



Avaya Solution & Interoperability Test Lab

Configuring SIP Trunks among Avaya Communication Server 1000E 6.0 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.

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.

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. A SIP trunk is used to connect the Avaya Communication Server 1000E to Avaya Meeting Exchange Enterprise over the LAN. Signaling messages are carried over the TCP-based SIP trunk 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 Enterprise is set to 44444.

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.

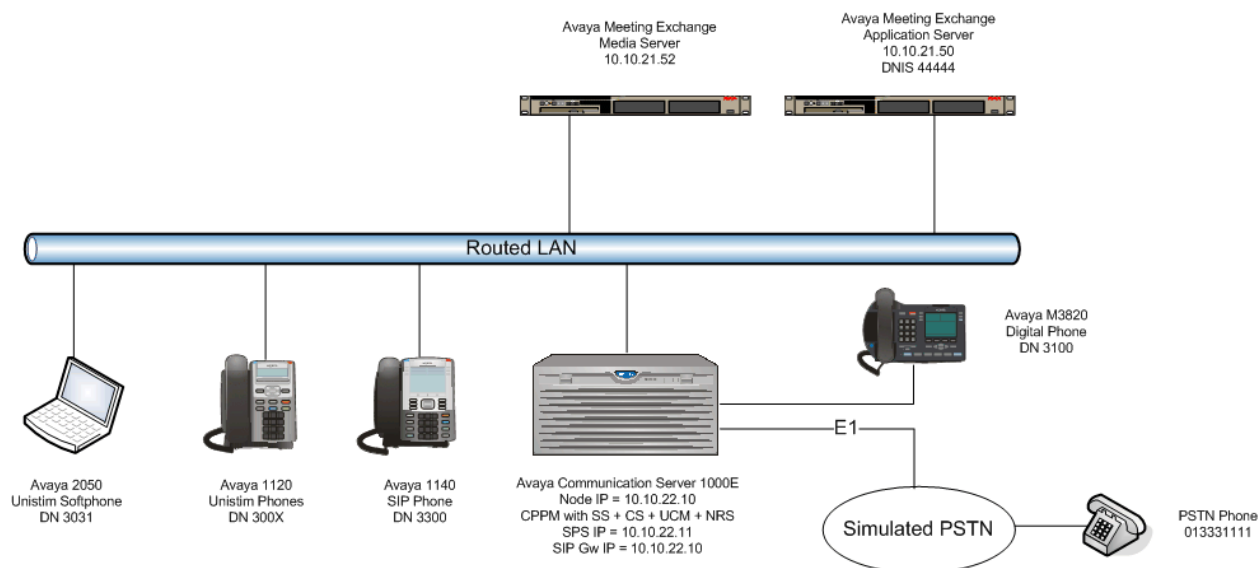


Figure 1 – Sample Configuration

These Application Notes will focus on the configuration of the SIP trunk and call routing. Detailed administration of the endpoint telephones and E1 PRI trunk will not be described (see the appropriate documents listed in **Section 8**).

2. Equipment and Software Validated

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

Equipment	Software Version
Avaya Communication Server 1000E	Release 600R, Version 4121
Avaya 1140 IP Phone	SIP 02.02.21.00
Avaya 1120E IP Phone	UNISim 0624C60
Avaya IP SoftPhone2050	UNISim 3.04.0003
Avaya M3820 Digital Phone	N/A
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Application Server S6200 R5.2 Build 5.2.1.0.4 (GA) and MX Patch Group 5.2.1.2.1
Avaya S8510 Server	Avaya Meeting Exchange Enterprise Edition Media Server S6200 R5.2 Build 5.2.1.0.4 (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 UNISim (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.

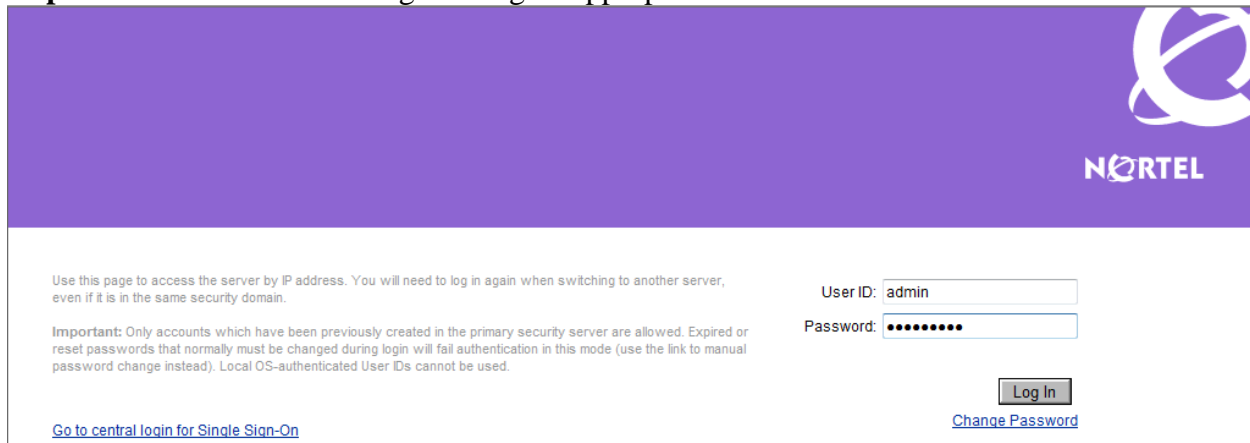
These Application Notes used the Coordinated Dial Plan (CDP) feature to route calls from the Avaya Communication Server 1000E, over a SIP trunk to Avaya Meeting Exchange. The CDP feature is assumed to be already enabled on Avaya Communication Server 1000, and therefore will not be described in detail.

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 Avaya Meeting Exchange:

- Log in 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:

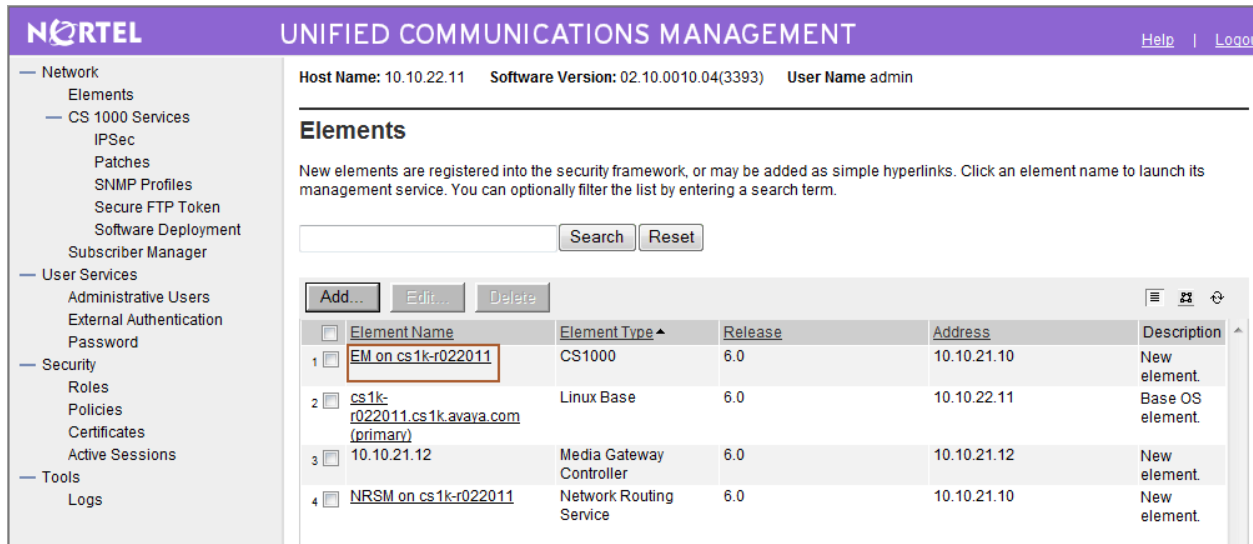
Password:

[Change Password](#)

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

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](#)

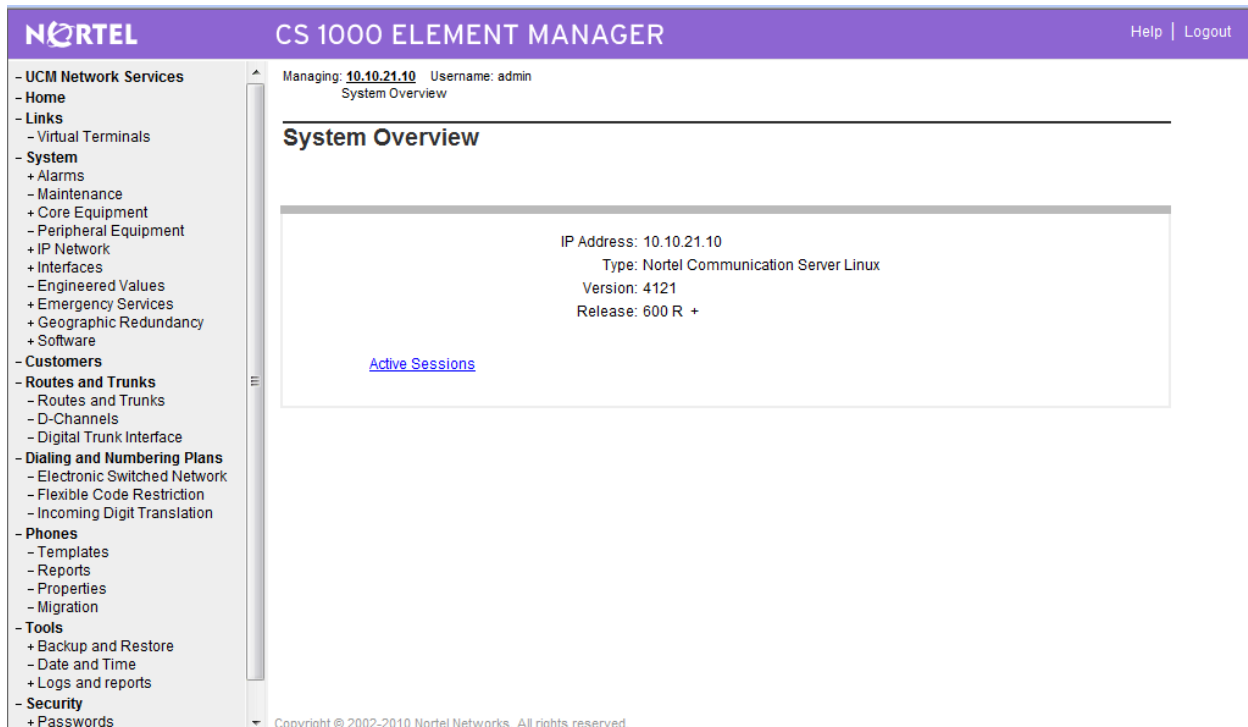
Host Name: 10.10.22.11 Software Version: 02.10.0010.04(3393) User Name admin

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"/>	1 EM on cs1k-r022011	CS1000	6.0	10.10.21.10	New element.
<input type="checkbox"/>	2 cs1k-r022011.cs1k.avaya.com (primary)	Linux Base	6.0	10.10.22.11	Base OS element.
<input type="checkbox"/>	3 10.10.21.12	Media Gateway Controller	6.0	10.10.21.12	New element.
<input type="checkbox"/>	4 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

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 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](#)

<input type="checkbox"/> Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
<input type="checkbox"/> 1000	1	SIP Line, LTSP, Gateway (SIPGw, H323Gw	-	10.10.22.10	Synchronized

Show: ☒ Nodes ☐ Component Servers and Cards

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

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.21.10 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](#)

<input type="checkbox"/> Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
<input type="checkbox"/> 1000	1	SIP Line, LTSP, Gateway (SIPGw, H323Gw	-	10.10.22.10	Synchronized

Show: ☒ Nodes ☐ Component Servers and Cards

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).

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

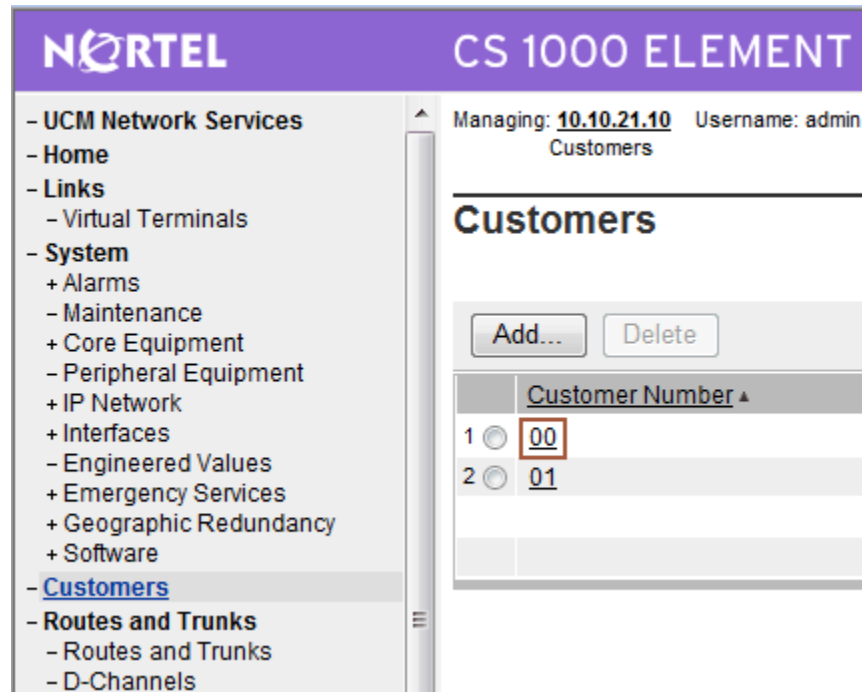
Select to add [Print](#) | [Refresh](#)

<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

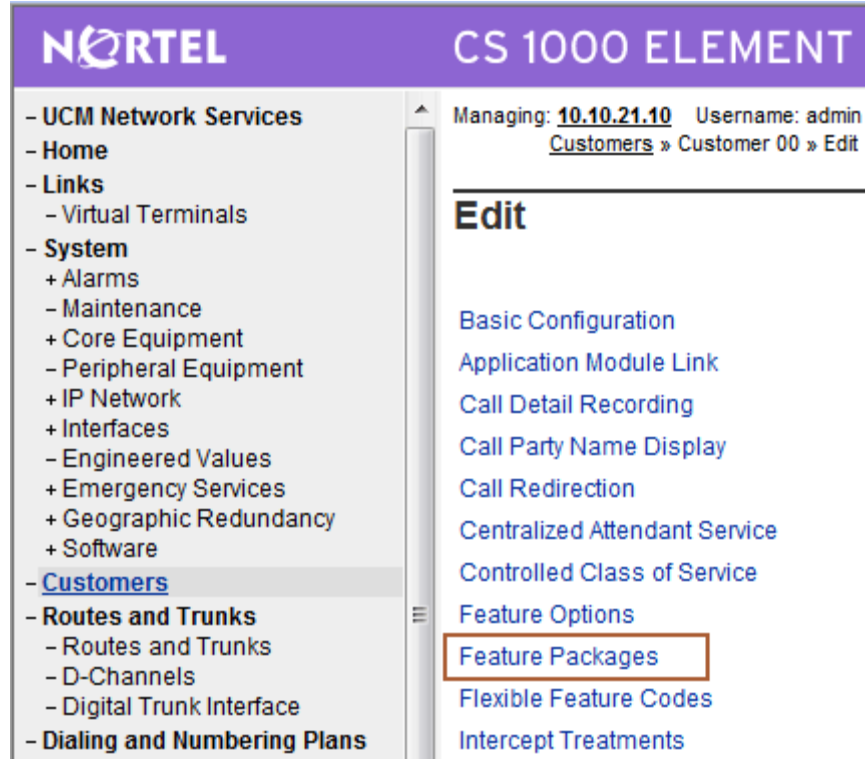
Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

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
- 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

+ Command Status Link	Package: 77
+ Pretranslation	Package: 92
+ Dialed Number Identification System	Package: 98
+ Malicious Call Trace	Package: 107
+ Incoming Digit Conversion	Package: 113
+ Directed Call Pickup	Package: 115
+ Enhanced Music	Package: 119
+ Station Camp-On	Package: 121
+ Flexible Tones and Cadences	Package: 125
+ Multifrequency Compelled Signaling	Package: 128
+ International Supplementary Features	Package: 131
+ Enhanced Night Service	Package: 133
+ Integrated Services Digital Network	Package: 145
+ Network Attendant Service	Package: 159

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).

NORTEL CS 1000 ELEMENT MANAGER

- 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

+ Multifrequency Compelled Signaling	Package: 128
+ International Supplementary Features	Package: 131
+ Enhanced Night Service	Package: 133
- Integrated Services Digital Network	Package: 145
+ Dial Access Prefix on CLID table entry option	
Integrated Services Digital Network: <input checked="" type="checkbox"/>	
- Virtual Private Network Identifier:	1 (1 - 16383)
- Private Network Identifier:	1 (1 - 16383)
- Node DN:	
- Multi-location Business Group:	0 (0 - 65535)

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. The top header is purple with the Nortel logo and the text 'CS 1000 ELEMENT MANAGER'. Below the header, a navigation pane on the left lists various system components, with 'D-Channels' highlighted under the 'Routes and Trunks' section. The main content area is titled 'D-Channels' and contains two sections: 'Maintenance' and 'Configuration'. The 'Maintenance' section lists several diagnostic links: 'D-Channel Diagnostics (LD 96)', 'Network and Peripheral Equipment (LD 32, Virtual D-Channels)', 'MSDL Diagnostics (LD 96)', and 'D-Channel Expansion Diagnostics (LD 48)'. The 'Configuration' section features a form with a dropdown menu for 'Choose a D-Channel Number' (currently showing '0'), a dropdown for 'and type:' (currently showing 'DCH'), and a button labeled 'to Add' which is highlighted with an orange border.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.10.21.10** Username: admin
Routes and Trunks » D-Channels

D-Channels

Maintenance

- [D-Channel Diagnostics \(LD 96\)](#)
- [Network and Peripheral Equipment \(LD 32, Virtual D-Channels\)](#)
- [MSDL Diagnostics \(LD 96\)](#)
- [D-Channel Expansion Diagnostics \(LD 48\)](#)

Configuration

Choose a D-Channel Number: and type:

The **D-Channels 20 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)

Click **Submit** (not shown).

NORTEL CS 1000 ELEMENT MANAGER Help |

Managing: 10.10.21.10 Username: admin
Routes and Trunks » D-Channels » D-Channels 20 Property Configuration

D-Channels 20 Property Configuration

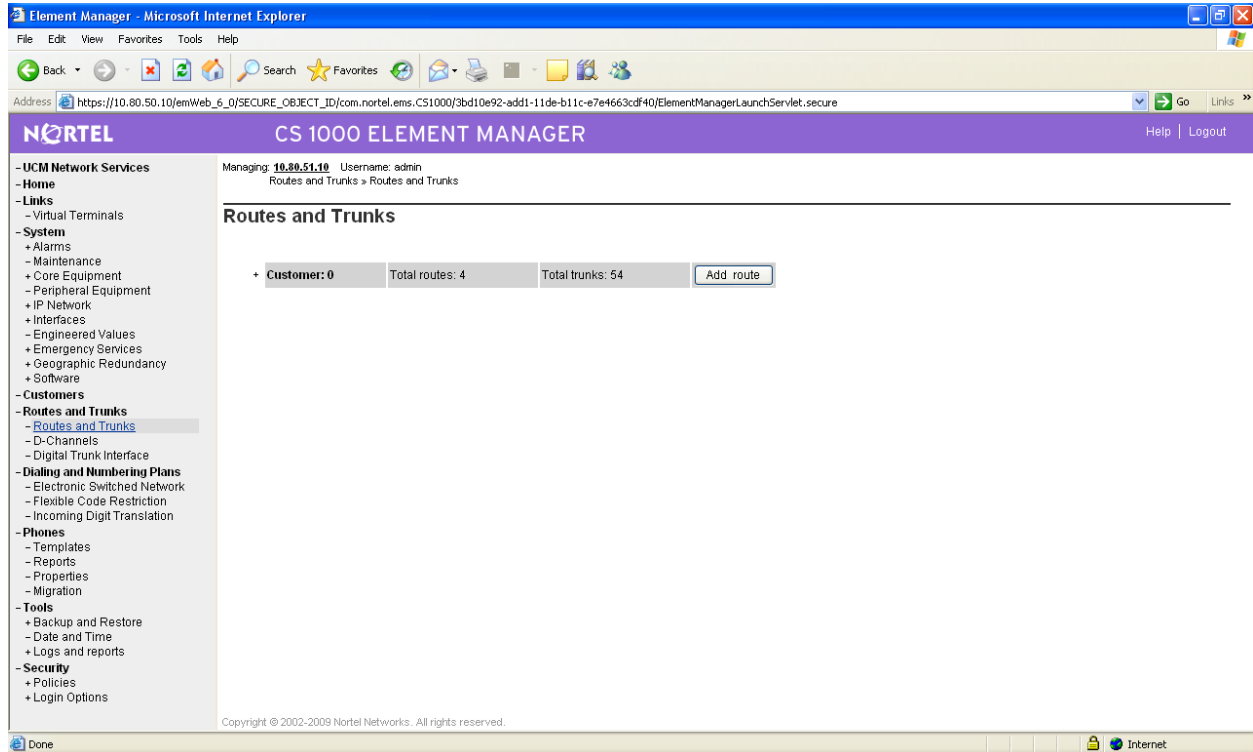
- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	VrtkNode1000
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"/> <button>more PRI</button>
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

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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.

NORTEL CS 1000 ELEMENT MANAGER

- 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
 - Properties
 - Migration
- Tools
 - + Backup and Restore
 - Date and Time

- 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) ☒

- 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) ☒

- Mode of operation (MODE)

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

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.

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):

D channel number (DCH):

Network calling name allowed (NCNA):

Network call redirection (NCRD):

Route uses ISDN Signaling Link (ISLD)

D-Channel number from **Section 3.4**

Check the field.

Check the field.

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.

NORTEL CS 1000 ELEMENT MANAGER

Managing: [10.10.21.10](#) Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

Customer	Total routes	Total trunks	Actions	
- 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: VTRKNode1001SIPL	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:

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

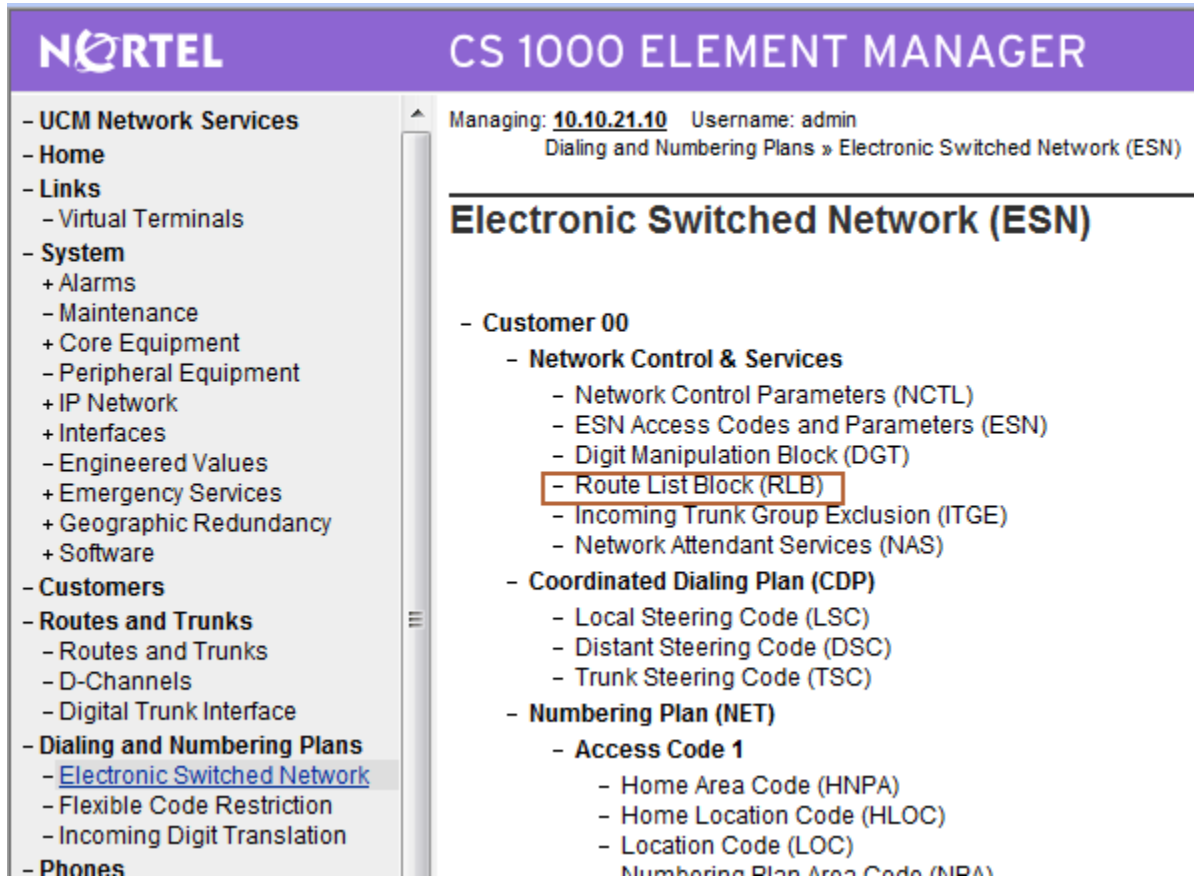
Customer 0, Route 15, Trunk 1 Property Configuration

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="VTRKCODE1000S"/>
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="text" value="Edit"/>

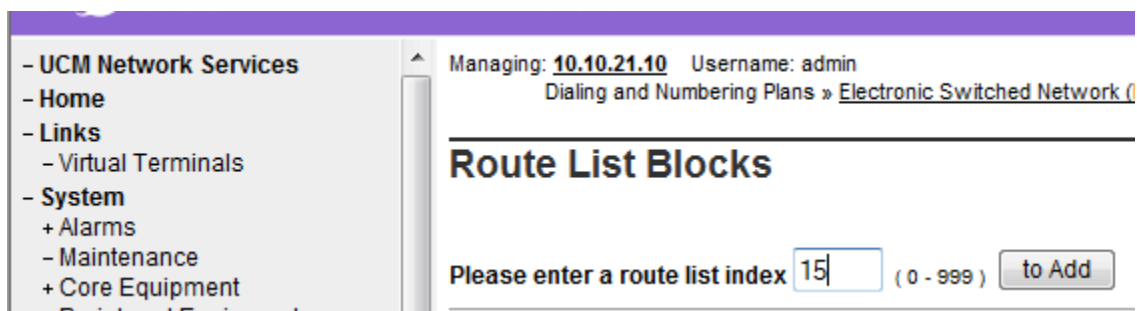
+ Advanced Trunk Configurations

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).

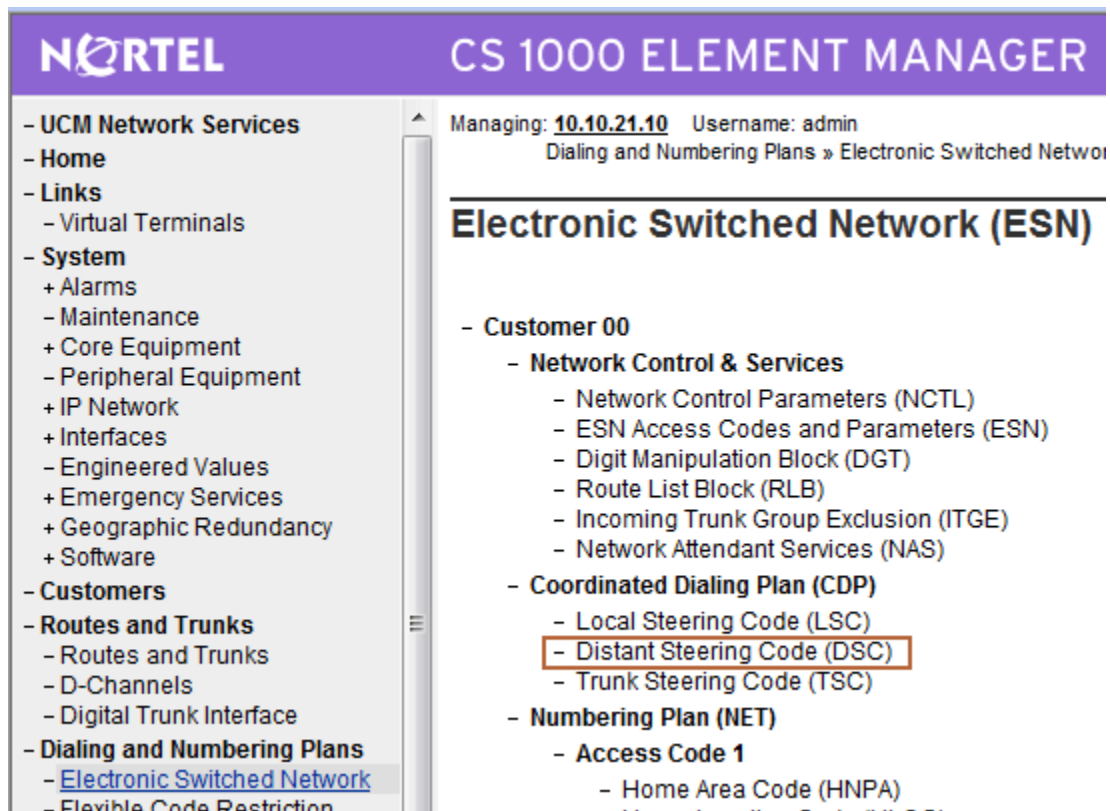
NORTEL CS 1000 ELEMENT MANAGER

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 44444 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 444) as the conference access number is 444444. 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. Save Cancel

Associated Signaling Servers & Cards

Select to add Print | Refresh

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, i.e. **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 of nodes. The node with ID '1000' is highlighted, showing its components and status.

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

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 displays configuration fields for the node, including Node ID, Call Server IP Address, and various IP addresses. The 'Gateway (SIPGw & H323Gw)' application is selected under 'Applications (click to edit configuration)'. Below the configuration fields is a table for '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

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 our 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 'Monitor IP Addresses' checkbox, a text area for IP 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**

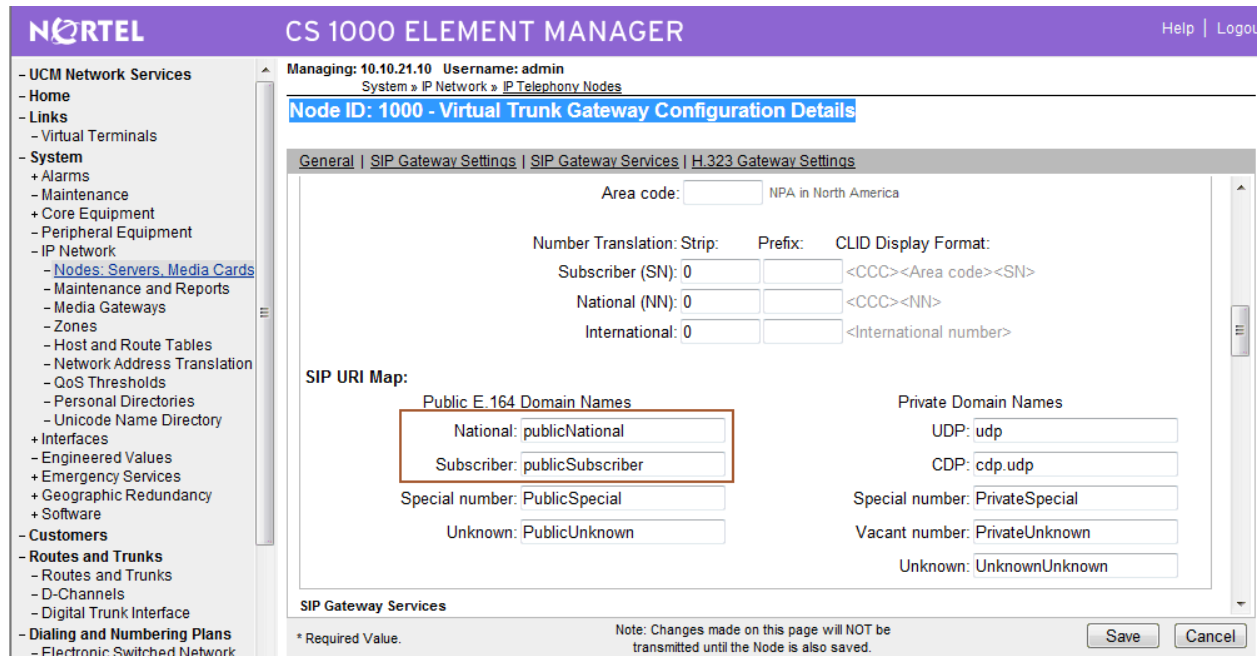
Options: Check **Support registration** and **Primary CDS Proxy**

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, System, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area is titled 'Node ID: 1000 - Virtual Trunk Gateway Configuration Details'. Below this, there are tabs for 'General', 'SIP Gateway Settings', 'SIP Gateway Services', and 'H.323 Gateway Settings'. The 'SIP Gateway Settings' tab is active, showing various configuration fields. A red box highlights the 'Proxy Or Redirect Server' section, which includes fields for Primary TLAN IP Address (10.10.22.11), Secondary TLAN IP Address (0.0.0.0), Port (5060), Transport protocol (TCP), and Options (Support registration and Primary CDS Proxy are checked). The 'TLS Security' section is set to 'Security Disabled'. The 'CLID Presentation' section is empty. At the bottom, there are 'Save' and 'Cancel' buttons, and a note stating 'Changes made on this page will NOT be transmitted until the Node is also saved.'

Scroll down the parameters box to the **SIP URI Map** section. Under **Public E.164 Domain Names**, for

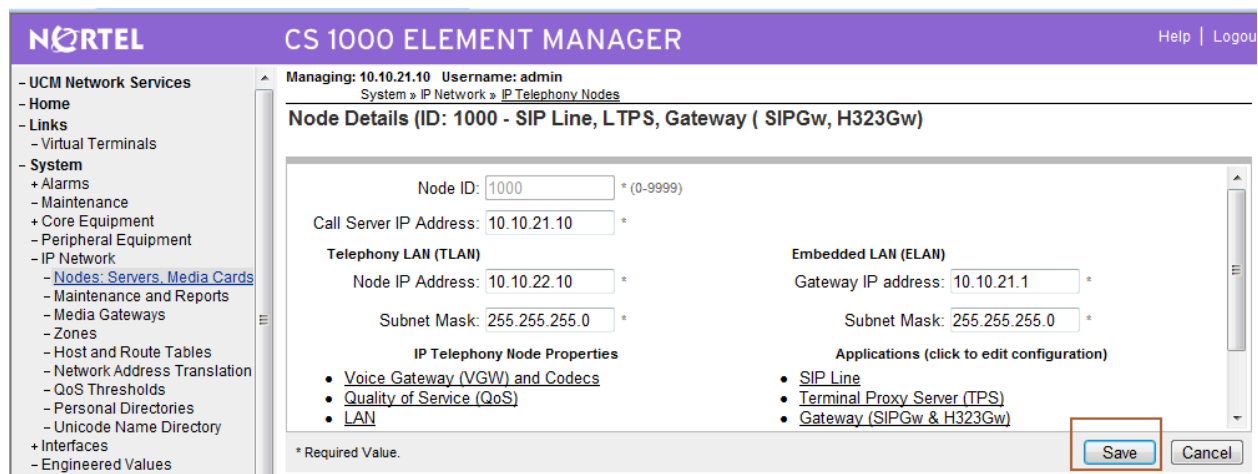
National: Enter **publicNational**
Subscriber: Enter **publicSubscriber**

The remaining fields can be left at their default values. Click on **Save**.



The screenshot shows the 'Node ID: 1000 - Virtual Trunk Gateway Configuration Details' page in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation tree with categories like UCM Network Services, Links, System, Customers, and Routes and Trunks. The main content area is titled 'Node ID: 1000 - Virtual Trunk Gateway Configuration Details' and includes tabs for General, SIP Gateway Settings, SIP Gateway Services, and H.323 Gateway Settings. The 'SIP Gateway Settings' tab is active, showing fields for Area code, Number Translation Strip, Prefix, and CLID Display Format. Below these is the 'SIP URI Map' section, which is highlighted with a red box. It contains two columns of fields: 'Public E.164 Domain Names' and 'Private Domain Names'. The 'Public E.164 Domain Names' column has fields for National (publicNational), Subscriber (publicSubscriber), Special number (PublicSpecial), and Unknown (PublicUnknown). The 'Private Domain Names' column has fields for UDP (udp), CDP (cdp.udp), Special number (PrivateSpecial), Vacant number (PrivateUnknown), and Unknown (UnknownUnknown). At the bottom, there is a 'SIP Gateway Services' section with a note: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' and 'Save' and 'Cancel' buttons.

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



The screenshot shows the 'Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))' page in the Nortel CS 1000 Element Manager. The left sidebar is the same as the previous screenshot. The main content area is titled 'Node Details (ID: 1000 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))' and contains fields for Node ID (1000), 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), and Subnet Mask (255.255.255.0). Below these are two sections: 'IP Telephony Node Properties' with links for Voice Gateway (VGW) and Codecs, Quality of Service (QoS), and LAN; and 'Applications (click to edit configuration)' with links for SIP Line, Terminal Proxy Server (TPS), and Gateway (SIPGw & H323Gw). At the bottom right, the 'Save' button is highlighted with a red box, along with a 'Cancel' button.

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

NORTEL CS 1000 ELEMENT MANAGER

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

Node Saved

Node ID: 1000 has been saved on the call server.

The new configuration must also be transferred to associated servers and media cards.

Transfer Now... You will be given an option to select individual servers, or transfer to all.

Show Nodes You may initiate a transfer manually at a later time.

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

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

Synchronize Configuration Files (Node ID <1000>)

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.

Start Sync **Cancel** **Restart Applications**

<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**.

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

Synchronize Configuration Files (Node ID <1000>)

Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart* of applications on affected server(s) when complete.

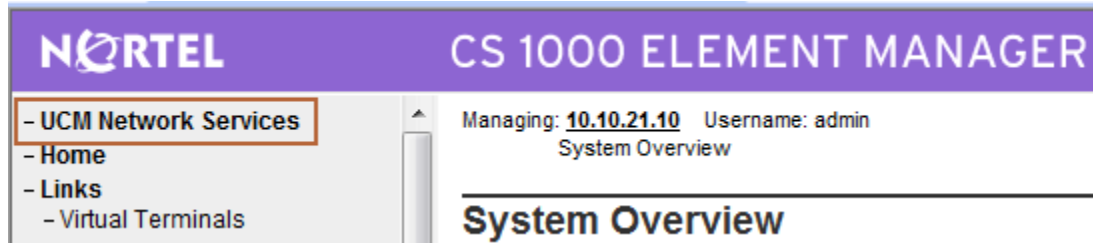
Start Sync **Cancel** **Restart Applications** [Print](#) [Refresh](#)

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

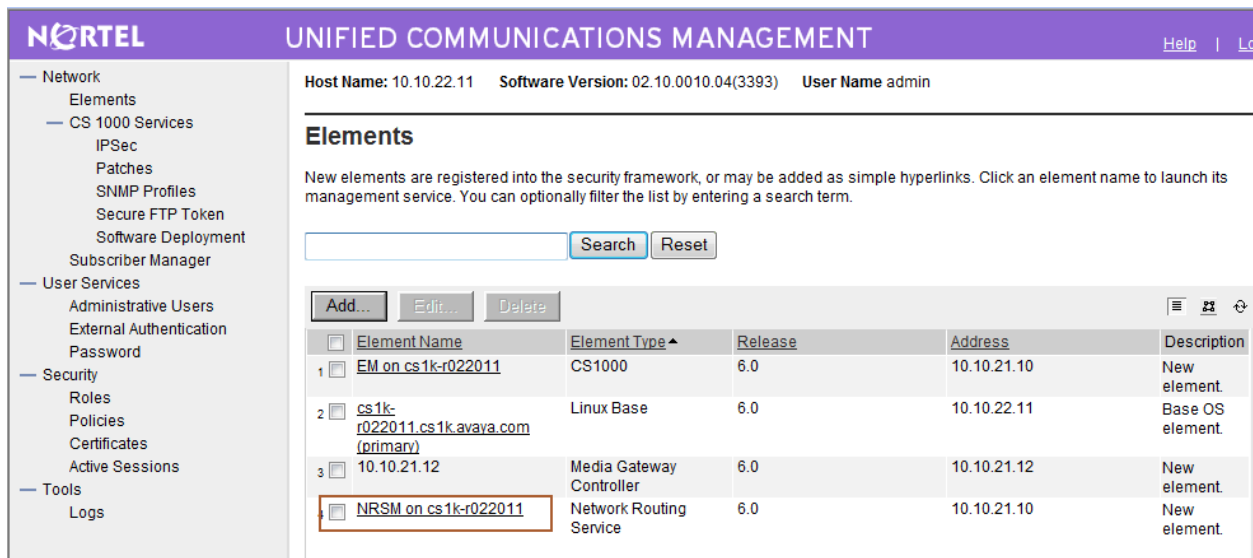
* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application 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.
4	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 Avaya CS1000E and Avaya 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**.

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

Managing: 10.10.21.10
System » [NRS Server](#) » Edit

Edit Server Configuration

SIP Server Settings

Public name for non-trusted networks:

Public number for non-trusted networks:

UDP Transport enabled: ☒

Primary server UDP IP:

Primary server UDP port:

Secondary server UDP IP:

Secondary server UDP port:

TCP Transport enabled: ☒

Primary server TCP IP:

Primary server TCP port:

Secondary server TCP IP:

Secondary server TCP port:

(Note: Any modification of NRS Server configuration would not take effect until you restart all the services.)

* Required value.

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.

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains**
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers

Managing: ☐ Active database ☒ Standby database 10.10.21.10
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)
---------------------	----------------------	----------------------

Add... Delete Refresh

The **Add Service Domain** screen is displayed (not shown). Enter the SIP domain name from **Section 3.7** into the **Domain name** field, and a descriptive text for the **Domain description** field. Click **Save**.

«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

Managing: ☐ Active database ☒ Standby database 10.10.21.10
Numbering Plans » Domains » Service Domains

Edit Service Domain

Domain name: cs1k.avaya.com *

Domain description: Calls to Meeting Exchange

* Required value. Save Cancel

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 8**.

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 don

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.

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 Avaya Meeting Exchange Application Server.

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
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing:
☐ Active database
 ☒ Standby database
 10.10.21.10
[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 (4)
User Endpoints (0)

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

NORTEL**NETWORK ROUTING SERVICE MANAGER**

«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: *

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:

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 Avaya 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**.

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: 10.10.21.50
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 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)

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

Configure two routing entries. The first entry uses the Avaya Meeting Exchange Application Server **Gateway Endpoint** to reach the conference access number (DNIS=44444). The second entry uses the Avaya SIP Signaling Gateway **Gateway Endpoint** to reach Avaya endpoints in the 3xxx extension range. 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 Avaya Meeting Exchange Application Server. Click on **Add**.

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
 - Backup
 - Restore
 - GK/NRS Data upgrade

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

Search for Routing Entries [Hide](#)

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

DN Prefix: * DN Type: All DN Types

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

Endpoint Name: mx

Results per page: 50 [Search](#)

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

[Add...](#) [Copy...](#) [Move...](#) [Import...](#) [Export...](#) [Routing test...](#) [Delete](#) [Refresh](#)

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)**

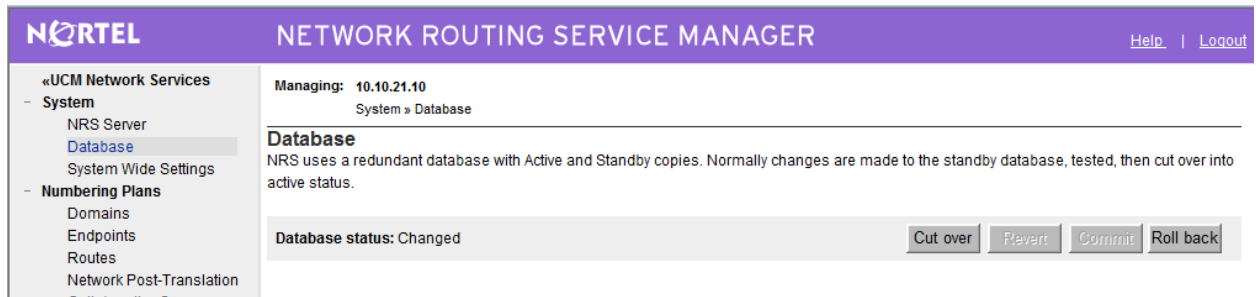
DN prefix: Dialed prefix digits to match on, in this case **44444**. **44444** is the conference access number.

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

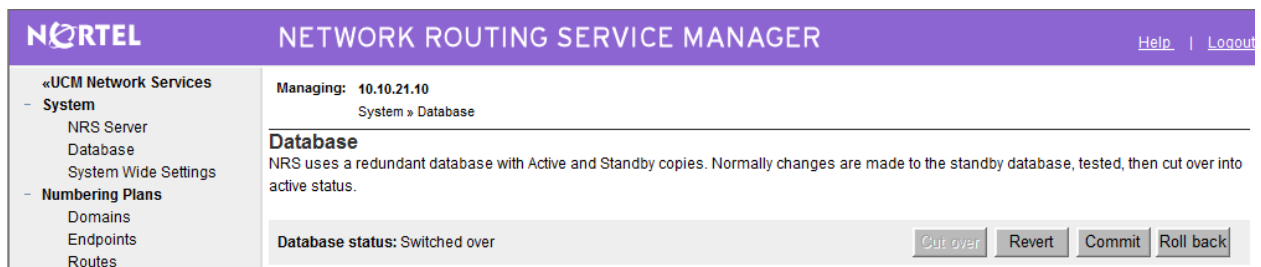
Repeat the same procedures to add a routing entry to reach the CS1000E endpoints with extension digits 3xxx behind the Avaya SIP Signaling Gateway Endpoint.

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 SIP trunks between Avaya Meeting Exchange Enterprise Application Server and Avaya Communication Server 1000E. 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 Avaya 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 Avaya 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 the SIP Proxy Server (SPS) component of the Avaya Communication Server 1000E 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**
 - Add the following lines to set the Min-SE timer to 900 seconds in SIP INVITE messages from the Meeting Exchange:
sessionRefreshTimerValue= 900
minSETimerValue= 900

Note: The values for the sessionRefreshTimerValue and the minSETimerValue are defined in seconds and should be provisioned to be greater than or equal to the value used by the CS1000 SIP Proxy Server. This setting is necessary to enable Dial-Out from the Meeting Exchange into the endpoint registered with CS1000.

```
# 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
MaxChannelCount=3200

# SIP settings
sessionRefreshTimer=900
minSETimerValue=900
```

4.2. Configure Dialout

The FQDN of the CS1000 SIP Proxy Server 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      10.10.22.11
```

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

Add the following line to the file to route outbound calls from the Avaya 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 Avaya Meeting Exchange Enterprise Application Server as follows:

- Add 44444 as a new DNIS entry using the following command:

```
[mx6200-a ~]# cbutil add 44444 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

44444	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:44444;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.

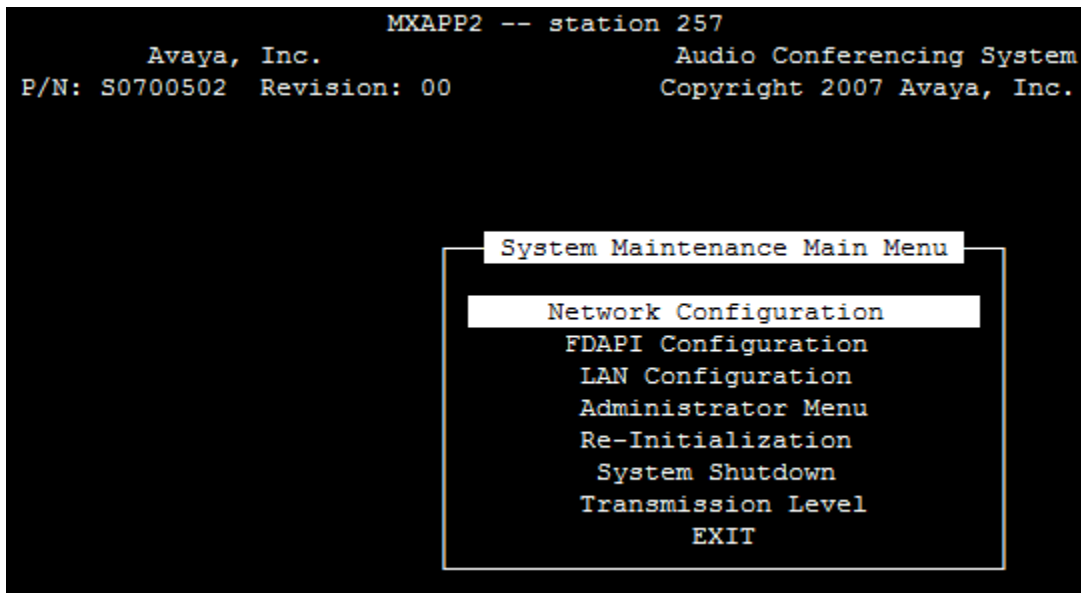
Edit the /usr/ipcb/config/UriToTelnum.tab file with a text editor. Enter the following two rules at the top of the list:

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

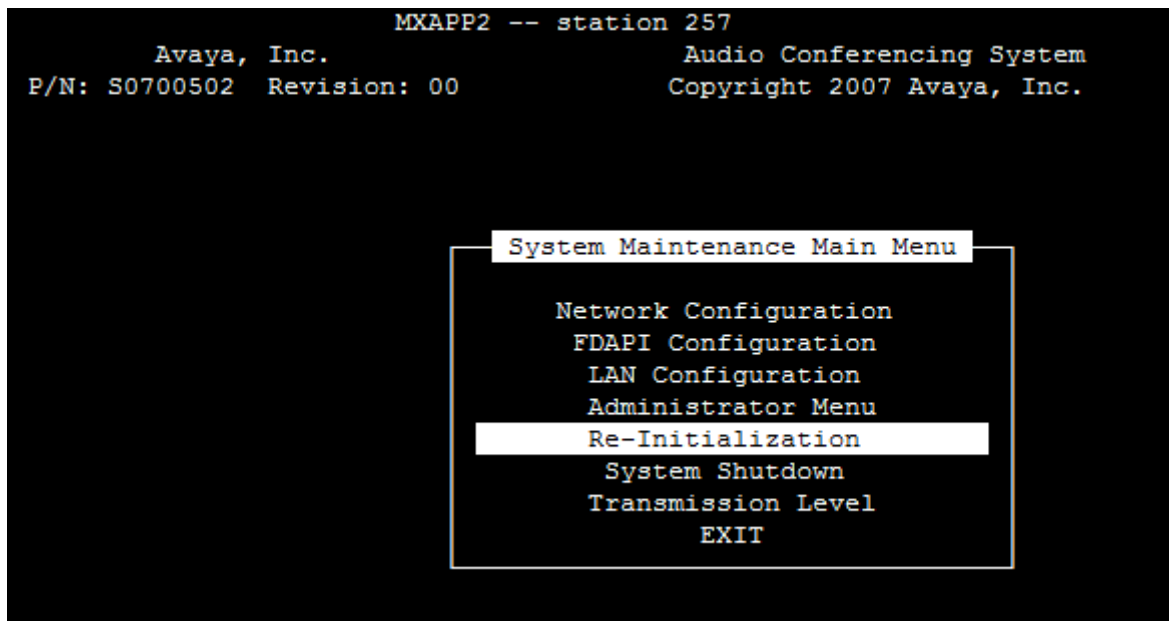
4.5. Restart the Conference Bridge

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. 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.

5.1. 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 Help | Logout

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	

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 Help | Logout

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
  
```

5.2. 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. 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.

7. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000E (formerly known as Nortel CS1000E) can interoperate with Avaya Meeting Exchange Enterprise Edition 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** parameters from the incoming INVITE on the Avaya Meeting Exchange Application Server.

8. 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>

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