



Avaya Solution & Interoperability Test Lab

Configuring SIP Trunks among Avaya Communication Server 1000E 6.0, Avaya Aura[™] Session Manager 5.2 and Avaya Meeting Exchange Enterprise Edition 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses SIP trunks to connect the Avaya Communication Server 1000E (formerly known as Nortel CS1000E) to Avaya Meeting Exchange Enterprise Edition via Avaya Aura[™] Session Manager 5.2.

For the sample configuration, Avaya Communication Server 1000E runs the SIP Proxy NRS hosted co-resident with the Signaling Server on a single CPPM card while the Avaya Meeting Exchange Application and Media Server run on two separate S8510 servers.

1. Introduction

These Application Notes present a sample configuration for a network that uses SIP trunks to connect the Avaya Communication Server 1000E (formerly known as Nortel CS1000E) to Avaya Meeting Exchange Enterprise Edition via Avaya AuraTM Session Manager 5.2.

As shown in **Figure 1**, the Avaya 2050 IP Softphone (UNISTim), Avaya M3820 Digital Telephone, Avaya 1120E UNISTim IP Telephone and Avaya 1140E SIP Telephone are supported by Avaya Communication Server 1000E. The Avaya 1140E SIP phones are registered to the SIP Line Gateway (SIPL) which is a SIP Registrar running on the Signaling Server component of the Communication Server 1000E. SIP trunks are used to connect Avaya AuraTM Session Manager to Communication Server 1000E and Avaya Meeting Exchange Enterprise over the LAN. Signaling messages are carried over the TCP-based SIP trunks while DTMF is transmitted within the RTP stream using RFC2833 compliant messages. An analogue phone is attached to a PSTN simulator which is connected with an E1 PRI trunk to the sample telephony network. This configuration enables PSTN users to participate in a conference with other enterprise users on the private network. The Telephones are configured in the 3xxx extension range, while the conference access number (DNIS) on the Avaya Meeting Exchange is set to 11111.

For the sample configuration, Avaya Communication Server 1000E runs on a single CPPM card while the Avaya Meeting Exchange Enterprise Application and Media Server run on two separate Avaya S8510 servers.

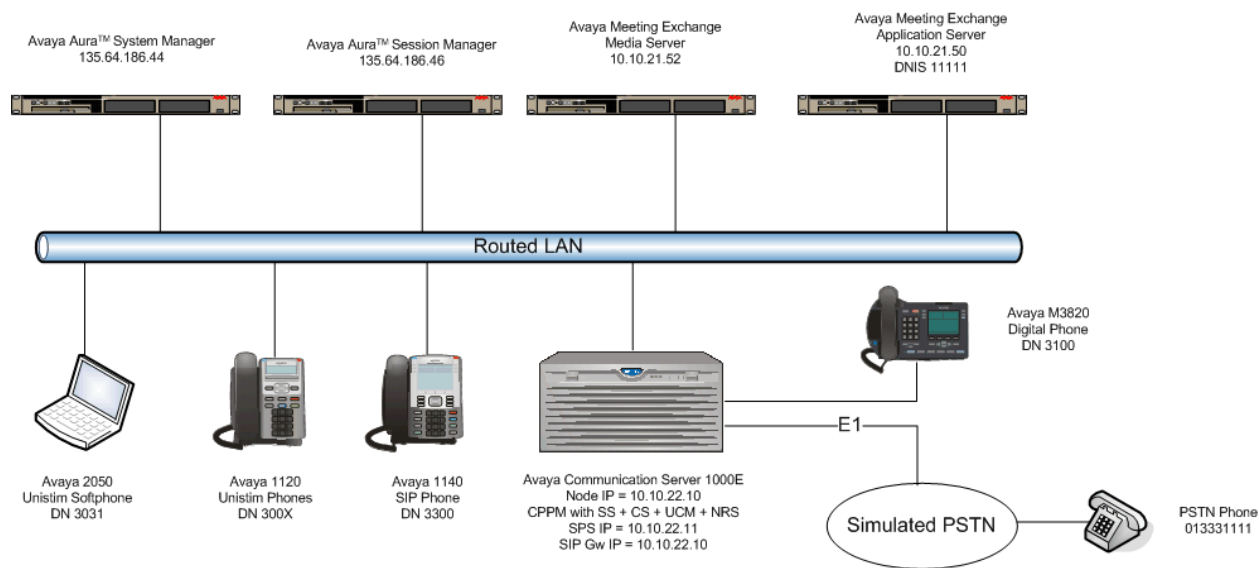


Figure 1 – Sample Configuration

These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones and E1 PRI trunk will not be described (see the appropriate documents listed in **Section 9**).

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software Version
Avaya S8510 Server	Avaya Aura™ Session Manager 5.2.0.1 VSP 1.1.0.0.8 with patch 1.1.0.4.8
Avaya S8510 Server	Avaya Aura™ System Manager 5.2.0.7.11
Avaya Communication Server 1000E	Release 600R, Version 4121
Avaya 1140 IP Phone	SIP 02.02.21.00
Avaya 1120E IP Phone	UNISTim 0624C60
Avaya IP SoftPhone2050	UNISTim 3.04.0003
Avaya M3820 Digital Phone	N/A
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Application Server S6200 R5.2 Build 5.2.0.0.22 (GA) and MX Patch Group 5.2.1.2.1
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Media Server S6200 R5.2 Build 5.2.0.0.22 (GA) and MX Patch Group 5.2.1.2.1

Table 1: Version Numbers of Equipment and Software

3. Configure Avaya Communication Server 1000E

The sample network uses a single CPPM card within the Avaya MG1000E gateway which runs all the software services necessary to route calls, administer the dial plan and provide signaling interfaces to external nodes (telephones and trunks). Avaya Communication Server 1000E uses the Signaling Server and Network Routing Service (NRS) to provide SIP, H.323 and UNISTim (Avaya proprietary) signaling interfaces to IP networks. The Call Server is another software component that resides on the CPPM card within the Avaya MG1000E gateway and controls the telephony features. Avaya provides a single web GUI which runs on the CPPM called Unified Communications Management (UCM) for the provisioning of the various telephony software components.

These Application Notes used the Coordinated Dial Plan (CDP) feature to route calls from the Avaya Communication Server 1000E, over a SIP trunk to Session Manager. The CDP feature is assumed to be already enabled on Avaya Communication Server 1000E, and therefore will not be described in detail.

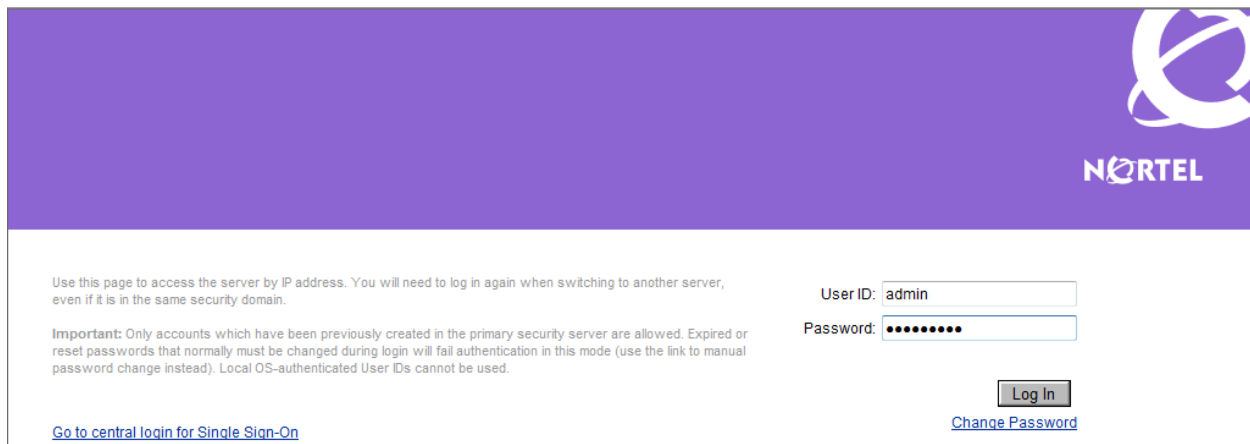
All configurations in this section are administered using a web browser. These Application Notes assume that the basic configuration has already been administered. For further information on Avaya Communications Server 1000E, please consult reference [1]. The

procedures below describe the details of configuring Avaya Communication Server 1000 with a SIP trunk to Session Manager:

- Login to the Unified Communications Management GUI
- Obtain Node IP address
- Administer ISDN
- Administer D-Channel
- Administer virtual SIP routes and trunks
- Administer route list block and distant steering code
- Administer Node Media and SIP parameters
- Launch NRS Manager
- Administer service domain
- Administer SIP Signaling Gateway endpoints
- Administer routing entries
- Cut over and commit changes

3.1. Log in to the Unified Communications Management GUI

Open an instance of a web browser and connect to the UCM GUI at the following address: `http://<node IP address>`. Log in using an appropriate Username and Password.



Use this page to access the server by IP address. You will need to log in again when switching to another server, even if it is in the same security domain.

Important: Only accounts which have been previously created in the primary security server are allowed. Expired or reset passwords that normally must be changed during login will fail authentication in this mode (use the link to manual password change instead). Local OS-authenticated User IDs cannot be used.

User ID:

Password:

[Go to central login for Single Sign-On](#)

[Change Password](#)

3.2. Obtain Node IP Address

The **Elements** screen is displayed. Click on the **Element Name** of the CS1000 Element.

NORTEL UNIFIED COMMUNICATIONS MANAGEMENT [Help](#) | [Logout](#)

Host Name: 10.10.22.11 Software Version: 02.10.0010.04(3393) 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.

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on cs1k-r022011	CS1000	6.0	10.10.21.10	New element.
<input type="checkbox"/>	cs1k-r022011.cs1k.avaya.com (primary)	Linux Base	6.0	10.10.22.11	Base OS element.
<input type="checkbox"/>	10.10.21.12	Media Gateway Controller	6.0	10.10.21.12	New element.
<input type="checkbox"/>	NRSM on cs1k-r022011	Network Routing Service	6.0	10.10.21.10	New element.

The CS 1000 Element Manager **System Overview** page is displayed.

NORTEL CS 1000 ELEMENT MANAGER [Help](#) | [Logout](#)

Managing: [10.10.21.10](#) Username: admin
System Overview

System Overview

IP Address: 10.10.21.10
Type: Nortel Communication Server Linux
Version: 4121
Release: 600 R +

[Active Sessions](#)

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Select IP Network → Nodes: Servers, Media Cards

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar contains a navigation menu with options like UCM Network Services, Home, Links, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, and Nodes: Servers, Media Cards. The main content area displays the 'IP Telephony Nodes' section. At the top, it says 'Managing: 10.10.21.10 Username: admin' and 'System » IP Network » IP Telephony Nodes'. Below this, there's a table with columns: Node ID, Components, Enabled Applications, ELAN IP, TLAN IP, and Status. A single node is listed with ID 1000, 1 component, SIP Line, LTPS, Gateway (SIPGw, H323Gw), ELAN IP 10.10.22.10, and Status Synchronized. Buttons for Add, Import, Export, and Delete are visible, along with Print and Refresh links.

Click on the **Node ID** of your CS1000 Element.

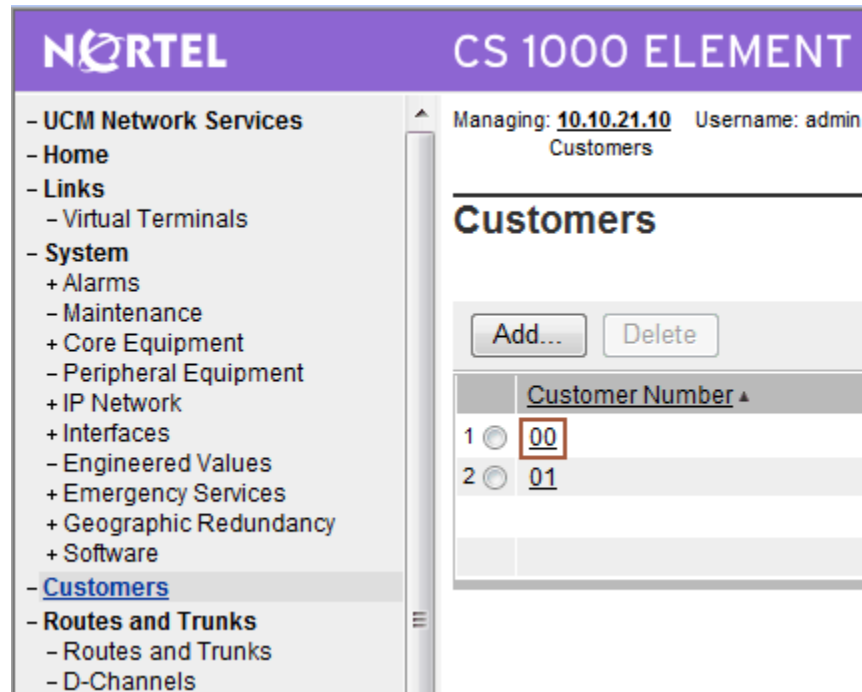
This screenshot is similar to the previous one, but the 'Node ID' 1000 in the table is highlighted with a red box, indicating it has been selected for further action.

The **Node Details** screen is displayed with the IP address of the CS1000 node. The **Node IP Address** is a virtual address which corresponds to the IP address of the Call Server, Signaling Server, SIP Signaling Gateway and UCM interface. The SIP Signaling Gateway uses the **Node IP Address** as a source address when registering with the SIP Proxy Server (SPS).

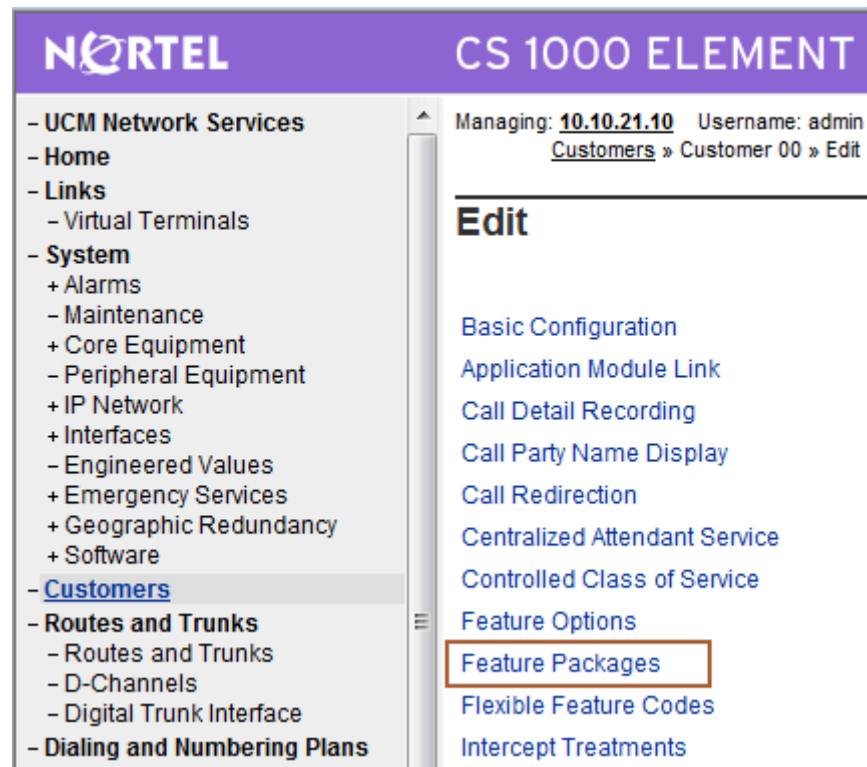
The screenshot shows the 'Node Details' screen for Node ID 1000. The title is 'Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))'. The form contains several fields: Node ID (1000), Call Server IP Address (10.10.21.10), and a section for 'Telephony LAN (TLAN)' with Node IP Address (10.10.22.10) and Subnet Mask (255.255.255.0). There's also an 'Embedded LAN (ELAN)' section with Gateway IP address (10.10.21.1) and Subnet Mask (255.255.255.0). Below these are two lists: 'IP Telephony Node Properties' (Voice Gateway (VGW) and Codecs, Quality of Service (QoS), LAN) and 'Applications (click to edit configuration)' (SIP Line, Terminal Proxy Server (TPS), Gateway (SIPGw & H323Gw)). At the bottom, there's a section for 'Associated Signaling Servers & Cards' with a table showing a single entry: cs1k-r022011, Signaling Server, SIP Line, LTPS, Gateway, PD, with ELAN IP 10.10.21.10, TLAN IP 10.10.22.11, and Role Leader. Buttons for Save and Cancel are present.

3.3. Administer ISDN

Select **Customers** in the left pane. The **Customers** screen is displayed. Click the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options.



The **Customer 00** screen is displayed next. Select **Feature Packages**.



The screen is updated with a listing of feature packages populated below **Feature Packages** (not all features shown below). Select **Integrated Services Digital Network** to edit its parameters.

NORTEL CS 1000 ELEMENT MANAGER		
- Home	+ Command Status Link	Package: 77
- Links	+ Pretranslation	Package: 92
- Virtual Terminals	+ Dialed Number Identification System	Package: 98
- System	+ Malicious Call Trace	Package: 107
+ Alarms	+ Incoming Digit Conversion	Package: 113
- Maintenance	+ Directed Call Pickup	Package: 115
+ Core Equipment	+ Enhanced Music	Package: 119
- Peripheral Equipment	+ Station Camp-On	Package: 121
+ IP Network	+ Flexible Tones and Cadences	Package: 125
+ Interfaces	+ Multifrequency Compelled Signaling	Package: 128
- Engineered Values	+ International Supplementary Features	Package: 131
+ Emergency Services	+ Enhanced Night Service	Package: 133
+ Geographic Redundancy	+ Integrated Services Digital Network	Package: 145
+ Software	+ Network Attendant Service	Package: 159
- Customers		
- Routes and Trunks		
- Routes and Trunks		
- D-Channels		
- Digital Trunk Interface		
- Dialing and Numbering Plans		
- Electronic Switched Network		
- Flexible Code Restriction		

The screen is updated with parameters populated below **Integrated Services Digital Network**. Check the **Integrated Services Digital Network (ISDN)** checkbox, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click **Save** (not shown).

The screenshot shows the Nortel CS 1000 Element Manager interface. On the left is a navigation tree with categories: Home, Links, System, Customers, and Routes and Trunks. The 'System' category is expanded, showing sub-items like Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, Interfaces, Engineered Values, Emergency Services, Geographic Redundancy, and Software. The 'Integrated Services Digital Network' checkbox is checked and highlighted with a red box. Below it are input fields for Virtual Private Network Identifier (1), Private Network Identifier (1), Node DN, and Multi-location Business Group (0). To the right, a list of packages is shown: Multifrequency Compelled Signaling (Package: 128), International Supplementary Features (Package: 131), Enhanced Night Service (Package: 133), and Integrated Services Digital Network (Package: 145).

3.4. Administer a Virtual D-Channel

Select **Routes and Trunks** → **D-Channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list. Click to **Add**.

The screenshot shows the Nortel CS 1000 Element Manager interface with the 'D-Channels' page selected. The left navigation tree shows 'Routes and Trunks' expanded, with 'D-Channels' selected. The main content area is titled 'D-Channels' and includes a 'Maintenance' section with links for D-Channel Diagnostics (LD 96), Network and Peripheral Equipment (LD 32, Virtual D-Channels), MSDI Diagnostics (LD 96), and D-Channel Expansion Diagnostics (LD 48). Below this is a 'Configuration' section with a 'Choose a D-Channel Number' dropdown menu (set to 0) and a 'type' dropdown menu (set to DCH). A red box highlights the 'to Add' button.

The **D-Channels 15 Property Configuration** screen is displayed next. Enter the following values for the specified fields, and retain the default values for the remaining fields.

D channel Card Type (CTYP): D-Channel is over IP (**DCIP**)

Designator (DES): A descriptive name.

Interface type for D-channel (IFC): Meridian Meridian1 (**SL1**)

Country (CNTY): ETS 300 = 102 basic protocol (**ETSI**)

Click **Submit** (not shown).

Managing: 10.10.21.10 Username: admin
Routes and Trunks » D-Channels » D-Channels 15 Property Configuration

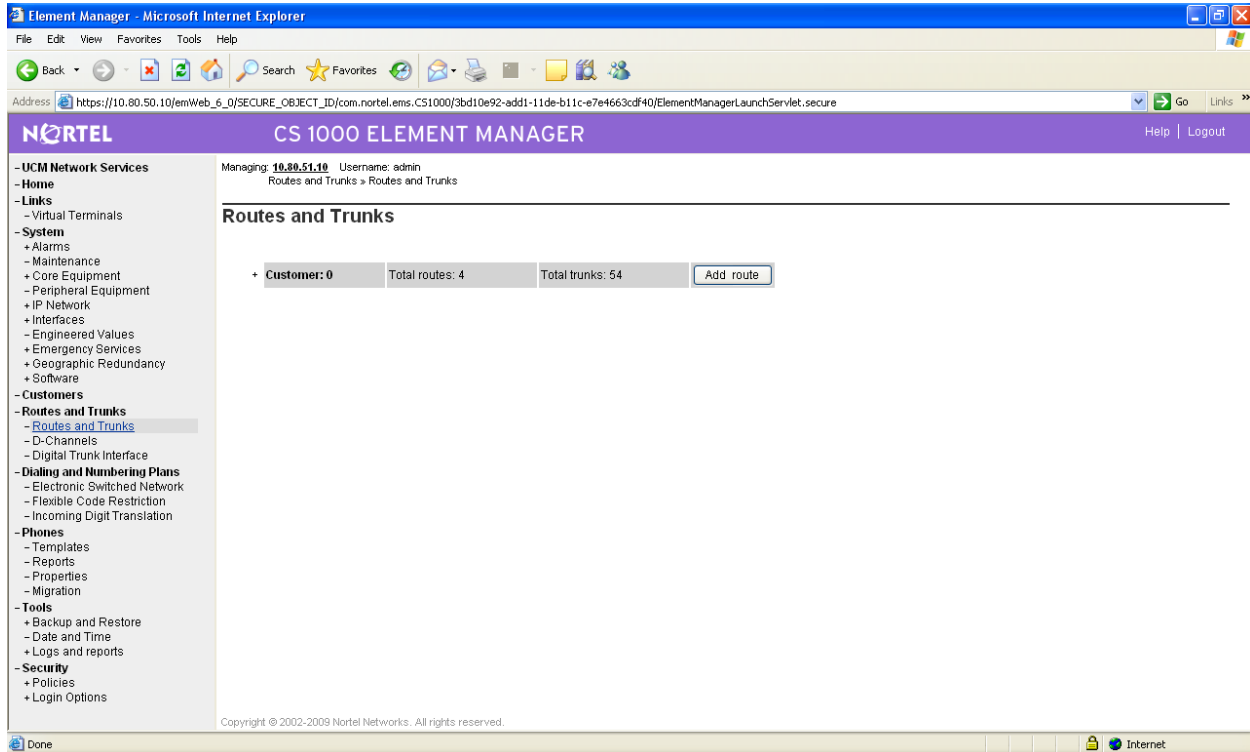
D-Channels 15 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	VtrkNode1000
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	25
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000

3.5. Administer Virtual SIP Routes and Trunks

Select **Routes and Trunks** → **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. Next to the applicable **Customer** row, click **Add route**.



The **Customer 0, New Route Configuration** screen is displayed next. Scroll down until the **Basic Configuration** section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields.

Route Number (ROUT):	Select an available route number
Designator field for trunk (DES):	A descriptive text
Trunk Type (TKTP):	TIE trunk data block (TIE)
Incoming and outgoing trunk (ICOG):	Incoming and Outgoing (IAO)
Access code for the trunk route (ACOD):	An available access code

Customer 0, Route 15 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P) ☐

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

Scroll down the screen, and check the field **The route is for a virtual trunk route (VTRK)**, to enable four additional fields to appear. For the **Zone for codec selection and bandwidth management (ZONE)** field, enter the default zone number (in this case **0**). For the **Node ID of signaling server of this route (NODE)** field, enter the node number from **Section 3.3**. Select **SIP (SIP)** from the drop-down list for the **Protocol ID for the route (PCID)** field.

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

- Zone for codec selection and bandwidth management (ZONE) Range: 0 - 255

- Node ID of signaling server of this route (NODE) Range: 0 - 9999

- Protocol ID for the route (PCID)

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

Integrated services digital network option (ISDN) ☒

Scroll down the screen, check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit** (not shown).

Mode of operation (MODE):

Route uses ISDN Signaling Link (ISLD)

D channel number (DCH):

D-Channel number from **Section 3.4**

Network calling name allowed (NCNA): Check the field

Network call redirection (NCRD): Check the field

Integrated services digital network option (ISDN) ☒

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

- D channel number (DCH) 15 Range: 0 - 254

- Interface type for route (IFC) Meridian M1 (SL1)

- Private network identifier (PNI) 00001 Range: 0 - 32700

- Network calling name allowed (NCNA) ☒

- Network call redirection (NCRD) ☒

The **Routes and Trunks** screen is displayed again, and updated with the newly added route. Click the **Add trunk** button next to the newly added route.

Managing: 10.10.21.10 Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

Customer	Total routes	Total trunks		
- Customer: 0	4	126	Add route	
+ Route: 15	Type: TIE	Description: VTRKNode1000SIP	Edit	Add trunk
+ Route: 16	Type: TIE	Description: VTRKNode1000H323	Edit	Add trunk
+ Route: 17	Type: TIE	Description: VTRKNode1001SIP	Edit	Add trunk
+ Route: 42	Type: TIE	Description: EURO_ETSI	Edit	Add trunk
- Customer: 1	0	0	Add route	

The **Customer 0, Route 15, Trunk 1 Property Configuration** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Save**. The **Multiple trunk input number (MTINPUT)** field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, four trunks were created.

Trunk data block (TYPE):

IP Trunk (IPTI)

Terminal Number (TN):

An available terminal number

Designator field for trunk (DES):

A descriptive text

Extended Trunk (XTRK):

Virtual trunk (VTRK)

Route number, Member number (RTMB):

Current route number and starting member

Card Density:

Select **8D**

Start arrangement Incoming (STRI):

Wink or Fast Flash (WNK)

Start arrangement Outgoing (STRO):

Wink or Fast Flash (WNK)

Trunk Group Access Restriction (TGAR):

Desired trunk group access restriction level

Channel ID for this trunk (CHID):

An available starting channel ID.

NORTEL CS 1000 ELEMENT MANAGER Help |

Routes and Trunks » Routes and Trunks » Customer 0, Route 15, Trunk 1 Property Configuration

Customer 0, Route 15, Trunk 1 Property Configuration

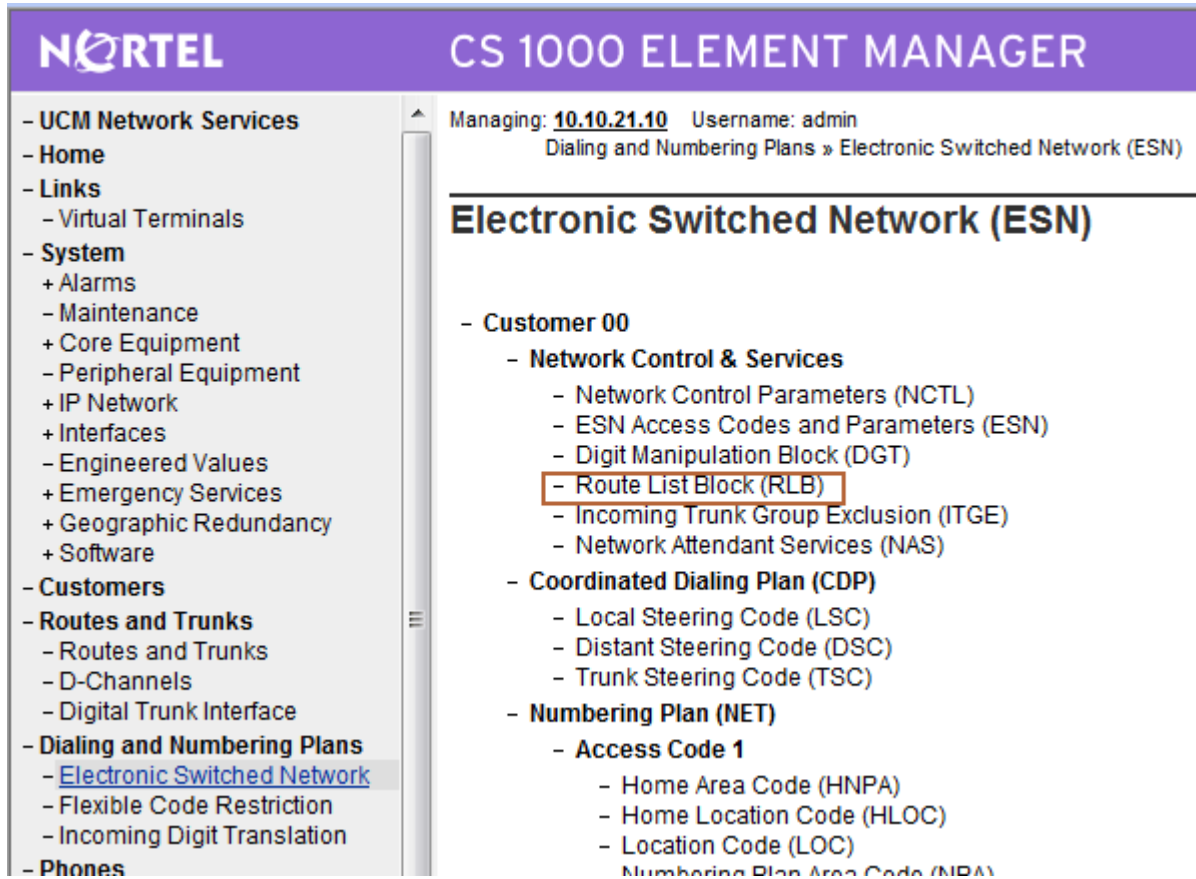
- Basic Configuration

Input Description	Input Value
Trunk data block (TYPE)	<input type="text" value="IPTI"/>
Terminal Number (TN)	<input type="text" value="252 0 15 00"/>
Designator field for trunk (DES)	<input type="text" value="VTRK"/>
Extended Trunk (XTRK)	<input type="text" value="VTRK"/>
Route number, Member number (RTMB)	<input type="text" value="15 1"/>
Level 3 Signaling (SIGL)	<input type="text" value=""/>
Card Density (CDEN)	<input type="text" value="8D"/>
Start arrangement Incoming (STRI)	<input type="text" value="Wink or Fast Flash (WNK)"/>
Start arrangement Outgoing (STRO)	<input type="text" value="Wink or Fast Flash (WNK)"/>
Trunk Group Access Restriction (TGAR)	<input type="text" value="1"/>
Channel ID for this trunk (CHID)	<input type="text" value="1"/>
Increase or decrease the member numbers (INC)	<input type="text" value="Increase channel and member number (YES)"/>
Class of Service (CLS)	<input type="button" value="Edit"/>

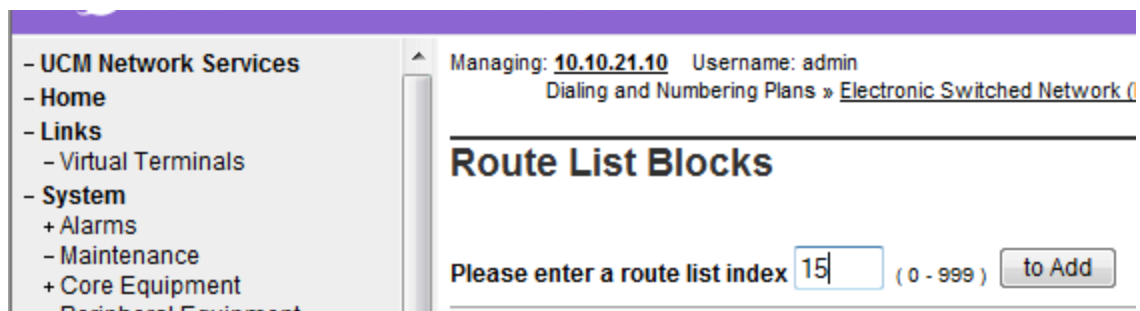
+ Advanced Trunk Configurations

3.6. Administer Route List Block and Distant Steering Code

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Route List Block (RLB)**.



The **Route List Blocks** screen is displayed. In the **Please enter a route list index** field, enter an available route list block number (in this case **15**). Click to **Add**.



The **Route List Block** screen is displayed with a listing of parameters. For the **Route Number (ROUT)** field, select the route number from **Section 3.6**. Retain the default values for the remaining fields, and scroll down to the bottom of the screen and click **Submit** (not shown).

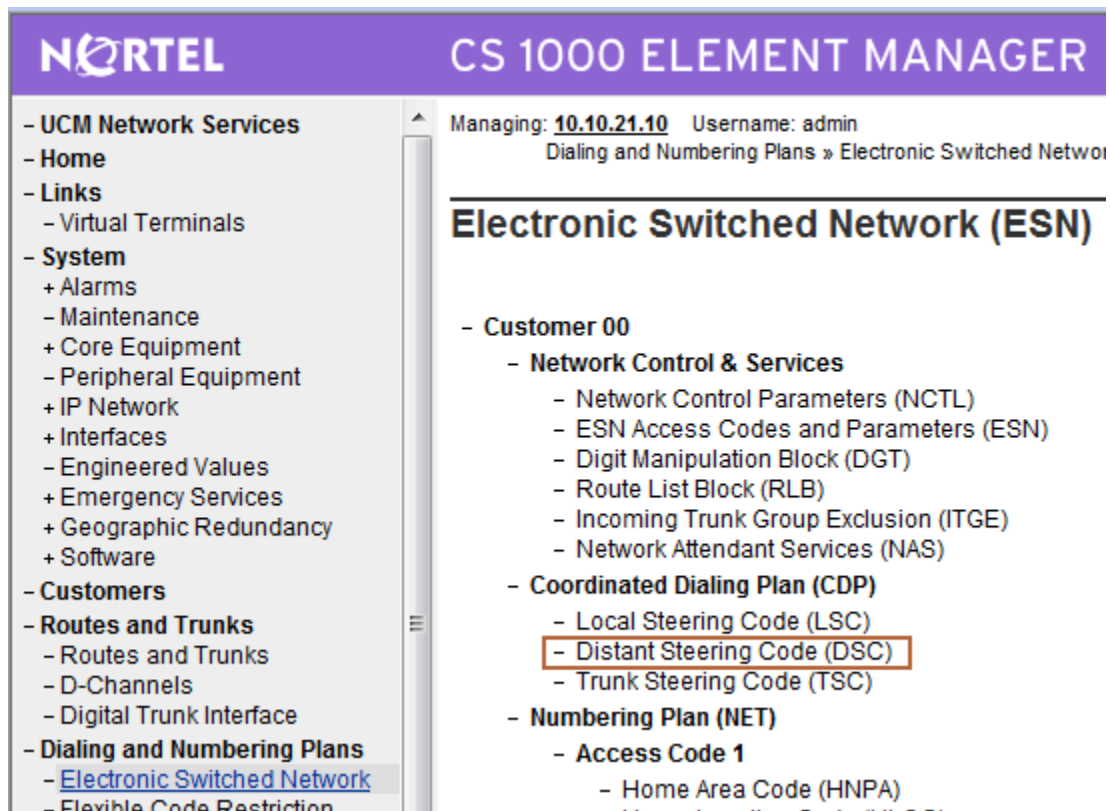
NORTEL CS 1000 ELEMENT MANAGER

Data Entry of a Route List Block

Route List Block Index: 15

Input Description	
Entry Number for the Route List (ENTR):	0
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	15
Skip Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0 (0 - 7)
Digit Manipulation Index (DMI):	0

Select **Dialing and Numbering Plans** → **Electronic Switched Network** again from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Distant Steering Code (DSC)** to add an entry to route 1111 calls to Avaya Meeting Exchange Application Server.



The **Distant Steering Code List** screen is displayed next (not shown). In the **Please enter a distant steering code** field, enter the dialed prefix digits to match on (in this case **111**) as the conference access number is 44444. Click to **Add**.

3.7. Administer Node SIP and Media Parameters

Select **IP Network** → **Nodes: Servers, Media Cards** → **Configuration** from the left pane, and in the **IP Telephony Nodes** screen displayed (not shown), select the node ID of this CS1000 system (see **Section 3.2**). The **Node Details** screen is displayed. Click on **Voice Gateway (VGW) and Codecs** (VGW) and Codecs.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))

Node ID: * (0-9999)

Call Server IP Address: *

Telephony LAN (TLAN)
Node IP Address: *
Subnet Mask: *

Embedded LAN (ELAN)
Gateway IP address: *
Subnet Mask: *

IP Telephony Node Properties

- [Voice Gateway \(VGW\) and Codecs](#)
- [Quality of Service \(QoS\)](#)
- [LAN](#)

Applications (click to edit configuration)

- [SIP Line](#)
- [Terminal Proxy Server \(TPS\)](#)
- [Gateway \(SIPGw & H323Gw\)](#)

* Required Value.

Associated Signaling Servers & Cards

Select to add [Print](#) | [Refresh](#)

<input type="checkbox"/>	Hostname ^	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input type="checkbox"/>	cs1k-r022011	Signaling Server	SIP Line, LTPS, Gateway, PD	10.10.21.10	10.10.22.11	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

In the following screen scroll down the parameters box and check the desired codecs under **Voice Codecs**. Note that G.711 and G.729 were verified for the sample configuration. G.711 is checked by default and cannot be unchecked. Click on **Save**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Voice Codecs

Codec G711: ☒ Enabled (required)
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)

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)

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

Save Cancel

Select **IP Network** → **Nodes: Servers, Media Cards**. Click the **Node ID** of the Avaya CS1000 Element (in this case **1000**).

The screenshot shows the 'CS 1000 ELEMENT MANAGER' interface. The left sidebar contains a navigation tree with 'IP Network' expanded and 'Nodes: Servers, Media Cards' selected. The main content area displays 'IP Telephony Nodes' with a table listing nodes. The node with ID '1000' is highlighted.

Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
1000	1	SIP Line, LTPS, Gateway (SIPGw, H323Gw)		10.10.22.10	Synchronized

Buttons at the top include 'Add...', 'Import...', 'Export...', and 'Delete'. A 'Show:' section has checkboxes for 'Nodes' (checked) and 'Component Servers and Cards'.

On the **Node Details** screen select **Gateway (SIPGw & H323Gw)**

The screenshot shows the 'Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))' screen. The left sidebar is the same as the previous screenshot. The main content area contains configuration fields for the node.

Node ID: 1000 * (0-9999)
 Call Server IP Address: 10.10.21.10 *
 Telephony LAN (TLAN)
 Node IP Address: 10.10.22.10 *
 Subnet Mask: 255.255.255.0 *
 Embedded LAN (ELAN)
 Gateway IP address: 10.10.21.1 *
 Subnet Mask: 255.255.255.0 *

IP Telephony Node Properties:

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN

Applications (click to edit configuration):

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw & H323Gw)

Buttons: Save, Cancel

Associated Signaling Servers & Cards

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
cs1k-r022011	Signaling Server	SIP Line, LTPS, Gateway, PD	10.10.21.10	10.10.22.11	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

Under **General** of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values for the specified fields, and retain the default values for the remaining fields.

Vtrk Gateway Application: Select **SIPGw and H.323Gw**

SIP Domain Name Domain name used for constructing the SIP URI in SIP messages (in this case **cs1k.avaya.com**)

Local SIP Port **5060**

Gateway endpoint name An appropriate name

The screenshot shows a web-based configuration interface for a network device. On the left is a navigation tree with categories like 'UCM Network Services', 'System', 'Interfaces', 'Customers', and 'Routes and Trunks'. The main area is titled 'Node ID: 1000 - Virtual Trunk Gateway Configuration Details'. It has tabs for 'General', 'SIP Gateway Settings', 'SIP Gateway Services', and 'H.323 Gateway Settings'. The 'General' tab is active, showing a 'Vtrk Gateway Application' dropdown set to 'SIPGw and H.323Gw', a 'SIP Domain name' field with 'cs1k.avaya.com', a 'Local SIP Port' field with '5060', a 'Gateway endpoint name' field with 'node1000', a 'Gateway password' field, an 'H.323 ID' field with 'node1000', and an 'Enable failsafe NRS' checkbox. To the right is a 'Virtual Trunk Network Health Monitor' section with a checkbox for 'Monitor IP Addresses', a text area for 'Monitor addresses', and 'Add' and 'Remove' buttons.

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Vtrk Gateway Application: ☒ Enable gateway service on this Node

General

Vtrk Gateway Application: SIPGw and H.323Gw

SIP Domain name: cs1k.avaya.com *

Local SIP Port: 5060 * (1 - 65535)

Gateway endpoint name: node1000 *

Gateway password: *

H.323 ID: node1000 *

Enable failsafe NRS: ☐

Virtual Trunk Network Health Monitor

☐ Monitor IP Addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP:

Monitor addresses:

Click on **SIP Gateway Settings**, and under **Proxy or Redirect Server**, enter the following values for the specified fields, and retain the default values for the remaining fields.

Primary TLAN IP Address: The IP address of the Signaling Server noted in **Section 3.2**.

Port **5060**

Transport Protocol **TCP**

Options Check **Support registration** and **Primary CDS Proxy**

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Virtual Trunk Gateway Configuration Details

General **SIP Gateway Settings** SIP Gateway Services H.323 Gateway Settings

SIP Gateway Settings

TLS Security: Security Disabled ▼

Port: 5061 (1 - 65535)

Number of Byte Re-negotiation: 0 ▼

Options: ☐ Client Authentication
☐ X509 certificate authority

Proxy Or Redirect Server:

Primary TLAN IP Address: 10.10.22.11 Secondary TLAN IP Address: 0.0.0.0

Port: 5060 (1 - 65535) Port: 5060 (1 - 65535)

Transport protocol: TCP ▼ Transport protocol: TCP ▼

Options: ☒ Support registration ☐ Support registration
☒ Primary CDS Proxy ☐ Secondary CDS Proxy

CLID Presentation:

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

Scroll down the parameters box to the **SIP URI Map** section. Under **Public E.164 Domain Names** heading, for
National Enter **publicNational**
Subscriber Enter **publicSubscriber**
The remaining fields can be left at their default values. Click on **Save**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

Area code: NPA in North America

Number Translation: Strip: Prefix: CLID Display Format:

Subscriber (SN): <CCC><Area code><SN>

National (NN): <CCC><NN>

International: <International number>

SIP URI Map:

Public E.164 Domain Names

National:

Subscriber:

Special number:

Unknown:

Private Domain Names

UDP:

CDP:

Special number:

Vacant number:

Unknown:

SIP Gateway Services

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

Return to the **Node Details** screen and click **Save**, as shown below.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))

Node ID: * (0-9999)

Call Server IP Address: *

Telephony LAN (TLAN)

Node IP Address: *

Subnet Mask: *

Embedded LAN (ELAN)

Gateway IP address: *

Subnet Mask: *

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw & H323Gw)

* Required Value. Save Cancel

The **Node Saved** screen is displayed. Click **Transfer Now...**

The screenshot shows the 'Node Saved' screen in the CS 1000 Element Manager. The left sidebar contains a navigation menu with options like 'UCM Network Services', 'Home', 'Links', 'System', and 'IP Network'. The main content area displays the message: 'Node ID: 1000 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards.' Below this message are two buttons: 'Transfer Now...' and 'Show Nodes'. The 'Transfer Now...' button is highlighted with a red box.

The **Synchronize Configuration Files** screen is displayed. Select the Signaling Server and click on **Start Sync**.

The screenshot shows the 'Synchronize Configuration Files (Node ID <1000>)' screen. The left sidebar is the same as the previous screenshot. The main content area has a title 'Synchronize Configuration Files (Node ID <1000>)' and a description: 'Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected restart* of applications on affected server(s) when complete.' Below the description are three buttons: 'Start Sync', 'Cancel', and 'Restart Applications'. The 'Start Sync' button is highlighted with a red box. Below the buttons is a table with the following data:

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	cs1k-r022011	Signaling Server	SIP Line, LTPS, Gateway, PD	Sync required

* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding servers.

When the synchronization completes, click on **Restart Applications**.

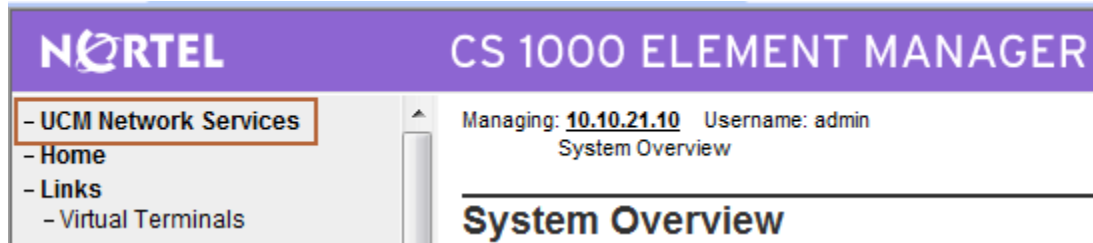
The screenshot shows the 'Synchronize Configuration Files (Node ID <1000>)' screen after the synchronization process has completed. The left sidebar is the same. The main content area has the same title and description. Below the description are three buttons: 'Start Sync', 'Cancel', and 'Restart Applications'. The 'Restart Applications' button is highlighted with a red box. Below the buttons is a table with the following data:

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	cs1k-r022011	Signaling Server	SIP Line, LTPS, Gateway, PD	Synchronized

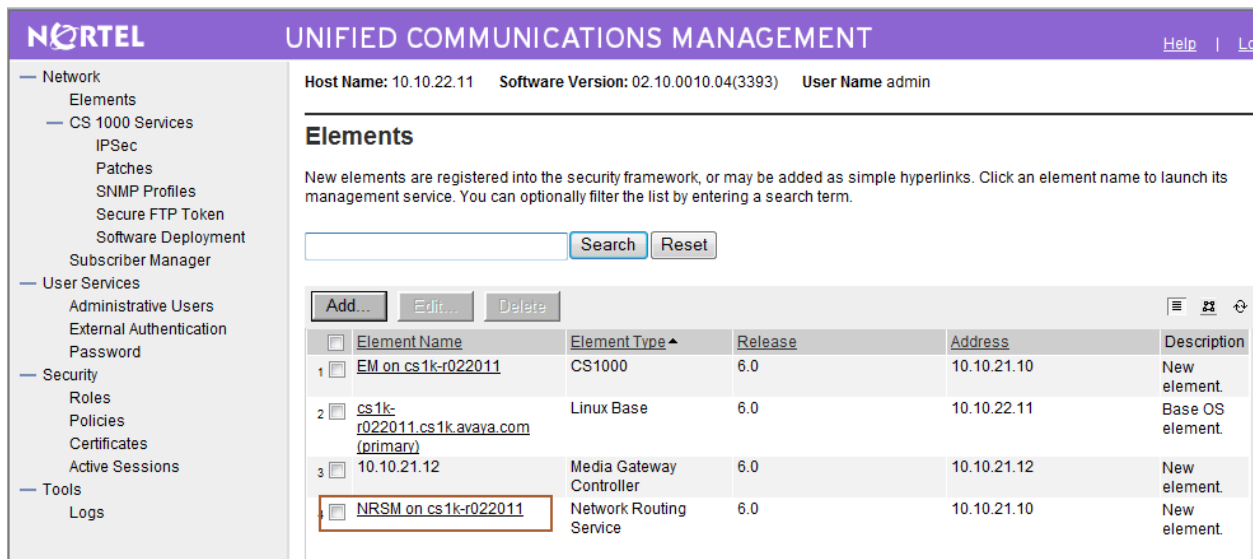
* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

3.8. Launch NRS Manager

Select **UCM Network Services** from the left pane, which will display the high level **Unified Communications Management** screen.



Click on the **Element Name** with **Element Type** “Network Routing Service”



Host Name: 10.10.22.11 Software Version: 02.10.0010.04(3393) 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.

	Element Name	Element Type	Release	Address	Description
1	EM on cs1k-r022011	CS1000	6.0	10.10.21.10	New element.
2	cs1k-r022011.cs1k.avaya.com (primary)	Linux Base	6.0	10.10.22.11	Base OS element.
3	10.10.21.12	Media Gateway Controller	6.0	10.10.21.12	New element.
	NRS on cs1k-r022011	Network Routing Service	6.0	10.10.21.10	New element.

The **NRS Server** screen is displayed. Click **Edit**.

NORTEL NETWORK ROUTING SERVICE MANAGER [Help](#) | [Logout](#)

Managing: 10.10.21.10
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 DublinNRS
Primary TLAN IP address 10.10.22.11
Secondary TLAN IP address 0.0.0.0
Secondary server host name SecondaryHostName
Control priority 40
Server mate communication port 5005
Realm name realmName
Server role Primary

H.323 Gatekeeper Settings

Location request (LRQ) response timeout 3

Under **SIP Server Settings**, enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Save** (not shown).

NOTE: For the scenario described in these Application Notes, SIP communication between the CS1000E and Meeting Exchange utilizes TCP.

TCP transport enabled: Check the checkbox

Primary Server TCP IP: Type the IP address of the Avaya SPS/NRS . All outgoing SIP messages from the Avaya SPS/NRS will use this address in the source field of the IP Header

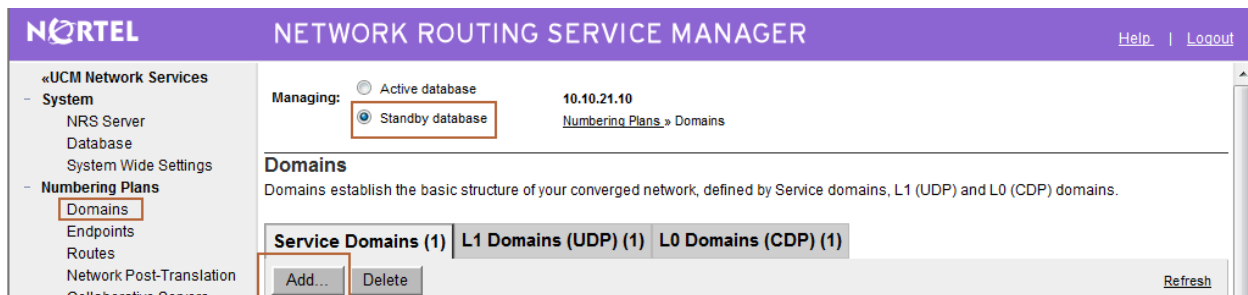
Primary Server TCP port: **5060**

Click on **Save**.

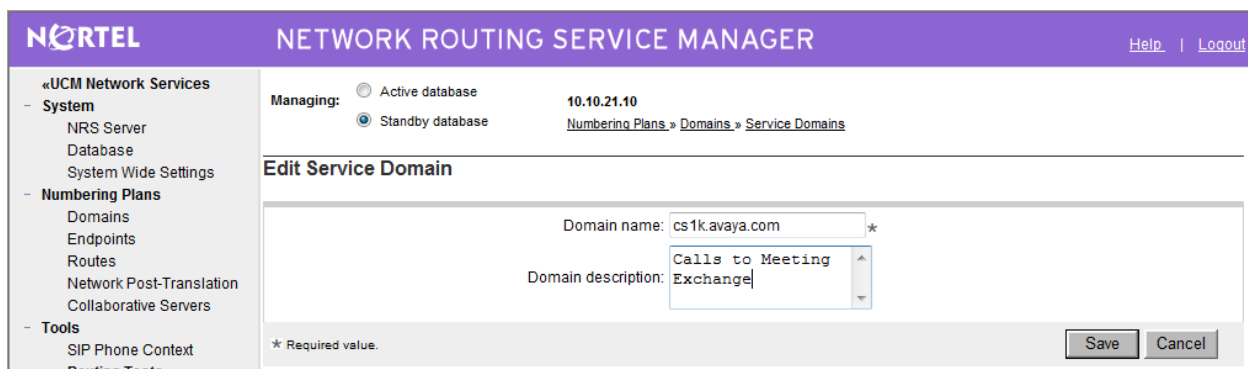
The screenshot shows the Nortel Network Routing Service Manager (NRS) interface. The top header is purple with the Nortel logo and the text "NETWORK ROUTING SERVICE MANAGER". On the right of the header are links for "Help" and "Logo". A left-hand navigation menu lists various configuration categories: «UCM Network Services», System (with sub-items NRS Server, Database, System Wide Settings), Numbering Plans (with sub-items Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers), Tools (with sub-items SIP Phone Context, Routing Tests (H.323, SIP), Backup, Restore, and GK/NRS Data upgrade). The main content area is titled "Edit Server Configuration" and shows "Managing: 10.10.21.10" with a link to "System » NRS Server » Edit". The "SIP Server Settings" section contains the following fields: "Public name for non-trusted networks" (unknown), "Public number for non-trusted networks" (000-000), "UDP Transport enabled" (checked), "Primary server UDP IP" (10.10.22.11), "Primary server UDP port" (5060), "Secondary server UDP IP" (0.0.0.0), "Secondary server UDP port" (5060), "TCP Transport enabled" (checked), "Primary server TCP IP" (10.10.22.11), "Primary server TCP port" (5060), "Secondary server TCP IP" (0.0.0.0), and "Secondary server TCP port" (5060). A note at the bottom states: "(Note: Any modification of NRS Server configuration would not take effect until you restart all the services.)". Below the note is a small text "* Required value." and two buttons: "Save" and "Cancel".

3.9. Administer Service Domain

The NRS hosts an active and a standby database. The active database is used for runtime queries, and the standby database is used for administrative modifications. Under **Numbering Plans** on the left, click on **Domains**, and the **Domains** screen will be displayed. To add a domain, first click on the **Standby database** radio button to switch to the standby database. Then the **Add** button will be added to the display. Click on it to add a domain.



The **Add Service Domain** screen is displayed. Enter the SIP domain name from **Section 3.8** into the **Domain name** field, and a descriptive text for the **Domain description** field. Click **Save**.



Select the **L1 Domains (UDP)** tab. Select the service domain just created for **Filter by Domain**, and click on **Add** to add a new L1 domain. The L1 and L0 domains are building blocks of the phone context for private addresses. For more information on L1 and L0 domains, refer to the Nortel documentation in **Section 9**.

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans » Domains](#)

Domains

Domains establish the basic structure of your converged network, defined by Service dom

Service Domains (1)	L1 Domains (UDP) (1)	L0 Domains (CDP) (1)
Filter by Domain : cs1k.avaya.com ▼		
<div>Add... Delete</div>		

The **Add L1 Domain (cs1k.avaya.com)** screen is displayed next (not shown). Enter a descriptive **Domain name** and **Domain description**. Retain the default value in the remaining fields, and click on **Save**.

NORTEL NETWORK ROUTING SERVICE MANAGER [Help](#) | [Logout](#)

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans » Domains » L1 Domain](#)

Edit L1 Domain (cs1k.avaya.com)

Domain name: udp *

Domain description:

Endpoint authentication enabled: Authentication off ▼

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)

Private L1 domain (UDP location) dialing access code:

* Required value

Save **Cancel**

Select the **L0 Domains (CDP)** tab. Select the service domain just created for **Filter by Domain** and **udp**, and click on **Add** to add a new L0 domain.

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans » Domains](#)

Domains

Domains establish the basic structure of your converged network, defined by Service domain

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

Filter by Domain : cs1k.avaya.com / udp

Add... **Delete**

The **Add L0 Domain (cs1k.avaya.com /udp)** screen is displayed next, (not shown). Enter a descriptive **Domain name** and **Domain description**. Retain the default values in the remaining fields and click **Save**.

NETWORK ROUTING SERVICE MANAGER[Help](#) | [Logout](#)

Managing: ☐ Active database **10.10.21.10**
☒ Standby database [Numbering Plans » Domains » L0 Domain](#)

Edit L0 Domain (cs1k.avaya.com / udp)

Domain name: *

Domain description:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

Private unqualified number label:

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)

★ Required value.

SaveCancel

3.10. Administer SIP Signaling Gateway Endpoints

Next, configure two SIP Signaling Gateway endpoints - one for the Avaya Meeting Exchange Application Server and another for the Avaya SIP Signaling Gateway. Under **Numbering Plans** on the left, click on **Endpoints**, and the **Search for Endpoints** screen will be displayed. For **Limit results to Domain**, select the service domain just created (**cs1k.avaya.com**), **udp** and **cdp**. Click **Add** to add a new gateway endpoint for Meeting Exchange Application Server.

«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 10.10.21.10
☒ 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: cs1k.avaya.com / udp / cdp

Results per page: 50 [Search](#)

Gateway Endpoints (4) **User Endpoints (0)**

[Add...](#) [Delete](#) [SIP phone context...](#) [Refresh](#)

Enter a descriptive **End point name** and **Description**.

«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 10.10.21.10
☒ Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint (cs1k.avaya.com / udp / cdp)

End point name: asmsil *

Description:

Trust Node: ☒

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Not Applicable

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:

* Required value [Save](#) [Cancel](#)

Scroll down the screen. Enter the following values for the specified fields, and retain the default values for the remaining fields.

Static endpoint address: IP address of the Meeting Exchange Application Server

SIP support: Static SIP endpoint

Select the **SIP TCP transport enabled** checkbox. Retain the default values in the remaining fields and click **Save**.

The screenshot displays the Nortel Network Routing Service Manager (NRS) web interface. The top header is purple with the Nortel logo and the text 'NETWORK ROUTING SERVICE MANAGER'. A 'Help' link is visible in the top right corner. On the left, a navigation menu lists various services and tools. The main content area is titled 'Edit Gateway Endpoint (cs1k.avaya.com / udp / cdp)'. It contains several configuration fields and checkboxes. The 'Static endpoint address type' is set to 'IP version 4'. The 'Static endpoint address' is '135.64.186.46'. 'H.323 support' is set to 'H.323 not supported'. 'SIP support' is set to 'Static SIP endpoint'. 'SIP Mode' has 'Proxy Mode' selected. 'SIP TCP transport enabled' is checked. 'SIP TCP port' is '5060'. 'SIP UDP transport enabled' is unchecked. 'SIP UDP port' is '5060'. 'SIP TLS transport enabled' is unchecked. 'SIP TLS port' is '5061'. 'Persistent TCP support enabled' is checked. 'End to end security support' is unchecked. 'Network Connection Server enabled' is unchecked. A 'Save' button is at the bottom right. A footer note indicates '* Required value'.

«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 **10.10.21.10**
☒ Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint (cs1k.avaya.com / udp / cdp)

Static endpoint address type: IP version 4
Static endpoint address: 135.64.186.46
H.323 support: H.323 not supported
SIP support: Static SIP endpoint
SIP Mode: ☒ Proxy Mode ☐ Redirect Mode
SIP TCP transport enabled: ☒
SIP TCP port: 5060
SIP UDP transport enabled: ☐
SIP UDP port: 5060
SIP TLS transport enabled: ☐
SIP TLS port: 5061
Persistent TCP support enabled: ☒
End to end security support: ☐
Network Connection Server enabled: ☐

* Required value Save

Repeat the procedures to add a **Gateway Endpoint** for the Avaya SIP Signaling Gateway as shown below.

NETWORK ROUTING SERVICE MANAGER[Help](#) | [Logout](#)

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans »](#) [Endpoints »](#) [Gateway Endpoint](#)

Edit Gateway Endpoint (cs1k.avaya.com / udp / cdp)

End point name: node1000 *

Description:

Trust Node: ☒

Tandem gateway endpoint name: Not Applicable ▾

Endpoint authentication enabled: Authentication off ▾

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:

* Required value

Save Cancel

Scroll down the screen. For the **SIP support** field, select **Dynamic SIP endpoint** from the drop-down list. Check the **SIP TCP transport enabled** field to match the SIP transport protocol from **Section 3.8**. Maintain the default values in the remaining fields, and click **Save**.

NETWORK ROUTING SERVICE MANAGERHelp | Logout

Managing: ☐ Active database **10.10.21.10**
☒ Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Edit Gateway Endpoint (cs1k.avaya.com / udp / cdp)

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

Static endpoint address type: **IP version 4** ▼

Static endpoint address:

H.323 support: **RAS H.323 endpoint** ▼

SIP support: **Dynamic SIP endpoint** ▼

SIP Mode
☒ Proxy Mode
☐ Redirect Mode

SIP TCP transport enabled: ☒

SIP TCP port: 5060

SIP UDP transport enabled: ☐

SIP UDP port: 5060

SIP TLS transport enabled: ☐

SIP TLS port: 5061

Persistent TCP support enabled ☒

End to end security support: ☐

Network Connection Server enabled: ☐

★ Required value

3.11. Administer Routing Entries

A single routing entry is required for Session Manager in order to reach the conference access number (DNIS=11111) on Meeting Exchange. Under **Numbering Plans** on the left, click on **Routes**, and the **Search for Endpoints** screen will be displayed. For **Limit results to Domain**, select the service domain just created, **udp** and **cdp**. Enter the **Endpoint name** corresponding to Session Manager. Click on **Add**.

«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 10.10.21.10
☒ Standby database [Numbering Plans » Routes](#)

Search for Routing Entries

Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a

DN Prefix: * DN Type: All DN Types

Limit results to Domain: cs1k.avaya.com / udp / cdp

Endpoint Name: asmsil

Re

Routing Entries (1) **Default Routes (0)**

Add... Copy... Move... Import... Export... Routing test... Delete

The **Add Routing Entry** screen is displayed next. Enter the following values for the specified fields, and retain the default values for the remaining fields. Click **Save**.

DN type: **Private level 0 regional (CDP steering code)**

DN prefix: Dialed prefix digits to match on, in this case **11**. **11111** is the conference access number.

Route cost (1 – 255): An appropriate cost value with 1 being least cost.

«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

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans » Routes » Routing Entry](#)

Edit Routing Entry (cs1k.avaya.com / udp / cdp / asmsil)

DN type: Private level 0 regional (CDP steering code) ▼

DN prefix: 11 *

Route cost: 1 * (1-255)

* Required value.

Repeat the same procedures to add a routing entry to reach the CS1000E endpoints with extension digits 3xxx behind the Avaya SIP Signaling Gateway Endpoint.

NORTEL NETWORK ROUTING SERVICE MANAGER [Help](#) | [Logout](#)

«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

Managing: ☐ Active database 10.10.21.10
☒ Standby database [Numbering Plans » Routes » Routing Entry](#)

Edit Routing Entry (cs1k.avaya.com / udp / cdp / node1000)

DN type: Private level 0 regional (CDP steering code) ▼

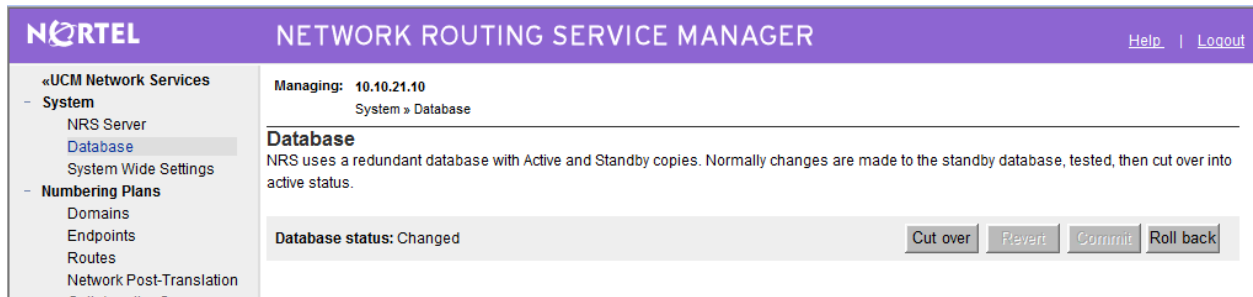
DN prefix: 3 *

Route cost: 1 * (1-255)

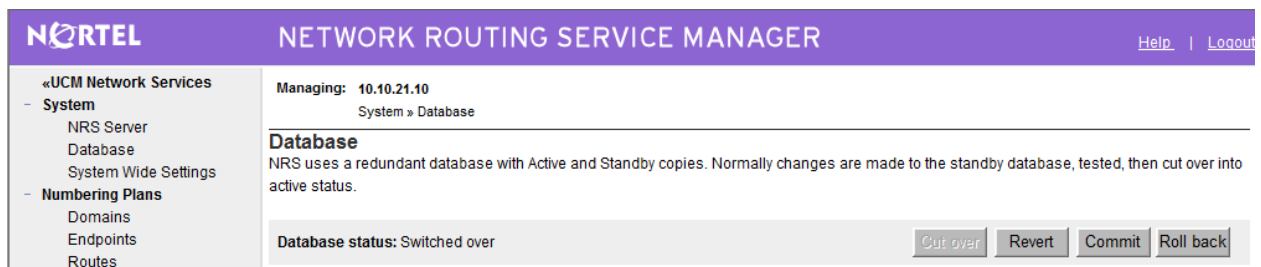
* Required value.

3.12. Cut Over and Commit Changes

Under **System** on the left, select **Database** to display the **Database** screen. Click on **Cut over**.



The **Database status** will change to **Switched over** and the **Commit** button will be enabled. Click on **Commit**.



4. Configure Avaya Meeting Exchange Enterprise Application Server

This section describes the steps for configuring a SIP trunk between Meeting Exchange Enterprise Application Server and Session Manager. This configuration will allow both moderators and participants to access a conference and also for operators to dial participants who have difficulties accessing a conference. It's assumed, that Meeting Exchange is installed, configured and licensed as per [3]. The following steps describe the administrative procedures for configuring Meeting Exchange Enterprise Application Server:

- Configure SIP Listener
- Configure Dialout
- Configure DNIS Mappings
- Configure Incoming SIP URI Conversion Rules
- Restart the Conference Bridge

The following instructions assume the user is logged in to the Meeting Exchange Enterprise Application Server Linux console using SSH.

4.1. Configure SIP Listener

The SIP signaling messages in the sample configuration are transmitted between the Avaya Meeting Exchange Enterprise Application Server and Session Manager over a TCP channel. Configure the following settings to enable SIP connectivity on the Meeting Exchange Enterprise Application Server:

- Edit `/usr/ipcb/config/system.cfg` using the Linux vi tool or download the file to your local machine using a Secure Copy Protocol (SCP) client (i.e.: WinSCP) for editing.
 - Add the IP address of the Meeting Exchange Enterprise Application Server: **IPAddress=10.10.21.50** as shown below
 - Add a line to populate the From Header Field in SIP INVITE messages. The following SIP URI will be displayed when the conference operator calls a participant: **MyListener=sip:6000@10.10.21.50**
 - Add a line to provide a SIP Device Contact address to use for acknowledging SIP messages: **respContact=sip:6000@10.10.21.50**

```
# ip address of the server
IPAddress=10.10.21.50

# request we will be listening to
MyListener=sip:6000@10.10.21.50

# if this setting is populated will Overwrite the contact field in responses
respContact=<sip:6000@10.10.21.50:5060;transport=tcp>
```

4.2. Configure Dialout

The FQDN of the Session Manager must be configured on Meeting Exchange Enterprise Application Server for dialout to work. The Application Server must be able to resolve the FQDN to an IP address, by either using its own hosts file or an external DNS server. For the

sample configuration an external DNS server was authoritative for the **cs1k.avaya.com** zone and had the following entry in its zone file:

```
cs1k.avaya.com      IN      A      135.64.186.46
```

Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor.

Add the following line to the file to route outbound calls from the Meeting Exchange Enterprise Application Server to the extension range administered on the CS1000's dial plan.

```
3*                  sip:$0@cs1k.avaya.com:5060;transport=tcp  default
```

4.3. Configure DNIS Mappings

The DNIS is the number that the phone users dial to access a conference. To map DNIS entries, run the **cbutil** utility on Meeting Exchange Enterprise Application Server as follows:

- Add 11111 as a new DNIS entry using the following command:

```
[mx6200-a ~]# cbutil add 11111 0 247 1 N SCAN
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

At the command prompt, enter **cbutil list** to verify the DNIS entries provisioned.

```
[mx6200-a ~]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

DNIS	Grp	Msg	PS	CP	Function	On	Failure	Line	Name	Company	Name	Room	Start	Room	End
-----	---	---	---	---	-----	---	-----	---	-----	-----	-----	-----	-----	-----	-----
11111	0	247	1	N	SCAN	ENTER						0		0	

4.4. Configure Incoming SIP URI Conversion Rules

The Meeting Exchange Enterprise Application Server rejects incoming SIP calls from the CS1000 SIP Proxy Server when CDP is used for call routing, because the CS1000 SIP Signaling Gateway includes a "phone-context=" and "user=phone" parameter in the SIP INVITE messages. In the sample configuration the To: field of the incoming INVITE includes the following values:

To: <sip:11111;phone-context=cdp.udp@cs1k.avaya.com;user=phone

The purpose of the **Incoming SIP URI Conversion Rule** is to extract the DNIS from the To: field and discard the **phone-context=cdp.udp@cs1k.avaya.com** and **user=phone** parameters.

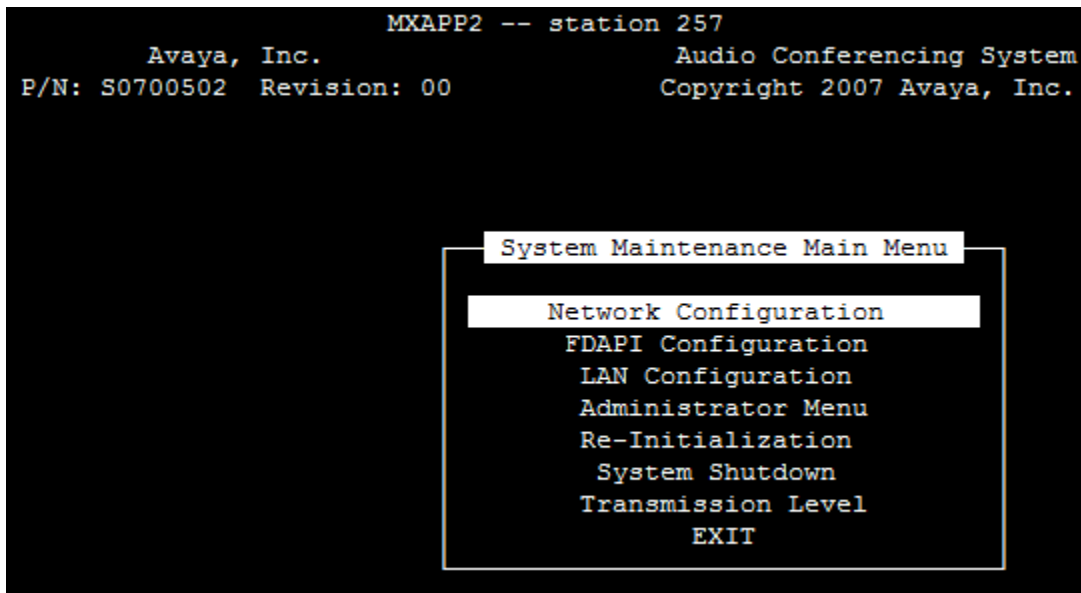
Edit the /usr/ipcb/config/UriToTelnum.tab file with a text editor. Enter the following two rules at the top of the list:

TelnumPattern	TelnumConversion	comment
"*sip:*;*@"	\$2	
"sip:*;*@"	\$1	

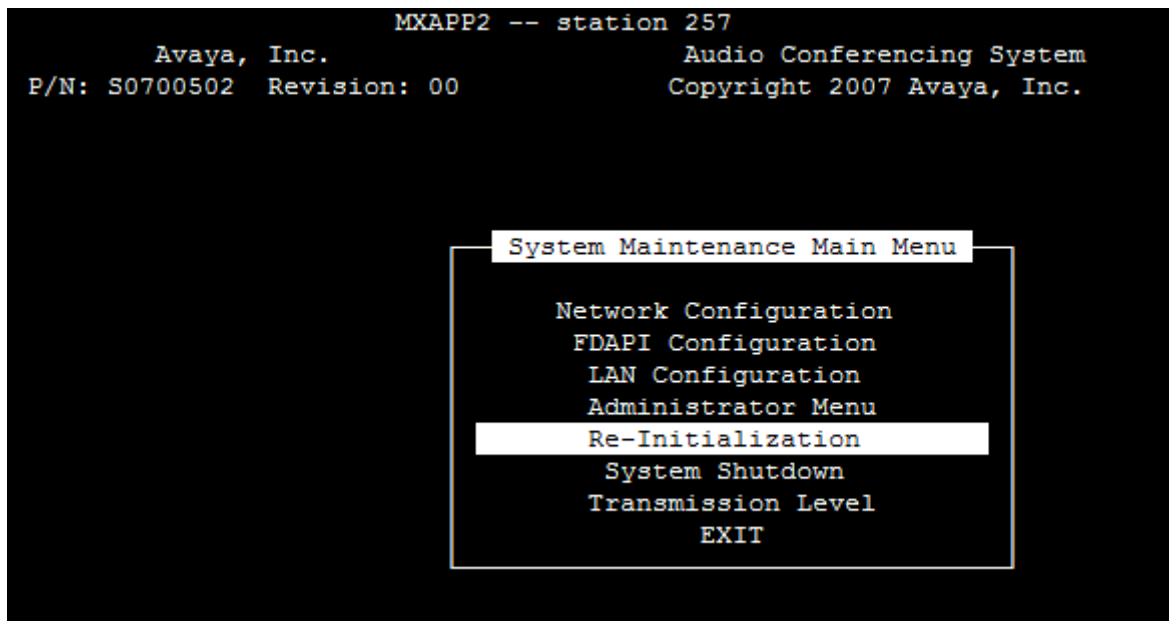
4.5. Restart the Conference Bridge

After the configuration changes are made, restart the Meeting Exchange Enterprise Application Server:

- Log in to the MX Application Server using the **dcbmaint** account.
- Issue the **dcbmaint** command. The **System Maintenance Main Menu** screen is displayed.



- Navigate to **Re-Initialization**.



- Press **Enter** and at the prompt type **yes**.

```
MXAPP2 -- station 257
Avaya, Inc. Audio Conferencing System
P/N: S0700502 Revision: 00 Copyright 2007 Avaya, Inc.

System Re-Initialization
ARE YOU SURE? (yes/NO):
```

- The **dcbmaint** utility terminates and the following message is displayed:

```
[dcbmaint@MXAPP2 sroot]$ dcbmaint
System Shutdown in Progress...
Please Stand By.

[dcbmaint@MXAPP2 sroot]$
```

5. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring the Session Manager and includes the following items:

- Log in to Avaya Aura™ System Manager using the GUI
- Administer SIP domain
- Define Locations for SIP Entities

For each SIP entity in the sample configuration:

- Define SIP Entity
- Define Entity Links
- Define Routing Policies
- Define Dial Patterns

5.1. Log in to Avaya Aura™ System Manager using the GUI

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “<https://<ip-address>/SMGR>”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials.

AVAYA Avaya Aura™ System Manager 5.2 [Help](#)

Home / Log On

Log On

You have successfully logged out.

Username :

Password :

5.2. Administer SIP Domain

Expand Network Routing Policy and select **SIP Domains**.

The screenshot shows the Avaya Aura System Manager 5.2 interface. The top header includes the Avaya logo, the product name "Avaya Aura™ System Manager 5.2", and a welcome message for user "admin" last logged on at Mar. 18, 2010 9:45 AM. A red breadcrumb trail shows "Home / Network Routing Policy". On the left, a navigation menu lists various sections, with "Network Routing Policy" expanded and "SIP Domains" highlighted. The main content area is titled "Introduction to Network Routing Policy (NRP)" and explains that NRP consists of several applications like "Domains", "Locations", "SIP Entities", etc. It provides a recommended workflow: Step 1: Create "Domains" of type SIP; Step 2: Create "Locations"; Step 3: Create "Adaptations"; Step 4: Create "SIP Entities" (including Outbound Proxies, Session Manager, CM, SIP/PSTN Gateways, SIP Trunks, and assigning Locations, Adaptations, and Outbound Proxies); Step 5: Create the "Entity Links" (between Session Managers and between Session Managers and other SIP Entities).

Click **New**.

The screenshot shows the Avaya Aura System Manager 5.2 interface at the "SIP Domains" page. The breadcrumb trail is "Home / Network Routing Policy / SIP Domains". The left navigation menu shows "Asset Management", "Communication System Management", and "User Management". The main content area is titled "Domain Management" and contains a row of buttons: "Edit", "New" (highlighted with a red box), "Duplicate", "Delete", and "More Actions".

On the **Domain Management** screen under **Name** add a descriptive name (in this case **cs1k.avaya.com**). Retain the default values for the remaining fields. Click **Commit** to save.

AVAYA Avaya Aura™ System Manager 5.2

Welcome, admin Last Logged on at Mar. 18, 2010 9:45 AM Help | Log off

Home / Network Routing Policy / SIP Domains

Domain Management

Commit Cancel

1 Item | Refresh Filter: Enable

Name	Type	Default	Notes
cs1k.avaya.com	sip	<input checked="" type="checkbox"/>	

* Input Required

Commit Cancel

5.3. Define Locations for SIP Entities

Expand Network Routing Policy and select **Locations**. Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. Click **New**.

AVAYA Avaya Aura™ System Manager 5.2

Welcome 9:45 AM

Home / Network Routing Policy / Locations

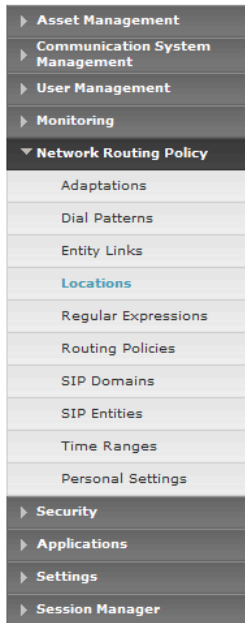
Location

Edit New Duplicate Delete More Actions Commit

8 Items | Refresh

	Name	Notes
<input type="checkbox"/>	Avaya	Lab, Test Domain, SILStack Domain
<input type="checkbox"/>	CUCME	
<input type="checkbox"/>	Interop-CME-7 1	

In the **General** Section, under **Name** add a descriptive name. Click on **Add**. In the **Location Pattern** Section under **IP Address Pattern** enter the IP Address of the SIP Proxy Server (SPS) of the CS1000E. In **Section 3.11** the SIP Mode of the Session Manager Endpoint was set to Proxy. This means that the SIP messages will be sourced from the IP address of the NRS/SPS. Click **Commit** to save.



Location Details

General

* Name:

Notes:

Managed Bandwidth:

* Average Bandwidth per Call: Kbit/sec ▾

* Time to Live (secs):

Location Pattern

1 Item | [Refresh](#)

Filter: [Enable](#)

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* <input type="text" value="10.10.22.11"/>	<input type="text"/>

Select : All, None (0 of 1 Selected)

Repeat the above steps for adding a **Location** for the Meeting Exchange Enterprise Application Server. Under **IP Address Pattern** enter the IP Address of the Meeting Exchange Enterprise Application Server as per **Figure 1**. Click **Commit** to save.

AVAYA

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Mar. 18, 2010 9:45 AM

Help | Log off

Home / Network Routing Policy / Locations / Location Details

Asset Management

Communication System Management

User Management

Monitoring

Network Routing Policy

Adaptations

Dial Patterns

Entity Links

Locations

Regular Expressions

Routing Policies

SIP Domains

SIP Entities

Time Ranges

Personal Settings

Security

Applications

Settings

Session Manager

Location Details

Commit

Cancel

General

* Name: Interop-MX

Notes:

Managed Bandwidth:

* Average Bandwidth per Call: 80 Kbit/sec

* Time to Live (secs): 3600

Location Pattern

Add

Remove

1 Item

Refresh

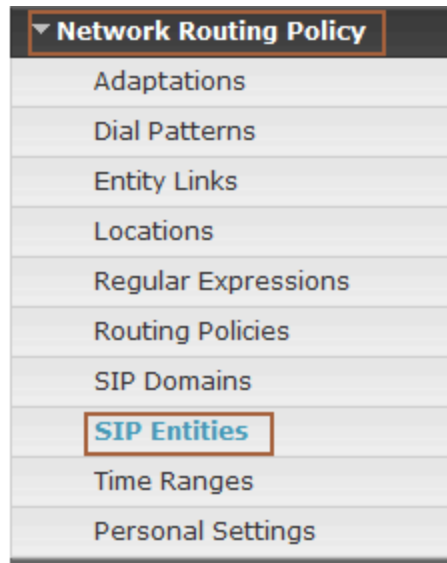
Filter: Enable

	IP Address Pattern	Notes
<input checked="" type="checkbox"/>	* 10.10.21.50	

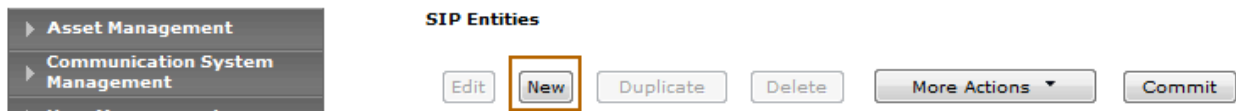
Select : All, None (0 of 1 Selected)

5.4. Define a SIP Entity for the Avaya CS1000E

Under **Network Routing Policy** in the left pane click **SIP Entities**.



The **SIP Entities** screen is displayed. Click **New**.



The following screen shows addition of the CS1000E as a SIP entity.

Under **General**:

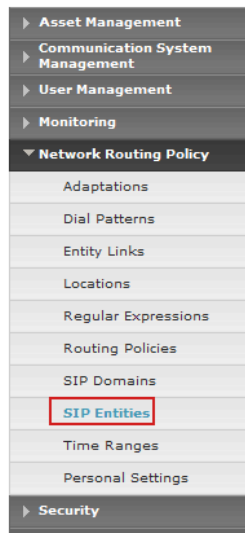
Name Type in a descriptive name for the CS1000E

FQDN or IP Address Type IP address of the NRS/SPS

Type Select **Other**

Notes (Optional) Type in description

Location Select the Location created in **Section 5.3**.



SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

5.5. Define a SIP Entity for Avaya Meeting Exchange Application Server

Repeat the steps from the previous section. Click **New** on the **SIP Entities** page. The following screen shows addition of the Meeting Exchange as a SIP entity. Under **General**:

Name Type in a descriptive name for the Meeting Exchange

FQDN or IP Address Type IP address of the Meeting Exchange Application Server

Type Select **Other**

Notes (Optional) Type in description

Location Select the Location we created in **Section 5.3**

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Mar, 18, 2010 9:45 AM [Help](#) | [Log off](#)

Home / Network Routing Policy / SIP Entities / SIP Entity Details

SIP Entity Details [Commit](#) [Cancel](#)

General

* Name: MX-Interop-Active

* FQDN or IP Address: 10.10.21.50

Type: Other

Notes:

Adaptation:

Location: Interop-MX

Time Zone: America/Denver

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

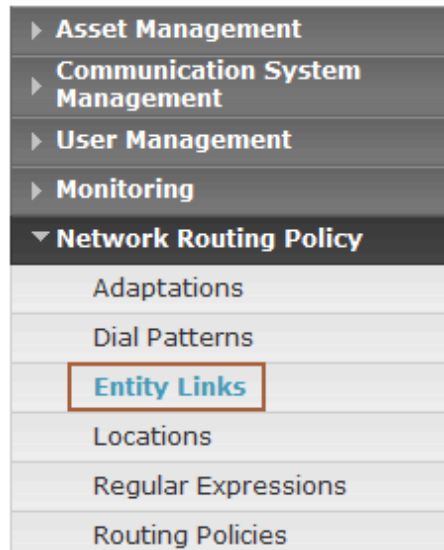
Credential name:

Call Detail Recording: none

SIP Link Monitoring

5.6. Create Entity Link for CS1000E

A SIP trunk between Session Manager and the NRS/SPS is described by an Entity Link. Entity Links are not required for SIP user-agent (IP Telephones). To add an Entity Link, select **Entity Links** on the left pane.



Click **New**.

Entity Links



The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

- Name:** A descriptive name, i.e. **CS1000**.
SIP Entity 1: Select the **Session Manager**.
Protocol: Select **TCP**.
Port: Type **5060**. The Session Manager will listen for SIP requests on TCP port 5060.
SIP Entity 2: Select the SIP Entity we created in **Section 5.4**

Entity Links Commit Cancel

1 Item [Refresh](#) Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2
CS1000	SessionManager	TCP	5060	CS1000

*** Input Required** Commit Cancel

Move the scroll bar to the right and fill in the remaining fields.

- Protocol:** Select **TCP**
Port: Type **5060**. The NRS/SPS will listen for SIP requests on TCP port 5060.
Trusted: Check this box, otherwise SIP calls will be denied to/from the NRS/SPS
Notes (Optional): A description for the Entity Link.

Click **Commit**.

Entity Links Commit Cancel

1 Item [Refresh](#) Filter: Enable

Protocol	Port	SIP Entity 2	Port	Trusted	Notes
TCP	5060	CS1000	5060	<input checked="" type="checkbox"/>	CS1000 Entity Link

*** Input Required** Commit Cancel

5.7. Create Entity Link for Avaya Meeting Exchange Enterprise Application Server

Repeat the steps from the previous section. Click **New** on the **Entity Links** page. The **Entity Links** screen is displayed. Fill in the following fields in the new row that is displayed:

- Name:** A descriptive name. In this case **MX-Interop-Active**
SIP Entity 1: Select the **Session Manager**
Protocol: Select **TCP**
Port: Type **5060**. The Session Manager will listen for SIP requests on TCP port 5060
SIP Entity 2: Select the SIP Entity we created in **Section 5.5**

Entity Links

Commit

Cancel

1 Item Refresh		Filter: Enable		
Name	SIP Entity 1	Protocol	Port	SIP Entity 2
* MX-Interop-Active	* SessionManager ▼	TCP ▼	* 5060	* MX-Interop-Active

* Input Required

Commit

Cancel

Move the scroll bar to the right and fill in the remaining fields.

Protocol: Select **TCP**.

Port: Type **5060**. The Meeting Exchange Application Server will listen for SIP requests on TCP port 5060.

Trusted: Check this box, otherwise SIP calls will be denied to/from the Meeting Exchange Application Server.

Notes (Optional): A description for the Entity Link.

Click **Commit**.

Entity Links

Commit

Cancel

Protocol	Port	SIP Entity 2	Port	Trusted	Notes
TCP	* 5060	* MX-Interop-Active	* 5060	<input checked="" type="checkbox"/>	

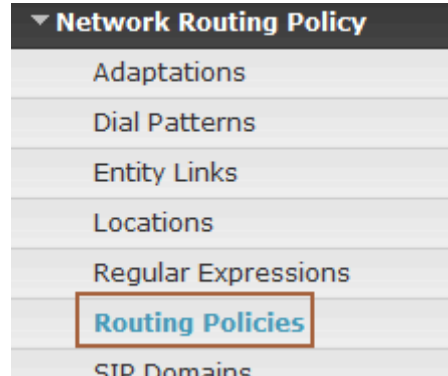
* Input Required

Commit

Cancel

5.8. Add Routing Policy for CS1000E

A routing policy describes the conditions under which calls will be routed to the CS1000E. To add a routing policy, select **Routing Policies** on the left pane.



Click **New**.

Routing Policies



The **Routing Policy Details** screen is displayed. Fill in the following under **General**:

Name Descriptive name. In this case **CS1000**.

Notes (Optional) Brief description.

Under **SIP Entity as Destination**:

Click **Select**, and then select the CS1000E SIP Entity we created in **Section 5.4** to which this routing policy applies.

Under **Time of Day**:

Click **Add**, and select the default **24/7** time range. Defaults can be used for the remaining fields. Click **Commit**.

Routing Policy Details

Commit

Cancel

General

* Name: CS1000

Disabled: ☐

Notes: Interop CS1000

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CS1000	10.10.22.11	Other	Nortel CS1000 R6 Node IP

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item Refresh		Filter: Enable										
<input type="checkbox"/>	Ranking 1 ▲	Name 2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input checked="" type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7
Select : All, None (0 of 1 Selected)												

5.9. Add Routing Policy for Avaya Meeting Exchange Enterprise Application Server

Repeat the steps from the previous section until the **Routing Policy Details** screen is displayed. Fill in the following under **General**:

Name Descriptive name. In this case MX-Interop-Active.

Notes (Optional) Brief description.

Under **SIP Entity as Destination**:

Click **Select**, and then select the Meeting Exchange SIP Entity we created in **Section 5.5** to which this routing policy applies. In this case **MX-Interop-Active**.

Under **Time of Day**:

Click **Add**, and select the default **24/7** time range. Defaults can be used for the remaining fields. Click **Commit**.

Routing Policy Details

CommitCancel

General

* Name: MX-Interop-Active

Disabled: ☐

Notes: from CS1K to MX_A

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
MX-Interop-Active	10.10.21.50	Other	

Time of Day

AddRemoveView Gaps/Overlaps

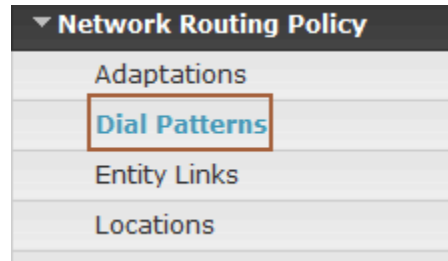
1 Item RefreshFilter: Enable

<input type="checkbox"/>	Ranking 1 ▲	Name 2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None (0 of 1 Selected)

5.10. Add Dial Pattern for CS1000E

Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 4-digit extensions beginning with **3** should be routed to the NRS/SPS. To add a dial pattern, select **Dial Patterns** on the left pane.



Click **New**.

Dial Patterns



The **Dial Pattern Details** screen is displayed. Under **General** fill in the following fields:

Pattern: Type **3xxx** as four digit extensions are used on the IP and Digital phones beginning with **3**.
Min: Minimum length of dialled number. Type **4**
Max: Maximum length of dialled number. Type **4**
SIP Domain: SIP domain specified in **Section 5.2**

Under **Originating Locations and Routing Policies**:

Click **Add**. Select the following entries:

Originating Location Name Select the Location created in **Section 5.3**

Routing Policy Name Select the Routing Policy created in **Section 5.8**

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern. The following screenshot shows the dial pattern for the CS1000E.

Dial Pattern Details

[Commit](#) [Cancel](#)

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination
<input type="checkbox"/>	Interop-MX		CS1000	0	<input type="checkbox"/>	CS1000

Select : All, None (0 of 1 Selected)

5.11. Add Dial Pattern for Avaya Meeting Exchange Enterprise Application Server

Repeat the steps from the previous section until the **Dial Pattern Details** screen is displayed.

Under **General** fill in the following fields:

Pattern: Type **1111** as the DNIS (conference access number) on the Meeting Exchange is set to 1111.

Min: Minimum length of dialled number. Type **5**

Max: Maximum length of dialled number. Type **5**

SIP Domain: SIP domain specified in **Section 5.2**

Under **Originating Locations and Routing Policies**:

Click **Add**. Select the following entries:

Originating Location Name Select **All**

Routing Policy Name Select the Routing Policy we created in **Section 5.9**. In this case **MX-Interop-Active**.

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern. The following screenshot shows the dial pattern for the Meeting Exchange.

Dial Pattern Details

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

2 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination
<input type="checkbox"/>	-ALL-	Any Locations	MX-Interop-Active	0	<input type="checkbox"/>	MX-Interop-Active

6. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Communication Server 1000E and Avaya Meeting Exchange Enterprise Edition.

6.1. Verify Avaya Communication Server 1000E

Select **IP Network** → **Nodes: Servers, Media Cards** → **Maintenance and Reports** on the left pane. Click **GEN CMD**.

Nortel CS 1000 ELEMENT MANAGER

Managing: 10.10.21.10 Username: admin
System » IP Network » Node Maintenance and Reports

Node Maintenance and Reports

Node ID: 1000	Node IP: 10.10.22.10	Total elements: 1		
Index	ELAN IP	Type	TN	ELAN
cs1k-r022011	10.10.21.10	Signaling Server-Nortel CPPMV1	NO TN	

Buttons: GEN CMD, SYS LOG, OM RPT, Reset, Virtual Terminal, Status

The **General Commands** page is displayed. Click on **Group** and from the drop-down list box select **Sip**. From the **Command** list box select **SIPGwShow** and click on **RUN**.

Nortel CS 1000 ELEMENT MANAGER

Managing: 10.10.21.10 Username: admin
System » IP Network » Node Maintenance and Reports » General Commands

General Commands

Element IP: 10.10.21.10 Element Type: Signaling Server-Nortel CPPMV1

Group: Sip Command: SIPGwShow Sip RUN

IP address: 10.10.21.10 Number of pings: 3 PING

Click a button to invoke a command.

Confirm that the SIP Signaling Gateway is registered to the SIP Proxy Server and that the SPS uses TCP port 5060 for SIP message transport.

Managing: **10.10.21.10** Username: admin
System » IP Network » Node Maintenance and Reports » General Commands

General Commands

Element IP : 10.10.21.10 Element Type : Signaling Server-Nortel CPPMv1

Group	Sip	Command	SIPGwShow	Sip	RUN
IP address	10.10.21.10	Number of pings	3	PING	

```
SIPNPM Status           : Active
Primary  Proxy IP address : 10.10.22.11
Primary  Proxy port       : 5060
Primary  Proxy Transport  : TCP
Secondary Proxy IP address : 0.0.0.0
Secondary Proxy port       : 5060
Secondary Proxy Transport  : TCP
Active   Proxy            : Primary :Registered
Time To Next Registration  : 290 Seconds
Channels Busy / Idle / Total : 0 / 40 / 40
Stack version              : 4.0.0.30
TLS Security Policy        : Security Disabled
SIP Gw Registration Trace  : OFF
Output Type Used           : TTY
Channel tracing            : -1
```

6.2. Verify Avaya Meeting Exchange Enterprise Edition

Verify that the **sipagent** process is running and that number **1** or number **2** is displayed after **/usr/dcb/bin/sipagent** in the following command output on the MX Application Server:

```
[craft@MXAPP1]$ ps -ef | grep -i sipagent
sroot      11079 10966  0 Feb18 ?          00:00:21 /usr/dcb/bin/sipagent 1
craft      25946 25926  0 14:00 pts/1    00:00:00 grep -i sipagent
```

Verify that the MX Application Server is listening for SIP requests on TCP port 5060:

```
[craft@MXAPP1]$ netstat -antu | grep 506
tcp          0          0 0.0.0.0:5060          0.0.0.0:*           LISTEN
```

6.3. Verify Avaya Aura™ Session Manager

Log in to System Manager. Navigate to **Session Manager** → **System Status** → **SIP Entity Monitoring** on the left pane to verify that none of the links to the defined SIP entities are down. The **SIP Entity Monitoring** screen is displayed. Click on the individual SIP Entities to check their status. In this sample configuration, **CS1000** and **MX-Interop-Active** were used.

▼ **Session Manager**

Session Manager Administration

▶ Network Configuration

▶ Device and Location Configuration

▶ Application Configuration

▼ **System Status**

▪ System State Administration

▪ **SIP Entity Monitoring**

▪ Managed Bandwidth Usage

▪ Security Module Status

▪ Data Replication Status

▪ RegistrationSummary

▪ User Registrations

▶ System Tools

Shortcuts

Change Password

Help for SIP Monitoring

Help for Page Fields

SessionManager2 0/0 0

All Monitored SIP Entities


Refresh

19 Items Filter: Enable

SIP Entity Name
AvayaCM
AvayaCM_MD
AvayaCMtom
Branch CM
Cisco
CS1000
feature
MM 3rd Party
ModularMessaging2_MD
ModularMessaging_MD
MX-Interop-Active
MX-Interop-Standby
MX-S6200
NewStackFeature
PSTN_CM

< Previous | Page 1 of 2 | Next >

The following screenshot displays the **SIP Entity Link Connection Status** of the CS1000 Entity. The **SIP Entity Resolved IP** column displays the IP address of the NRS/SPS. Check the **Link Status** column and make sure it displays **Up**. Repeat these steps for the Meeting Exchange Entity Link.



Avaya Aura™ System Manager
5.2

Welcome, **admin** Last Logged on at Feb. 22, 2010 1:00 PM

[Help](#) [Log off](#)

[Home](#) / [Session Manager](#) / [System Status](#) / [SIP Entity Monitoring](#) / [SIP Entity Link Status](#)

- ▶ Asset Management
- ▶ Communication System Management
- ▶ User Management
- ▶ Monitoring
- ▶ Network Routing Policy
- ▶ Security
- ▶ Applications
- ▶ Settings
- ▼ Session Manager
 - Session Manager Administration
 - ▶ Network Configuration
 - ▶ Device and Location Configuration
 - ▶ Application Configuration
- ▼ System Status
 - System State Administration
 - ▶ SIP Entity Monitoring
 - Managed Bandwidth Usage

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: CS1000

Refresh
Summary View

Details		SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▶ Show		10.10.22.11	5060	TCP	Up	200 OK	Up

7. Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Conference calls between various telephones on the Avaya Communication Server 1000E can be made using G.711MU/A, G.729 and G.722.
- Proper display of the calling and called party name and number information was verified for all telephones.
- Dialout from the Operator phone to conference participants was verified.

8. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000E (formerly known as Nortel CS1000E) can interoperate with Avaya Meeting Exchange Enterprise Edition via Avaya Aura™ Session Manager using SIP trunks. The following is a list of interoperability items to note:

- MX Patch Group 5.2.1.2.1 needs to be applied on the Meeting Exchange Application Server for this configuration to work.
- The UriToTelnum.cfg file on the Avaya Meeting Exchange Application Server must contain the required regular expression entries in order to remove the **phone-context** and **user=phone** parameter from the incoming INVITE on the Avaya Meeting Exchange Application Server.

9. Additional References

Avaya CS1000E Support Documents:

- [1] *Network Routing Service Fundamentals*, Nortel Communication Server 1000 Release 6, Document Number NN43001-130, Version 1.03, May, 2009, available on the Nortel Communication Server Electronic Reference Library CD. <http://support.nortel.com>
- [2] *IP Peer Networking Installation and Commissioning*, Nortel Communication Server 1000 Release 6, Document Number NN43001-313, Version 3.02, May, 2009, available on the Nortel Communication Server Electronic Reference Library CD. <http://support.nortel.com>

Avaya Meeting Exchange Support Documents:

- [3] *Administering Meeting Exchange™ 5.2 Servers*, Doc ID 04-603419, 16-Nov-2009, available at <http://support.avaya.com/css/P8/documents/100068644>
- [4] *Administering Meeting Exchange™ 5.2 Applications*, Doc ID 04-603420, 16-Nov-2009, available at <http://support.avaya.com/css/P8/documents/100068646>

Avaya Aura™ Session Manager:

- [5] *Avaya Aura™ Session Manager Overview*, Doc ID 03-603473, available at <http://support.avaya.com>.
- [6] *Installing and Upgrading Avaya Aura™ Session Manager*, Doc ID 03-603324, available at <http://support.avaya.com>.
- [7] *Maintaining and Troubleshooting Avaya Aura™ Session Manager*, Doc ID 03-603325, available at <http://support.avaya.com>.

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