

Avaya CS1000E Release 7.65 using SIP trunk to Cisco Unified Communications Manager Release 10.5

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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.5 to interoperate with the Avaya CS1000E 7.65 using SIP Early-Offer.

Key Results

- Basic call, call transfer, call forwarding, conference call, and hold and resume work successfully.
- Centralized voicemail, using Unity Connection server integrated to Cisco UCM with SCCP and SIP was tested. This voicemail solution can provide centralized voicemail services, supporting both Avaya and Cisco end-users.

The following items were tested:

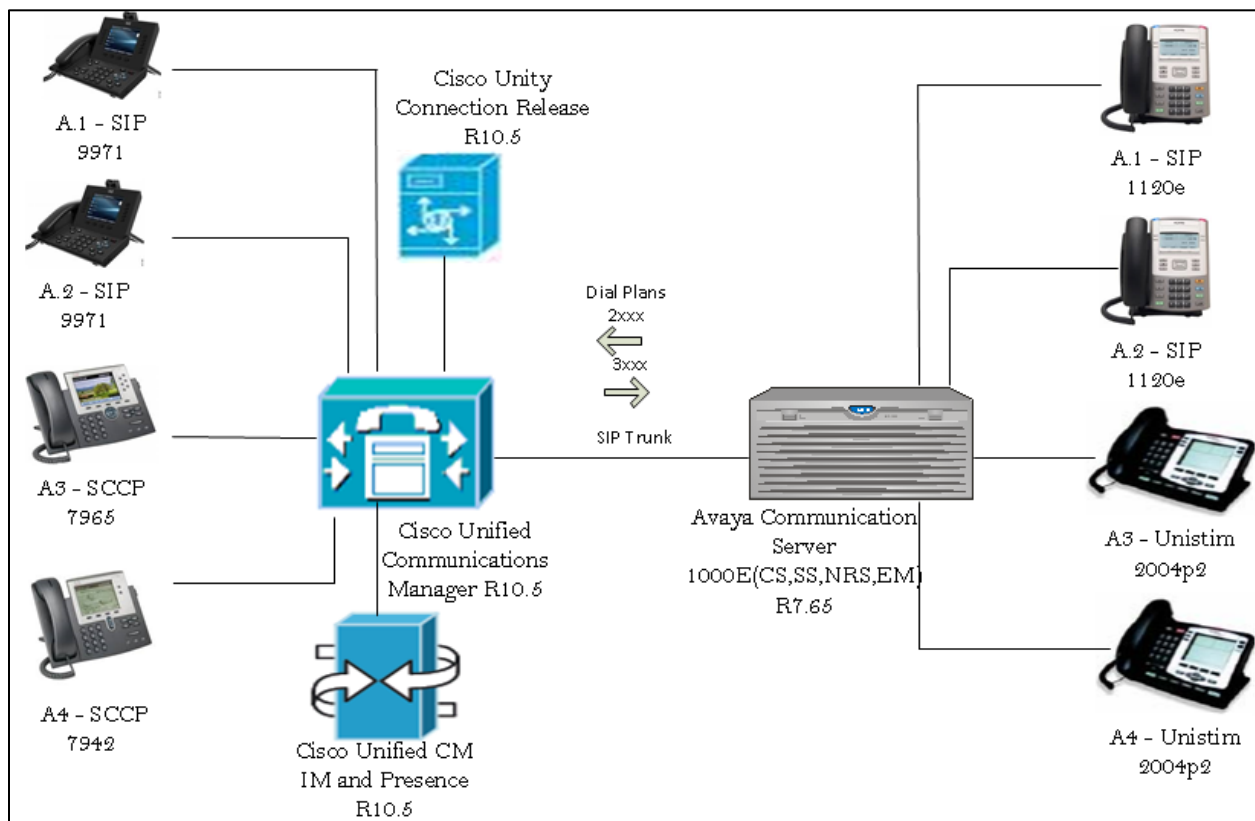
- Basic call between the two systems and verification of voice path, using both SIP and Unistim phones on the Avaya side, SIP and SCCP IP phones on the Cisco side
- CLIP/CLIR/CNIP/CNIR features: calling party name and number delivery (allowed and restricted)
- COLP/CONP/COLR/CONR features: connected name and number delivery (allowed and restricted)
- Call transfer: attended, and early attended
- Alerting Name Identification
- Call Park
- Call forwarding: call forward unconditional (CFU), call forward busy (CFB), and call forward no answer (CFNA)
- Hold and resume with music on hold
- Three-way conferencing
- Voice messaging and MWI activation-deactivation
- Audio Codec Preference List
- DTMF-relay via RFC2833 -verification of DTMF- relay by accessing each other's VM system and responding to prompts using the keypad to send RTP Telephone Event (RFC2833) of digits pressed

Listed below are the highlights of the integration issues:

- Basic calls worked from Cisco UCM to Avaya CS1000E and vice versa. The Avaya CS1000E only supports early offer to set its media attribute to send/receive mode. Thus, for calls from Cisco UCM to Avaya CS1000E, the Cisco UCM must be set to send SIP Invite with SDP. This will ensure two-way audio once the call is connected.
- CLIR/CNIR - Restriction of calling number on Avaya CS1000E Unistim phones is achieved by configuring the Avaya station's class of service. Setting the class of service (CLS) to DDGD sets the SIP P-Asserted Identity setting to privacy = id. This restricts the calling number information. Setting the class of service to NAMD sets the SIP P-Asserted Identity setting to privacy = user. Restriction of calling name and number on the Cisco UCM can be done on the Route Pattern or SIP Trunk page. Calling name and number restrictions are honored by both sides.

Network Topology

Basic Call Setup



System Components

Hardware Requirements

The following hardware was used:

- Cisco Unified Communications Manager: Cisco UCS-C240 VMware ESXI.5.5.0
- Cisco Unity Connection Cisco: UCS-C240 VMware ESXI.5.5.0 server
- Cisco Unified CM IM and Presence: Cisco UCS-C240 VMware ESXI.5.5.0
- Cisco 7965, 7942, 9971 IP phones
- Avaya Communication Server 1000E/CPPM Linux , CPPM - Pentium M 1.4 GHz(NTDW61BAE5)
- Avaya Media Gateway Controller - NTDW60BB
- Avaya 2004P2, 1120e IP Phones

Software Requirements

The following software is required:

- Cisco UCSC-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Cisco Unified Communications Manager release 10.5 -10.5.1.10000-7
- Cisco Unified Communications Manager IM & P release 10.5 - 10.5.1.10000-9
- Cisco Unity Connection release 10.5 -10.5.1.10000-7
- Avaya Communication Server 1000E - RELEASE 7 ISSUE 65 P - (Application configuration: CS+SS+NRS+EM_SubM)
- Avaya Media Gateway Controller - LOADWARE VERSION: PSWV 100+
- Cisco Jabber Client – Version 10.5.0 Build 37889

Features

This section lists supported and unsupported features. Deviance from the configuration presented in this guide is not supported by Cisco. Please see the Limitations section on page 7 for more information.

Features Supported

- CLIP—calling line (number) identification presentation
- CLIR—calling line (number) identification restriction.(Refer to Integration issues section)
- CNIP—calling name identification presentation
- CNIR—calling name identification restriction. (Refer to Integration issues section)
- Alerting name.(Refer to Limitation Section)
- Attended call transfer. (Refer to Limitation Section)
- Early attended call transfer. (Refer to Limitation Section)
- CFU—call forwarding unconditional. (Refer to Limitation Section)
- CFB—call forwarding busy. (Refer to Limitation Section)
- CFNA—call forwarding no answer. (Refer to Limitation Section)
- COLP—connected line (number) identification presentation.(Refer to Integration issues section)
- COLR—connected line (number) identification restriction
- CONP—connected name identification presentation
- CONR—connected name identification restriction. (Refer to Integration issues section)
- Hold and resume
- Conference call
- MWI—Message Waiting Indicator (lamp ON, lamp OFF)
- Audio Codec Preference List
- Call Park (Refer to Limitation Section)
- DTMF-relay using RFC2833

Features Not Supported or Not Tested

- Call completion (callback, automatic callback)
- Blind Call Transfer
- Inter-working Test Cases with Various Calling/Connected Name and Number
- Shared Line - Hold & Resume with MOH
- Interworking Test Cases for Call Transfer

Limitations

These are the known limitations, caveats, or integration issues:

- Avaya CS1000E doesn't support Alerting Name feature. Although the Cisco UCM sends P-Asserted Identity (PAI) header with the alerting name(180 ringing) and connected name(200 ok), this information is not displayed by the Avaya SIP phones, However Avaya Unistim phones displayed the Alerting and connected name details.
- The Avaya PBX uses the History-Info field to send redirecting number information, while the Cisco UCM uses the Diversion header. This affects how calls are treated when redirected to a voice mail system over an SIP trunk. Since release 8.5, Cisco UCM provides the ability to translate either Diversion headers into History-Info headers or History-Info headers to Diversion headers via SIP Normalization Script. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- Avaya phone is configured to restrict connected name and number, it was observed that the SIP response to Cisco UCM only sets the privacy=user. However, the Cisco UCM only recognizes privacy=id to restrict presentation of both connected name and number. Cisco UCM provides the ability to covert the Privacy=user to Privacy=Id using normalization script.
- For integration where Cisco Unity is the centralized voice messaging system, a SIP normalization script is required to enable/disable MWI on Avaya phones. Please refer to the configuration section of this document for more details on the actual normalization script used for this testing.
- During a conference call hosted by the Avaya CS1000E SIP telephone, if the SIP telephone is hung up/dropped out of the conference, the conference call is dropped. The behavior is not seen with Unistim phones.
- Call Park: While retrieving Avaya CS1000E parked call from Cisco UCM, the call has been disconnected, Cisco parked calls has been retrieved successfully from Avaya CS1000E.
- Both systems support call forwarding and call transfer features. There are some call forward and transfer scenarios where the calling name and number and/or connected name and number are not updated after the call has been transferred or forwarded. This issue is found primarily when an Avaya phone is the forwarding or transferring party to a Cisco phone via the SIP trunk.

Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager and Avaya CS1000E PBX's. The deployment will interconnect the UC systems using SIP. No PSTN connectivity has been tested with this integration. The following sections provide the required configurations for a successful integration.

Configuring Sequence and Tasks:

Avaya Communication Server 1000E:

Configure the IP D-channel (signaling channel) between the call server and the signaling server —LD 17.

1. Zone Configuration —LD 117.
2. Configure the SIP route — LD 16.
3. Configure the SIP virtual trunks to the signal — LD 14.
4. Configure for the virtual lines for the Avaya IP phone — LD 20.
5. Configure the route list block for the virtual trunk route — LD 86.
6. Configure CDP steering codes — LD 87.
7. List trunk member — LD 21.
8. Avaya SIP Line Configuration.

Signaling Server Setup via the Avaya CS1000E Node Summary:

1. Configure a new IP telephony node summary.
2. Configure the VGW and IP phone codec profile section.
3. Configure the SIP GW setting section.
4. Configure the quality of service (QoS) section.

Network Routing Server:

1. Configure the system-wide settings.
2. Configure the NRS server settings.
3. Configure a service domain.
4. Configure an L1 domain (UDP).
5. Configure an L0 domain (CDP).
6. Configure a gateway endpoint gateway.
7. Configure the routing entries.

Cisco Unified Communications Manager:

1. Device setting SIP profile.
2. Media resource group and media resource group list.
3. Assign media resource group list (MRGL) in the default device pool.
4. SIP trunk to Avaya CS1000E PBX.
5. SIP Trunk Normalization Script.
6. SIP and SCCP phones device configuration.
7. Route pattern to the Avaya CS1000E PBX.
8. Call Manager Service Parameter "Duplex Streaming Enabled" set to "True".
9. Audio Codec Preference Configuration.
10. Region Configuration.
11. Cisco UCM Unity Integration.

Configuring the Avaya CS1000E

Avaya CS1000E Software Version – Issue and Release

```
REQ iss
VERSION 4121
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:      1
IPMGs Unregistered:   0
IPMGs Configured/unregistered: 0
RELEASE 7
ISSUE 65 P
IDLE_SET_DISPLAY tekvizion
IPMG TYPE CSP/SW MSP APP FPGA BOOT DBL1 DBL2
0 1 MGC DC04 AB02 BA18 AA22 BA18 DSP1AB07 N/A
```

```
REQ issp
02/12/14 02:03:33
TID: XXXXXXXXX
VERSION 4121
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:      1
IPMGs Unregistered:   0
IPMGs Configured/unregistered: 0
RELEASE 7
ISSUE 65 P
IDLE_SET_DISPLAY tekvizion
SYSTEM HAS NO IN-SERVICE DEPLISTS
MDP>LAST SUCCESSFUL MDP REFRESH :2014-11-03 12:46:57(Local Time)
MDP>USING DEPLIST ZIP FILE DOWNLOADED :2014-11-03 10:39:29(est)
SYSTEM HAS NO USER SELECTED PEPS IN-SERVICE
```

```
LOADWARE VERSION: PSWV 100+
INSTALLED LOADWARE PEPS : 7
```

PAT#	CR #	PATCH REF #	NAME	DATE	FILENAME
00	Q01981776	ISS1:1OF1	udtcab25	03/11/2014	udtcab25.lw
01	wi01171831	ISS1:1OF1	MGCCDC04	03/11/2014	MGCCDC04.LW
02	wi01057886	ISS1:1OF1	DSP1AB07	03/11/2014	DSP1AB07.LW
03	wi01057886	ISS1:1OF1	DSP2AB07	03/11/2014	DSP2AB07.LW
04	wi01057886	ISS1:1OF1	DSP3AB07	03/11/2014	DSP3AB07.LW
05	wi01057886	ISS1:1OF1	DSP4AB07	03/11/2014	DSP4AB07.LW

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ENABLED PLUGINS : 1

PLUGIN	STATUS	PRS/CR_NUM	MPLR_NUM	DESCRIPTION
501	ENABLED	Q02138637	MPLR30070	Enables blind transfer to a SIP endpoint even if SIP UPDATE is not supported by the far end

501 ENABLED Q02138637 MPLR30070 Enables blind transfer to a SIP endpoint even if SIP UPDATE is not supported by the far end

IP D-Channel Configuration

Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server – LD 17. The SIP Gateway application requires a D-channel over IP to communicate with the CS 1000E system. The SIP routes are associated with the D-channels and the SIP Gateway application running on a Linux server. The SIP Gateway route is used to communicate with the Call Server.

D channel Card Type (CYTP) list, select D-Channel is over IP (DCIP).

Set User = Integrated Services Signaling Link Dedicated (ISLD).

Set Interface type for D-channel (IFC) list = Meridian Meridian1 (SL1).

Set Meridian 1 Node Type = Slave to the Controller (USR).

AVAYA CS1000 Element Manager | Help | Logout

Managing: 10.0.0.1 Username: admin
Routes and Trunks » D-Channels » D-Channels 15 Property Configuration

D-Channels 15 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	vtrk
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> more PRI
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	1800 Range: 0 - 3700

Zone Configuration

Navigation Path: CS1000 Element Manager → System → IP Network → Zone → BandwidthZones

Zones are used to group related information for either bandwidth or dial plan numbering purposes. Zone 6(MO) and Zone 7(VTRK) were used for Best Bandwidth (G729). Zone 3(MO) or 0(MO) and Zone 1(VTRK) were used for Best Quality (G711).

Bandwidth Zones

	Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description	Location Name	Reserved BW Block Size
1	1	100000	BQ	100000	BQ	SHARED	VTRK	ZONE1		0
2	2	100000	BQ	100000	BQ	SHARED	VTRK			0
3	3	1000000	BQ	1000000	BQ	SHARED	MO			0
4	4	1000000	BQ	1000000	BB	SHARED	MO			0
5	5	1000000	BQ	1000000	BQ	SHARED	MO			0
6	6	1000000	BB	1000000	BB	SHARED	MO			0
7	7	1000000	BB	1000000	BB	SHARED	VTRK			0
8	100	1000000	BQ	1000000	BB	SHARED	VTRK			0

SIP Route Configuration

Navigation Path: CS1000 Element Manager → Routes and Trunk

Set Route Data Block (RDB) = RDB.

Set Customer Number (CUST) = 0. This is used for this testing.

Set Route Number (ROUT) = 10. This is used for this testing.

Set Trunk type (TKTP) = TIE trunk data block (TIE).

Set Incoming and outgoing trunk (ICOG) = Incoming and Outgoing (IAO).

Set Access code for the trunk route (ACOD) = 7088. This is used for this testing.

Set Node ID of signaling server of this route (NODE) = 1. This is used for this testing.

Set Protocol ID for the route (PCID) = SIP.

Set Mode of Operation (MODE) = ISDN Signaling Link (ISLD).

Set D channel number (DCH) = 15. This is used for this testing.

Set Interface type for route (IFC) = Meridian M1 (SL1).

Check Network calling name allowed (NCNA).

Check Network call redirection (NCRD).

Avaya CS1000E SIP Trunk to Cisco UCM Configuration (Continued)

AVAYA CS1000 Element Manager Help | Logout

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - [Peripheral Equipment](#)
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - [Routes and Trunks](#)
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Views
 - Lists
 - Properties
 - Migration
- Tools
 - + Backup and Restore
 - Date and Time
 - + Logs and reports
- Security
 - + Passwords
 - + Policies
 - + Login Options

Customer 0, Route 10 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE):	RDB
Customer number (CUST):	00
Route number (ROUT):	10
Designator field for trunk (DES):	VTRK
Trunk type (TKTP):	TIE
Incoming and outgoing trunk (ICOG):	Incoming and Outgoing (IAO) ▼
Access code for the trunk route (ACOD):	7088 *

Trunk type M911P (M911P):

The route is for a virtual trunk route (VTRK): <input checked="" type="checkbox"/>	
- Zone for codec selection and bandwidth management (ZONE):	00001 (0 - 8000)
- Node ID of signaling server of this route (NODE):	1 (0 - 9999)
- Protocol ID for the route (PCID):	SIP (SIP) ▼

- Print correlation ID in CDR for the route (CRID):

- Enable Shared Bandwidth Management for the route (SBWM):

Integrated services digital network option (ISDN):

- Mode of operation (MODE):	Route uses ISDN Signaling Link (ISLD) ▼
- D channel number (DCH):	15 (0 - 254)
- Interface type for route (IFC):	Meridian M1 (SL1) ▼
- Private network identifier (PNI):	00001 (0 - 32700)

- Network calling name allowed (NCNA):

- Network call redirection (NCRD):

Avaya CS1000E SIP Trunk to Cisco UCM Configuration (Continued)

<ul style="list-style-type: none">- UCM Network Services- Home- Links- Virtual Terminals- System+ Alarms- Maintenance+ Core Equipment- Peripheral Equipment+ IP Network+ Interfaces- Engineered Values+ Emergency Services+ Geographic Redundancy+ Software- Customers- Routes and Trunks<ul style="list-style-type: none">- Routes and Trunks- D-Channels- Digital Trunk Interface- Dialing and Numbering Plans- Electronic Switched Network- Flexible Code Restriction- Incoming Digit Translation- Phones- Templates- Reports- Views- Lists- Properties- Migration- Tools+ Backup and Restore- Date and Time+ Logs and reports- Security+ Passwords+ Policies+ Login Options	- Mode of operation (MODE) : <input type="text" value="Route uses ISDN Signaling Link (ISLD)"/>
	- D channel number (DCH) : <input type="text" value="15"/> (0 - 254)
	- Interface type for route (IFC) : <input type="text" value="Meridian M1 (SL1)"/>
	- Private network identifier (PNI) : <input type="text" value="00001"/> (0 - 32700)
	- Network calling name allowed (NCNA) : <input checked="" type="checkbox"/>
	- Network call redirection (NCRD) : <input checked="" type="checkbox"/>
	- Trunk route optimization (TRO) : <input type="checkbox"/>
	- Recognition of DTI2 ABCD FALT signal for ISL (FALT) : <input checked="" type="checkbox"/>
	- Channel type (CHTY) : <input type="text" value="B-channel (BCH)"/>
	- Call type for outgoing direct dialed TIE route (CTYP) : <input type="text" value="Unknown Call type (UKWN)"/>
	- Insert ESN access code (INAC) : <input type="checkbox"/>
	- Integrated service access route (ISAR) : <input type="checkbox"/>
	- Display of access prefix on CLID (DAPC) : <input type="checkbox"/>
	- Mobile extension route (MBXR) : <input checked="" type="checkbox"/>
	- Screen indicator (SIND) : <input checked="" type="checkbox"/>
	- Mobile extension outgoing type (MBXOT) : <input type="text" value="National number (NPA)"/>
	- Mobile extension timer (MBXT) : <input type="text" value="0"/> (0 - 8000 milliseconds)
	Calling number dialing plan (CNDP) : <input type="text" value="Unknown (UKWN)"/>
	+ Basic Route Options
	+ Network Options
	+ General Options
	+ Advanced Configurations
	<input type="button" value="Submit"/> <input type="button" value="Refresh"/> <input type="button" value="Delete"/> <input type="button" value="Cancel"/>

SIP Virtual trunk Configuration

Navigation Path: CS1000 Element Manager → Routes and Trunk

The screenshot shows the CS1000 Element Manager interface. On the left is a navigation tree with categories like UCM Network Services, System, Customers, Routes and Trunks, and Phones. The main area displays a list of routes and trunks. A red box highlights the configuration for Route 10, which is of Type TIE and Description VTRK. Underneath, it shows 8 trunks (Trunk 1 to Trunk 8), each with a TN (Terminal Number) and Description (VTRK). Buttons for 'Edit' and 'Add trunk' are visible for each entry.

- Set Route Data Block (RDB) = RDB.
- Set Trunk data block = IP Trunk (IPTI).
- Set Terminal Number = 100 0 01 00. This is used for this testing.
- Set Designator field for trunk = VTRK.
- Set Member number = 1. This is used for this testing.
- Set Start arrangement Incoming = Immediate (IMM).
- Set arrangement Outgoing = Immediate (IMM).
- Set Trunk Group Access Restriction = 1. This is used for this testing.
- Set Channel ID for this trunk = 1. This is used for this testing.

The screenshot shows the 'Customer 0, Route 10, Trunk 1 Property Configuration' dialog box. The 'Basic Configuration' section is highlighted with a red box and contains the following fields:

- Auto increment member number:
- Trunk data block: IPTI
- Terminal number: 100 0 01 00
- Designator field for trunk: VTRK
- Extended trunk: VTRK
- Member number: 1
- Level 3 Signaling: [Dropdown]
- Card density: 8D
- Start arrangement Incoming: Immediate (IMM)
- Start arrangement Outgoing: Immediate (IMM)
- Trunk group access restriction: 0
- Channel ID for this trunk: 1
- Class of Service: [Edit]

 Below this section is the '+ Advanced Trunk Configurations' section, which is currently collapsed. At the bottom of the dialog are 'Save', 'Delete', and 'Cancel' buttons.

Virtual Lines for Avaya CS1000E IP Phones

Avaya Unistim Phone

```
>ld 20
REQ: prt
TYPE: tn
TYPE TNB
TN 108 1 5 0
DATE
PAGE
DES
DES 2004P2
TN 108 1 05 00 VIRTUAL
TYPE 2004P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00000
CUR_ZONE 00000
MRT
ERL 0
ECL 0
FDN 2004
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXD ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
AHD DDGA NAMA
DRDD EXR0
USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC MCBN
VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
```

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Avaya CS1000E Unistim Phone Configuration (Continued)

```
CPND_LANG ENG
RCO 0
EFD 2004
HUNT 2004
EHT 2004
LHK 0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 3005 0  MARP
  CPND
    NAME cs1kunis1
    XPLN 9
    DISPLAY_FMT FIRST,LAST
01
02
03
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16 2004
```

- 20 RGA
- 21 PRK
- 22 RNP
- 23
- 24 PRS
- 25 CHG
- 26 CPN
- 27
- 28
- 29
- 30
- 31

DATE 25 NOV 2014

Route List Block Configuration

Navigation Path: CS1000 Element Manager → Digital and Numbering Plans → Route List Block(RLB)

The screenshot displays the AVAYA CS1000 Element Manager interface. The left-hand navigation menu includes categories such as Emergency Services, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, Tools, and Security. The main content area is titled "Route List Block Index -- 26" and contains the following configuration details:

- Initial Set: 0
- Number of Alternate Routing Attempts: 5
- Set Minimum Facility Restriction Level: 0
- Data Entry Index -- 0
- Route Number: 10
- Expensive Route: N
- Facility Restriction Level: 0
- Digit Manipulation Index: 0
- ISL D-Channel Down Digit Manipulation Index: 0
- Free Calling Area Screening Index: 0
- Free Special Number Screening Index: 0

Below the configuration details, a list of other Route List Block Indices is shown, each with an "Edit" button:

- Route List Block Index -- 27
- Route List Block Index -- 30
- Route List Block Index -- 50
- Route List Block Index -- 51
- Route List Block Index -- 52
- Route List Block Index -- 85

At the bottom of the interface, the copyright notice reads: "Copyright © 2002-2013 Avaya Inc. All rights reserved."

CDP Steering code configuration

Navigation Path: CS1000 Element Manager → Digital and Numbering Plans → Coordinated Dialing Plan(CDP) list → Distance Steering code(DSC)

The screenshot shows the Avaya CS1000 Element Manager interface. The top navigation bar includes the Avaya logo, the title "CS1000 Element Manager", and "Help | Logout" links. The left sidebar contains a tree view of system components, with "Dialing and Numbering Plans" selected. The main content area displays the "Distant Steering Code List" configuration page. At the top, it shows the current user and session information: "Managing: 10.0.0.1 Username: admin" and the breadcrumb path "Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List". Below this is a "Display" dropdown menu. The configuration fields include "Starting Distant Steering Code" set to 20, "Number of Steering Codes to display" set to 2, and a "View" button. A table of steering code lists is shown below, with the first entry "Distant Steering Code List -- 20" highlighted in a red box. This entry has an "Edit" button and the following details: "Flexible Length number of digits: 4", "Display: LSC", "Route List to be accessed for trunk steering code: 26", "Maximum 7 digit NPA code allowed:", and "Maximum 7 digit NXX code allowed:". A second entry "Distant Steering Code List -- 214" is partially visible below with its own "Edit" button.

Avaya SIP Line Configuration

Navigation Path: CS1000 Element Manager → System → Customers → SIP Line Service

In the User agent DN prefix field, enter a DN prefix to build the HOT U key information for SIP Line phones. The DN cannot conflict with the current system. The User Agent DN prefix field is the same as the UAPR prompt in LD 15 and is used on the Phones.

A HOT U key label is available when UXTY is SIPL. The HOT U key is also known as the User Agent Directory Number (UADN) key. The UADN key is used to make and receive calls between the SIP Line Gateway and the Universal Extensions. However, this key is used only by the SIP Line Gateway (SLG) application. (The UADN is not dialed by end users. It is only used internally between the Call Server and the SIP Line Gateway application.)

Avaya CS1000E SIP Line Configuration (Continued)

The screenshot shows the Avaya CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, System, and Customers. The main content area is titled 'SIP Line Service' and shows a configuration form. A red box highlights the 'SIP Line Service' checkbox, which is checked, and the 'User agent DN prefix' field, which contains the value '23'. Below this, there are optional features for 'Nortel Multimedia' and 'Cancel' buttons. The top of the page shows the Avaya logo and 'CS1000 Element Manager' title.

SIP Line Configuration Details

Navigation Path: CS1000 Element Manager → System → Node ID → SIP Line under Applications

Check Enable gateway service on this node

Set domain Name * = lab.tekvizion.com. This is used for this example

Set SLG Local SIP port = 5070

Set SLG Local TLS port = 5071

The screenshot shows the Avaya CS1000 Element Manager interface for 'Node ID: 1 - SIP Line Configuration Details'. The left sidebar is the same as in the previous screenshot. The main content area is titled 'Node ID: 1 - SIP Line Configuration Details' and has sub-tabs for 'General', 'SIP Line Gateway Settings', and 'SIP Line Gateway Service'. A red box highlights the 'SIP Line Gateway Application: Enable gateway service on this node' checkbox, which is checked. Below this, there are two columns: 'General' and 'Virtual Trunk Network Health Monitor'. The 'General' column contains fields for 'SIP domain name: lab.tekvizion.com', 'SLG endpoint name: lab', 'SLG Group ID', 'SLG Local Sip port: 5070', and 'SLG Local Tls port: 5071'. The 'Virtual Trunk Network Health Monitor' column contains a checked checkbox for 'Monitor IP addresses (listed below)' and a list of monitor IP addresses. The bottom section is 'SIP Line Gateway Settings' with fields for 'Security policy: Security Disabled', 'Number of byte re-negotiation: 0', and options for 'Client authentication' and 'x509 Certificate authentication enabled'.

Avaya CS1000E SIP Line Configuration (Continued)

System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

Node ID: 1 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Service

Branch / GR Office Settings:

SLG role:

SLG mode:

MO SLG IPv4 address:
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

MO SLG IPv6 address:

MO SLG port: (1 - 65535)

MO SLG transport:

GR SLG IPv4 address:
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

GR SLG IPv6 address:

GR SLG port: (1 - 65535)

GR SLG Transport:

IVR Settings:

SLG IVR proxy IP:

SLG IVR proxy port: (1 - 65535)

SLG IVR proxy transport:

SLG IVR CDP URI map:

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

D – Channel Configuration

Navigation Path: CS1000 Element Manager → Routes and Trunks → D-Channels

The SIP Line Gateway (SLG) application requires a D-channel over IP to communicate with the Avaya CS1000E system. The SIP Line routes are associated with the D-channels and the SLG application running on a Linux server. The SIP Line route is used to communicate with the Call Server.

- D channel Card Type (CYTP) list, select D-Channel is over IP (DCIP)
- Set User = Integrated Services Signaling Link Dedicated (ISLD)
- Set Interface type for D-channel (IFC) list = Meridian Meridian1 (SL1)
- Set Meridian 1 Node Type = Slave to the Controller (USR)

Avaya CS1000E D channel Configuration (Continued)

Managing: 10.0.0.1 Username: admin
Routes and Trunks » D-Channels » D-Channels 15 Property Configuration

D-Channels 15 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	vtrk
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <input type="button" value="more PRI"/>
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	1800 Range: 0 - 3700

Application Module Link Details

Navigation Path: CS1000 Element Manager → System → Interfaces → Application Module Link

The SLG application uses the AML over ELAN link to establish a pbxlink (AML over ELAN) connection with the CS 1000 system. The SLG application can control the SIPL UEXT using AML messages with the pbxlink established. Application Module Link page, in the Port number field, enter the port number. The SIP Line service uses ports 32 to 127.

Description = enter a description for the AML.

AVAYA CS1000 Element Manager Help | Logout

Managing: 10.0.0.1 Username: admin
System » Interfaces » Application Module Link » Application Module Link Details 33

Application Module Link Details 33

Link Type: ELAN
Description: sipol
Maximum octets: 512 (per HDLC frame)

Save Cancel

VAS ID association with AML over ELAN link

Navigation Path: CS1000 Element Manager → System → Interfaces → Value Added server

Every AML over ELAN link configured on the Avaya CS1000E system requires a Value Added Server (VAS) ID for the AML messages to be sent. Use the following procedure to associate a Value Added Server (VAS) with AML over ELAN.

Set Ethernet LAN Link = 33. This is used for this example.

Set Application Security check box is cleared.

AVAYA CS1000 Element Manager Help | Logout

Managing: 10.0.0.1 Username: admin
System » Interfaces » Value Added Server » Edit Value Added Server 033

Edit Value Added Server 033

Ethernet LAN Link: 033
ELAN port configured in ADAN

Application security:
Interval: 1
Time interval for checking the link for overload in five second increments

Message count threshold: 9999 * (10 - 9999)

* Required value.

Save Cancel

SIP Line Route Configuration

Navigation Path: CS1000 Element Manager → Routes and Trunk

Configure a SIP Line route similar to the way to configure a virtual trunk route, such as SIP. A virtual trunk zone is required for the SIP Line route to work. Ensure to configure a virtual trunk zone.

Set Route Data Block (RDB) = RDB.

Set Customer Number (CUST) = 0. This is used for this testing.

Set Route Number (ROUT) = 20. This is used for this testing.

Set Trunk type (TKTP) = TIE trunk data block (TIE).

Set Incoming and outgoing trunk (ICOG) = Incoming and Outgoing (IAO).

Set Access code for the trunk route (ACOD) = 7020. This is used for this testing.

Set Node ID of signaling server of this route (NODE) = 1. This is used for this testing.

Set Protocol ID for the route (PCID) = SIP Line (SIPL).

Set Mode of Operation (MODE) = ISDN Signaling Link (ISLD).

Set D channel number (DCH) = 15. This is used for this testing.

Set Interface type for route (IFC) = Meridian M1 (SL1).

Check Network calling name allowed (NCNA).

Check Network call redirection (NCRD).

AVAYA CS1000 Element Manager Help | Logout

Customer 0, Route 20 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE):	RDB
Customer number (CUST):	00
Route number (ROUT):	20
Designator field for trunk (DES):	SIP
Trunk type (TKTP):	TIE
Incoming and outgoing trunk (ICOG):	Incoming and Outgoing (IAO) ▼
Access code for the trunk route (ACOD):	7020 *

Trunk type M911P (M911P):

The route is for a virtual trunk route (VTRK):

- Zone for codec selection and bandwidth management (ZONE):	00001 (0 - 8000)
- Node ID of signaling server of this route (NODE):	1 (0 - 9999)
- Protocol ID for the route (PCID):	SIP Line (SIPL) ▼

Integrated services digital network option (ISDN):

- Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD) ▼

Avaya CS1000E SIP Line Configuration (Continued)

<ul style="list-style-type: none"> - Interfaces - Application Module Link - Value Added Server - Property Management System - Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks <ul style="list-style-type: none"> - Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans <ul style="list-style-type: none"> - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones <ul style="list-style-type: none"> - Templates - Reports - Views - Lists - Properties - Migration - Tools <ul style="list-style-type: none"> + Backup and Restore - Date and Time + Logs and reports - Security <ul style="list-style-type: none"> + Passwords 	<ul style="list-style-type: none"> - D channel number (DCH) : 15 (0 - 254) - Interface type for route (IFC) : Meridian M1 (SL1) ▼ - Private network identifier (PNI) : 00000 (0 - 32700) - Network calling name allowed (NCNA) : <input checked="" type="checkbox"/> - Network call redirection (NCRD) : <input checked="" type="checkbox"/> -- Trunk route optimization (TRO) : <input type="checkbox"/> - Recognition of DT12 ABCD FALT signal for ISL (FALT) : <input type="checkbox"/> <ul style="list-style-type: none"> - Channel type (CHTY) : B-channel (BCH) ▼ - Call type for outgoing direct dialed TIE route (CTYP) : Unknown Call type (UKWN) ▼ <ul style="list-style-type: none"> - Insert ESN access code (INAC) : <input type="checkbox"/> - Integrated service access route (ISAR) : <input type="checkbox"/> - Display of access prefix on CLID (DAPC) : <input type="checkbox"/> - Mobile extension route (MBXR) : <input checked="" type="checkbox"/> - Screen indicator (SIND) : <input checked="" type="checkbox"/> - Mobile extension outgoing type (MBXOT) : National number (NPA) ▼ - Mobile extension timer (MBXT) : 0 (0 - 8000 milliseconds)
--	--

<ul style="list-style-type: none"> - Reports - Views - Lists - Properties - Migration - Tools <ul style="list-style-type: none"> + Backup and Restore - Date and Time + Logs and reports - Security <ul style="list-style-type: none"> + Passwords + Policies 	<p>Calling number dialing plan (CNDP) : Unknown (UKWN) ▼</p> <ul style="list-style-type: none"> + Basic Route Options + Network Options + General Options + Advanced Configurations <p style="text-align: center;"> <input type="button" value="Submit"/> <input type="button" value="Refresh"/> <input type="button" value="Delete"/> <input type="button" value="Cancel"/> </p>
---	---

SIP Line Trunk Configuration

Navigation Path: CS1000 Element Manager → Routes and Trunk

The screenshot shows the AVAYA CS1000 Element Manager interface. The main content area displays the configuration for Route 20, which is a TIE type with a description of SIP. It lists 10 trunks, each with a TN (Terminal Number) and a description of SIPL. The interface includes a navigation menu on the left, a main content area with a table of trunks, and a right-hand sidebar with buttons for 'Edit' and 'Add trunk'.

Trunk	TN	Description	Action
- Route: 20	Type: TIE	Description: SIP	Edit Add trunk
- Trunk: 1 - 10	Total trunks: 10		
- Trunk: 1	TN: 100 0 02 00	Description: SIPL	Edit Multi - Del
- Trunk: 2	TN: 100 0 02 01	Description: SIPL	Edit
- Trunk: 3	TN: 100 0 02 02	Description: SIPL	Edit
- Trunk: 4	TN: 100 0 02 03	Description: SIPL	Edit
- Trunk: 5	TN: 100 0 02 04	Description: SIPL	Edit
- Trunk: 6	TN: 100 0 02 05	Description: SIPL	Edit
- Trunk: 7	TN: 100 0 02 06	Description: SIPL	Edit
- Trunk: 8	TN: 100 0 02 07	Description: SIPL	Edit
- Trunk: 9	TN: 100 0 02 08	Description: SIPL	Edit
- Trunk: 10	TN: 100 0 02 09	Description: SIPL	Edit

Set Route Data Block (RDB) = RDB.

Set Trunk data block = IP Trunk (IPTI).

Set Terminal Number = 100 0 02 00. This is used for this testing.

Set Designator field for trunk = SIPL

Set Member number = 1. This is used for this testing.

Set Start arrangement Incoming = Immediate (IMM)

Set arrangement Outgoing = Immediate (IMM)

Set Trunk Group Access Restriction = 1. This is used for this testing.

Set Channel ID for this trunk = 10. This is used for this testing.

Avaya CS1000E SIP Line Configuration (Continued)

AVAYA CS1000 Element Manager Help | Logout

Customer 0, Route 20, Trunk 1 Property Configuration

- Basic Configuration

Auto increment member number:	<input checked="" type="checkbox"/>
Trunk data block:	<input type="text" value="IPTI"/>
Terminal number:	<input type="text" value="100 0 02 00"/>
Designator field for trunk:	<input type="text" value="SIPL"/>
Extended trunk:	<input type="text" value="VTRK"/>
Member number:	<input type="text" value="1"/> *
Level 3 Signaling:	<input type="text"/>
Card density:	<input type="text" value="8D"/>
Start arrangement Incoming:	<input type="text" value="Immediate (IMM)"/>
Start arrangement Outgoing:	<input type="text" value="Immediate (IMM)"/>
Trunk group access restriction:	<input type="text" value="1"/>
Channel ID for this trunk:	<input type="text" value="10"/>
Class of Service:	<input type="button" value="Edit"/>

+ Advanced Trunk Configurations

Avaya SIP Line Phone Configuration

Ld 20
REQ: prt
TYPE: tn
TYPE TNB
TN 108 1 5 29
DATE
PAGE
DES
DES SIP1
TN 108 1 05 29 VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL
MCCL YES
SIPN 1
SIP3 0
FMCL 0
TL SV 0
SIPU 3003
NDID 3003
SUPR NO
SUBR DFLT MWI RGA CWI MSB
UXID 3003
NUID
NHTN
CFG_ZONE 00003
CUR_ZONE 00003
MRT
ERL 0
ECL 0
FDN 2003
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 3003
CLS CTD FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTA SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD

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Avaya CS1000E SIPL Phone Configuration (Continued)

ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXA ARHD CLTD ASCD
CPFA CPTA ABDD CFHA FICD NAID BUZZ AGRD MOAD
AHA DDGA NAMA
DRDD EXR0
USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC MCBN
VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
MSNV FRA PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
RCO 0
EFD 2003
HUNT 2003
EHT 2003
LHK 0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 3003 0 MARP
 CPND
 NAME cs1ksip1
 XPLN 8
 DISPLAY_FMT FIRST, LAST
01 HOT U 233003 MARP 0
02
03
04
05
06
07
08
09
10
11
12
13

Avaya CS1000E SIPL Phone Configuration (Continued)

15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31

DATE 25 NOV 2014

NACT

[Avaya CS1000E CLI Configuration](#)

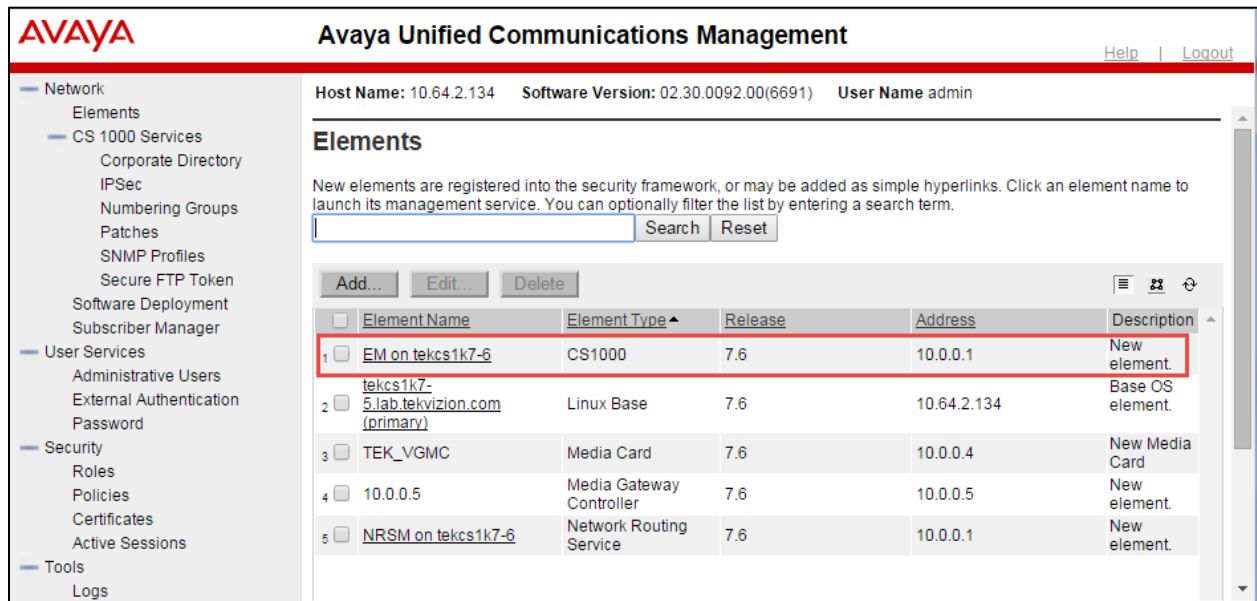
Attached is the LD prints for Avaya CS1000E Configuration for reference



CS1k
Configuration.txt

Avaya Unified Communication Management

The UCM security domain provides central authentication, authorization, auditing, certificate management, and secure navigation functionality between managed elements. All elements within the same security domain appear in a single navigation tree.



AVAYA Avaya Unified Communications Management Help | Logout

Host Name: 10.64.2.134 Software Version: 02.30.0092.00(6691) User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Search Reset

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on tekcs1k7-6	CS1000	7.6	10.0.0.1	New element.
<input type="checkbox"/>	tekcs1k7-5.lab.tekvizion.com (primary)	Linux Base	7.6	10.64.2.134	Base OS element.
<input type="checkbox"/>	TEK_VGMC	Media Card	7.6	10.0.0.4	New Media Card
<input type="checkbox"/>	10.0.0.5	Media Gateway Controller	7.6	10.0.0.5	New element.
<input type="checkbox"/>	NRSM on tekcs1k7-6	Network Routing Service	7.6	10.0.0.1	New element.

Avaya CS1000E Element Manager

Element Manager is a Web-based user interface used to configure and maintain Avaya CS1000E components. Element Manager is deployed with the Avaya Unified Communications Management solution on a Linux based operating system. Avaya UCM provides logon and security features for Element Manager.

- UCM Network Services**- Home****- Links****- Virtual Terminals****- System**

+ Alarms

- Maintenance

+ Core Equipment

- Peripheral Equipment

+ IP Network

+ Interfaces

- Engineered Values

+ Emergency Services

+ Geographic Redundancy

+ Software

- Customers**- Routes and Trunks**

- Routes and Trunks

- D-Channels

- Digital Trunk Interface

- Dialing and Numbering Plans

- Electronic Switched Network

- Flexible Code Restriction

- Incoming Digit Translation

- Phones

- Templates

- Reports

- Views

- Lists

- Properties

- Migration

Managing: **10.0.0.1** Username: admin
System Overview

System Overview

IP Address: 10.0.0.1

Type: Avaya Communication Server 1000E CPPM Linux

Version: 4121

Release: 765 P

IP Telephony Nodes

Navigation Path: CS1000 Element Manager → IP Network → Nodes:Servers,Media Cards

Managing: 10.0.0.1 Username: admin
System » IP Network » IP Telephony Nodes

IP Telephony Nodes

Click the Node ID to view or edit its properties.

Add... Import... Export... Delete Print | Refresh

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
1	1	SIP Line, LTPS, Gateway (SIPGw)	-	10.64.2.131	-	Synchronized

Show: Nodes Component servers and cards IPv6 address

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Set Node ID = 1. This is used for this testing.

Set Call server IP address = 10.0.0.1. This is used for this testing.

Set Gateway IP address= 10.0.0.10.This is used for this testing.

Set Node IPV4 address = 10.64.2.131. This is used for this testing.

Node Details (ID: 1 - SIP Line, LTPS, Gateway (SIPGw))

Node ID: 1 * (0-9999)

Call server IP address: 10.0.0.1 *

TLAN address type: IPv4 only
 IPv4 and IPv6

Embedded LAN (ELAN)

Gateway IP address: 10.0.0.10 *

Subnet mask: 255.255.255.0 *

Telephony LAN (TLAN)

Node IPv4 address: 10.64.2.131 *

Subnet mask: 255.255.0.0 *

Node IPv6 address:

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN
- SNTP
- Numbering Zones
- MCDN Alternative Routing Treatment (MALT) Causes

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)
- Personal Directories (PD)
- Presence Publisher
- IP Media Services

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Add Remove Make Leader Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
tekcs1k7-5	Signaling_Server	SIP Line, LTPS, Gateway (SIPH323), PD, Presence Publisher, IP Media Services	10.0.0.1	10.64.2.134	Leader

Show: IPv6 address

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Voice Gateway (VGW) and Codecs

Navigation Path: CS1000 Element Manager → IP Network → Nodes:Servers,Media Cards

Check Codec G711 Enabled required.

Check Codec G729 Enabled required.

AVAYA CS1000 Element Manager Help | Logout

Managing: 10.0.0.1 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 1 - Voice Gateway (VGW) and Codecs

General | **Voice Codescs** | Fax

General

Echo cancellation: Use canceller, with tail delay: 128
 Dynamic attenuation

Voice activity detection threshold: -17 (-20 - +10 DBM)
Idle noise level: -65 (-327 - +327 DBM)

Signaling options: DTMF tone detection
 Low latency mode
 Remove DTMF delay (squelch DTMF from TDM to IP)
 Modem/Fax pass-through
 V.21 Fax tone detection
 R factor calculation

Voice Codescs

Codec G711: **Enabled (required)**
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum

AVAYA CS1000 Element Manager Help | Logout

Managing: 10.0.0.1 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 1 - Voice Gateway (VGW) and Codecs

General | **Voice Codescs** | Fax

Voice Codescs

Voice Activity Detection (VAD)

Codec G722: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Codec G729: **Enabled**
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Voice Activity Detection (VAD)

Avaya CS1000E VGW and Codecs (Continued)

Managing: 10.0.0.1 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 1 - Voice Gateway (VGW) and Codecs

General | Voice Codecs | Fax

Voice Activity Detection (VAD)

Codec G723.1: Enabled

Voice payload size: 30 (milliseconds per frame)

Voice playout (jitter buffer) delay: (milliseconds)

Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Coding rate: (kbps)

Fax

Codec name: T.38 FAX

Maximum rate: (bps)

Fax TCF method:

Fax playout nominal delay: (0 - 300 milliseconds)

FAX no activity timeout: (10 - 32000 milliseconds)

Packet size: (bps)

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Avaya CS1000E SIP Gateway

Navigation Path: CS1000 Element Manager → IP Network → Nodes:Servers,Media Cards → Select Node → Gateway(SIPGw)

Check VTRK gateway application Enable gateway application on this node.

Set Vtrk gateway application = SIP Gateway (SIPGw).

Set SIP domain name = lab.tekvizion.com. This is used for this testing.

Set Local SIP port = 5060.

Set Gateway endpoint name = nortel. This is used for this testing.

Application node ID = 1. This is used for this testing.

In the Proxy or Redirect Server set the Primary TLAN IP address = 10.64.2.134. This is used for this testing.

Set Port = 5060.

Set Transport Protocol= TCP. This is the transport protocol used for SIP message exchange between the Gateway and Redirect/Proxy Server. The two options are TCP and UDP. TCP is the default option.

The screenshot displays the Avaya CS1000 Element Manager web interface. The top navigation bar includes the Avaya logo, the title "CS1000 Element Manager", and "Help | Logout" links. The breadcrumb trail shows the path: "System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration". The main content area is titled "Node ID: 1 - Virtual Trunk Gateway Configuration Details" and has tabs for "General", "SIP Gateway Settings", and "SIP Gateway Services".

Key configuration details are highlighted with red boxes:

- Vtrk gateway application:** Enable gateway service on this node
- Vtrk gateway application:** SIP Gateway (SIPGw)
- SIP domain name:** lab.tekvizion.com
- Local SIP port:** 5060
- Gateway endpoint name:** nortel
- Application node ID:** 1

Other visible settings include:

- Virtual Trunk Network Health Monitor:** Monitor IP addresses (listed below)
- Monitor IP:** (empty field) [Add]
- Monitor addresses:** (empty list) [Remove]
- Enable failsafe NRS:**

A note at the bottom states: "Note: FailSafe NRS cannot be enabled, if all servers in the node have NRS application deployed."

Avaya CS1000E SIP Gateway (Continued)

- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface

SIP ANAT: IPv4
 IPv6

SIP Gateway Settings

TLS Security: Security Disabled

Port: 5061 (1 - 65535)
Number of byte re-negotiation: 0
Options: Client authentication
 X509 certificate authority

Direct SIP Route
 Enforce Direct SIP Route to Microsoft Mediation Server
FQDN of Microsoft Mediation Server:
Port: 5060 (1 - 65535)
Transport protocol: TCP

- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels

Shared Bandwidth Management:

Enable Shared Bandwidth Management

Proxy Or Redirect Server:
Proxy Server Route 1:

Primary TLAN IP address: 10.64.2.134
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)
Transport protocol: UDP
Options: Support registration
 Primary CDS proxy

Secondary TLAN IP address: 10.70.2.6
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)

Avaya CS1000E SIP Gateway (Continued)

<ul style="list-style-type: none"> - Maintenance + Core Equipment - Peripheral Equipment - IP Network <ul style="list-style-type: none"> - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translator - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software Customers Routes and Trunks <ul style="list-style-type: none"> - Routes and Trunks - D-Channels - Digital Trunk Interface 	<p>Transport protocol: TCP ▼</p> <p>Options: <input type="checkbox"/> Support registration <input checked="" type="checkbox"/> Secondary CDS proxy</p> <p>Tertiary IP address: <input type="text" value="0.0.0.0"/></p> <p>Port: <input type="text" value="5060"/> (1 - 65535)</p> <p>Transport protocol: TCP ▼</p> <p>Options: <input type="checkbox"/> Support registration <input type="checkbox"/> Tertiary CDS proxy</p> <p>Proxy Server Route 2:</p> <p>Primary TLAN IP address: <input type="text" value="10.70.2.6"/> <small>The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"</small></p> <p>Port: <input type="text" value="5060"/> (1 - 65535)</p> <p>Transport protocol: TCP ▼</p>
---	--

<ul style="list-style-type: none"> + Core Equipment - Peripheral Equipment - IP Network <ul style="list-style-type: none"> - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translator - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software 	<p>Options: <input type="checkbox"/> Registration not supported <input checked="" type="checkbox"/> Primary CDS proxy</p> <p>CLID Presentation:</p> <p>Country code (CCC): <input type="text" value="1"/></p> <p>Area code: <input type="text" value="719"/> NPA in North America</p> <p>Number translation: Strip: Prefix: CLID display format:</p> <p>Subscriber (SN): <input type="text" value="0"/> <input type="text"/> <CCC><Area code><SN></p> <p>National (NN): <input type="text" value="0"/> <input type="text"/> <CCC><NN></p> <p>International: <input type="text" value="0"/> <input type="text"/> <International number></p>
--	--

Quality of Service

Navigation Path: CS1000 Element Manager → IP Network → Nodes:Servers,Media Cards → Select Node → Quality of Service(QoS)

The screenshot displays the AVAYA CS1000 Element Manager interface. The top header shows the AVAYA logo, the title "CS1000 Element Manager", and "Help | Logout" links. The breadcrumb trail indicates the path: "System » IP Network » IP Telephony Nodes » Node Details » Quality of Service (QoS)".

The left navigation menu includes the following items:

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - IP Network
 - Nodes: Servers, Media Cards
 - Maintenance and Reports
 - Media Gateways
 - Zones
 - Host and Route Tables
 - Network Address Translation
 - QoS Thresholds
 - Personal Directories
 - Unicode Name Directory
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network

The main content area is titled "Node ID: 1 - Quality of Service (QoS)". It contains a configuration form for "Diffserv Codepoint (DSCP)" with the following fields:

- Enable Avaya automatic QoS:
- Control packets: (0-63)
- Voice packets: (0-63)
- VLAN tagging: 802.1Q support
- 802.1Q bits value (802.1P): (0-7)

At the bottom of the form, there is a note: "Note: Changes made on this page will NOT be transmitted until the Node is also saved." and two buttons: "Save" and "Cancel". A small asterisk indicates that the "Control packets" and "Voice packets" fields are required values.

Network Routing Service Manager

Navigation Path: CS1000 NRSM → Numbering Plans → Domains

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
 - Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans » Domains](#)

Domains

Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1) **L1 Domains (UDP) (1)** **L0 Domains (CDP) (1)**

[Refresh](#)

<input type="checkbox"/>	Domain Name	Description	# of L1 Domains	# of L0 Domains	# of Gateway Endpoints
1 <input type="checkbox"/>	lab.tekvizion.com		1	1	6

1 - 1 of 1 Service Domain(s) Page 1 of 1 [First](#) | [Previous](#) | [Next](#) | [Last](#)

L1 Domain

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
 - Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans » Domains](#)

Domains

Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1) **L1 Domains (UDP) (1)** **L0 Domains (CDP) (1)**

Filter by Domain: lab.tekvizion.com

[Refresh](#)

<input type="checkbox"/>	ID	Description	# of L0 Domains	# of Gateway Endpoints	# of Routing Entries	Context
1 <input type="checkbox"/>	udp		1	6	32	lab.tekvizion.com

1 - 1 of 1 L1 Domain(s) Page 1 of 1 [First](#) | [Previous](#) | [Next](#) | [Last](#)

L0 Domain

AVAYA Network Routing Service Manager Help | Logout

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans » Domains](#)

Domains
 Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1) L1 Domains (UDP) (1) L0 Domains (CDP) (1)

Filter by Domain: lab.tekvizion.com / udp

Refresh

ID	Description	# of Gateway Endpoints	# of Routing Entries	Context
1	cdp	6	32	lab.tekvizion.com / udp

1 - 1 of 1 L0 Domain(s) Page 1 of 1 First| Previous| Next| Last

System Wide Settings

AVAYA Network Routing Service Manager Help | Logout

Managing: 10.0.0.1
 System » System Wide Settings

System Wide Settings

SIP registration time to live timer: 300 (30-3600 Seconds)

H.323 gatekeeper registration time to live timer: 300 (30-3600 Seconds)

H.323 alias name: * (Required value)

Auto backup time: 23:49 (HH:MM)

Auto backup to secure FTP site enabled:

Auto backup to secure FTP site's IP address:

Auto backup secure FTP site's path:

Auto backup secure FTP user name: admin

Auto backup secure FTP password:

Call Server Type: CS1000

* Required value. Save Cancel

NRS Server

AVAYA Network Routing Service Manager Help | Logout

«UCM Network Services
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 - Routing Tests
 H.323
 SIP
 Backup
 Restore
 GK/NRS Data upgrade

Managing: 10.0.0.1
 System » NRS Server

NRS Server

Service Status
 Enable Graceful disable Restart

	Service Name	Service Status
1 <input type="checkbox"/>	SIP Proxy Server (SPS)	In service
2 <input type="checkbox"/>	Gatekeeper (GK)	In service
3 <input type="checkbox"/>	Network Connection Server (NCS)	In service

Server Configuration Edit...

NRS Setting

Host name TekCS1kNRS
 Address type IPv4 only
 Primary TLAN IPv4 address 10.64.2.134
 Secondary TLAN IPv4 address 0.0.0.0
 Secondary server host name SecondaryHostName
 Control priority 40
 Server mate communication port 5005
 Realm name realmName
 Server role Primary

Backup
 Restore
 GK/NRS Data upgrade

H.323 Gatekeeper Settings

Location request (LRQ) response timeout 3

SIP Server Settings

Public name for non-trusted networks unknown
 Public number for non-trusted networks 000-000

UDP Transport enabled
 Primary server UDP IPv4 10.64.2.134
 Primary server UDP port 5060
 Secondary server UDP IPv4 0.0.0.0
 Secondary server UDP port 5060

SIP
 Backup
 Restore
 GK/NRS Data upgrade

TCP Transport enabled
 Primary server TCP IPv4 10.64.2.134
 Primary server TCP port 5060
 Secondary server TCP IPv4 0.0.0.0
 Secondary server TCP port 5060

TLS Transport enabled
 Primary server TLS IPv4 10.64.2.134
 Primary server TLS port 5061
 Secondary server TLS IPv4 0.0.0.0
 Secondary server TLS port 5061

Avaya CS1000E NRS Server Configuration (Continued)

SIP Backup Restore GK/NRS Data upgrade	Secondary server TCP port 5060 TLS Transport enabled <input type="checkbox"/> Primary server TLS IPv4 10.64.2.134 Primary server TLS port 5061 Secondary server TLS IPv4 0.0.0.0 Secondary server TLS port 5061
---	--

SIP Backup Restore GK/NRS Data upgrade	Transport Layer Security (TLS) Settings Maximum session cache 2048000 Session cache timeout 600 Renegotiation in byte 2048000 X509 Certificate authentication <input type="checkbox"/> Client authentication <input type="checkbox"/> Network Connection Server (NCS) Settings Primary NCS port 16500 Secondary NCS port 16500 Primary NCS timeout 10
---	--

Endpoints

AVAYA Network Routing Service Manager Help | Logout

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans](#) » Endpoints

Search for Endpoints Hide

Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID:

Limit results to Domain: / /

Results per page:

Gateway Endpoints (6) User Endpoints (0)

<input type="checkbox"/>	ID	Supported Protocols	SIP mode:	Call Signaling IP	Description	# of Routing Entries	Context
<input type="checkbox"/>	Genband	Static SIP endpoint	Proxy Mode	10.70.76.10		11	lab.tekvizion.com / udp / cdp
<input type="checkbox"/>	Lynclabsd	Static SIP endpoint	Proxy Mode	10.64.3.166		2	lab.tekvizion.com / udp / cdp
<input type="checkbox"/>	ShoreTel_Mobility	Static SIP endpoint	Proxy Mode	10.64.2.159	ShoreTel MR sip trk	2	lab.tekvizion.com / udp / cdp
<input type="checkbox"/>	Verizon	Static SIP endpoint	Proxy Mode	10.70.2.201	Verizon SIP testing	1	lab.tekvizion.com / udp / cdp
<input type="checkbox"/>	cucm	Static SIP endpoint	Redirect Mode	10.80.16.3		3	lab.tekvizion.com / udp / cdp
<input type="checkbox"/>	nortel	Dynamic SIP endpoint	Redirect Mode	Not available		17	lab.tekvizion.com / udp / cdp

1 - 6 of 6 Gateway Endpoint(s) Page 1 of 1 First| Previous Next| Last

Cisco UCM Endpoint

Navigation Path: CS1000 NRSM → Numbering Plans → Endpoints → cucm

AVAYA Network Routing Service Manager Help | Logout

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans](#) » Endpoints » Gateway Endpoint

Edit Gateway Endpoint lab.tekvizion.com / udp / cdp)

End point name: *

Description:

Trust Node:

Tandem gateway endpoint name:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Avaya CS1000E Endpoint Configuration (Continued)

«UCM Network Services - System NRS Server Database System Wide Settings - Numbering Plans Domains Endpoints Routes Network Post-Translation Collaborative Servers - Tools SIP Phone Context - Routing Tests H.323 SIP Backup Restore GK/NRS Data upgrade	Managing: <input type="radio"/> Active database 10.0.0.1 <input checked="" type="radio"/> Standby database Numbering Plans » Endpoints » Gateway Endpoint
	Edit Gateway Endpoint lab.tekvizion.com / udp / cdp) Private L1 domain (UDP location) dialing access code: <input type="text"/> Private L1 domain (UDP location) dialing code length: <input type="text"/> (0-99) Private Special number 1: <input type="text"/> Private Special number 1 dialing code length: <input type="text"/> (0-31) Private Special number 2: <input type="text"/> Private Special number 2 dialing code length: <input type="text"/> (0-31) Static endpoint address type: IP version 4 ▾ Static endpoint address: 10.80.16.3 H.323 support: H.323 not supported ▾ SIP support: Static SIP endpoint ▾ SIP mode: <input type="radio"/> Proxy Mode <input checked="" type="radio"/> Redirect Mode SIP TCP transport enabled: <input type="checkbox"/> SIP TCP port: 5060 SIP UDP transport enabled: <input checked="" type="checkbox"/> SIP UDP port: 5060 SIP TLS transport enabled: <input type="checkbox"/> SIP TLS port: 5061 Persistent TCP support enabled: <input checked="" type="checkbox"/> End to end security support: <input type="checkbox"/> Network Connection Server enabled: <input type="checkbox"/> Redundancy enabled: Not Configured ▾ Main endpoint name: Not Applicable ▾ Redundant endpoint name: Not Applicable ▾ Virtual Private Networks Identifier: <input type="text"/> (1-16383) Bandwidth Zone: <input type="text"/> (0-8000) User Parameter(s): <input type="text"/>

* Required value

Cisco CS1000E Endpoint

Navigation Path: CS1000 NRSM → Numbering Plans → Endpoints → Avaya

AVAYA Network Routing Service Manager [Help](#) | [Logout](#)

«UCM Network Services
- **System**
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Collaborative Servers
- **Tools**
SIP Phone Context
- **Routing Tests**
H.323
SIP
Backup
Restore
GK/NRS Data upgrade

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint lab.tekvizion.com / udp / cdp)

End point name: nortel *

Description:

Trust Node:

Tandem gateway endpoint name: Not Applicable ▾

Endpoint authentication enabled: Authentication off ▾

Authentication password:

E.164 country code: 1

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

E.164 local (subscriber) dialing code length: (0-99)

Private L1 domain (UDP location) dialing access code:

Private L1 domain (UDP location) dialing code length: (0-99)

Private Special number 1:

Private Special number 1 dialing code length: (0-31)

Private Special number 2:

Private Special number 2 dialing code length: (0-31)

Static endpoint address type: IP version 4 ▾

Static endpoint address:

H.323 support: H.323 not supported ▾

SIP support: Dynamic SIP endpoint ▾

SIP mode: Proxy Mode
 Redirect Mode

SIP TCP transport enabled:

SIP TCP port: 5060

Cisco UCM Endpoint Configuration (Continued)

<ul style="list-style-type: none"> Domains Endpoints Routes Network Post-Translati Collaborative Servers - Tools SIP Phone Context - Routing Tests H.323 SIP Backup Restore GK/NRS Data upgrade 	SIP UDP transport enabled: <input checked="" type="checkbox"/>
	SIP UDP port: 5060
	SIP TLS transport enabled: <input type="checkbox"/>
	SIP TLS port: 5061
	Persistent TCP support enabled: <input checked="" type="checkbox"/>
	End to end security support: <input type="checkbox"/>
	Network Connection Server enabled: <input type="checkbox"/>
	Redundancy enabled: Not Configured
	Main endpoint name: Not Applicable
	Redundant endpoint name: Not Applicable
	Virtual Private Networks Identifier: <input type="text"/> (1-16383)
	Bandwidth Zone: <input type="text"/> (0-8000)
	User Parameter(s): <input type="text"/>
	* Required value

Save Cancel

Cisco UCM Routing Entry

AVAYA		Network Routing Service Manager		Help Logout
<ul style="list-style-type: none"> «UCM Network Services - System NRS Server Database System Wide Settings - Numbering Plans Domains Endpoints Routes Network Post-Translati Collaborative Servers - Tools SIP Phone Context - Routing Tests 	Managing: <input type="radio"/> Active database	10.0.0.1		
	<input checked="" type="radio"/> Standby database	Numbering Plans » Routes » Routing Entry		
	Edit Routing Entry (lab.tekvizion.com / udp / cdp / cucm)			
	DN type: Private level 0 regional (CDP steering code)			
	DN prefix: 20 *			
	Route cost: 1 * (1-255)			
	* Required value.			
	Save Cancel			

Avaya CS1000E Routing Entry

AVAYA **Network Routing Service Manager** [Help](#) | [Logout](#)

«UCM Network Services
– System
NRS Server
Database
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Collaborative Servers
– Tools
SIP Phone Context
– **Routing Tests**

Managing: Active database 10.0.0.1
 Standby database [Numbering Plans](#) » [Routes](#) » [Routing Entry](#)

Edit Routing Entry (lab.tekvizion.com / udp / cdp / nortel)

DN type: Private level 0 regional (CDP steering code) ▼
DN prefix: 3 *
Route cost: 1 * (1-255)

* Required value.

Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version

CISCO **Cisco Unified CM Administration** [Navigation](#) Cisco Unified CM Administration
For Cisco Unified Communications Solutions **administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

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

Cisco Unified CM Administration

System version: 10.5.1.10000-7

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHZ, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned

Last Successful Logon: Thursday, December 4, 2014 1:17:30 AM CST

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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 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.

Cisco UCM SIP Trunk Security Profile

- Set Name* = Non Secure SIP Trunk Profile. This is used for this example.
- Set Description = This text is used to identify this SIP Trunk Security Profile.
- Check Accept out of dialog refer.
- Check Accept unsolicited notification.
- Check Accept replaces header.
- All other values are default.

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration" and it includes a navigation bar with "Cisco Unified CM Administration" and "administrator" roles. The configuration form is titled "SIP Trunk Security Profile Information" and contains the following fields and options:

Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
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 checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

At the bottom of the form, there are buttons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New".

Cisco UCM SIP Profile

Set Name*= Early Offer SIP Profile. This is used for this example.

Set Description = This text is used to identify this SIP Profile.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for SIP Profile Configuration. The page is titled "SIP Profile Configuration" and includes a navigation menu at the top. The main content area is divided into several sections:

- Status:** Shows "Status: Ready" and a note: "All SIP devices using this profile must be restarted before any changes will take affect."
- SIP Profile Information:** This section contains the following fields:
 - Name*:** CS1000_Standard SIP Profile
 - Description:** Default SIP Profile
 - Default MTP Telephony Event Payload Type*:** 101
 - Early Offer for G.Clear Calls*:** Disabled
 - User-Agent and Server header information*:** Send Unified CM Version Information as User-Agen
 - Version in User Agent and Server Header*:** Major And Minor
 - Dial String Interpretation*:** Phone number consists of characters 0-9, *, #, anc
 - Confidential Access Level Headers*:** Disabled
 - Redirect by Application
 - Disable Early Media on 180
 - Outgoing T.38 INVITE include audio mline
 - Use Fully Qualified Domain Name in SIP Requests
 - Assured Services SIP conformance
- SDP Information:** This section contains the following fields:
 - SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*:** TIAS and AS
 - SDP Transparency Profile:** Pass all unknown SDP attributes
 - Accept Audio Codec Preferences in Received Offer*:** Default
 - Require SDP Inactive Exchange for Mid-Call Media Change
 - Allow RR/RS bandwidth modifier (RFC 3556)
- Parameters used in Phone:** This section contains the following fields:
 - 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

Cisco Unified Communications Manager SIP Profile (Continued)

These values are default.

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
Resource Priority Namespace	< None >
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 type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP

Check Early Offer support for voice and video calls (insert MTP if needed)

All other values are default.

<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	
Normalization Script	
Normalization Script	< None >
<input type="checkbox"/> Enable Trace	
Parameter Name	Parameter Value
1	
Incoming Requests FROM URI Settings	
Caller ID DN	
Caller Name	
Trunk Specific Configuration	
Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)

Cisco Unified Communications Manager SIP Profile (Continued)

Check Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Check Send send-receive SDP in mid-call INVITE.

All other values are default.

Enable ANAT

Deliver Conference Bridge Identifier

Allow Passthrough of Configured Line Device Caller Information

Reject Anonymous Incoming Calls

Reject Anonymous Outgoing Calls

Send ILS Learned Destination Route String

SIP OPTIONS Ping

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

Cisco UCM SIP Trunk to Avaya Configuration

Set Device Name* = Avaya_CS1000. This is used for this example.

Set Description = This text is used to identify this Trunk Group.

Set Device Pool* = G711 Preferred. This is used for this example.

Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Cisco Unified CM Administration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "Trunk Configuration" with a "Related Links" section containing "Back To Find/List".

At the top of the configuration area, there are buttons for Save, Delete, Reset, and Add New. Below this, the "Status" section shows "Status: Ready". The "SIP Trunk Status" section indicates "Service Status: Full Service" and "Duration: Time In Full Service: 7 days 23 hours 4 minutes".

The "Device Information" section is the primary focus, with several fields highlighted by red boxes:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Nortel_CS1000
Description	
Device Pool*	G711 Preferred
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_SW_MTP
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	

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

All other values are default.

MLPP and Confidential Access Level Information	
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Routing Information	
<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	Default
SIP Privacy*	Default
Inbound Calls	
Significant Digits*	4
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

Check Redirecting Diversion Header Delivery – Inbound

Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

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

Clear Prefix Settings Default Prefix Settings

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

Incoming Called Party Settings

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

Clear Prefix Settings Default Prefix Settings

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

Connected Party Settings

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

Set Calling and Connected Party Info Format* = Deliver URI and DN in connected party, if available.
 Check Redirecting Diversion Header Delivery – Outbound.
 All other values are default.

Outbound Calls

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Calling and Connected Party Info Format* Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

Cisco Unified Communications Manager SIP Trunk to Avaya Configuration (Continued)

Set Destination Address = 10.64.2.131. This is used in this example.

Set SIP Trunk Security Profile*= Non Secure SIP Trunk Profile.

Set SIP Profile*= Early Offer SIP Profile.

Set DTMF Signaling Method*= No Preference.

Set Normalization Script = nortel_Script_As_Is. The Cisco UCM-Software Script should be applied at SIP trunk toward Avaya PBX. This script normalizes the SIP messaging to/from the Avaya for UC Voice Mail center MWI, History-Info to Diversion Header conversion, Diversion Header to History- Info header conversion and privacy=user to privacy=id conversion.

All other values are default.

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.64.2.131		5060	up

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

i *- indicates required item.

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

Cisco UCM SIP Trunk Normalization Script

Set Name*= Nortel_Script_DIV_PRIVACY. This is used for this example.
Set Description = This text is used to identify this SIP Normalization Script.
Set Content*= Please see full contents on next page.

All Other values are default

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Normalization Script. The page title is "SIP Normalization Script Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The configuration page has a toolbar with Save, Delete, Reset, Add New, and Import File buttons. The "SIP Normalization Script Info" section contains the following fields:

- Name*: Nortel_Script_DIV_PRIVACY
- Description: (empty)
- Content*:

```
Nortel = {}
trace.enable()
-- Tested with Nortel CS1000E release 7.65
Nortel.allowHeaders = {"History-Info"}
local mwi_number = scriptParameters.getValue("mwi-number")
if not mwi_number
then
mwi_number = "1001"
end
local function adjustRedirectInfo(msg)
local di = msg.getHeader("Diversion")
if not di
then
return
end
msg:convertDiversionToHI()

msg:removeHeader("Diversion")
local historyInfos = msg.getHeaderValues("History-Info")
msg:removeHeader("History-Info")
-- For debugging purposes, dump out what the Diversion header contained and dump out the
list of History-Info headers
-- produced by msg:convertDiversionToHI. These extra headers will help debug but should
be ignored by Nortel. Trace
-- should be disabled via Admin UI unless a problem is being debugged. Therefore, under
normal operating conditions,
-- the debug headers won't be included in the message.
if trace.enabled()
then
msg:addHeader("X-Debug-Diversion", di)
for i, hi in ipairs(historyInfos)
do
```

SIP Normalization script Configuration (Continued)

	<pre>msg:addHeader("X-Debug-History-Info", hi) end end -- Example: -- Original Diversion header generated natively by CUCM might have been this: -- Diversion: <sip:1000@10.10.10.100>;reason=unconditional;privacy=off;screen=yes -- -- The call to convertDiversionToHI will produce these: -- History-Info: <sip:1000@10.10.10.100:5060?Reason=sip;cause=302; text="unconditional">;index=1 -- History-Info: <sip:3005@10.10.10.200:5060>;index=1.1 -- -- However, Nortel needs something that looks like this: -- History-Info:<sip:1000@10.10.10.100?reason=sip%3Bcause%3D302%3Btext %3D%22Moved%20Temporarily%22>;index=1</pre>
	<pre>-- History-Info: <sip:3005@10.10.10.200>;index=2 -- -- This loop generates the additional History-Info header and uses the index value for the first header generated by -- convertDiversionToHI. Each header uses the index from the next. The last header uses the last value plus one. -- While processing each header, it also removes the port number from the URI and does any necessary conversion of -- to special characters to the escaped value for the embedded header. for i, hi in ipairs(historyInfos) do local uri = string.match(hi, '<(.*)%?') or string.match(hi, "<(.*)>;index=.*") or "" -- Strip out the port number. uri = string.gsub(uri, "@(.*):%d+", "@%1") -- Get the embedded header but without the ?Reason=sip part.</pre>
	<pre>if embed_header then embed_header = string.gsub(embed_header, "unconditional", "Moved Temporarily") embed_header = string.gsub(embed_header, ";", "%3B") embed_header = string.gsub(embed_header, "=", "%3D") embed_header = string.gsub(embed_header, "\\", "%22") embed_header = string.gsub(embed_header, " ", "%20") embed_header = string.format("?reason=sip%s", embed_header) end hi = string.format("<%s%>;index=%s", uri, embed_header or "", i) msg:addHeader("History-Info", hi) end end -- Remove OPTIONS from outbound INVITE requests. -- Convert Diversion to History-Info.</pre>
	<pre>function Nortel.outbound_INVITE(msg) msg:removeHeaderValue("Allow", "OPTIONS") adjustRedirectInfo(msg) end -- Remove OPTIONS from any outbound request function Nortel.outbound_ANY(msg) msg:removeHeaderValue("Allow", "OPTIONS") end -- Remove OPTIONS from any outbound response to any request function Nortel_outbound_ANY_ANY(msg) msg:removeHeaderValue("Allow", "OPTIONS") end -- Modify the From header so that the userpart is numeric. CUCM will natively send -- 'voicemail' as the userpart. Nortel does not handle that. This code changes -- the user part to 1000 or the value of the configured script parameter: mwi-number.</pre>

SIP Normalization script Configuration (Continued)

	<pre>function Nortel.outbound_NOTIFY(msg) msg:removeHeaderValue("Allow", "OPTIONS") local from = msg:getHeader("From") if from then from = from:gsub("voicemail", mwi_number) msg:modifyHeader("From", from) msg:addHeaderUriParameter("From", "user", "phone") end end -- Convert History-Info to Diversion for inbound invites. Also, remove the -- phone-context userpart parameter and user=phone URI parameter if either -- is present. function Nortel.inbound_INVITE(msg) msg:removeHeaderValue("Allow", "UPDATE")</pre>
	<pre>local hist = msg:getHeader("History-Info") if not hist then return end msg:convertHIToDiversion() msg:removeHeader("History-Info") local diversion = msg:getHeader("Diversion") if diversion then -- This first regex will remove the phone-context userpart parameter if there -- are other parameters after it but before the @. diversion = diversion:gsub(";phone%-context=[^;]*;([^@]*)@", ";%1@") -- This second regex will remove the phone-context userpart parameter if it -- is immediately before the @.</pre>
	<pre>diversion = diversion:gsub(";phone%-context=[^@]*@", "@") -- Remove user=phone URI parameter. diversion = diversion:gsub(";user=phone", "") diversion = diversion:gsub(";reason=deflection", ";reason=no-answer") -- Save the changes. msg:modifyHeader("Diversion", diversion) end end function Nortel.inbound_ANY_INVITE(msg) msg:removeHeaderValue("Allow", "UPDATE") end function Nortel.outbound_ANY_INVITE(msg) msg:removeHeaderValue("Allow", "OPTIONS") end</pre>
	<pre>function Nortel.inbound_200_INVITE(msg) -- Modify Privacy header value user to id local privacyValues = msg:getHeaderValues("Privacy") local paramName = string.match(privacyValues[1], "user") if paramName == "user" then msg:modifyHeader("Privacy", "id") end end end return Nortel</pre>

Note: The Cisco UCM-Software Script should be applied at SIP trunk toward Avaya PBX. This script normalizes the SIP messaging to/from the Avaya for UC Voice Mail center MWI, History-Info to Diversion Header conversion, Diversion Header to History-Info header conversion, Omitting Option and Update from Allow header.

Download the script “Normalization Script” at Cisco Downloads Home > Products > Unified Communications > Call Control > Cisco Unified Communications Manager (CallManager) > Cisco Unified Communications Manager Version 10.5 > SIP Normalization and Transparency Scripts:

<https://software.cisco.com/download/release.html?i=ly&mdfid=285963825&softwareid=284695022&release=Scripts&os=>

Cisco UCM Service Parameter

Set Duplex Streaming Enabled* = True. See Note under capture for more info.

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Service Parameter Configuration" and it is for the "administrator" user. The "Clusterwide Parameters (Service)" section is expanded, showing a list of parameters. The "Duplex Streaming Enabled*" parameter is highlighted with a red box and is currently set to "True".

Parameter Name	Current Value	Default Value
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
Duplex Streaming Enabled *	True	False
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage *	95	95

Note: Cisco Unified Communications Manager Service Parameter “Duplex Streaming Enabled” should be set to “True” in order for MoH and ring back to work properly during call transfers/conferences initiated by Cisco stations to Avaya IP endpoints.

Cisco UCM Media Resource Group

The screenshot shows the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below the navigation bar, a menu contains "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Find and List Media Resource Groups". It features a toolbar with icons for "Add New", "Select All", "Clear All", and "Delete Selected". Below this is a "Status" section indicating "2 records found".

The table below displays the list of Media Resource Groups. The first row, "MRG_SW_MTP", is highlighted with a red border. The table has columns for "Name", "Description", "Multi-cast", and "Copy".

<input type="checkbox"/>	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	MRG_SW_MTP	MRG_SW_MTP	false	
<input type="checkbox"/>	MRG_SW_noMTP	MRG_SW_noMTP	false	

At the bottom of the table area, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Media Resource Group MRG_MTP

Set Name*= MRG_SW_MTP. This is used for this example.

Set Description = This text is used to identify this Media Resource Group.

Set all resources in the Selected Media Resources* Box.

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration". The breadcrumb navigation shows: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The user is logged in as "administrator".

Media Resource Group Status
Media Resource Group: MRG_SW_MTP (used by 9 devices)

Media Resource Group Information

Name*	MRG_SW_MTP
Description	MRG_SW_MTP

Devices for this Group

Available Media Resources**

- ANN_2
- ANN_4
- CFB_2
- CFB_4
- EXTMTP

Selected Media Resources*

- ANN_3 (ANN)
- CFB_3 (CFB)
- MOH_3 (MOH)
- MTP_3 (MTP)

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

Buttons: Save, Delete, Copy, Add New

i *- indicates required item.

Cisco UCM Media Resource Group List

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu contains various system management options like "System", "Call Routing", "Media Resources", etc.

The current page is titled "Find and List Media Resource Group Lists". It features a toolbar with actions: "Add New", "Select All", "Clear All", and "Delete Selected". A status box indicates "2 records found".

The main content area shows a table of Media Resource Group Lists. The table has a search filter set to "Name begins with" and a "Rows per Page" dropdown set to 50. The table lists two entries:

<input type="checkbox"/>	Name ^	Copy
<input type="checkbox"/>	MRGL_SW_MTP	
<input type="checkbox"/>	MRGL_SW_noMTP	

At the bottom of the table area, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Set Name*= MRGL_SW_MTP This is used for this example.
Set Selected Media Resource Groups= MRG_SW_MTP

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration". The navigation bar includes "Navigation" (Cisco Unified CM Administration), "administrator", "Search Documentation", "About", and "Logout". The breadcrumb trail shows: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The "Related Links" section contains "Back To Find/List" and "Go".

Below the navigation, there are icons for "Save", "Delete", "Copy", and "Add New". The main content area is divided into three sections:

- Media Resource Group List Status:** Media Resource Group List: MRGL_SW_MTP (used by 9 devices)
- Media Resource Group List Information:** Name* MRGL_SW_MTP (highlighted with a red box)
- Media Resource Groups for this List:**
 - Available Media Resource Groups: MRG_SW_noMTP
 - Selected Media Resource Groups: MRG_SW_MTP (highlighted with a red box)

At the bottom, there are buttons for "Save", "Delete", "Copy", and "Add New". A footer note states: "i *- indicates required item."

Note: This Media Resource Group List was added to provide early offer on the invite from Cisco to Avaya for SCCP phones.

Cisco UCM Route Pattern to Avaya

Set Route Pattern* =3XXX. This is used to route Avaya in this example.
Set Description = this text is used to identify this Route Pattern.
Set Gateway/Route List* = Avaya_CS1000. This is used for this example.
All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a user role of "administrator". A secondary navigation bar lists various system functions like "System", "Call Routing", "Media Resources", etc. The main content area is titled "Route Pattern Configuration" and includes a toolbar with "Save", "Delete", "Copy", and "Add New" buttons. Below this, the "Status" section shows "Status: Ready". The "Pattern Definition" section contains several fields: "Route Pattern*" (3XXX), "Route Partition" (< None >), "Description" (call to CS1000), "Numbering Plan" (-- Not Selected --), "Route Filter" (< None >), "MLPP Precedence*" (Default), "Apply Call Blocking Percentage" (checkbox), "Resource Priority Namespace Network Domain" (< None >), "Route Class*" (Default), "Gateway/Route List*" (Nortel_CS1000), and "Route Option" (Route this pattern). Red boxes highlight the "Route Pattern*", "Description", and "Gateway/Route List*" fields. An "(Edit)" link is visible next to the "Gateway/Route List*" field.

Field	Value
Route Pattern*	3XXX
Route Partition	< None >
Description	call to CS1000
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	<input type="checkbox"/>
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	Nortel_CS1000
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern

Route Patter Configuration for 3xxx (Continued)

All other values are default.

Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			
Calling Party Transformations			
<input type="checkbox"/> Use Calling Party's External Phone Number Mask			
Calling Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Calling Line ID Presentation*	Default		
Calling Name Presentation*	Default		
Calling Party Number Type*	Cisco CallManager		
Calling Party Numbering Plan*	Cisco CallManager		
Connected Party Transformations			
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		
Called Party Transformations			
Discard Digits	< None >		
Called Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		
ISDN Network-Specific Facilities Information Element			
Network Service Protocol	-- Not Selected --		
Carrier Identification Code			
Network Service	Service Parameter Name	Service F	
-- Not Selected --	< Not Exist >		
Save	Delete	Copy	Add New

Cisco UCM SIP Phone Ext. 2003 Device Level Configuration

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes "Cisco Unified CM Administration" and "administrator". The main menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Find and List Phones" section is active, showing "7 records found". The "Query Information" section states: "Searching on a directory number may show the same device name multiple times depending on the number of lines configured per device." The "Phone (1 - 7 of 7)" section shows a search for "Directory Number" "begins with" "200". The table below lists the search results:

Device Name(Line)	Description	Device Pool	Extension	Partition	Device Protocol	Status	IPv4 Address	Copy	Super Copy
SEPFCFBFBCAF26D(1)	SEPFCFBFBCAF26D	Default	2000		SIP	None	None		
SEP1C17D337D19F(1)	SEP1C17D337D19F	G729 Preferred	2001		SIP	None	None		
SEPFCFBFB115842(1)	SEPFCFBFB115842	Default	2002		SCCP	None	None		
SEPC07BBCA1B846(1)	9971_SIP_1	Default	2003		SIP	Registered with clus26sub1	172.16.31.139		
SEP64D814A44CF7(1)	SCCP-7942	Default	2004		SCCP	Registered with clus26sub1	172.16.31.190		
SEP0C2724315FB9(1)	7965_SCCP	Default	2005		SCCP	None	None		
SEPC07BBCA1B872(1)	9971_SIP_2	Default	2006		SIP	None	None		

Set MAC Address* = C07BBCA1B846. This is used in this example.
 Set Description = 9971_SIP_1. This text is used to identify this Phone.
 Set Device Pool* = Default. This is used in this example.
 Set Phone Button Template* = Standard 9971 SIP. This is used in this example.
 All other values are default.

Cisco Unified Communications Manager SIP Phone Ext. 5000 Device Level Configuration (Continued)

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP phone. The page title is "Cisco Unified CM Administration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help.

The main section is "Phone Configuration" with a "Related Links" dropdown set to "Back To Find/List". Below this are action buttons: Save, Delete, Copy, Reset, Apply Config, and Add New.

On the left, there is a "Modify Button Items" section with a list of 20 items. Item 1 is "Line [1] - 2003 (no partition)", and items 3, 4, and 5 are "Add a new SD".

The right-hand side contains configuration details for a "Cisco 9971" device with "SIP" protocol. It includes a "Real-time Device Status" section with registration information (Registered with Cisco Unified Communications Manager clus26sub1, IPv4 Address: 172.16.31.139, Active Load ID: sip9971.9-4-1-9, Inactive Load ID: sip9971.9-3-2-10, Download Status: None).

The "Device Information" section shows that the device is active and trusted. Two red boxes highlight specific fields:

- The first box highlights the MAC Address* (C07BBCA1B846), Description (9971_SIP_1), and Device Pool* (Default).
- The second box highlights the Phone Button Template* (Standard 9971 SIP), Softkey Template (Standard User park), and Common Phone Profile* (Standard Common Phone Profile).

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

All other values are default.

Set Media Resource Group List = MRGL_SW_MTP. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

21	None
22	None
23	None
24	None
25	None
26	None
27	None
28	None
29	None
30	None
31	None
32	None
33	None
34	None
35	None
36	None
37	None
38	None
39	None
40	None
41	None

Media Resource Group List	MRGL_SW_MTP
User Hold MOH Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	1-SampleAudioSource
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default View Current Device Mobility Settings
Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
Owner User ID	
Phone Personalization*	Default
Services Provisioning*	Default
Phone Load Name	
Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default

42	None
43	None
44	None
45	None
46	None
47	None
48	None
49	None
50	None
51	None
52	None
53	None
54	None
55	None
56	None
57	None
58	None
59	None
60	None
61	None
62	None
63	None
64	None
65	None

BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
Feature Control Policy	Call Park
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
<input checked="" type="checkbox"/> Allow Control of Device from CTI	
<input checked="" type="checkbox"/> Logged Into Hunt Group	
<input type="checkbox"/> Remote Device	
<input type="checkbox"/> Protected Device****	
<input type="checkbox"/> Require off-premise location	

Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

43	None
44	None
45	None
46	None
47	None
48	None
49	None
50	None
51	None
52	None
53	None
54	None
55	None
56	None
57	None
58	None
59	None
60	None
61	None
62	None
63	None
64	None
65	None

BLF Audible Alert Setting (Phone Busy)* Default

Always Use Prime Line* Default

Always Use Prime Line for Voice Message* Default

Geolocation < None >

Feature Control Policy Call Park

Ignore Presentation Indicators (internal calls only)

Allow Control of Device from CTI

Logged Into Hunt Group

Remote Device

Protected Device****

Require off-premise location

Number Presentation Transformation

Caller ID For Calls From This Phone

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

Remote Number

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

85	None
86	None
87	None
88	None
89	None
90	None
91	None
92	None
93	None
94	None
95	None
96	None
97	None
98	None
99	None
100	None
101	None
102	None
103	None

Certification Authority Proxy Function (CAPF) Information

Certificate Operation* No Pending Operation

Authentication Mode* By Null String

Authentication String

Key Size (Bits)* 1024

Operation Completes By 2014 12 14 12 (YYYY:MM:DD:HH)

Certificate Operation Status: None

Note: Security Profile Contains Addition CAPF Settings.

Expansion Module Information

Module 1 < None >

Module 1 Load Name

Module 2 < None >

Module 2 Load Name

Module 3 < None >

Module 3 Load Name

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

104	None
105	None
106	None
107	None
108	None
109	None
110	None
111	None
112	None
113	None
114	None
----- Unassigned Associated Items -----	
115	Add a new SD
116	Line [2] - Add a new DN
117	All Calls
118	Add a new BLF Directed Call Park
119	Call Pickup
120	CallBack

External Data Locations Information (Leave blank to use default)

Information

Directory

Messages

Services

Authentication Server

Proxy Server

Idle

Idle Timer (seconds)

Secure Authentication URL

Secure Directory URL

Secure Idle URL

Secure Information URL

Secure Messages URL

Secure Services URL

121	Group Call Pickup
122	Hunt Group Logout
123	Intercom [1] - Add a new Intercom
124	Malicious Call Identification
125	Meet Me Conference
126	Mobility
127	Other Pickup
128	Quality Reporting Tool
129	Redial
130	Add a new SURL
131	Add a new BLF SD
132	Answer Oldest
133	Do Not Disturb
134	Services
135	Record
136	Alerting Calls
137	Queue Status
138	Privacy
139	None

Extension Information

Enable Extension Mobility

Log Out Profile

Log in Time

Log out Time

MLPP and Confidential Access Level Information

MLPP Domain

MLPP Indication*

MLPP Preemption*

Confidential Access Mode

Confidential Access Level

Do Not Disturb

Do Not Disturb

DND Option*

DND Incoming Call Alert

Secure Shell Information

Secure Shell User

Secure Shell Password

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

Product Specific Configuration Layout		Parameter Value	Override Common Settings
		?	
<input type="checkbox"/>	Disable Speakerphone		
<input type="checkbox"/>	Disable Speakerphone and Headset		
	PC Port *	Enabled	
	Back USB Port*	Enabled	<input type="checkbox"/>
	Side USB Port*	Enabled	<input type="checkbox"/>
	Cisco Camera*	Disabled	<input type="checkbox"/>
	Console Access*	Disabled	<input type="checkbox"/>
	Video Capabilities*	Enabled	<input checked="" type="checkbox"/>
	Enable/Disable USB Classes	Mass Storage Human Interface Device Audio Class	<input type="checkbox"/>
	SDIO *	Disabled	<input type="checkbox"/>
	Bluetooth *	Enabled	<input type="checkbox"/>
	Wifi *	Enabled	<input type="checkbox"/>
	Bluetooth Profiles*	Handsfree Human Interface Device	<input type="checkbox"/>
	Settings Access*	Enabled	<input type="checkbox"/>
	Gratuitous ARP*	Disabled	
	PC Voice VLAN Access *	Enabled	

	Web Access*	Disabled	<input type="checkbox"/>
	Show All Calls on Primary Line*	Disabled	
	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"/>
	HTTPS Server*	http and https Enabled	<input type="checkbox"/>
	Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
	Phone On Time	00:00	<input type="checkbox"/>
	Phone Off Time	24:00	<input type="checkbox"/>
	Phone Off Idle Timeout*	60	<input type="checkbox"/>
	<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
	EnergyWise Domain		<input type="checkbox"/>
	EnergyWise Endpoint Security Secret		<input type="checkbox"/>

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

<input type="checkbox"/>	Allow EnergyWise Overrides		<input type="checkbox"/>
	Span to PC Port*	Disabled	<input type="checkbox"/>
	Logging Display*	Disabled	<input type="checkbox"/>
	Load Server	<input type="text"/>	<input type="checkbox"/>
	IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
	Recording Tone*	Disabled	<input type="checkbox"/>
	Recording Tone Local Volume*	100	<input type="checkbox"/>
	Recording Tone Remote Volume*	50	<input type="checkbox"/>
	Recording Tone Duration	<input type="text"/>	<input type="checkbox"/>
	Display On When Incoming Call*	Enabled	<input type="checkbox"/>
	RTCP*	Disabled	<input type="checkbox"/>
	Log Server	<input type="text"/>	<input type="checkbox"/>
	IPv6 Log Server	<input type="text"/>	<input type="checkbox"/>
	Remote Log*	Disabled	<input type="checkbox"/>
	Log Profile	Default Preset Telephony	<input type="checkbox"/>
	Advertise G.722 and iSAC Codecs *	Use System Default	<input type="checkbox"/>
	Wideband Headset UI Control*	Enabled	<input type="checkbox"/>

	Wideband Headset*	Enabled	<input type="checkbox"/>
	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	<input type="text"/>	<input type="checkbox"/>
	LLDP Power Priority*	Unknown	<input type="checkbox"/>
	802.1x Authentication *	User Controlled	<input type="checkbox"/>
	FIPS Mode*	Disabled	<input type="checkbox"/>
	Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
	Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>

Cisco Unified Communications Manager SIP Phone Ext. 2003 Device Level Configuration (Continued)

PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>
Power Negotiation*	Enabled	<input type="checkbox"/>
Restrict Data Rates*	Disabled	<input type="checkbox"/>
SSH Access*	Disabled	<input type="checkbox"/>
Incoming Call Toast Timer*	5	<input type="checkbox"/>
Provide Dial Tone from Release Button*	Disabled	<input type="checkbox"/>
Hide Video By Default*	Disabled	<input type="checkbox"/>
Background Image	<input type="text"/>	<input type="checkbox"/>
Simplified New Call UI*	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC	<input type="text"/>	<input type="checkbox"/>
VXC VPN Option*	Dual Tunnel	<input type="checkbox"/>
VXC Challenge*	Challenge	<input type="checkbox"/>
VXC-M Servers	<input type="text"/>	<input type="checkbox"/>
Revert to All Calls*	Disabled	<input type="checkbox"/>
80-bit SRTCP*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input type="checkbox"/>
Record Call Log from Shared Line*	Disabled	<input type="checkbox"/>

Show Call History for Selected Line Only.*	Disabled	<input type="checkbox"/>
Actionable Incoming Call Alert*	Disabled	<input type="checkbox"/>
DF bit*	0	<input type="checkbox"/>
Default Line Filter	<input type="text"/>	<input type="checkbox"/>
Separate Audio and Video Mute*	Disabled	<input type="checkbox"/>
Softkey Control*	Feature Control Policy	<input type="checkbox"/>
Start Video Port	<input type="text"/>	<input type="checkbox"/>
Stop Video Port	<input type="text"/>	<input type="checkbox"/>
Lowest Alerting Line State Priority*	Disabled	<input type="checkbox"/>
TLS Resumption Timer*	3600	<input type="checkbox"/>

i *- indicates required item.

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

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

i ****Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

Cisco UCM SCCP Phone Ext. 2004 Device Level Configuration

Set MAC Address* = 64D814A44CF7. This is used in this example.

Set Description = SCCP-7942. This text is used to identify this Phone

Set Device Pool* = Default. This is used in this example.

Set Phone Button Template* = Standard 7942G SCCP. This is used in this example

Set Media Resource Group List = MRGL_SW_MTP. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

All other values are default.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Association

Modify Button Items

- Line [1] - 2004 (no partition)
- Line [2] - Add a new DN
- Unassigned Associated Items
- Add a new SURL
- Add a new BLF SD
- Add a new SD
- Add a new BLF Directed Call Park
- CallBack
- Call Park
- Call Pickup
- Conference List
- Conference
- Do Not Disturb
- End Call
- Forward All
- Group Call Pickup
- Hold
- Hunt Group Logout

Phone Type

Product Type: Cisco 7942
Device Protocol: SCCP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager clus26sub1
IPv4 Address: 172.16.31.190
Active Load ID: SCCP42.9-3-1SR3-1S
Download Status: None

Device Information

Device is Active
 Device is trusted

MAC Address* 64D814A44CF7
Description SCCP-7942
Device Pool* Default [View Details](#)
Common Device Configuration < None > [View Details](#)
Phone Button Template* Standard 7942G SCCP
Softkey Template Standard User
Common Phone Profile* Standard Common Phone Profile [View Details](#)
Calling Search Space < None >

Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

<ul style="list-style-type: none"> 20 Meet Me Conference 21 Mobility 22 New Call 23 Other Pickup 24 Quality Reporting Tool 25 Redial 26 Remove Last Participant 27 Transfer 28 Video Mode 29 Queue Status 30 Privacy 31 None 	<p>Media Resource Group List <input type="text" value="MRGL_SW_MTP"/></p> <p>User Hold MOH Audio Source <input type="text" value="1-SampleAudioSource"/></p> <p>Network Hold MOH Audio Source <input type="text" value="1-SampleAudioSource"/></p> <p>Location* <input type="text" value="Hub_None"/></p> <p>AAR Group <input type="text" value="< None >"/></p> <p>User Locale <input type="text" value="< None >"/></p> <p>Network Locale <input type="text" value="< None >"/></p> <p>Built In Bridge* <input type="text" value="Default"/></p> <p>Privacy* <input type="text" value="Default"/></p> <p>Device Mobility Mode* <input type="text" value="Default"/> View Current Device Mobility Settings</p> <p>Owner <input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)</p> <p>Owner User ID <input type="text"/></p> <p>Phone Personalization* <input type="text" value="Default"/></p> <p>Services Provisioning* <input type="text" value="Default"/></p> <p>Phone Load Name <input type="text"/></p> <p>Single Button Barge <input type="text" value="Default"/></p> <p>Join Across Lines <input type="text" value="Default"/></p> <p>Use Trusted Relay Point* <input type="text" value="Default"/></p>
--	---

	<p>BLF Audible Alert Setting (Phone Idle)* <input type="text" value="Default"/></p> <p>BLF Audible Alert Setting (Phone Busy)* <input type="text" value="Default"/></p> <p>Always Use Prime Line* <input type="text" value="Default"/></p> <p>Always Use Prime Line for Voice Message* <input type="text" value="Default"/></p> <p>Geolocation <input type="text" value="< None >"/></p> <p><input checked="" type="checkbox"/> Retry Video Call as Audio</p> <p><input type="checkbox"/> Ignore Presentation Indicators (internal calls only)</p> <p><input checked="" type="checkbox"/> Allow Control of Device from CTI</p> <p><input checked="" type="checkbox"/> Logged Into Hunt Group</p> <p><input type="checkbox"/> Remote Device</p> <p><input type="checkbox"/> Protected Device****</p> <p><input type="checkbox"/> Hot line Device*****</p> <p><input type="checkbox"/> Require off-premise location</p>
--	---

Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

<input type="checkbox"/> RFC2833 Disabled	
Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Key Size (Bits)*	1024
Operation Completes By	2014 12 14 12 (YYYY:MM:DD:HH)
Certificate Operation Status:	None
Note: Security Profile Contains Addition CAPF Settings.	
External Data Locations Information (Leave blank to use default)	
Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

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 and Confidential Access Level Information	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >
Confidential Access Level	< None >

Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

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="text"/>
Product Specific Configuration Layout	
?	Parameter Value
<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"/>
Override Common Settings	

Auto Line Select *	Disabled	
Web Access*	Disabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain	<input type="text"/>	<input type="checkbox"/>
EnergyWise Endpoint Security Secret	<input type="text"/>	<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	PC Controlled	
Load Server	<input type="text"/>	<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	


Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)


Recording Tone Duration	<input type="text"/>	
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server	<input type="text"/>	<input type="checkbox"/>
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):	Enabled	<input type="checkbox"/>


Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID	<input type="text"/>	
LLDP Power Priority*	Unknown	
Wireless Headset Hookswitch Control*	Disabled	
Display Refresh Rate*	Normal	
IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	
802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
Minimum Ring Volume*	0-Silent	
Headset Sidetone Level*	Default	
Headset Send Gain*	Default	
HTTPS Server*	http and https Enabled	<input type="checkbox"/>

Cisco Unified Communications Manager SCCP Phone Ext. 2004 Device Level Configuration (Continued)

HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Handset/Headset Monitor*	Enabled	<input type="checkbox"/>
Headset Recording*	Disabled	<input type="checkbox"/>
Enbloc Dialing*	Enabled	<input type="checkbox"/>
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>
SSH Access*	Disabled	<input type="checkbox"/>
LOGIN Access*	Enabled	<input type="checkbox"/>
FIPS Mode*	Disabled	<input type="checkbox"/>
80-bit SRTCP*	Disabled	<input type="checkbox"/>

 *- indicates required item.

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

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

Cisco UCM Audio Codec Preference List Configuration

Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Service Parameter Configuration" and it is for "Parameters for All Servers". The user is logged in as "administrator". The page contains a list of service parameters with their current values and factory defaults. The parameter "Accept Audio Codec Preferences in Received Offer" is highlighted with a red box and is set to "Off".

Parameter Name	Current Value	Factory Default
(Includes Audio) *	304	---
Default Interregion Max Video Call Bit Rate (Includes Audio) *	384	384
Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Default Interregion Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Use Video BandwidthPool for Immersive Video Calls *	True	True
Default Intraregion and Interregion Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override *	False	False

Clusterwide Parameters (System - CCM Automated Alternate Routing)

Automated Alternate Routing Enable *	False	False
--------------------------------------	-------	-------

Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

G711 Preferred and G729 Preferred Audio Codec Preference List created in this example. All other values are default.

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and the user role "administrator". Below the navigation bar, a menu contains options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Find and List Audio Codec Preference Lists". It features a toolbar with "Add New", "Select All", "Clear All", and "Delete Selected" buttons. A status box indicates "4 records found".

The table below lists the audio codec preference lists. The "G711 G729" and "G729 G711" rows are highlighted with a red border. The table has columns for "Name", "Description", and "Copy".

<input type="checkbox"/>	Name ^	Description	Copy
<input type="checkbox"/>	Factory Default lossy	Lossy Codec List	
<input type="checkbox"/>	Factory Default low loss	Low Loss Codec List	
<input type="checkbox"/>	G711 G729	G711 G729	
<input type="checkbox"/>	G729 G711	G729 G711	

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name*= G711 G729. This is used for this example.

Set Description*= G711 G729. This text is used to identify this Audio Codec Preference List.

Set Codec in List*= G.711 U-Law 64k. First choice in this example.

Set Codec in List*= G.711 A-Law 64k. Second choice in this example.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below this is a menu bar with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Audio Codec Preference List Configuration" and includes a "Related Links" section with a "Back To Find/List" button. A toolbar contains "Save", "Delete", "Copy", and "Add New" icons. The "Status" section shows "Status: Ready". The "Audio Codec Preference List Information" section contains a form with the following fields:

Name*	G711 G729
Description*	G711 G729
Codecs in List*	<ul style="list-style-type: none">G.711 U-Law 64kG.711 A-Law 64kG.729 8kG.729a 8kG.729b 8kAMR-WB (7k-24k)AMR (5k-13k)MP4A-LATM 128kAAC-LD (MP4A Generic)MP4A-LATM 64kMP4A-LATM 56kL16 256kMP4A-LATM 48kG.722 64kISAC 32kMP4A-LATM 32kG.722.1 32kG.722 56kG.722.1 24kG.722 48kMP4A-LATM 24k

Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

Set Name* = G729 G711. This is used for this example.

Set Description* = G729 G711. This text is used to identify this Audio Codec Preference List.

Set Codec in List* = G.729 8k. First choice for this example.

Set Codec in List* = G.729a 8k. Second choice for this example.

Set Codec in List* = G.729b 8k. Second choice for this example.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below the navigation bar is a menu with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Audio Codec Preference List Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a toolbar with "Save", "Delete", "Copy", and "Add New" icons. The "Status" section shows "Status: Ready". The "Audio Codec Preference List Information" section contains a form with the following fields:

Name*	G729 G711
Description*	G729 G711
Codecs in List*	<ul style="list-style-type: none">G.729 8kG.729a 8kG.729b 8kG.711 U-Law 64kG.711 A-Law 56kAMR-WB (7k-24k)AMR (5k-13k)MP4A-LATM 128kAAC-LD (MP4A Generic)MP4A-LATM 64kMP4A-LATM 56kL16 256kMP4A-LATM 48kG.722 64kISAC 32kMP4A-LATM 32kG.722.1 32kG.722 56kG.722.1 24kG.722 48kMP4A-LATM 24k

Cisco UCM Region Configuration

G711 Region and G729 Region created in this example.
All other values are default.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and the user role "administrator". Below the navigation bar, there are several menu items: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk Administration. The main content area is titled "Find and List Regions". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, a status box indicates "3 records found". The main table displays a list of regions with the following columns: a checkbox for selection and a "Name" column. The table contains three rows: "Default", "G711 Preferred", and "G729 Preferred". The "G711 Preferred" and "G729 Preferred" entries are highlighted with a red box. Below the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

	Name ^
<input type="checkbox"/>	Default
<input type="checkbox"/>	G711 Preferred
<input type="checkbox"/>	G729 Preferred

Cisco Unified Communications Manager Region Configuration (Continued)

Set Name*= G711 Preferred. This is used in this example.

Set Region= G711 Preferred. This is used in this example

Set Audio Codec Preference List= G711 Preferred.

Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.

Region Configuration Navigation: Cisco Unified CM Administration

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

Related Links:

Save

Region Information

Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711 G729	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G711 Preferred	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<input type="checkbox"/> Default <input checked="" type="checkbox"/> G711 Preferred <input type="checkbox"/> G729 Preferred	<input type="radio"/> Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> Use System Default <input type="radio"/> None	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps

Cisco Unified Communications Manager Region Configuration (Continued)

Set Name*= G729 Region. This is used in this example.

Set Audio Codec Preference List= G729 Preferred. This is used in this example

Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.

Region Information

Name* | G729 Preferred

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G729 G711	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Preferred	Factory Default low loss	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default G711 Preferred G729 Preferred	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> [] kbps	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> [] kbps

Cisco UCM Device Pool Configuration

G711 Preferred Pool and G729 Preferred Pool created in this example.
All other values are default.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

Find and List Device Pools

+ Add New | Select All | Clear All | Delete Selected

Status
3 records found

Device Pool (1 - 3 of 3) Rows per Page 50

Find Device Pool where Device Pool Name begins with Find Clear Filter + -

<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	Default	Default	Default	CMLocal	
<input type="checkbox"/>	G711 Preferred	Default	G711 Preferred	CMLocal	
<input type="checkbox"/>	G729 Preferred	Default	G729 Preferred	CMLocal	

+ Add New | Select All | Clear All | Delete Selected

Ring Schedule

All the time
 As specified below

Monday All Day No Office Hours to No Office Hours

Tuesday All Day No Office Hours to No Office Hours

Wednesday All Day No Office Hours to No Office Hours

Thursday All Day No Office Hours to No Office Hours

Friday All Day No Office Hours to No Office Hours

Saturday All Day No Office Hours to No Office Hours

Sunday All Day No Office Hours to No Office Hours

Time Zone* (GMT) Etc/GMT

When receiving a call during the above ring schedule

Always ring this destination
 Ring this destination only if caller is in -- Not Selected -- [View Details](#)
 Do not ring this destination if caller is in -- Not Selected -- [View Details](#)

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name* = G711 Preferred. This is used in this example.

Set Cisco Unified Communications Manager Group* = Default.

Set Date/Time Group* = CMLocal.

Set Region* = G711 Preferred. This is used in this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a Device Pool. The page title is "Device Pool Configuration" and the user is logged in as "administrator". The configuration is for a device pool named "G711 Preferred" which currently has 1 member. The configuration is divided into several sections:

- Device Pool Information:** Shows the device pool name and member count.
- Device Pool Settings:** Contains several dropdown menus:
 - Device Pool Name*: G711 Preferred (highlighted with a red box)
 - Cisco Unified Communications Manager Group*: Default
 - Calling Search Space for Auto-registration: < None >
 - Adjunct CSS: < None >
 - Reverted Call Focus Priority: Default
 - Intercompany Media Services Enrolled Group: < None >
- Local Route Group Settings:** Shows the Standard Local Route Group set to < None >.
- Roaming Sensitive Settings:** Contains several dropdown menus:
 - Date/Time Group*: CMLocal (highlighted with a red box)
 - Region*: G711 Preferred (highlighted with a red box)
 - Media Resource Group List: < None >
 - Location: < None >
 - Network Locale: < None >
 - SRST Reference*: Disable

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Connection Monitor Duration***	<input type="text"/>
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > View Details

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >
Calling Party Transformation CSS	< None >
Called Party Transformation CSS	< None >

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

Call Routing Information

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
National Number	Default	<input type="text"/>	< None >
International Number	Default	<input type="text"/>	< None >
Unknown Number	Default	<input type="text"/>	< None >
Subscriber Number	Default	<input type="text"/>	< None >

Incoming Called 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
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Subscriber Number	Default	0	< None >
Phone Settings			
Caller ID For Calls From This Phone			
Calling Party Transformation CSS < None >			
Connected Party Settings			
Connected Party Transformation CSS < None >			
Redirecting Party Settings			
Redirecting Party Transformation CSS < None >			
Save Delete Copy Reset Apply Config Add New			
 *- indicates required item.			
 **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.			
 ***Leave the field blank or enter -1 to use the configuration from the enterprise parameter.			

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Set Device Pool Name* = G729 Preferred. This is used in this example.

Set Cisco Unified Communications Manager Group* = Default.

Set Date/Time Group* = CMLocal.

Set Region* = G729 Preferred. This is used in this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a Device Pool. The page is titled "Device Pool Configuration" and includes a navigation menu at the top. The main content area is divided into several sections:

- Device Pool Information:** Shows the Device Pool name as "G729 Preferred" with 1 member.
- Device Pool Settings:** Contains several dropdown menus:
 - Device Pool Name*: G729 Preferred (highlighted with a red box)
 - Cisco Unified Communications Manager Group*: Default
 - Calling Search Space for Auto-registration: < None >
 - Adjunct CSS: < None >
 - Reverted Call Focus Priority: Default
 - Intercompany Media Services Enrolled Group: < None >
- Local Route Group Settings:** Shows the Standard Local Route Group set to < None >.
- Roaming Sensitive Settings:** Contains several dropdown menus:
 - Date/Time Group*: CMLocal (highlighted with a red box)
 - Region*: G729 Preferred (highlighted with a red box)
 - Media Resource Group List: < None >
 - Location: < None >
 - Network Locale: < None >
 - SRST Reference*: Disable

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Connection Monitor Duration***	<input type="text"/>
Single Button Barge*	Default <input type="button" value="v"/>
Join Across Lines*	Default <input type="button" value="v"/>
Physical Location	< None > <input type="button" value="v"/>
Device Mobility Group	< None > <input type="button" value="v"/>
Wireless LAN Profile Group	< None > <input type="button" value="v"/> View Details

Device Mobility Related Information****

Device Mobility Calling Search Space	< None > <input type="button" value="v"/>
AAR Calling Search Space	< None > <input type="button" value="v"/>
AAR Group	< None > <input type="button" value="v"/>
Calling Party Transformation CSS	< None > <input type="button" value="v"/>
Called Party Transformation CSS	< None > <input type="button" value="v"/>

Geolocation Configuration

Geolocation	< None > <input type="button" value="v"/>
Geolocation Filter	< None > <input type="button" value="v"/>

Call Routing Information

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
National Number	Default <input type="text"/>	<input type="text"/>	< None > <input type="button" value="v"/>
International Number	Default <input type="text"/>	<input type="text"/>	< None > <input type="button" value="v"/>
Unknown Number	Default <input type="text"/>	<input type="text"/>	< None > <input type="button" value="v"/>
Subscriber Number	Default <input type="text"/>	<input type="text"/>	< None > <input type="button" value="v"/>

Incoming Called 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.

Cisco Unified Communications Manager Device Pool Configuration (Continued)

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	< None >

Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS

Cisco UCM Extent and Connect

Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications which interoperate with any endpoint. With Extend and Connect, users can Leverage the benefits of UC applications from any location using any device. This feature also allows Interoperability between newer UC solutions and legacy systems, so customers can migrate to newer UC Solutions over time as existing hardware is deprecated.

Cisco UCM end user configuration

Add user to Cisco UCM

Navigation Path: User management → End user

Set User ID*= user2. This is used for this example.

Set Last Name = Jabber2. This is used for this example.

Check Home Cluster.

The screenshot displays the Cisco Unified CM Administration web interface. The page title is "End User Configuration" and it shows the configuration for an "Active Local User". The user's status is "Ready". The "User Information" section contains the following fields:

User Status	Active Local User
User ID*	user2
Password
Confirm Password
Self-Service User ID	2500
PIN
Confirm PIN
Last name*	Jabber2
Middle name	
First name	
Title	
Directory URI	user2@lab.tekvizion.com
Telephone Number	
Home Number	
Mobile Number	

Buttons for "Edit Credential" are visible next to the Password and PIN fields.

Cisco UCM end user Configuration (Continued)

Pager Number	<input type="text"/>
Mail ID	<input type="text"/>
Manager User ID	<input type="text"/>
Department	<input type="text"/>
User Locale	English, United States
Associated PC	<input type="text"/>
Digest Credentials	<input type="text"/>
Confirm Digest Credentials	<input type="text"/>
User Profile	Use System Default("Standard (Factory Default) U" View Details

Service Settings

Home Cluster
 Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)

Include meeting information in presence(Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

[Presence Viewer for User](#)

UC Service Profile: Use System Default("Jabber_Services") [View Details](#)

Set Controlled Devices = CTIRDuser2. This is used for this example.

Controlled Devices	CTIRDuser2	Device Association
Available Profiles	<input type="text"/>	Line Appearance Association for Presence
CTI Controlled Device Profiles	<input type="text"/>	

Extension Mobility

Available Profiles:

Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI

Select the Primary Extension for this user.2007 is used for this example.

The screenshot displays the configuration interface for an end user in Cisco UCM. It is divided into two main sections: **Extension Mobility** and **Directory Number Associations**.

Extension Mobility Section:

- Available Profiles:** An empty list box.
- Controlled Profiles:** An empty list box with up and down arrow icons to its right.
- Default Profile:** A dropdown menu showing "-- Not Selected --".
- BLF Presence Group*:** A dropdown menu showing "Standard Presence group".
- SUBSCRIBE Calling Search Space:** A dropdown menu showing "< None >".
- Checkboxes:**
 - Allow Control of Device from CTI (This checkbox is highlighted with a red box in the image.)
 - Enable Extension Mobility Cross Cluster

Directory Number Associations Section:

- Primary Extension:** A dropdown menu showing "2007" (This dropdown is highlighted with a red box in the image.)

Check Enable Mobility

Mobility Information
 Enable Mobility
 Enable Mobile Voice Access
Maximum Wait Time for Desk Pickup* 10000
Remote Destination Limit* 4
Remote Destination Profiles
[View Details](#)

Multilevel Precedence and Preemption Authorization
MLPP User Identification Number
MLPP Password
Confirm MLPP Password
MLPP Precedence Authorization Level Default

CAPF Information
Associated CAPF Profiles
[View Details](#)

Add the following permissions for Standard Users:

- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration

Permissions Information

Groups	Roles
Standard CCM End Users Standard CTI Enabled	Standard CCM End Users Standard CCMUSER Administration Standard CTI Enabled

[View Details](#) [View Details](#)

[Add to Access Control Group](#) [Remove from Access Control Group](#)

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

i *- indicates required item.

Add Phone: CTI Remote Device

The CTI Remote Device type represents the user's remote device(s) .

Select the desired Owner User ID .User2 is used in this example.

Set the Device name populated automatically. Modify if desired - CTIRDuser2 used this example.

Set Device Pool: Default. This is used in this example.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation dropdown menu set to "Cisco Unified CM Administration". The user is logged in as "administrator". The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "Phone Configuration", with a "Related Links" dropdown set to "Back To Find/List".

Below the navigation is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The main content area is divided into several sections:

- Status:** Shows "Status: Ready".
- Association:** A list of two lines:
 - 1. Line [1] - 2007 (no partition)
 - 2. Line [2] - Add a new DN
- Phone Type:** Product Type: CTI Remote Device
- Real-time Device Status:** Registration: Registered with Cisco Unified Communications Manager clus26pub; IPv4 Address: (empty)
- Device Information:** A form with the following fields:
 - Device is Active:
 - Device is not trusted:
 - Active Remote Destination: none
 - Owner User ID*: user2
 - Device Name*: CTIRDuser2
 - Description: test
 - Device Pool*: Default (with a "View Details" link)
 - Calling Search Space: < None >
 - User Hold MOH Audio Source: 1-SampleAudioSource
 - Network Hold MOH Audio Source: 1-SampleAudioSource
 - Location*: Hub_None

Cisco UCM CTI remote device Configuration (Continued)

User Locale	English, United States
Network Locale	United States
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)	
Number Presentation Transformation	
Caller ID For Calls From This Phone	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
Remote Number	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	
Protocol Specific Information	
BLF Presence Group*	Standard Presence group
SUBSCRIBE Calling Search Space	< None >
Rerouting Calling Search Space	< None >

Set RD*= 3003. This is used for this example.3003 is the Avaya extension.

Associated Remote Destinations	
<input type="checkbox"/> Route calls to all remote destinations when client is not connected	
Name	Destination Number
RD	3003
Add a New Remote Destination	
Do Not Disturb	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Call Reject

Remote Destination Configuration

Set Destination Number*= 3003. This is used for this example.
Check Enable Extend and Connect.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go
administrator | Search Documentation | About | Logout

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

Remote Destination Configuration

Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Line	Line Association
Line [1] - 2007 (no partition)	<input checked="" type="checkbox"/>

Remote Destination Information

Name: RD

Destination Number*: 3003

Owner User ID*: user2

Enable Unified Mobility features
Remote Destination Profile*: -- Not Selected --
Single Number Reach Voicemail Policy*: Use System Default

Enable Single Number Reach
Ring this phone and my business phone at the same time when my business line(s) is dialed.

Enable Move to Mobile
If this is a mobile phone, transfer active calls to this phone when the mobility button on your Cisco IP Phone is pressed.

Enable Extend and Connect
Allow this phone to be controlled by CTI applications (e.g. Jabber)
CTI Remote Device*: CTIRDuser2

Timer Information

Wait* 4.0 seconds before ringing this phone when my business line is dialed.*

Prevent this call from going straight to this phone's voicemail by using a time delay of* 1.5 seconds to detect when straight to voicemail.*

Stop ringing this phone after* 19.0 seconds to avoid connecting to this phone's voicemail.*

Cisco UCM UC service Configuration

Navigation Path: User management → User setting → UC Service

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation menu includes 'System', 'Call Routing', 'Media Resources', 'Advanced Features', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'User Management' menu is expanded, showing 'UC Service' as the selected option. The main content area is titled 'Find and List UC Services'. It features a toolbar with 'Add New', 'Select All', 'Clear All', and 'Delete Selected' buttons. Below the toolbar, a status bar indicates '3 records found'. The main table, titled 'UC Service (1 - 3 of 3)', has a search filter set to 'Name begins with'. The table contains three rows, each with a checkbox, a name, a UC Service Type, a Product Type, a Host/IP Address, a Port, and a Protocol. The first two rows, 'CTI_SRV' and 'CTI_SUB1', are highlighted with a red border. The third row, 'IMP_SRV', is not highlighted. Below the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	CTI_SRV	CTI	CTI	10.80.16.2	2748	TCP
<input type="checkbox"/>	CTI_SUB1	CTI	CTI	10.80.16.3	2748	TCP
<input type="checkbox"/>	IMP_SRV	IM and Presence	Unified CM (IM and Presence)	10.80.16.6		

Cisco UCM service Profile Configuration

Navigation Path: User management → User setting → Service Profile

The screenshot shows the Cisco Unified CM Administration interface for configuring a Service Profile. The page title is "Service Profile Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

At the top right, there is a "Related Links" section with a dropdown menu set to "Back To Find/List" and a "Go" button. Below this is a toolbar with icons for Save, Delete, Copy, and Add New.

The configuration is organized into several sections:

- Status:** Shows "Status: Ready" with an information icon.
- Service Profile Information:** Contains a "Name*" field with the value "Jabber_SVC_Profile" (highlighted with a red box), a "Description" field with the value "Jabber Service Profile", and a checkbox for "Make this the default service profile for the system".
- Voicemail Profile:** Includes dropdown menus for "Primary", "Secondary", and "Tertiary" (all set to "<None>"), and a "Credentials source for voicemail service*" dropdown set to "Not set".
- MailStore Profile:** Includes dropdown menus for "Primary", "Secondary", and "Tertiary" (all set to "<None>"), and text input fields for "Inbox Folder*" (value: INBOX), "Trash Folder*" (value: Deleted Items), and "Polling Interval (in seconds)*" (value: 60).

Cisco UCM service profile Configuration (Continued)

[Allow dual folder mode](#)

Conferencing Profile

Primary

Secondary

Tertiary

Server Certificate Verification

[Credentials source for web conference service*](#)

Directory Profile

Primary

Secondary

Tertiary

[Use UDS for Contact Resolution](#)

[Use Logged On User Credential](#)

[Username](#)

[Password](#)

[Search Base 1](#)

[Search Base 2](#)

[Search Base 3](#)

[Recursive Search on All Search Bases](#)

[Search Timeout \(seconds\)*](#)

[Base Filter \(Only used for Advance Directory\)](#)

[Predictive Search Filter \(Only used for Advance Directory\)](#)

IM and Presence Profile

Primary

Secondary

Tertiary

CTI Profile

Primary

Secondary

Tertiary

Video Conference Scheduling Portal Profile

Primary

Secondary

Tertiary

Cisco Unified CM IM Presence – CCMCIP Profile Configuration

Navigation Path: Application → Legacy Clients → CCMCIP Profile

Set Name *: remotedesk, this is used in this example.

Set Primary CCMCIP Host *: 10.80.16.2.Cisco Publisher IP. This is used in this example.

Set Backup CCMCIP Host *: 10.80.16.3.Cisco Publisher IP. This is used in this example.

Add Users to Profile: user2.This is used in this example.

The screenshot displays the Cisco Unified CM IM and Presence Administration interface. The page title is "CCMCIP Profile Configuration". The navigation path is "Application → Legacy Clients → CCMCIP Profile". The status is "Ready".

CCMCIP Profile Settings

- Name*: remotedesk
- Description:
- Primary CCMCIP Host*: 10.80.16.2
- Backup CCMCIP Host*: 10.80.16.3
- Server Certificate Verification*: Any Certificate
- Make this the default CCMCIP Profile for the system.

Users in Profile

	User ID	Firstname	Lastname	Department
<input type="checkbox"/>	user1		cisco	
<input checked="" type="checkbox"/>	user2		Jabber2	
<input type="checkbox"/>	user3		jabber3	

Buttons: Add Users to Profile, Select All, Clear All, Delete Selected. Rows per Page: 50

Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

Navigation Path: Device → Trunk

Set Device Name* = IMPTrunk. This is used for this example.

Set Description = This text is used to identify this Trunk Group.

Set Device Pool* = Default. This is used for this example.

Set Media Resource Group List = MRGL_SW_MTP. This is used for this example.

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the user is logged in as "administrator". The interface includes a navigation menu at the top and a toolbar with "Save", "Delete", "Reset", and "Add New" buttons. The configuration is organized into sections: "Status" (Ready), "SIP Trunk Status" (Service Status: Unknown - OPTIONS Ping not enabled, Duration: Unknown), and "Device Information". The "Device Information" section contains the following fields:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	IMPTrunk
Description	IMPTrunk
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_SW_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes

Red boxes highlight the "Device Name*", "Device Pool*", "Media Resource Group List", and "Location*" fields.

Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

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 checked="" type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input checked="" type="checkbox"/> Unattended Port	
<input checked="" 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	

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

All other values are default.

MLPP and Confidential Access Level Information	
MLPP Domain	< None >
Confidential Access Mode	< None >
Confidential Access Level	< None >

Call Routing Information	
<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	Default
SIP Privacy*	Default

Inbound Calls	
Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

Cisco UCM SIP Trunk to CUP Configuration (Continued)

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.				
<input type="button" value="Clear Prefix Settings"/>		<input type="button" value="Default Prefix Settings"/>		
Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Incoming Called 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.				
<input type="button" value="Clear Prefix Settings"/>		<input type="button" value="Default Prefix Settings"/>		
Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Connected Party Settings	
Connected Party Transformation CSS	< None >
<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
Calling and Connected Party Info Format*	Deliver DN only in connected party
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	

Caller Information	
Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers	

Cisco UCM SIP Trunk to CUP Configuration (Continued)

Set Destination Address = 10.80.16.6. This is used in this example.

Set SIP Trunk Security Profile* = Non Secure SIP Trunk Profile.

Set SIP Profile* = Standard SIP Profile.

Set DTMF Signaling Method* = No Preference.

All other values are default.

SIP Information
Destination
 Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.16.6		0

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script
Normalization Script
 Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Normalization Script
Normalization Script
 Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Recording Information
 None
 This trunk connects to a recording-enabled gateway
 This trunk connects to other clusters with recording-enabled gateways
Geolocation Configuration
Geolocation
Geolocation Filter
 Send Geolocation Information

Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

CUC Version

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unity Connection Administration Go
administrator Search Documentation About Sign Out

Cisco Unity Connection

- Users
 - Users
 - Import Users
 - Synch Users
- Class of Service
 - Class of Service
 - Class of Service Membership
- Templates
 - User Templates
 - Call Handler Templates
 - Contact Templates
 - Notification Templates
- Contacts
 - Contacts
- Distribution Lists
 - System Distribution Lists
- Call Management
 - System Call Handlers
 - Directory Handlers
 - Interview Handlers
 - Custom Recordings
- Call Routing

Cisco Unity Connection Administration

Version 10.5.1.10000-7

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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.

CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

Phone System Name* = PhoneSystem. This is used for this example

The screenshot displays the Cisco Unity Connection Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unity Connection Administration", and the user role "administrator". A left-hand navigation menu lists various configuration options under "Cisco Unity Connection". The main content area is titled "Search Phone Systems" and shows a status message: "Found 1 Phone System(s)". Below this, a table lists the phone systems. The table has columns for "Display Name" and "Port Count". One entry is visible: "PhoneSystem" with a port count of "1". The "PhoneSystem" text in the table is highlighted with a red box. Below the table are buttons for "Delete Selected" and "Add New".

Display Name	Port Count
PhoneSystem	1

CUC Port Group

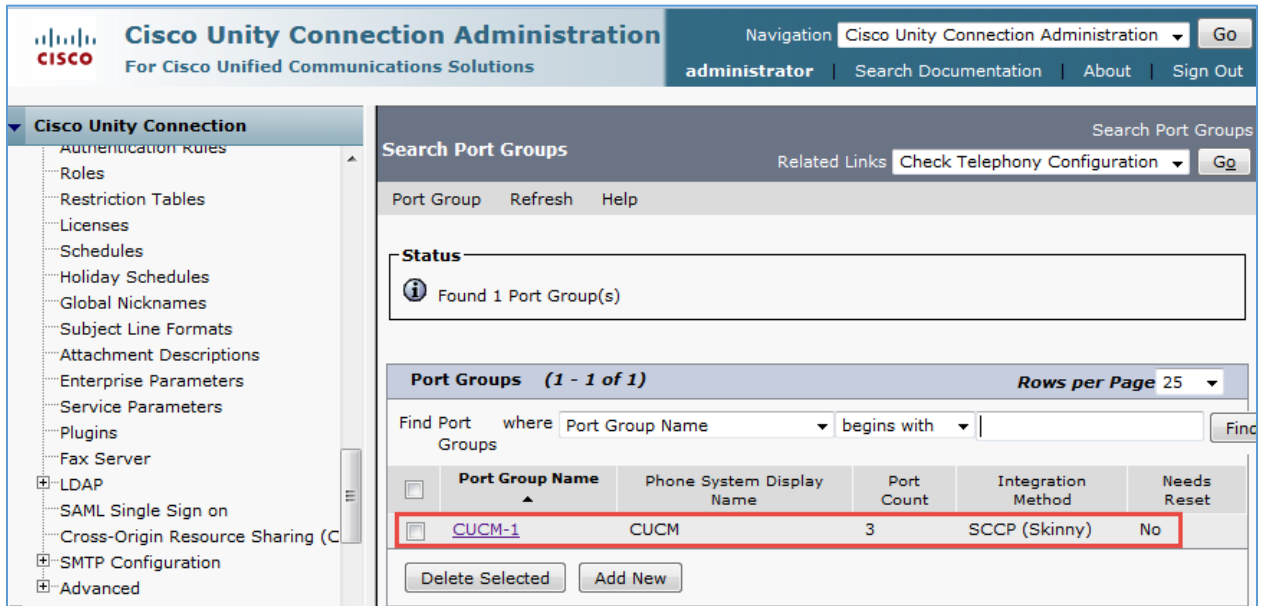
Navigation: Telephony Integration → Port Group

Set Display Name* = CUCM-1. This is used in this example.

Check Enable Message waiting indicators.

Set MWI on Extension = 1001. This is used in this example.

Set MWI off Extension= 1002. This is used in this example.



The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation menu with categories like Authentication Rules, Roles, Restriction Tables, Licenses, Schedules, Holiday Schedules, Global Nicknames, Subject Line Formats, Attachment Descriptions, Enterprise Parameters, Service Parameters, Plugins, Fax Server, LDAP, SAML Single Sign on, Cross-Origin Resource Sharing (C), SMTP Configuration, and Advanced.

The main content area is titled 'Search Port Groups' and includes a search bar and a table of port groups. The table has the following columns: Port Group Name, Phone System Display Name, Port Count, Integration Method, and Needs Reset. One row is highlighted with a red border:

Port Group Name	Phone System Display Name	Port Count	Integration Method	Needs Reset
CUCM-1	CUCM	3	SCCP (Skinny)	No

Below the table are buttons for 'Delete Selected' and 'Add New'.

Cisco Unified Connections Port Configuration (Continued)

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Search Port Groups | Port Group Basics (CUCM-1) | Related Links: Add Ports

Port Group Edit Refresh Help

Save Delete Previous Next

Port Group

Display Name* CUCM-1
Integration Method SCCP (Skinny)
Device Name Prefix* CiscoUM1-VI

Reset Status: Reset Not Required | Reset

Message Waiting Indicator Settings

Enable Message Waiting Indicators
MWI On Extension: 1001
MWI Off Extension: 1002

Delay between Requests: 0 milliseconds
Maximum Concurrent Requests: 0
Retries After Successful Attempt: 0
Retry Interval After Successful Attempt: 5 milliseconds

Save Delete Previous Next

Fields marked with an asterisk (*) are required.

CUC Port Settings

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration | administrator | Search Documentation | About | Sign Out

Search Ports | Related Links: Check Telephony Configuration

Port Refresh Help

Status: Found 3 Port(s)

Port (1 - 3 of 3) | Rows per Page: 25

Find Port where: Display Name begins with | Find

Display Name	Phone System Display Name	Extension	Server	Enabled	Answer Calls	Message Notification	Dialout MWI	TRAP Connection	Security Mode
CUCM-1-001	CUCM		clus26unity	X	X	X	X	X	Non-secure
CUCM-1-002	CUCM		clus26unity	X	X	X	X	X	Non-secure
CUCM-1-003	CUCM		clus26unity	X	X	X	X	X	Non-secure

Delete Selected | Add New

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CUC Sample User Basic Settings

Navigation: Cisco Unity connection → Users → Users

Set Alias = 2003. This is one of the extension used for this testing.

Set Extension = 2003. This is used for this example.

The screenshot displays the Cisco Unity Connection Administration interface. The main content area is titled "Edit User Basics (2003)". On the left, a navigation tree shows the "Users" section expanded. The form contains the following fields and options:

- Name:**
 - Alias*: 2003 (highlighted with a red box)
 - First Name: [Empty]
 - Last Name: [Empty]
 - Display Name: 2003
- SMTP Address:** 2003 @clus26unity
- Initials:** [Empty]
- Title:** [Empty]
- Employee ID:** [Empty]
- LDAP Integration Status:**
 - Integrate with LDAP Directory
 - Do Not Integrate with LDAP Directory
- Phone:**
 - Extension*: 2003 (highlighted with a red box)
 - Cross-Server Transfer Extension or URI: [Empty]

Note: Need to configure Alternate extension for Cisco Extend and connect enabled remote destination DN to retrieve successful mail access from Avaya CS1000E.

CUC Sample User Basic Settings (Contd...)

Set Partition = clus23-unity partition. This is used for this example.

Select Search Scope = clus23-unity Search Scope.

Select Phone System = CUCM.

The screenshot displays the Cisco Unified Communications Manager (CUCM) user configuration interface. The left sidebar shows a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, and Networking. The main content area is split into two panels. The top panel, titled 'Users', contains the following fields and options:

- Outgoing Fax Number: [Empty]
- Outgoing Fax Server: --- Not Selected ---
- Partition: clus26unity Partition (highlighted in a red box)
- Search Scope: clus26unity Search Space (highlighted in a red box)
- Phone System: CUCM (highlighted in a red box)
- Class of Service: Voice Mail User COS
- Active Schedule: Weekdays [View]
- Set for Self-enrollment at Next Sign-In
- List in Directory
- Send Non-Delivery Receipts on Failed Message Delivery
- Skip PIN When Calling From a Known Extension
- Caution!** Security risk. See Help for This Page for details.
- Use Short Calendar Caching Poll Interval
- Recorded Name: [Play/Record]
- Location section:
 - Address: [Empty]
 - Building: [Empty]
 - City: [Empty]
 - State: [Empty]
 - Postal Code: [Empty]
 - Country: United States
 - Use System Default Time Zone

The bottom panel, titled 'Contacts', contains the following fields and options:

- Time Zone: (GMT-06:00) America/Chicago
- Language: Use System Default Language
- English(United States)
- Department: [Empty]
- Manager: [Empty]
- Billing ID: [Empty]
- Corporate Email Address: [Empty]
- Generate SMTP Proxy Address From Corporate Email Address
- Directory URI: [Empty]
- Corporate Phone Number: [Empty]
- [Save] [Delete] [Previous] [Next]
- Fields marked with an asterisk (*) are required.

Cisco UCM Voice Mail Port

Navigation: Advanced Feature → Voice Mail → Voice Mail Port

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

Find and List Voice Mail Ports

+ Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected

Status
3 records found

Voice Mail Port (1 - 3 of 3) Rows per Page 50

Find Voice Mail Port where Device Name begins with | Find | Clear Filter | Select item or enter search text

Device Name	Description	Device Pool	Device Security Mode	Calling Search Space	Extension	Partition	Status	IPv4 Address	Copy
CiscoUM1-VI1	CUC	Default	Non Secure Voice Mail Port		1003		Registered with clus26sub1	10.80.16.5	
CiscoUM1-VI2	CUC	Default	Non Secure Voice Mail Port		1004		Registered with clus26sub1	10.80.16.5	
CiscoUM1-VI3	CUC	Default	Non Secure Voice Mail Port		1005		Registered with clus26sub1	10.80.16.5	

+ Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation

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

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

Save | Delete | Copy | Reset | Apply Config | Add New

Status
Status: Ready

Device Information

Registration: Registered with Cisco Unified Communications Manager clus26sub1
IPv4 Address: 10.80.16.5
Device is trusted

Port Name* CiscoUM1-VI1
Description CUC

Device Pool* Default
Common Device Configuration < None >
Calling Search Space < None >
AAR Calling Search Space < None >
Location* Hub_None
Device Security Mode* Non Secure Voice Mail Port
Use Trusted Relay Point* Default
Geolocation < None >

Cisco Unified Connections Voice mail port configuration Settings (Contd...)

Directory Number Information

Directory Number*	1003
Partition	< None >
Calling Search Space	< None >
AAR Group	< None >
Internal Caller ID Display	VoiceMail
Internal Caller ID Display (ASCII format)	VoiceMail
External Number Mask	

*- indicates required item.

Cisco UCM Message Waiting Numbers

Navigation: Advanced Feature → Voice Mail → Message Waiting

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

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

Find and List Message Waiting Numbers

Status

2 records found

Message Waiting Numbers (1 - 2 of 2) Rows per Page 50 ▾

Message Waiting Numbers where Directory Number begins with and where Message Waiting Indicator is Both

	Directory Number ^	Description	Partition	Calling Search Space	Copy
<input type="checkbox"/>	1001	MWI on			
<input type="checkbox"/>	1002	MWI Off			

Cisco UCM Voice Mail Pilot

Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout

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

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

Save Delete Add New

Status
Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number: 1000
Calling Search Space: < None >
Description: Default
 Make this the default Voice Mail Pilot for the system

Save Delete Add New

*- indicates required item.

Cisco UCM Voice Mail Pilot

Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout

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

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

Save Delete Add New

Status
Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number: 1000
Calling Search Space: < None >
Description: Default
 Make this the default Voice Mail Pilot for the system

Save Delete Add New

*- indicates required item.

Cisco UCM Voice Mail Profile

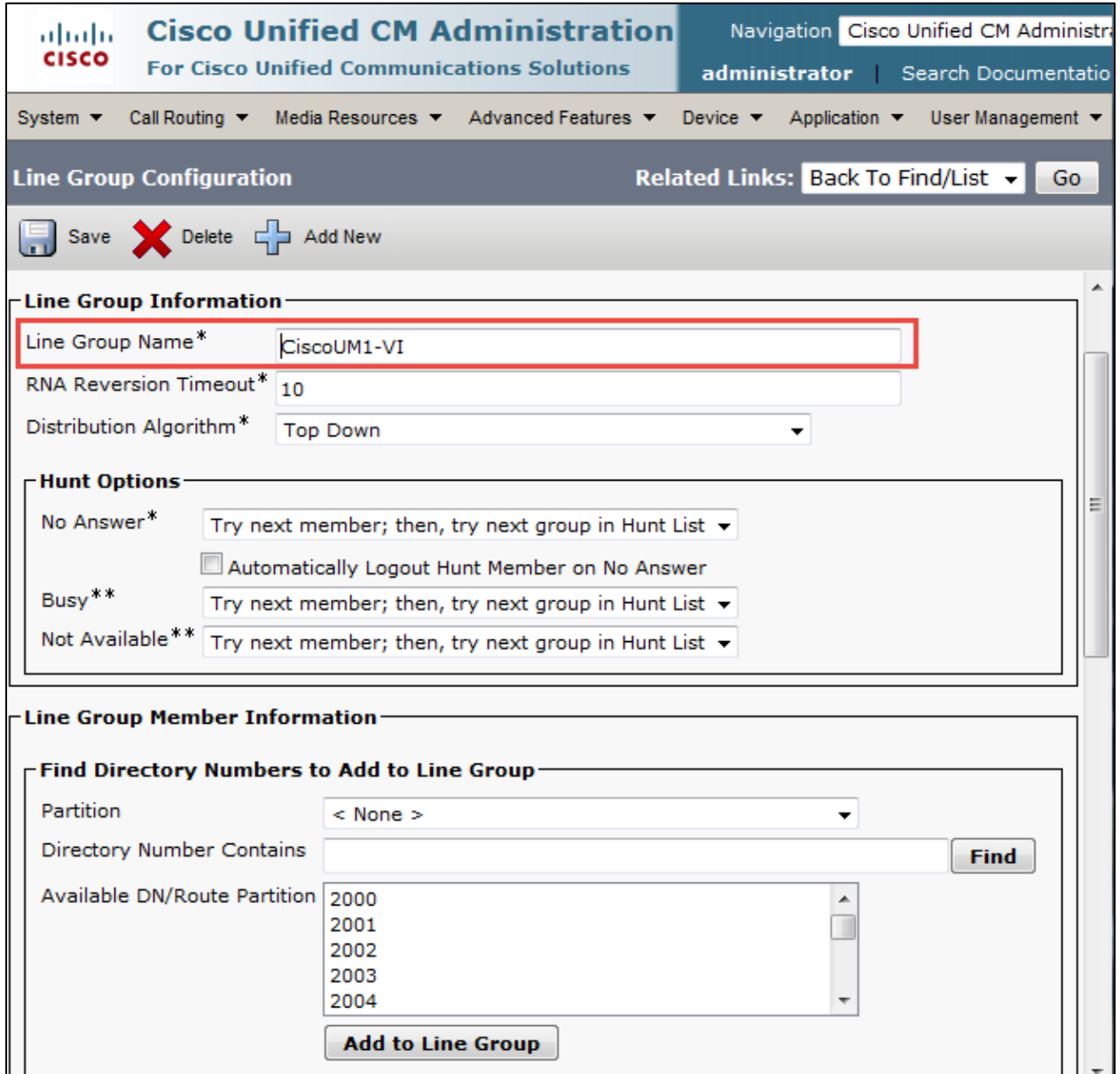
Navigation: Advanced Feature → Voice Mail → Voice Mail Pilot

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Voice Mail Profile. The page title is "Voice Mail Profile Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. The "Voice Mail Profile Configuration" page shows the following fields and options:

- Status:** Status: Ready
- Voice Mail Profile Information:**
 - Voice Mail Profile: Default (used by 12 devices)
 - Voice Mail Profile Name*: Default
 - Description: Default voice messaging profile
 - Voice Mail Pilot**:** 1000/< None > (This field is highlighted with a red box)
 - Voice Mail Box Mask: [Empty text field]
 - Make this the default Voice Mail Profile for the System

At the bottom of the configuration area, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New. A "Related Links" section at the top right of the configuration area contains a "Back To Find/List" link and a "Go" button.

Navigation: Call Routing → Route/Hunt → Line Group





Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration
administrator | Search Documentation

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

Line Group Configuration Related Links: Back To Find/List ▾ Go

Save  Delete  Add New

Line Group Information

Line Group Name* CiscoUM1-VI

RNA Reversion Timeout* 10

Distribution Algorithm* Top Down ▾

Hunt Options

No Answer* Try next member; then, try next group in Hunt List ▾

Automatically Logout Hunt Member on No Answer

Busy** Try next member; then, try next group in Hunt List ▾

Not Available** Try next member; then, try next group in Hunt List ▾

Line Group Member Information

Find Directory Numbers to Add to Line Group

Partition < None > ▾

Directory Number Contains Find

Available DN/Route Partition

2000
2001
2002
2003
2004

Add to Line Group

Cisco Unified Connections Line Group Member configuration (Contd...)

Current Line Group Members

Reverse Order of Selected DN/Route Partitions

Selected DN/Route Partition

1003	▲
1004	
1005	▼

▼ ▲

Removed DN/Route Partition

	▲
	▼

Directory Numbers

- 7718 1003 (no partition)
■ 7719
- 7718 1004 (no partition)
■ 7719
- 7718 1005 (no partition)
■ 7719

Save Delete Add New

Cisco UCM Hunt List

Navigation: Call Routing → Route/Hunt → Hunt List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration
administrator | Search Documentation | About

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

Hunt List Configuration Related Links: Back To Find/List ▾ Go

Save Delete Copy Reset Apply Config Add New

Hunt List Information

Device is trusted

Name* VM_HUNT

Description

Cisco Unified Communications Manager Group* Default ▾

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

For Voice Mail Usage

Hunt List Member Information

Add Line Group

Selected Groups** CiscoUM1-VI

Removed Groups***

Hunt List Details

CiscoUM1-VI

Cisco UCM Hunt Pilot

Navigation: Call Routing → Route/Hunt → Hunt Pilot

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration administrator | Search Documents

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

Hunt Pilot Configuration Related Links: Back To Find/List ▾ Go

Save ~~X~~ Delete Copy + Add New

Status
Status: Ready

Pattern Definition

Hunt Pilot*	1000
Route Partition	< None >
Description	
Numbering Plan	< None >
Route Filter	< None >
MLPP Precedence*	Default
Hunt List*	VM_HUNT (Edit)
Call Pickup Group	< None >
Alerting Name	
ASCII Alerting Name	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
<input checked="" type="checkbox"/> Provide Outside Dial Tone	
<input type="checkbox"/> Urgent Priority	

Cisco Unified Hunt Pilot configuration (Contd...)

All Other Values are Default

Hunt Call Treatment Settings
Forward Hunt No Answer
 Do Not Forward Unanswered Calls
 Use Forward Settings of Line Group Member
 Forward Unanswered Calls to
Destination
Calling Search Space
Maximum Hunt Timer
Forward Hunt Busy
 Do Not Forward Busy Calls
 Use Forward Settings of Line Group Member
 Forward Busy Calls to
Destination
Calling Search Space

Acronyms

Acronym	Definition
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
Cisco UCM	Cisco Unified Communications Manager
CUP	Cisco Unified IM and Presence Server
DNS	Domain Name Server
FQDN	Fully Qualified Domain Name
MWI	Message Waiting Indicator
MRGL	Media Resource Group List
MTP	Media Termination Point
PSTN	Public Switched Telephone Network
SIP	Session Initiated Protocol

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