Aastra MX-ONE (Ericsson MD110) version 3.2 SP1 to Cisco Unified Communications Manager 7.0 using Cisco Unified Border Element 1.2 (CUBE) for SIP-to-SIP interworking

April 6th 2009, Rev# 2

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Introduction

- This application note describes the necessary steps and configurations for connectivity between an Ericsson (Aastra) MD110 (MX-One) Version 3.2 SP1 and a Cisco Unified Communications Manager (CUCM) 7.0 using CUBE 1.2 with SIP trunks.

- The network topology diagram (Figures 1) shows the test setup for end-to-end interoperability between Cisco Unified Communications Manager Release 7.0 connected to the Aastra MX-One (Ericsson MD110) PBX via Cisco UBE 1.2 using SIP trunks. A gigabit Ethernet port is used as the SIP trunking interface. Features tested are basic call, 3-way (ad-hoc) conference, call transfer (attended and unattended), call forward (all, busy and no answer), hold/resume, and DTMF internetworking.

- During testing, a Cisco 3825 voice gateway was used to run the CUBE features set, however other Cisco voice gateways can be used and the decision to choose what Cisco gateway model to use is left to the customer. The customer should choose a Cisco IOS gateway model based on the capabilities and the capacity that will be required based on the planned network deployment. Here is a list of Cisco IOS products capable of running CUBE.
  
  Cisco 2800 Series Integrated Services Routers  
  Cisco 3800 Series Integrated Services Routers  
  Cisco 7200VXR Routers  
  Cisco 7301 Routers  
  Cisco AS5350XM Universal Gateway  
  Cisco AS5400XM Universal Gateway
**Network Topology**

![Network Topology Diagram]

**Figure 1.** Basic Call Setup
Capabilities

- Voice calls including supplementary services can be successfully established between phones controlled by the Ericsson and IP phones controlled by the Cisco Unified Communications Manager.

Limitations

- Connected name and connected number cannot be presented during a call. As of the tested software the Aastra MX-ONE PBX does not support P-asserted-Id (PAI) nor Remote-party-Id (RPID) SIP headers.

- During external call transfer, name and number updates are not supported. PAI and RPID are not supported on Aastra PBX, as of the tested software.

- The Aastra PBX does not support external call forward unconditional (Follow-me across SIP trunk).

- A call forward busy or no answer, initiated from a user on the Aastra PBX, includes a History-info SIP header “without” a cause code. The omission of the cause code from the History-info header prevents CUBE from converting the History-info header to a Diversion SIP header. Due to this caveat centralized voice mail integration, using Cisco Unity, Unity Connection or Unity Express, is not possible. Cisco’s UM platforms require the SIP Diversion header for correct functionality.

- DTMF interwork limitation: CUBE translates SIP INFO to RFC2833, but cannot translate RFC2833 to SIP INFO. Since the Aastra PBX supports SIP INFO and does not support RFC2833, DTMF interworking will not work when a CUCM IP phone user attempts to send DTMF towards the Aastra PBX.

- When a user on the CUCM side invokes call hold feature CUCM sends a Re-INVITE with Audio-attribute “inactive” to stop RTP media from flowing. When the call resumes, CUCM sends SDP’less Re-INVITE and requires a 200OK response with SDP fresh offer listing capable codecs with attribute equal to sendrecv. Aastra PBX does not support the SDP’less Re-INVITE and returns a 200K message with Audio-attribute remaining in “inactive” state. The workaround for this limitation is to alter the Audio-attribute which is possible with CUBE using SIP profiles. The initial INVITE sent from CUCM, when the IP phone user places the call on hold, is modified by CUBE at the SDP level so that the “attribute” value is changed from “inactive” to “sendrecv, before the INVITE is forwarded to the Aastra PBX. Using this workaround, when call is RESUMED, voice is established both ways. See configuration section for details.

- Applying the SIP profiles recommended above to address the limitation of SDP’less Re-INVITE during mid-call features causes a corner case failure during a three-way conference that is initiated by a Aastra side phone user. The three-way conference call is successful but is not able to sustain when the conference initiator drops. One of the SIP profiles needed for MoH to work during a hold is to translate the Audio attribute in response 200 OK from “sendonly” to “sendrecv”. This SIP profile cli should be removed if it is required for the remaining parties of a terminated three-way conference to remain connected, but be aware that CUCM MoH will no longer work. (Scenario: Aastra PBX being the conferencing leader with the other two CUCM conferencing members and then conferencing leader dropping out.) With this SIP profile, voice is established only one way. See configuration section for details.
**System Components**

**Hardware Components**

- Cisco MCS 7800 Unified Communications Manager Appliance
- Cisco 3825 voice gateway
- 2 Cisco Unified IP phone 7960 configured as SCCP phones
- 2 Cisco Unified IP phone 7970 configured as SIP phones
- Aastra MX-One (Ericsson MD110) PBX

**Software Requirements**

The following software is required:

- Cisco Unified Communications Manager Release 7.0
- Cisco IOS Release 12.4(22)T
- Ericsson MD100 MX-One Version 3.2 (See configuration section for details)
Features

This section lists supported and unsupported features.

Features Supported

- Basic calls
- CLIP-Calling line (Number) identification presentation
- CLIR-Calling line (Number) identification restriction
- CNIP-Calling name identification presentation
- CNIR-Calling name identification restriction
- Consultation transfer – Local and Network/External (See Limitations section for details.)
- Early Attended transfer – Local and Network/External (See Limitations section for details.)
- Call forward unconditional by join – Local
- Call forward unconditional by join – Network/External (See Limitations section for details.)
- Call forward busy by join – Local
- Call forward busy by join – Network/External
- Call forward no reply by join – Local
- Call forward no reply by join – Network/External
- Hold and resume
- DTMF interworking (see limitations section for details)
- 3-way conference (ad-hoc)

Features Not Supported (See limitations)

- COLP-Connected line (Number) identification presentation
- COLR- Connected line (Number) identification restriction
- CONP-Connected name identification presentation
- CONR- Connected name identification restriction
- Centralized Message center voicemail integration
Configuration

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

Configuring the Aastra MX-One

1. System Software Release
2. ROCAI
3. RODAI
4. ROEQI
5. RODDI
System software release:

> ts_about;
====== MX-ONE Telephony System ======
Version: 3.2 SP1 build16

RPM Packages

Telephony Server 12.45.6:
  eri_sn-12.45.6-MR
Media Gateway File system 2.0:
  egx_rfs-2.0-1
Media Gateway 10.15:
  egx_sw-10.15-1
Media Gateway Classic 1.4_5:
  lsue_sw-1.4_5-1
Manager 8.48.1:
  eri_om-8.48.1-1

ROCAI:
Route category initiate
Set up internal characteristics for the route; for example, traffic direction, services, or bearer capabilities.

ROCAP:
Route category data print
> ROCAP:ROU=ALL;
ROUTE CATEGORY DATA
ROU SEL TRM SERV NODG DIST DISL TRAF SIG BCAP
10 7110000000700010 4 3110000010 0 30 128 00151515 111110000011 001100
50 7110000000000010 5 2110030000 0 30 128 03151515 111110000031 001100
70 7110000000000010 5 2110030000 0 30 128 03151515 111110000031 001100
100 7110000000000010 5 3110000001 0 30 128 03151515 111110000091 001100
101 7110000000000010 4 3110000001 0 30 128 00151515 1111100000A1 001100
102 7110000000000010 4 3110000001 0 30 128 00151515 1111100000A1 001100
103 7110000000000010 4 3110000001 0 30 128 00151515 1111100000A1 001100
104 7110000000000010 4 3110000001 0 30 128 00151515 1111100000A1 001100
END

RODAI:
Route data initiate

RODAC:
Route data print
> RODAP:ROU=104;
ROUTE DATA
ROU TYPE VARC VARI VARO FILTER
104 TL66 H'00000000 H'00000000 H'00000011 NO
END

SIP Route:
> sip_route -set -route 104 -uristring "sip:?@172.20.8.30"

> sip_route -print -route 104

Route data for SIP destination

Route : 104
Protocol = udp
Invite URI string 0 = sip:?@172.20.8.30
Invite URI string 1 =
Invite URI string 2 =
Invite URI string 3 =
Invite URI string 4 =
Invite URI string 5 =
Invite URI string 6 =
Invite URI string 7 =
From URI string 0 =
From URI string 1 =
From URI string 2 =
From URI string 3 =
From URI string 4 =
From URI string 5 =
From URI string 6 =
From URI string 7 =
Port = default
Accepted calls = ALL
Priority = 255
Match this =
context B-party 0 =
context B-party 1 =
context B-party 2 =
context B-party 3 =
context B-party 4 =
context B-party 5 =
context B-party 6 =
context B-party 7 =
context A-party 0 =
context A-party 1 =
context A-party 2 =
context A-party 3 =
context A-party 4 =
context A-party 5 =
context A-party 6 =
context A-party 7 =
Authentication username =
Password =
Registration type = NO_REG
Trusted privacy domain = no
SOS a subscriber number =
SOS type of number =
SOS destination number =

ROEQI:

1 CUBE’s IP address
Route equipment initiate

**ROEDP:**
Route equipment data print
> ROEDP:ROU=104,TRU=ALL;
ROUTE EQUIPMENT DATA
<table>
<thead>
<tr>
<th>ROU</th>
<th>TRU</th>
<th>EQU</th>
<th>IP ADDRESS</th>
<th>SQU</th>
<th>INDDAT</th>
<th>CNTRL</th>
</tr>
</thead>
<tbody>
<tr>
<td>104</td>
<td>001-1</td>
<td>H'000000000000</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
END

**RODDP:**
Route external destination date initiate

**RODDP:**
Route external destination data print
> RODDP:DEST=ALL;
EXTERNAL DESTINATION ROUTE DATA
<table>
<thead>
<tr>
<th>DEST</th>
<th>DRN</th>
<th>ROU</th>
<th>CHO</th>
<th>CUST</th>
<th>ADC</th>
<th>TRC</th>
<th>SRT</th>
<th>NUMACK</th>
<th>PRE</th>
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<tbody>
<tr>
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<td>000000000000025010500110000</td>
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<td>170</td>
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<td>0</td>
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<td></td>
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<td></td>
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<td>42</td>
<td>70</td>
<td>150</td>
<td>000000000000025010500110100</td>
<td>0</td>
<td>1</td>
<td>0</td>
<td></td>
<td></td>
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<td>50</td>
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<td>70</td>
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<td>0</td>
<td>1</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
END

**KSDDP:**
Key system directory data print

> KSDDP:DIR=ALL;
KEY SYSTEM DIRECTORY DATA
<table>
<thead>
<tr>
<th>DIR</th>
<th>CUST</th>
<th>EQU</th>
<th>CAT</th>
<th>ADN</th>
<th>ODN</th>
<th>CALALT</th>
<th>TIMER</th>
</tr>
</thead>
<tbody>
<tr>
<td>1251</td>
<td>001-0-22-00</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1252</td>
<td>001-0-22-01</td>
<td>-</td>
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<td></td>
<td></td>
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</tr>
<tr>
<td>1253</td>
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<td>-</td>
<td>10</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>1255</td>
<td>001-0-22-04</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1256</td>
<td>001-0-22-05</td>
<td>-</td>
<td>10</td>
<td>0</td>
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<td></td>
</tr>
<tr>
<td>1257</td>
<td>001-0-22-06</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1258</td>
<td>001-0-22-07</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
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</tr>
<tr>
<td>1259</td>
<td>001-0-22-08</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1260</td>
<td>001-0-22-09</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1261</td>
<td>001-0-22-10</td>
<td>-</td>
<td>10</td>
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<td></td>
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<tr>
<td>1262</td>
<td>001-0-22-11</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>1265</td>
<td>001-0-22-14</td>
<td>-</td>
<td>10</td>
<td>0</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
END
### Calling name/connected name restrictions

To configure Calling/Connected Name and Number Restricted, use the following command:

```plaintext
<KSCAP:DIR=1251&1252;
KEY SYSTEM CATEGORY PRINT
DIR TRAF SERV CDIV ROC ITYPE TRM ADC LANG BSEC
1251 03151515 0211120700 011151111 720004 20 0 00100013010000 0 0
1252 03151515 0215120700 011151111 720004 20 0 00100013010000 0 0
END
```

To configure Calling/Connected Name and Number Allowed, use the following command:

```plaintext
<KSCAP:DIR=1251&1252,ADC=00
010013010000;
```

### Call diversion on busy/no reply

To enable/disable diversion on busy/no reply, use the following command:

```plaintext
> CDIDP:DIR=1251&1252;
```

To enable/disable diversion on busy/no reply, use the following command:

```plaintext
> CDINI:DIR=1251,DIV=64001;       //Call diversion individual number initiate
> CDINE:DIR=1251;                           //Call diversion individual number end
```
Configuring the Cisco Unified Communications Manager

1. Cisco Unified Communications Manager Version
2. Region configuration
3. Device pool configuration
4. Media Termination Point registration
5. Conference Bridge registration
6. Media Resource Group List configuration
7. Media Resource Group configuration
8. SIP Trunk Configuration
9. Cisco IP Phone 7960 SCCP Configuration
10. Cisco IP Phone 7971 SIP Configuration
11. Enbloc Route Configuration
12. Calling name and restriction configuration
13. Connected name and number restriction configuration
### Configuring the Region

**Navigation Path:** System -> Region

#### Region Information

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Code</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phone971</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>phone9729</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**NOTE:** Regions(s) not displayed

#### Modify Relationship to Other Regions

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Code</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.711</td>
<td>64</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>phone971</td>
<td>G.722</td>
<td>64</td>
<td>Keep Current Setting</td>
</tr>
</tbody>
</table>

---

* indicates required item.

**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.*
### Configuration of Device pool (1 of 2)

**Navigation Path:** System -> Device pool

| Device Pool Configuration
<table>
<thead>
<tr>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Status</strong></td>
</tr>
<tr>
<td><em>Status: Ready</em></td>
</tr>
</tbody>
</table>

#### Device Pool Information

- **Device Pool:** SIP_trunk (7 members***)

#### Device Pool Settings

- **Device Pool Name:** SIP_trunk
- **Cisco Unified Communications Manager Group:** Default
- **Calling Search Space for Auto-registration:** Default
- **Reverted Call Setup Priority:** Default
- **Local Route Group:** Default

#### Roaming Sensitive Settings

- **Default Time Group:** < None >
- **Region:** < None >
- **Media Resource Group List:** < None >
- **Network Locales:** < None >
- **SIP Reference:** Use Default Gateway
- **Connection Monitor Duration:** < None >
- **Single Button Barge:** Default
- **Join Across Lines:** Default
- **Physical Location:** < None >
- **Device Mobility Group:** < None >

#### Device Mobility Related Information****

- **Device Mobility Calling Search Space:** < None >
- **All Calling Search Space:** < None >
- **All Group:** < None >
- **Call Router Configuration:** < None >
Configuration of Device pool (2 of 2)

Navigation Path: System -> Device pool

[Image of Cisco Unified CM Administration interface]

- **Device Mobility Related Information**: [Details of device mobility settings]
  - Device Mobility Calling Search Space
  - ARS Calling Search Space
  - Ask Group
  - Calling Party Transformation CSS
  - Called Party Transformation CSS

- **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 used.

  - Incoming Calling Party National Number Prefix
  - Incoming Calling Party International Number Prefix
  - Incoming Calling Party Unknown Number Prefix
  - Incoming Calling Party Subscriber Number Prefix

- **Related Links**: [Back To Find List]

---

### Important notices:

- The image contains a network of devices, with each device represented by a small icon. The Cisco logo is visible in the top left corner.

- The device pool configuration interface includes fields for setting connection monitor duration, single button range, join across access, physical location, and device mobility group.

- The device mobility settings include options for calling search spaces, group settings, and transformation CSSes.

- The incoming calling party settings allow for configuration of national, international, unknown, and subscriber number prefixes.

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**Configuration of the conference bridge**

**Navigation Path:** Media Resources -> Conference Bridge

---

**Note:** This configuration page refers to the conference bridge configured on the CUBE. Configuration is below:

```plaintext
scp ccm group 1
bind interface GigabitEthernet0/0
associate ccm 1 priority 1
associate profile 2 register cfb0021d8150930
!
dspfarm profile 2 conference
codec g711ulaw
codecs g711alaw
codecs g729ar8
codecs g729abr8
codecs g729r8
codecs g729br8
maximum sessions 6
associate application SCCP
```
### Media Resource Group List configuration

**Navigation Path:** Media Resources -> Media Resource Group List

<table>
<thead>
<tr>
<th>Media Resource Group List Configuration</th>
<th>Related Links: Back To FindList</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Status</strong></td>
<td></td>
</tr>
<tr>
<td>Status: Ready</td>
<td></td>
</tr>
<tr>
<td><strong>Media Resource Group List Status</strong></td>
<td></td>
</tr>
<tr>
<td>Media Resource Group List: MRGL (used by 12 devices)</td>
<td></td>
</tr>
<tr>
<td><strong>Media Resource Group List Information</strong></td>
<td></td>
</tr>
<tr>
<td>Name: MRGL</td>
<td></td>
</tr>
<tr>
<td><strong>Media Resource Groups for this List</strong></td>
<td></td>
</tr>
<tr>
<td>Available Media Resource Groups:</td>
<td></td>
</tr>
<tr>
<td>[SP_trunk_MRGL]</td>
<td></td>
</tr>
<tr>
<td>Selected Media Resource Groups:</td>
<td></td>
</tr>
<tr>
<td>[MRG]</td>
<td></td>
</tr>
</tbody>
</table>

- Save    | Delete    | Copy    | Add New |

* indicates required item.
### Media Resource Group configuration

**Navigation Path:** Media Resources -> Media Resource Group

#### Media Resource Group Status

- Media Resource Group: MRG (used by 12 devices)

#### Media Resource Group Information

- **Name:** MRG
- **Description:**

#### Devices for this Group

- **Available Media Resources:**
  - CPA, 2
  - MTP
  - MTS:001:8150930

- **Selected Media Resources:**
  - CPA, 2 (ANN)
  - MTP (MHD)
  - MTS:001:8150930 (CM)

- Use Multicast for MOH Audio (if at least one multicast MOH resource is available)

---

* indicates required item.

**Includes:** Announcers (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MHD), and Transcoders (XCODE).
SIP Trunk Configuration (1 of 3)

Navigation Path: Device -> Trunk

Status
- Update successful

Device Information
- SIP Trunk: SIP
- Device Protocol: SIP
- Device Name: CUBE_TRUNK
- Description: SIP_TRUNK_TO_CUBE
- Device Pool: SIP_trunk
- Common Device Configuration: < None >
- Call Classification: < None >
- Media Resource Group List: Other
- Location: Hub
- AAR Group: None
- Packet Capture Mode: None
- Packet Capture Duration: 0

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

Incoming Calling Party Unknown Number Prefix
- Default

Multilevel Precedence and Preemption (MLPP) Information
- Default
## Trunk Configuration (2 of 3)

**Navigation Path:** Device -> Trunk

### Multilevel Precedence and Preemption (MLPP) Information
- **MLPP Domain:** [Select]

### Call Routing Information
- **Remote-Party-ID**
  - [On]
  - [Off]
- **Asserted-Identity**
- **Asserted-Type**
  - Default
- **SIP Privacy**
  - Default

### Inbound Calls
- **Significant Digits**
  - [All]
- **Connected Line ID Presentation**
  - Default
- **Connected Name Presentation**
  - Default
- **Calling Search Space**
  - [Default]
- **AAR Calling Search Space**
  - [Default]
- **Prefixed DN**
  - [Default]
- **Redirecting Divergence Header Delivery - Inbound**

### Outbound Calls
- **Called Party Transformation CSS**
  - [Default]
- **Use Device Pool Called Party Transformation CSS**
  - [Default]
- **Calling Party Transformation CSS**
  - [Default]
- **Use Device Pool Calling Party Transformation CSS**
- **Calling Party Selection**
  - [Originator]
- **Calling Line ID Presentation**
  - [Default]
- **Calling Name Presentation**
  - [Default]
- **Caller DN**
  - [Default]
- **Caller Name**
  - [Default]
- **Redirecting Divergence Header Delivery - Outbound**

---

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Page 21 of 48
Trunk Configuration (3 of 3)

Navigation Path: Device -> Trunk

[Image of the Cisco Unified CM Administration interface]

- **Called Party Transformation CSS**: <None>
- **User Device Pool Called Party Transformation CSS**: <None>
- **Calling Party Transformation CSS**: <None>
- **User Device Pool Calling Party Transformation CSS**: <None>
- **Calling Party Selection**: Originator
- **Calling User ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Caller ID ON**: False
- **Caller Name**: False
- **Redirecting Diversion Header Delivery - Outbound**: False

**SIP Information**

- **Destination Address**: 172.168.28.28
- **Destination Port**: 5060
- **MTP Preferred Originating Codes**: G723/G729a
- **Presence Group**: Standard Presence Group
- **SIP Trunk Security Profile**: Non-Secure SIP Trunk Profile
- **Reverting Calling Search Space**: <None>
- **Out-Of-Dialing Calling Search Space**: <None>
- **SUBSCRIBE Calling Search Space**: <None>
- **SIP Profile**: Standard SIP Profile
- **OTMF Signaling Method**: COB and RFC 2933

* indicates required item.
** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Cisco IP Phone 7960 SCCP Configuration (1 of 7)

Navigation Path: Device -> Phone

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

### Phone Type
- **Product Type:** Cisco 7960
- **Device Protocol:** SCCP

### Device Information
- **Registration:** Registered with Cisco Unified Communications Manager CM-CMVSA
- **IP Address:** 172.20.0.65
- **MAC Address:** 00DA4B05529
- **Description:** Auto 4611
- **Device Pool:** phones
- **Common Device Configuration:** < None >
- **Phone Button Template:** Standard 7960 SCCP
- **Softkey Template:** Standard User CALLBACK
- **Common Phone Profile:** Standard User CALLBACK
- **Calling Search Space:** < None >
- **Interdigit Tone:** < None >
- **Interdigit Tone Time:** < None >
- **Interdigit Tone Volume:** < None >
- **Ringing:** < None >
- **Ring Volume:** < None >
- **Ring Time:** < None >
- **Ring Type:** < None >
- **Ring Detect Tone Volume:** < None >
- **Ring Detect Tone Time:** < None >
- **Ring Detect Tone:** < None >
- **Ring Detect Time:** < None >
- **Ring Detect:** < None >
- **Auto Answer:** < None >
- **Auto Answer Time:** < None >
- **Auto Answer Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
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- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
- **Auto Answer Tone Time:** < None >
- **Auto Answer Tone Volume:** < None >
- **Auto Answer Tone:** < None >
Cisco IP Phone 7960 SCCP Configuration (2 of 7)

Navigation Path: Device -> Phone

### Phone Configuration

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Idle)</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>Is Active</td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td></td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td></td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td></td>
</tr>
<tr>
<td>Logged Into Hunt Group</td>
<td></td>
</tr>
<tr>
<td>Remote Device</td>
<td></td>
</tr>
</tbody>
</table>

### Protocol Specific Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td></td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 7960 - Standard SCCP Non-Secure Profile</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>Require OTHY Reception</td>
<td></td>
</tr>
<tr>
<td>RSCC/383 Disabled</td>
<td></td>
</tr>
</tbody>
</table>

### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
<td>No Pending Operation</td>
</tr>
<tr>
<td>Authentication Mode</td>
<td>By neat String</td>
</tr>
<tr>
<td>Authentication String</td>
<td></td>
</tr>
<tr>
<td>Key Size (bits)</td>
<td>2048</td>
</tr>
<tr>
<td>Operation Completed By</td>
<td>09/09/08 16:32 (YYYY/MM/DD hh:mm)</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>Note</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7960 SCCP Configuration (3 of 7)
Navigation Path: Device -> Phone

<table>
<thead>
<tr>
<th>Expansion Module Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1:</td>
</tr>
<tr>
<td>Module 1 Load Name:</td>
</tr>
<tr>
<td>Module 2:</td>
</tr>
<tr>
<td>Module 2 Load Name:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>External Data Locations Information (Leave blank to use default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory:</td>
</tr>
<tr>
<td>Messages:</td>
</tr>
<tr>
<td>Services:</td>
</tr>
<tr>
<td>Authentication Server:</td>
</tr>
<tr>
<td>Proxy Server:</td>
</tr>
<tr>
<td>Idle:</td>
</tr>
<tr>
<td>Idle Timer (seconds):</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Extension Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility:</td>
</tr>
<tr>
<td>Log Out Profile:</td>
</tr>
<tr>
<td>Log in Time:</td>
</tr>
<tr>
<td>Log out Time:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SIP Profile Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>MUPI Domain:</td>
</tr>
<tr>
<td>MUPI Identification:</td>
</tr>
<tr>
<td>MUPI Presence:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb:</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7960 SCCP Configuration (4 of 7)

Navigation Path: Device -> Phone

- **MLPP Information**
  - **MLPP Domain**: <None>
  - **MLPP Indication**: Default
  - **MLPP Preemption**: Default

- **Do Not Disturb**
  - **Do Not Disturb Option**: Ringer Off
  - **DND Incoming Call Alert**: <None>

- **Product Specific Configuration Layout**
  - **Disable Speakerphone**
  - **Enable Speakerphone and Headset**
  - **PC Port**: Enabled
  - **Settings Access**: Enabled
  - **Administrative API**: Enabled
  - **PC Voice VLAN Access**: Enabled
  - **Video Capabilities**: Disabled
  - **Auto Line Select**: Disabled
  - **Web Access**: Enabled

---

* indicates required item.
** indicates device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
***Note: Security Profile Contains Additional CAP Settings.
****Note: A new softkey template without supplementary service softkeys must be used for a protected phone.
Cisco IP Phone 7960 SCCP Configuration (6 of 7)

Navigation Path: Device -> Phone -> Modify/Add Button

Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td></td>
<td>Use System Default</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Secondary Call</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CTI Failure</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

No Answer Ring Duration (seconds)

Call Pickup Group

< None >

MIP Alternate Party Settings

Target (Destination)

MIP Calling Search Space

< None >

MIP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Extension Ring Duration (seconds)

Setting the Hold Extension Ring Duration to zero will disable the feature

Hold Extension Notification Interval (seconds)

Setting the Hold Extension Notification Interval to zero will disable the feature

Display (Internal Caller ID)

PhoneC1

Display text for a line appearance is intended for displaying text such as name instead of a directory number for...
**Cisco IP Phone 7960 SCCP Configuration (7 of 7)**

**Navigation Path:** Device -> Phone -> Modify/Add Button

**Directory Number Configuration**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASCII Display (Internal Caller ID)</td>
<td>PhoneC1</td>
</tr>
<tr>
<td>Line Text Label</td>
<td>PhoneC1</td>
</tr>
<tr>
<td>ASCII Line Text Label</td>
<td>PhoneC1</td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy *</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle) *</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>None</td>
</tr>
</tbody>
</table>

---

**Multiple Call/Call Waiting Settings on Device SEP000A41688539**

- **Note:** The range to select for the Max Number of calls is 1-200
- **Busy Trigger:**
  - Max Call (Less than or equal to Max Calls)

**Forwarded Call Information Display on Device SEP000A41688539**

- **Caller Name**
- **Caller Number**
- **Redirected Number**
- **Dialed Number**

---

**Users Associated with Line**

<table>
<thead>
<tr>
<th>Action</th>
<th>User</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
<td></td>
</tr>
<tr>
<td>Delete</td>
<td></td>
</tr>
<tr>
<td>Reset</td>
<td></td>
</tr>
<tr>
<td>Add</td>
<td></td>
</tr>
</tbody>
</table>

---

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Cisco IP Phone 7971 SIP Configuration (2 of 8)

Navigation Path: Device -> Phone

### Phone Configuration

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone Line Name</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Trust Relay Port</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Idle)</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>

#### Protocol Specific Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MIP Preferred Originating Codes</td>
<td>Default</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 7971 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td>Proxy Server Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SDP</td>
<td>Standard SDP Profile</td>
</tr>
<tr>
<td>Digest User</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Certificates Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Authority</td>
<td></td>
</tr>
<tr>
<td>Certificate Authority Port</td>
<td></td>
</tr>
<tr>
<td>Require DTP Reception</td>
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</tr>
</tbody>
</table>
## Cisco IP Phone 7971 SIP Configuration (5 of 8)

**Navigation Path:** Device -> Phone

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Recording Tone On/Off</td>
<td>Enabled</td>
</tr>
<tr>
<td>Recording Tone Volume</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Enabled</td>
</tr>
<tr>
<td>PSTN*</td>
<td>Disabled</td>
</tr>
<tr>
<td>&quot;more* Soft Key Timer</td>
<td>Enabled</td>
</tr>
<tr>
<td>Auto Call Select*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Voice Server</td>
<td>Enabled</td>
</tr>
<tr>
<td>Advertisement G.722 Codec*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Wideband Headset at Contact*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Wideband Headset 10 Contact*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Wideband Headset 15 Contact*</td>
<td>Disabled</td>
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<tr>
<td>Wideband Headset 20 Contact*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): FC Port*</td>
<td>Disabled</td>
</tr>
<tr>
<td>LLDP Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*</td>
<td>Disabled</td>
</tr>
<tr>
<td>LLDP Discovery Protocol (LLDP): FC Port*</td>
<td>Disabled</td>
</tr>
<tr>
<td>LLDP ESSID*</td>
<td>Unknown</td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td>Normal</td>
</tr>
</tbody>
</table>

* indicates required item.

** Note: Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

** Note: Security Profile Contains Additional CAP Settings.

*** Note: A new softkey template without supplementary service softkeys must be used for a protected phone.
Cisco IP Phone 7971 SIP Configuration (6 of 8)

**Navigation Path:** Device -> Phone -> Modify/Add Button

### Directory Number Configuration

**Status**
- Status: Ready

### Directory Number Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory Number</td>
<td>4000</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; Name &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Alerting Name</td>
<td>Phone2</td>
</tr>
<tr>
<td>ASCII Alerting Name</td>
<td>Phone2</td>
</tr>
<tr>
<td>Associated Devices</td>
<td>SEP001PRD0CD0</td>
</tr>
</tbody>
</table>

**Edit Device**

**Edit Line Appearance**

### Directory Number Settings

- **Voice Mail Profile:** < Name > (Choose <None> to use system default)
- **Calling Search Space:** < Name >
- **Presence Group:** Standard Presence group
- **User Hold MOH Audio Source:** < Name >
- **Network Hold MOH Audio Source:** < Name >
- **Auto Answer:** Auto Answer Off

### AAR Settings

<table>
<thead>
<tr>
<th>AAR</th>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>&lt; Name &gt;</td>
</tr>
</tbody>
</table>

- Retain this destination in the call forwarding history.
## Enbloc Route Configuration 51XX (1 of 2)

**Navigation Path:** Call Routing -> Route/Hunt -> Route Pattern

### Route Pattern Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pattern Definition</td>
<td></td>
</tr>
<tr>
<td>Route Pattern</td>
<td>51XX</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>To-Cube</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>RUEF Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>&lt; None &gt; (Full)</td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
<tr>
<td>Call Classification</td>
<td>Outlet</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td></td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td></td>
</tr>
<tr>
<td>Urgent Priority</td>
<td></td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td></td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- Use Calling Party's External Phone Number Mask:
  - Prefix Digits (Outgoing Calls) |  |
  - Calling Party Transform Mask |  |
  - Calling Name Presentation | Default |
  - Calling Party Number Type | Cisco CallManager |
  - Calling Party Numbering Plan | Cisco CallManager |

**Connected Party Transformations**

- Use Connected Party's External Phone Number Mask:
  - Prefix Digits (Outgoing Calls) |  |
  - Connected Party Transform Mask |  |
  - Connected Name Presentation | Default |
  - Connected Party Number Type | Cisco CallManager |
  - Connected Party Numbering Plan | Cisco CallManager |
### Enbloc Route Configuration 51XX (2 of 2)

**Navigation Path:** Call Routing -> Route/Hunt -> Route Pattern

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Related Links: Back To Find/Save</th>
<th>Go</th>
</tr>
</thead>
</table>

**Calling Party Transformations**

- Use Calling Party's External Phone Number 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**

- Called ID
- Called Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Called Party Number Type
- Called Party Numbering Plan

**ISDN Network-Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
</table>

- Network Service: Not Selected
- Network Service Protocol: Not Selected

---

* indicates required item.
### Calling line name and number restriction configuration

**Navigation Path:** Device -> Trunk

<table>
<thead>
<tr>
<th>Inbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Significant Digits</strong></td>
<td>All</td>
</tr>
<tr>
<td><strong>Connected Line ID Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Connected Name Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>AAA Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Prefix DN</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Redirecting Diversion Header Delivery - Inbound</strong></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Outbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Use Device Pool Calling Party Transformation CSS</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Calling Party Transformation CSS</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Use Device Pool Calling Party Transformation CSS</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Calling Party Selection</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Calling Line ID Presentation</strong></td>
<td>Restricted</td>
</tr>
<tr>
<td><strong>Calling Name Presentation</strong></td>
<td>Restricted</td>
</tr>
<tr>
<td><strong>Caller ID DN</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Caller Name</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Redirecting Diversion Header Delivery - Outbound</strong></td>
<td></td>
</tr>
</tbody>
</table>

### SIP Information

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>172.16.8.30</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Destination Port</strong></td>
<td>5060</td>
</tr>
<tr>
<td><strong>MTP Preferred Originating Codec</strong></td>
<td>G729a</td>
</tr>
<tr>
<td><strong>Destination Codec</strong></td>
<td></td>
</tr>
</tbody>
</table>
**Connected name and number restriction configuration**

**Navigation Path:** Device -> Trunk

<table>
<thead>
<tr>
<th><strong>Inbound Calls</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits*</td>
<td>All</td>
</tr>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Restricted</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Restricted</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAP Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Outbound Calls</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party selection*</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

* SIP Information

---

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Configuring CUBE

CUBE#show version
Cisco IOS Software, 3800 Software (C3825-ADVENTERPRISEK9_IVS-M), Version 12.4(22)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2008 by Cisco Systems, Inc.
Compiled Fri 10-Oct-08 06:43 by prod_rel_team

ROM: System Bootstrap, Version 12.4(13r)T, RELEASE SOFTWARE (fc1)

CUBE uptime is 5 hours, 27 minutes
System returned to ROM by reload at 21:18:11 UTC Mon Apr 6 2009
System image file is “flash:c3825-adventerprisek9_ivs-mz.124-22.T.bin”

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:http://www.cisco.com/wwl/export/crypto/tool/stqr.html

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco 3825 (revision 1.2) with 225280K/36864K bytes of memory.
Processor board ID FTX1227A3CR
2 Gigabit Ethernet interfaces
4 Channelized (E1 or T1)/PRI ports
1 Virtual Private Network (VPN) Module
DRAM configuration is 64 bits wide with parity enabled.
479K bytes of NVRAM.
62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

CUBE#show run
Building configuration...

Current configuration : 3182 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE
!
boot-start-marker
boot-end-marker
!
! logging message-counter syslog
logging buffered 10000000
no logging console
enable password cisco
!
no Aa new-model
!
dot11 syslog
ip source-route
ip cef
!
no ip domain lookup
no ipv6 cef
!
multilink bundle-name authenticated
!
voice-card 0
dspfarm
dsp services dspfarm
!
voice service voip
  address-hiding
  allow-connections sip to sip
  redirect ip2ip
  h323
  sip
  history-info
  midcall-signaling passthru
  g729 annexb-all
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class sip-profiles 1
  request REINVITE sdp-header Audio-Attribute modify "inactive" "sendrecv"\(^2\)
  request ACK sdp-header Audio-Attribute modify "sendonly" "sendrecv"\(^1\)
  response 200 sdp-header Audio-Attribute modify "sendonly" "sendrecv"\(^1\)
!
voice translation-rule 5
  rule 5 /^5+/ //
!
voice translation-rule 6
  rule 6 /^6+/ //
!
voice translation-profile out-to-CallManager
  translate called 6
!
voice translation-profile out-to-Ericsson
  translate called 5
!
\(^2\) This SIP profile CLI is required for Call transfer and hold/resume to function properly
\(^3\) This command is needed for MoH to work. But enabling this command results in one-way conference when the conferencing phone drops out during the Network/External conference and the phone at the Ericsson end is conferencing.
archive
log config
hidekeys
!
interface GigabitEthernet0/0
ip address 172.20.8.30 255.255.255.0
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 172.20.8.1
no ip http server
no ip http secure-server
!
control-plane
!
ccm-manager fax protocol cisco
!
mgcp fax t38 ecm
!
sccp local GigabitEthernet0/0
sccp ccm 172.20.8.254 identifier 1 version 7.0
sccp
!
sccp ccm group 1
bind interface GigabitEthernet0/0
associate ccm 1 priority 1
associate profile 2 register cbf0021d8150930
!
dspfarm profile 2 conference
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729br8
codec g729r8
codec g729br8
maximum sessions 6
associate application SCCP
!
dial-peer voice 1200 voip
description To Ericsson
destination-pattern 12..
voice-class codec 1
session protocol sipv2
session target ipv4:172.20.244.254
dtmf-relay sip-notify
!
dial-peer voice 4000 voip
description To CUCM
destination-pattern 40..
voice-class codec 1
session protocol sipv2
session target ipv4:172.20.8.254
dtmf-relay rtp-nte
!
dial-peer voice 1201 voip
description Translates incoming number out to Ericsson
translation-profile incoming out-to-Ericsson
answer-address 4...
voice-class codec 1
session protocol sipv2
incoming called-number 512..
dtmf-relay rtp-nte
!
dial-peer voice 4001 voip
description Translates incoming number out to CUCM
translation-profile incoming out-to-CUCM
answer-address 1...
voice-class codec 1
session protocol sipv2
incoming called-number 640..
dtmf-relay rtp-nte
!
sip-ua
!
gatekeeper
shutdown
!
line con 0
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
scheduler allocate 20000 1000
end
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANF-PR</td>
<td>Additional Network Feature Path Replacement</td>
</tr>
<tr>
<td>AOC</td>
<td>Advice-of-charge. Information element is sent with the connection setup information for incoming Euro-ISDN connections. The AOC IE is used for call charge calculation.</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CCBS</td>
<td>Call Completion to Busy Subscriber</td>
</tr>
<tr>
<td>CCNR</td>
<td>Call Completion on No Reply</td>
</tr>
<tr>
<td>CFB</td>
<td>Call Forwarding on Busy</td>
</tr>
<tr>
<td>CFNR</td>
<td>Call Forwarding No Reply</td>
</tr>
<tr>
<td>CFU</td>
<td>Call Forwarding Unconditional</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CNIP</td>
<td>Calling Name Identification Presentation</td>
</tr>
<tr>
<td>CNIR</td>
<td>Calling Name Identification Restriction</td>
</tr>
<tr>
<td>COLP</td>
<td>Connected Line (Number) Identification Presentation</td>
</tr>
<tr>
<td>COLR</td>
<td>Connected Line (Number) Identification Restriction</td>
</tr>
<tr>
<td>CONP</td>
<td>Connected Name Identification Presentation</td>
</tr>
<tr>
<td>CONR</td>
<td>Connected Name Identification Restriction</td>
</tr>
<tr>
<td>CT</td>
<td>Call Transfer</td>
</tr>
<tr>
<td>MWI</td>
<td>Message Waiting Indicator</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
</tbody>
</table>
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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>HArlbergpark</td>
<td>170 West Tasman Drive</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>HArlbergweg  13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td>USA</td>
<td>1101 CH Amsterdam</td>
<td>USA</td>
<td>Singapore 068912</td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>www-europe.cisco.com</td>
<td>Tel: 408 526-7660</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td></td>
<td>Fax: 408 527-0883</td>
<td>Fax: +65 317 7799</td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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