
Set Type	ANALOG
Entity Number	1
Set Function	Default
Identifier of Domain	0
Language Id.	1
Secret Code	****
Can Be Called By Name	YES
Phone book Name (Call by name)	Test Fax Sta
Phone book First Name	
Displayed Name	Test Fax Sta

General Characteristics PIN Assoc.Sets Rights Profile VoiceMail

Facilities Set Characteristics Hotel SIP Miscellaneous All Action

1

Configuration: Nextiraone



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

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

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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

ccmadministrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### Cisco Unified CM Administration

System version: 8.5.1.10000-26

Last Successful Logon: Jan 26, 2011 8:48:02 AM

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



## Configuration of Device Pool to Region mapping

Navigation Path: System → Region

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**Region Configuration** Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

**Region Information**

Name\* Default

**Region Relationships**

Region	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
Default	64 kbps (G.722, G.711)	384	Use System Default
NOTE: Region(s) not displayed	Use System Default	Use System Default	Use System Default

**Modify Relationship to other Regions**

Regions	Max Audio Bit Rate	Max Video Call Bit Rate (Includes Audio)	Link Loss Type
Default	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> kbps	Keep Current Setting

Save Delete Reset Apply Config Add New

## Configuration of SIP Profile used by SIP trunks

Navigation Path: Device → Device Settings → SIP Profile

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

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System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

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

Save Delete Copy Reset Apply Config Add New

**SIP Profile Information**

Name\* Early Media SIP Profile

Description SIP Profile with Early Media Enabled

Default MTP Telephony Event Payload Type\* 101

Resource Priority Namespace List < None >

Early Offer for G.Clear Calls\* Disabled

☒ Redirect by Application

☒ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

☐ Enable ANAT

☐ Require SDP Inactive Exchange for Mid-Call Media Change



Parameters used in Phone	
Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled	
<input checked="" type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	

Trunk Specific Configuration	
Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Disabled
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input checked="" type="checkbox"/> Early Offer support for voice and video calls (insert MTP if needed)	
<input checked="" type="checkbox"/> Send send-receive SDP in mid-call INVITE	

SIP OPTIONS Ping	
<input type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

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



## Configuration of SIP Trunk Security Profile used by SIP trunks

Navigation Path: System → Security → SIP Trunk Security Profile

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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**SIP Trunk Security Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**  

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Str
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application Level Authorization	
<input type="checkbox"/> Accept Presence Subscription	
<input checked="" type="checkbox"/> Accept Out-of-Dialog REFER**	
<input checked="" type="checkbox"/> Accept Unsolicited Notification	
<input type="checkbox"/> Accept Replaces Header	
<input type="checkbox"/> Transmit Security Status	

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



## Configuration of SIP trunk to Alcatel PBX

Navigation Path: Device → Trunk

**Cisco Unified CM Administration**  
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**Trunk Configuration** Related Links: [Back To Find/List](#)

Save Delete Reset Add New

**Device Information**

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type

None(Default)

Device Name\*

Alcatel\_SIP

Description

SIP trunk to Alcatel PBX

Device Pool\*

Default

Common Device Configuration

< None >

Call Classification\*

Use System Default

Media Resource Group List

< None >

Location\*

Hub\_None

AAR Group

< None >

Tunneled Protocol\*

None

QSIG Variant\*

No Changes

ASN.1 ROSE OID Encoding\*

No Changes

Packet Capture Mode\*

None

Packet Capture Duration

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\*

When using both sRTP and TLS

Route Class Signaling Enabled\*

Default

Use Trusted Relay Point\*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes



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



Outbound Calls		
Called Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
Calling Party Selection *	Originator	
Calling Line ID Presentation *	Default	
Calling Name Presentation *	Default	
Caller ID DN		
Caller Name		
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound		

SIP Information		
<b>Destination</b>		
<input type="checkbox"/> Destination Address is an SRV		
1 *	Destination Address	Destination Address IPv6      Destination Port
	172.20.9.251	5060
MTP Preferred Originating Codec *	711ulaw	
Presence Group *	Standard Presence group	
SIP Trunk Security Profile *	Non Secure SIP Trunk Profile	
Rerouting Calling Search Space	< None >	
Out-Of-Dialog Refer Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile *	Early Media SIP Profile	
DTMF Signaling Method *	RFC 2833	

Normalization Script		
Normalization Script < None >		
<input type="checkbox"/> Enable Trace		
1	Parameter Name	Parameter Value





## Configuration of SIP trunk to Cisco UCM

Navigation Path: Device → Trunk

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**Trunk Configuration** Related Links: [Back To Find/List](#)

Save Delete Reset Add New

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="CM_POLARIS_SIP"/>
Description	<input type="text" value="SIP Trunk to CM-Polaris"/>
Device Pool*	<input type="text" value="Default"/>
Common Device Configuration	<input type="text" value=" &lt; None &gt;"/>
Call Classification*	<input type="text" value=" Use System Default"/>
Media Resource Group List	<input type="text" value=" &lt; None &gt;"/>
Location*	<input type="text" value=" Hub_None"/>
AAR Group	<input type="text" value=" &lt; None &gt;"/>
Tunneled Protocol*	<input type="text" value=" None"/>
QSIG Variant*	<input type="text" value=" No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value=" No Changes"/>
Packet Capture Mode*	<input type="text" value=" None"/>
Packet Capture Duration	<input type="text" value=" 0"/>
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	<input type="text" value=" When using both sRTP and TLS"/>
Route Class Signaling Enabled*	<input type="text" value=" Default"/>
Use Trusted Relay Point*	<input type="text" value=" Default"/>
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

**Intercompany Media Engine (IME)**E.164 Transformation Profile **Multilevel Precedence and Preemption (MLPP) Information**MLPP Domain **Call Routing Information**☒ Remote-Party-Id☒ Asserted-IdentityAsserted-Type\* SIP Privacy\* **Inbound Calls**Significant Digits\* Connected Line ID Presentation\* Connected Name Presentation\* Calling Search Space AAR Calling Search Space Prefix DN ☒ Redirecting Diversion Header Delivery - Inbound**Incoming Calling Party Settings**

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

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value=" Default"/>	<input type="text" value=" 0"/>	<input type="text" value=" &lt; None &gt;"/>	<input checked="" type="checkbox"/>

**Connected Party Settings**Connected Party Transformation CSS ☒ Use Device Pool Connected Party Transformation CSS



<b>Outbound Calls</b>	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	


**SIP Information**

<b>Destination</b>		
<input type="checkbox"/> Destination Address is an SRV		
<b>Destination Address</b>	<b>Destination Address IPv6</b>	<b>Destination Port</b>
1* 172.20.236.50		5060 + -
MTP Preferred Originating Codec*	711ulaw	
Presence Group*	Standard Presence group	
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile	
Rerouting Calling Search Space	< None >	
Out-Of-Dialog Refer Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile*	Early Media SIP Profile	
DTMF Signaling Method*	RFC 2833	
<b>Normalization Script</b>		
Normalization Script < None >		
<input type="checkbox"/> Enable Trace		
<b>Parameter Name</b>	<b>Parameter Value</b>	
1		+ -



## Configuration of Route Patterns – To Alcatel PBX

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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

ccmadministrator | Search Documentation | About | Logout

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Route Pattern Configuration

Related Links: Back To Find/List

Save Delete Copy Add New

**Pattern Definition**

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*

Route Option

Call Classification\*

☐ Allow Device Override

☐ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level\*

☐ Require Client Matter Code

30XX

< None >

To Alcatel PBX

-- Not Selected --

< None >

Default

< None >

Default

Alcatel\_SIP

☒ Route this pattern

☐ Block this pattern

No Error

OnNet

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

Default

Default

Cisco CallManager

Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

Default

Default

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

< None >

Cisco CallManager

Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

-- Not Selected --

< Not Exist >



## Configuration of Route Patterns – To Cisco UCM

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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

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Route Pattern Configuration

Related Links: Back To Find/List

Save Delete Copy Add New

**Pattern Definition**

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*

Route Option

Call Classification\*

☐ Allow Device Override

☐ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level\*

☐ Require Client Matter Code

50XX

< None >

Route to CM-Polaris

-- Not Selected --

< None >

Default

< None >

Default

CM\_POLARIS\_SIP

☒ Route this pattern

☐ Block this pattern

No Error

OnNet

☐ Allow Device Override

☐ Provide Outside Dial Tone

☐ Allow Overlap Sending

☐ Urgent Priority

☐ Require Forced Authorization Code

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\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value



## Configuration of SIP Route Pattern for processing redirect messages from Alcatel PBX

Navigation Path: Call Routing → SIP Route Pattern

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

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SIP Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Status

Update successful

Pattern Definition

Pattern Usage IPAddress Routing

IPv4 Pattern\* 172.20.9.251

IPv6 Pattern

Description SIP RP for redirection messages from Alcatel PBX

Route Partition < None >

SIP Trunk\* Alcatel\_SIP

☐ Block Pattern

Calling Party Transformations

☐ Use Calling Party's External Phone Mask

Calling Party Transformation Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default

Calling Line Name Presentation\* Default

Connected Party Transformations

Connected Line ID Presentation\* Default

Connected Line Name Presentation\* Default



## Configuring the Cisco Unified Communications Manager

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

## Cisco Unified Communications Manager version

**Cisco Unified CM Administration**  
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### Cisco Unified CM Administration

System version: 8.5.1.10000-26

Last Successful Logon: Jan 26, 2011 8:48:02 AM

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A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

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## Configuration of Device Pool to Region mapping

Navigation Path: System → Region

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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**Region Configuration** Related Links: Back To Find/List Go

Save Delete Reset Apply Config Add New

**Region Information**  
Name\* Default

**Region Relationships**

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.729	800	Use System Default
G711_Region	G.711	384	Use System Default
NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default			

**Modify Relationship to other Regions**

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default G711_Region	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps	Keep Current Setting

Save Delete Reset Apply Config Add New





**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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

**Region Configuration** Related Links: Back To Find/List Go

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**Region Information**  
Name\* G711\_Region

**Region Relationships**

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.711	384	Use System Default
G711_Region	G.711	384	Use System Default

NOTE: Regions(s) not displayed      Use System Default      Use System Default      Use System Default

**Modify Relationship to other Regions**

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default G711_Region	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps	Keep Current Setting

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## Configuration of Conference Bridge

Navigation Path: Media Resources → Conference Bridge

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**Conference Bridge Configuration** Related Links: Back To Find/List Go

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**Status**  
Status: Ready

**Conference Bridge Information**  
Conference Bridge : CFB112233445566 (Conference Bridge on IOS DSP Farm)  
Registration Registered with Cisco Unified Communications Manager CM-Polaris  
IPv4 Address 172.20.236.101

**IOS Conference Bridge Info**  
Conference Bridge Type\* Cisco IOS Enhanced Conference Bridge  
☒ Device is trusted  
Conference Bridge Name\* CFB112233445566  
Description Conference Bridge on IOS DSP Farm  
Device Pool\* Default  
Common Device Configuration < None >  
Location\* Hub\_None  
Device Security Mode\* Non Secure Conference Bridge  
Use Trusted Relay Point\* Default

Save Delete Copy Reset Apply Config Add New



### Conference Bridge IOS configuration:

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

### Configuration of Media Resource Group

Navigation Path: Media Resources → Media Resource Group

Cisco

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Media Resource Group ConfigurationRelated Links: Back To Find/ListGo

SaveDeleteCopyAdd New

Status

Status: Ready

Media Resource Group Status

Media Resource Group: MRG\_Polaris (used by 125 devices)

Media Resource Group Information

Name\*MRG\_Polaris

DescriptionMRG\_Polaris

Devices for this Group

Available Media Resources\*\*

C0300115C28E8BC  
CFB0001C9D93A99  
CFB\_2  
MTP\_2

Selected Media Resources\*

ANN\_2 (ANN)  
CFB112233445566 (CFB)  
MOH\_2 (MOH)  
MTP0015F90D0970 (XCODE)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)



## Configuration of Media Resource Group List

Navigation Path: Media Resources → Media Resource Group List

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**Media Resource Group List Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Media Resource Group List Status**  
Media Resource Group List: MRGL\_Polaris (used by 115 devices)

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

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

MRG\_Polaris

Save Delete Copy Add New



## Configuration of SIP Profile

Navigation Path: Device → Device Settings → SIP Profile

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**SIP Profile Configuration** Related Links: Back To Find/List

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**SIP Profile Information**

Name*	Early Media and Options SIP Profile
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Resource Priority Namespace List	< None >
Early Offer for G.Clear Calls*	Disabled
<input type="checkbox"/> Redirect by Application	
<input checked="" type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	

**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



Do Not Disturb Control*	User
Telnet Level for 7940 and 7960 *	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled	
<input checked="" type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	

---

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on *	Never
RSVP Over SIP *	Local RSVP
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Disabled
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input checked="" type="checkbox"/> Early Offer support for voice and video calls (insert MTP if needed)	
<input checked="" type="checkbox"/> Send send-receive SDP in mid-call INVITE	

---

**SIP OPTIONS Ping**

<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6



## Configuration of SIP Trunk to SME

Navigation Path: Device → Trunk

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**Trunk Configuration** Related Links: Back To Find/List

Save Delete Reset Add New

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CM_TITAN_SIP
Description	SIP Trunk to CM-Titan
Device Pool*	G711_Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Polaris
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	



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



<b>Outbound Calls</b>	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection *	Originator
Calling Line ID Presentation *	Default
Calling Name Presentation *	Default
Caller ID DN	
Caller Name	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

#### SIP Information

<b>Destination</b>		
<input type="checkbox"/> Destination Address is an SRV		
1 *	Destination Address	Destination Address IPv6
	172.20.236.252	
		Destination Port
		5060
MTP Preferred Originating Codec *	711ulaw	
Presence Group *	Standard Presence group	
SIP Trunk Security Profile *	SIP Trunks	
Rerouting Calling Search Space	< None >	
Out-Of-Dialog Refer Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
SIP Profile *	Early Media and Options SIP Profile	
DTMF Signaling Method *	RFC 2833	
<b>Normalization Script</b>		
Normalization Script < None >		
<input type="checkbox"/> Enable Trace		
1	Parameter Name	Parameter Value





## Configuration of Route Pattern to Alcatel PBX through SME

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

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**Route Pattern Configuration** Related Links: Back To Find/List

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**Pattern Definition**

Route Pattern\*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence\*

Resource Priority Namespace Network Domain

Route Class\*

Gateway/Route List\*

Route Option

30XX

route\_p

To Alcatel OmniPCX 4400

-- Not Selected --

< None >

Default

< None >

Default

CM\_TITAN\_SIP

Route this pattern

Block this pattern No Error

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\*

Calling Name Presentation\*

Calling Party Number Type\*

Calling Party Numbering Plan\*

Default

Default

Cisco CallManager

Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

Default

Default

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

< None >

Cisco CallManager

Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service

Service Parameter Name

Service Parameter Value

-- Not Selected --

-- Not Selected --

< Not Exist >





25	None	Device Mobility Mode*	Default	<a href="#">View Current Device</a>
26	None		<a href="#">Mobility Settings</a>	
27	None	Owner User ID	< None >	
28	None	Phone Personalization*	Default	
29	None	Services Provisioning*	Default	
30	None	Phone Load Name		
31	None	Single Button Barge	Default	
32	None	Join Across Lines	Default	
33	None	Use Trusted Relay Point*	Default	
34	None	BLF Audible Alert Setting (Phone Idle)*	Default	
35	None	BLF Audible Alert Setting (Phone Busy)*	Default	
37	None	Always Use Prime Line*	Default	
38	None	Always Use Prime Line for Voice Message*	Default	
39	None	Calling Party Transformation CSS	< None >	
40	None	Geolocation	< None >	
42	None	<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
43	None	<input checked="" type="checkbox"/> Retry Video Call as Audio		
44	None	<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
45	None	<input checked="" type="checkbox"/> Allow Control of Device from CTI		
46	None	<input type="checkbox"/> Logged Into Hunt Group		
47	None	<input type="checkbox"/> Remote Device		
48	None	<input type="checkbox"/> Protected Device****		
49	None	<input type="checkbox"/> Hot line Device*****		
51	None			
52	None	<b>Protocol Specific Information</b>		
53	None	Packet Capture Mode*	None	
54	None	Packet Capture Duration	0	
55	None	Presence Group*	Standard Presence group	
	----- Unassigned Associated Items -----	Device Security Profile*	Cisco 7965 - Standard SCCP Non-Secure Profile	
56	None	SUBSCRIBE Calling Search Space	< None >	

[Add a new SD](#)

[Add a new SURL](#)



57	<a href="#">Add a new BLF SD</a>	<input type="checkbox"/> Unattended Port
58	<a href="#">Add a new BLF Directed Call Park</a>	<input type="checkbox"/> Require DTMF Reception
59	CallBack	<input type="checkbox"/> RFC2833 Disabled
60	Call Park	
61	Call Pickup	<b>Certification Authority Proxy Function (CAPF) Information</b>
62	Conference List	Certificate Operation* <input type="text" value="No Pending Operation"/>
63	Conference	Authentication Mode* <input type="text" value="By Null String"/>
64	Do Not Disturb	Authentication String <input type="text"/>
65	End Call	<input type="button" value="Generate String"/>
66	Forward All	Key Size (Bits)* <input type="text" value="1024"/>
67	Group Call Pickup	Operation Completes By <input type="text" value="2011"/> <input type="text" value="4"/> <input type="text" value="11"/> <input type="text" value="12"/> (YYYY:MM:DD:HH)
68	Hold	Certificate Operation Status: None
69	Hunt Group Logout	Note: Security Profile Contains Addition CAPF Settings.
70	<a href="#">Intercom [1] - Add a new Intercom</a>	
71	Malicious Call Identification	<b>Expansion Module Information</b>
72	Meet Me Conference	Module 1 <input type="text" value="&lt; None &gt;"/>
73	Mobility	Module 1 Load Name <input type="text"/>
74	New Call	Module 2 <input type="text" value="&lt; None &gt;"/>
75	Other Pickup	Module 2 Load Name <input type="text"/>
76	Quality Reporting Tool	
77	Redial	<b>External Data Locations Information (Leave blank to use default)</b>
78	Remove Last Participant	Information <input type="text"/>
79	Transfer	Directory <input type="text"/>
80	Video Mode	Messages <input type="text"/>
81	Privacy	Services <input type="text"/>
82	None	Authentication Server <input type="text"/>
		Proxy Server <input type="text"/>
		Idle <input type="text"/>
		Idle Timer (seconds) <input type="text"/>
		Secure Authentication URL <input type="text"/>
		Secure Directory URL <input type="text"/>



Secure Idle URL	<input type="text"/>
Secure Information URL	<input type="text"/>
Secure Messages URL	<input type="text"/>
Secure Services URL	<input type="text"/>

#### Extension Information

<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

#### MLPP Information

MLPP Domain	< None >
MLPP Indication *	Default
MLPP Preemption *	Default

#### Do Not Disturb

<input type="checkbox"/> Do Not Disturb	
DND Option *	Use Common Phone Profile Setting
DND Incoming Call Alert	< None >

#### Secure Shell Information

Secure Shell User	<input type="text"/>
Secure Shell Password	<input type="password"/>

#### Product Specific Configuration Layout

		Param	Override Common Settings
<input type="checkbox"/> Disable Speakerphone			
<input type="checkbox"/> Disable Speakerphone and Headset			
Forwarding Delay *		Disabled	
PC Port *		Enabled	



Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Video Capabilities*	Disabled	<input type="checkbox"/>
Auto Line Select*	Disabled	
Web Access*	Disabled	<input type="checkbox"/>
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	PC Controlled	
Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Disabled	
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server		
Advertise G.722 Codec*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID		
LLDP Power Priority*	Unknown	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		<input type="checkbox"/>
IPv6 Log Server		
802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	
Minimum Ring Volume*	0-Silent	
HTTPS Server*	http and https Enabled	
Handset/Headset Monitor*	Enabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>



## Configuration of Cisco SIP 7965 Phone

Navigation Path: Device → Phone

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Phone Configuration

Related Links: Back To Find/List

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Association Information

Modify Button Items

1	<a href="#">Line [1] - 5015 in phones</a>
2	<a href="#">Line [2] - 4089333321 in phones</a>
3	<a href="#">Line [3] - 7323204085 in phones</a>
4	<a href="#">Add a new SD</a>
5	<a href="#">Add a new SD</a>
6	<a href="#">Add a new SD</a>
----- Add On Module(s) -----	
7	<a href="#">Add a new SD</a>
8	None
9	None
10	None
11	None
12	None
13	None
14	None
15	None
16	None
17	None
18	None
19	None
20	None
21	None
22	None
23	None
24	None

Phone Type

Product Type: Cisco 7965

Device Protocol: SIP

Device Information

Registered with Cisco Unified Communications Manager CM-Polaris

IP Address [172.20.236.32](#)

Active Load ID SIP45.9-1-1SR1S

☒ Device is Active

☒ Device is trusted

MAC Address\*

Description

Device Pool\*  [View Details](#)

Common Device Configuration  [View Details](#)

Phone Button Template\*

Softkey Template

Common Phone Profile\*

Calling Search Space

AAR Calling Search Space

Media Resource Group List

User Hold MOH Audio Source

Network Hold MOH Audio Source

Location\*

AAR Group

User Locale

Network Locale

Built In Bridge\*

Privacy\*



24	None	Device Mobility Mode*	Default	<a href="#">View Current Device</a>
25	None		<a href="#">Mobility Settings</a>	
26	None	Owner User ID	< None >	
27	None	Phone Personalization*	Default	
28	None	Services Provisioning*	Default	
29	None	Phone Load Name		
30	None	Single Button Barge	Default	
31	None	Join Across Lines	Default	
32	None	Use Trusted Relay Point*	Default	
33	None	BLF Audible Alert Setting (Phone Idle)*	Default	
34	None	BLF Audible Alert Setting (Phone Busy)*	Default	
35	None	Always Use Prime Line*	Default	
36	None	Always Use Prime Line for Voice Message*	Default	
37	None	Calling Party Transformation CSS	< None >	
38	None	Geolocation	< None >	
39	None	<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
40	None	<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
41	None	<input checked="" type="checkbox"/> Allow Control of Device from CTI		
42	None	<input checked="" type="checkbox"/> Logged Into Hunt Group		
43	None	<input type="checkbox"/> Remote Device		
44	None	<input type="checkbox"/> Protected Device*****		
45	None	<input type="checkbox"/> Hot line Device*****		
46	None	<b>Protocol Specific Information</b>		
47	None	Packet Capture Mode*	None	
48	None	Packet Capture Duration	0	
49	None	Presence Group*	Standard Presence group	
50	None	SIP Dial Rules	< None >	
51	None	MTP Preferred Originating Codec*	711ulaw	
52	None	Device Security Profile*	Cisco 7965 - Standard SIP Non-Secure Profile	
----- Unassigned Associated Items -----				
55	<a href="#">Line [4] - Add a new DN</a>			
56	<a href="#">Add a new SD</a>			





57	<a href="#">Add a new SURL</a>	Rerouting Calling Search Space	< None >
58	<a href="#">Add a new BLF SD</a>	SUBSCRIBE Calling Search Space	< None >
59	<a href="#">Add a new BLF Directed Call Park</a>	SIP Profile*	Standard SIP Profile
60	Do Not Disturb	Digest User	< None >
61	<a href="#">Intercom [1] - Add a new Intercom</a>	<input type="checkbox"/> Media Termination Point Required	
62	Call Park	<input type="checkbox"/> Unattended Port	
63	Call Pickup	<input type="checkbox"/> Require DTMF Reception	
64	CallBack	<b>Certification Authority Proxy Function (CAPF) Information</b>	
65	Conference List	Certificate Operation*	No Pending Operation
66	Conference	Authentication Mode*	By Null String
67	End Call	Authentication String	
68	Forward All	<input type="button" value="Generate String"/>	
69	Group Call Pickup	Key Size (Bits)*	1024
70	Hold	Operation Completes By	2011 4 11 12 (YYYY:MM:DD:HH)
71	Hunt Group Logout	Certificate Operation Status: None	
72	Malicious Call Identification	Note: Security Profile Contains Addition CAPF Settings.	
73	Meet Me Conference	<b>Expansion Module Information</b>	
74	Mobility	Module 1	< None >
75	New Call	Module 1 Load Name	
76	Other Pickup	Module 2	< None >
77	Quality Reporting Tool	Module 2 Load Name	
78	Redial	<b>External Data Locations Information (Leave blank to use default)</b>	
79	Remove Last Participant	Information	
80	Transfer	Directory	
81	Privacy	Messages	
82	None	Services	
		Authentication Server	
		Proxy Server	



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

---

**Extension Information**

☐ Enable Extension Mobility

Log Out Profile

Log in Time < None >

Log out Time < None >

---

**MLPP Information**

MLPP Domain

---

**Do Not Disturb**

☐ Do Not Disturb

DND Option\*

DND Incoming Call Alert

---

**Secure Shell Information**

Secure Shell User

Secure Shell Password

---

**Product Specific Configuration Layout**

?

Param

Override  
Common  
Settings

☐ Disable Speakerphone

☐ Disable Speakerphone and Headset

Forwarding Delay\*



PC Port *	Enabled	
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Auto Line Select*	Disabled	
Web Access*	Disabled	<input type="checkbox"/>
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	PC Controlled	
Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Disabled	
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server		
Advertise G.722 Codec*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID		
LLDP Power Priority*	Unknown	
Wireless Headset Hookswitch Control*	Disabled	
IPv6 Load Server		<input type="checkbox"/>
IPv6 Log Server		
802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	
Minimum Ring Volume*	0-Silent	
HTTPS Server*	http and https Enabled	
Handset/Headset Monitor*	Disabled	
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>



## Configuration of MGCP FAX Gateway

Navigation Path: Device → Gateway

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**Gateway Configuration** Related Links: [Back To Find/List](#)

Save Delete Reset Apply Config Add New

**Status**  
 Status: Ready

**Gateway Details**

Product	Cisco 3825
Gateway	3825DSPfarm.pbxmlab.org
Protocol	MGCP
Device is not trusted	
Domain Name*	<input type="text" value="3825DSPfarm.pbxmlab.org"/>
Description	<input type="text" value="3825 on bench 8 - 172.20.236.101"/>
Cisco Unified Communications Manager Group*	<input type="text" value="Default"/>

**Configured Slots, VICs and Endpoints**

Module in Slot 0 NM-4VWIC-MBRD

Subunit 0 VIC2-2FXO 0/0/ 0 0/0/ 1

Subunit 1 VIC2-2FXS 0/1/ 0 0/1/ 1

Subunit 2 < None >

Subunit 3 < None >

Module in Slot 1 NM-HDV2-2PORT-T1

Subunit 0 < None > Begin Port

Subunit 1 < None > Begin Port

Module in Slot 2 NM-HDV

Subunit 0 < None >

**Product Specific Configuration Layout**

Global ISDN Switch Type 4ESS

Switchback Timing\* Graceful

Switchback uptime-delay (min)

Switchback schedule (hh:mm)

Type Of DTMF Relay\* Current GW Config

Modem Passthrough\* Enable

Cisco Fax Relay\* Disable

T38 Fax Relay\* Enable

RTP Package Capability\* Enable

MT Package Capability\* Enable

RES Package Capability\* Enable

PRE Package Capability\* Enable

SST Package Capability\* Enable

RTP Unreachable OnOff\* Enable

RTP Unreachable timeout (ms)\*

RTCP Report Interval (secs)\*


Simple SDP\* Disable

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## Configuration of MGCP FAX Gateway Analog Endpoint

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Gateway Configuration

Related Links: Back to MGCP Configuration

Save Delete Reset Apply Config Add New

**Status**  
Status: Ready

**Directory Number Information**  
[798 Line \[1\] - 5014 in phones](#)

**Device Information**

Product	Cisco MGCP FXS Port
Gateway	3825DSPfarm.pbxmlab.org
Device Protocol	Analog Access
⚠ Device is not trusted	
Registration	Registered with Cisco Unified Communications Manager CM-Polaris
IP Address	172.20.236.101
End-Point Name *	AALN/S0/SU1/0@3825DSPfarm.pbxmlab.org
Description	AALN/S0/SU1/0@3825DSPfarm.pbxmlab.org
Device Pool*	G711_Pool
Common Device Configuration	< None >
Media Resource Group List	MRGL_Polaris
Packet Capture Mode*	None
Packet Capture Duration	0
Calling Search Space	rp_phones
AAR Calling Search Space	rp_phones
Location*	Hub_None
AAR Group	< None >
Network Locale	< None >
Use Trusted Relay Point*	Default
Geolocation	< None >
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
<input type="checkbox"/> Hot line Device	

**Multilevel Precedence and Preemption (MLPP) Information**

MLPP Domain	< None >
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

**Port Information (Loop Start)**

Port Direction*	Bothways
Attendant DN*	5015
Prefix DN	
<input type="checkbox"/> Unattended Port	

**Product Specific Configuration Layout**

?

Hookflash Timer (50-1550ms)*	50
Guard-Out Timer (300-3000ms)*	1000
Inter-digit Duration Timer (50-500 ms)*	100
Input Gain (-6..14 db)*	0
Output Attenuation (-6..14 db)*	0
Echo Cancellation Enable*	Enable
Echo Cancellation Coverage (ms)*	64
Ring Number*	Default
Impedance*	Default GW config



## Acronyms

Acronym	Definitions
CCBS	Call Completion to Busy Subscriber
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol



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