



Configuring SIP Trunks among Avaya Aura™ Session Manager 5.2, Avaya Aura™ Communication Manager 5.2.1, and Nortel Communication Server 1000 6.0 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Communication Manager and Nortel Communication Server 1000 using SIP trunks.

For the sample configuration, Avaya Aura™ Session Manager runs on an Avaya S8510 Server, Avaya Aura™ Communication Manager runs on an Avaya S8720 Server with Avaya G650 Media Gateway, and Nortel Communication Server 1000 runs on Nortel Communication Server 1000e. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura™ Communication Manager 5.2.1 and later.

1 Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Communication Manager and Nortel Communication Server 1000 using SIP trunks.

As shown in **Figure 1**, the Avaya 9630 IP Telephone (H.323) and 6408D+ Digital Telephone are supported by Communication Manager, which serves as an *Access Element* within the Session Manager architecture. The Nortel i2004 H.323 Telephone and 3904 Digital Telephone are supported by Nortel Communication Server 1000. SIP trunks are used to connect these two systems to Session Manager, using its SM-100 (Security Module) network interface. All inter-system calls are carried over these SIP trunks. Session Manager can support flexible inter-system call routing based on dialed number, calling number and system location, and can also provide protocol adaptation to allow multi-vendor systems to interoperate. It is managed by a separate Avaya Aura™ System Manager, which can manage multiple Session Managers by communicating with their management network interfaces. Session Manager also supports SIP telephones, but this configuration is not addressed in these application notes.¹

For the sample configuration, Session Manager runs on an Avaya S8510 Server, Communication Manager runs on Avaya S8720 Servers with Avaya G650 Media Gateway, and Nortel Communication Server 1000 runs on Nortel Communication Server 1000e. The results in these Application Notes should be applicable to other Avaya Aura™ servers and Media Gateways.

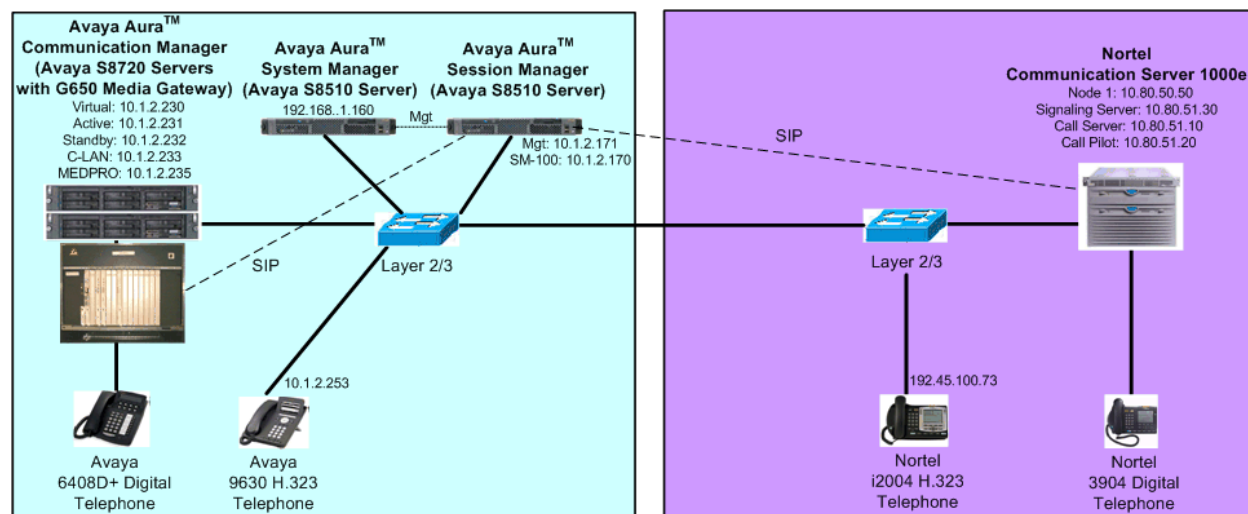


Figure 1 – Sample Configuration

Communication Manager Uniform Dial Plan (UDP) and Nortel Communication Server 1000 Coordinated Dial Plan (CDP) features are used to implement extension-extension dialing

¹ See Reference [7] for application notes on configuring Session Manager and Communication Manager as a Feature Server to support SIP telephones.

between systems. Unique extension ranges are associated with Communication Manager (3xxxx) and Nortel Communication Server 1000 (777xxxx).

These Application Notes will focus on configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones will not be described except where it affects specific feature operations, e.g., telephone name/number display (see the appropriate documentation listed in **Section 8**).

2 Equipment and Software Validated

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

Hardware Component	Software Version
Avaya S8510 Servers	Avaya Aura™ Session Manager R5.2 Load 5.2.0.0.520011 (GA)
	Avaya Aura™ System Manager R5.2 Load 5.2.0.0.520008 (GA)
	VSP patch 1.1.0.4.8
Avaya S8720 Servers with G650 Media Gateway	Avaya Aura™ Communication Manager 5.2.1, Load 16.4 (GA)
Avaya 9630 IP Telephone (H.323)	2.0
Avaya 6408D+ Digital Telephone	-
Nortel Communication Server 1000e	Release 600R, Version 4121
Nortel 3904 Digital Telephone	NA
Nortel I2004 H.323 Telephone	C502B41

3 Configure Avaya Aura™ Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify Avaya Aura™ Communication Manager license
- Configure system parameters features
- Configure IP node names
- Configure IP interface for C-LAN
- Configure IP codec set and network region
- Configure SIP signaling group and trunk group
- Configure route pattern
- Configure location and public unknown numbering
- Configure uniform dial plan and AAR analysis
- Save Translations

Some administration screens have been abbreviated for clarity.

3.1 Verify Avaya Aura™ Communication Manager License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

display system-parameters customer-options		Page	2 of 10
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		800	200
Maximum Concurrently Registered IP Stations:		18000	2
Maximum Administered Remote Office Trunks:		0	0
Maximum Concurrently Registered Remote Office Stations:		0	0
Maximum Concurrently Registered IP eCons:		0	0
Max Concur Registered Unauthenticated H.323 Stations:		0	0
Maximum Video Capable H.323 Stations:		0	0
Maximum Video Capable IP Softphones:		0	0
Maximum Administered SIP Trunks:		800	57

3.2 Configure System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers. This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch. For simplicity, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 8** for more details. Submit the change.

change system-parameters features		Page	1 of 18
FEATURE-RELATED SYSTEM PARAMETERS			
Self Station Display Enabled? y			
Trunk-to-Trunk Transfer: all			
Automatic Callback with Called Party Queuing? n			
Automatic Callback - No Answer Timeout Interval (rings): 3			
Call Park Timeout Interval (minutes): 10			
Off-Premises Tone Detect Timeout Interval (seconds): 20			
AAR/ARS Dial Tone Required? y			

3.3 Configure IP Node Names

Use the “change node-names ip” command to add entries for the C-LAN that will be used for connectivity, its default gateway, and Session Manager. In this case, “clan1” and “10.1.2.233” are entered as **Name** and **IP Address** for the C-LAN, “asm” and “10.1.2.170” are entered for the Session Manager Security Module (SM-100) interface, and “Gateway001” and “10.1.2.1” are entered for the default gateway. Note that “Gateway001” will be used in the form used to configure the IP interface for the C-LAN (see **Section 3.4**). The actual node names and IP addresses may vary. Submit these changes.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
clan1	10.1.2.233	
asm	10.1.2.170	
Gateway001	10.1.2.1	

3.4 Configure IP Interface for C-LAN

Add the C-LAN to the system configuration using the “add ip-interface 1a02” command. The actual slot number may vary. In this case, “1a02” is used as the **Slot** number. Enter the C-LAN node name assigned from **Section 3.3** into the **Node Name** field.

Enter proper values for the **Subnet Mask** and **Gateway Node Name** fields. In this case, “/24” and “Gateway001” are used to correspond to the network configuration in these Application Notes. Set the **Enable Interface** and **Allow H.323 Endpoints** fields to “y”. Default values may be used in the remaining fields. Submit these changes.

change ip-interface 1a02		Page 1 of 3
IP INTERFACES		
Type: C-LAN		
Slot: 01A02	Target socket load and Warning level: 400	
Code/Suffix: TN799 D	Receive Buffer TCP Window Size: 8320	
Enable Interface? y	Allow H.323 Endpoints? y	
VLAN: n	Allow H.248 Gateways? y	
Network Region: 1	Gatekeeper Priority: 5	
IPV4 PARAMETERS		
Node Name: clan1		
Subnet Mask: /24		
Gateway Node Name: Gateway001		
Ethernet Link: 1		
Network uses 1's for Broadcast Addresses? y		

3.5 Configure IP Codec Set and Network Region

Configure the IP codec set to use for calls to the Nortel Communication Server 1000. Use the “change ip-codec-set n” command, where “n” is an existing codec set number to be used for interoperability. Enter the desired audio codec type in the **Audio Codec** field. Retain the default values for the remaining fields and submit these changes.

In addition to the “G.711MU” codec shown below, G.729 has also been verified to be interoperable with Nortel Communication Server 1000 via SIP trunks.

change ip-codec-set 1				Page	1 of	2
IP Codec Set						
Codec Set: 1						
Audio	Silence	Frames	Packet			
Codec	Suppression	Per Pkt	Size(ms)			
1: G.711MU	n	2	20			
2:						
3:						

In the test configuration, network region “1” was used for calls to the Nortel Communication Server 1000 via Session Manager. Use the “change ip-network-region 1” command to configure this network region. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise network (See **Section 4.1**). This value is used to populate the SIP domain in the From header of SIP INVITE messages for outbound calls. It also must match the SIP domain in the request URI of incoming INVITES from other systems. Enter a descriptive **Name**. For the **Codec Set** field, enter the corresponding audio codec set configured above in this section. Enable the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio**. These settings will enable direct media between Avaya IP telephones and the far end. Retain the default values for the remaining fields, and submit these changes.

change ip-network-region 1				Page	1 of	19
IP NETWORK REGION						
Region: 1						
Location: Authoritative Domain: avaya.com						
Name: ASM to Nortel						
MEDIA PARAMETERS			Intra-region IP-IP Direct Audio: yes			
Codec Set: 1			Inter-region IP-IP Direct Audio: yes			
UDP Port Min: 2048			IP Audio Hairpinning? n			
UDP Port Max: 10001						
DIFFSERV/TOS PARAMETERS			RTCP Reporting Enabled? y			
Call Control PHB Value: 46			RTCP MONITOR SERVER PARAMETERS			
Audio PHB Value: 46			Use Default Server Parameters? y			
Video PHB Value: 26						

3.6 Configure SIP Signaling Group and Trunk Group

3.6.1 SIP Signaling Group

In the test configuration, trunk group “32” and signaling group “32” were used to reach Session Manager. Use the “add signaling-group n” command, where “n” is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** “clan1” C-LAN node name from **Section 3.3**.
- **Far-end Node Name:** “asm” Session Manager node name from **Section 3.3**.
- **Near-end Listen Port:** “5061”
- **Far-end Listen Port:** “5061”
- **Far-end Network Region:** Network region number “1” from **Section 3.5**.
- **Far-end Domain:** “avaya.com” SIP domain name from **Section 4.1**.

add signaling-group 32		Page 1 of 1
SIGNALING GROUP		
Group Number: 32	Group Type: sip	
	Transport Method: tls	
IMS Enabled? n		
Near-end Node Name: clan1	Far-end Node Name: asm	
Near-end Listen Port: 5061	Far-end Listen Port: 5061	
Far-end Domain: avaya.com	Far-end Network Region: 1	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early Media? n	
	Alternate Route Timer(sec): 6	

3.6.2 SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”
- **Signaling Group** The signaling group number defined in the previous section
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager (must be within the limits of the total trunks configure in **Section 3.1**).

```
add trunk-group 32                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 32                Group Type: sip          CDR Reports: y
  Group Name: To ASM              COR: 1                TN: 1          TAC: 132
    Direction: two-way          Outgoing Display? n
    Dial Access? n
    Queue Length: 0
  Service Type: tie              Auth Code? n
                                     Signaling Group: 32
                                     Number of Members: 4
```

Navigate to **Page 3**, and enter “public” for the **Numbering Format** field as shown below. Use default values for all other fields. Submit these changes.

```
add trunk-group 32                                     Page 3 of 21
TRUNK FEATURES
    ACA Assignment? n          Measured: none
                                Maintenance Tests? y
                                Numbering Format: public
                                UUI Treatment: service-provider
                                Replace Restricted Numbers? n
                                Replace Unavailable Numbers? n
```


3.7 Configure Route Pattern

Configure a route pattern to correspond to the newly added SIP trunk group. Use the “change route-pattern n” command, where “n” is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name.
- **Grp No:** The trunk group number from **Section 3.6.2**.
- **FRL:** Enter a level that allows access to this trunk, with 0 being least restrictive.

change route-pattern 32															Page 1 of 3		
Pattern Number: 32 Pattern Name: To ASM																	
SCCAN? n Secure SIP? n																	
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted								DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits								QSIG		
								Dgts								Intw	
1:	32	0													n	user	
2:																n	user
3:																n	user
4:																n	user
5:																n	user
6:																n	user
		BCC VALUE		TSC	CA-TSC		ITC BCIE		Service/Feature		PARM	No.	Numbering		LAR		
		0	1	2	M	4	W	Request				Dgts Format					
														Subaddress			
1:	y	y	y	y	y	n	n	rest							none		

3.8 Configure Location and Public Unknown Numbering

Use the “change locations” command to specify the SIP route pattern to be used as a default SIP route for the location corresponding to the Main site. In this way, calls to non-numeric users or unknown domains will still be routed to Session Manager. Add an entry for the Main site if one does not exist already, enter the following values for the specified fields, and retain default values for the remaining fields. Submit these changes.

- **Name:** A descriptive name to denote the Main site.
- **Timezone:** An appropriate timezone offset.
- **Rule:** An appropriate daylight savings rule.
- **Proxy Sel. Rte. Pat.:** The Avaya route pattern number from **Section 3.7**.

change locations					Page 1 of 1	
LOCATIONS						
ARS Prefix 1 Required For 10-Digit NANP Calls? y						
Loc No	Name	Timezone Offset	Rule	NPA	Proxy Sel Rte	Pat
1:	Main	+ 00:00	0		32	

Use the “change public-unknown-numbering 0” command, to define the calling party number to be sent to Nortel Communication Server 1000. Add an entry for the trunk group defined in **Section 3.6.2** to reach Nortel endpoints. In the example shown below, all calls originating from a 5-digit extension beginning with 3 and routed to trunk group 32 will result in a 5-digit calling number. The calling party number will be in the SIP “From” header. Submit these changes.

change public-unknown-numbering 0					Page 1 of 2	
NUMBERING - PUBLIC/UNKNOWN FORMAT						
				Total		
Ext	Ext	Trk	CPN	CPN		
Len	Code	Grp(s)	Prefix	Len		
				Total Administered: 2		
5	3			Maximum Entries: 9999		
				5		

3.9 Administer Uniform Dial Plan and AAR Analysis

This section provides sample Automatic Alternate Routing (AAR) used for routing calls with dialed digits 777xxxx to Nortel Communication Server 1000. Note that other methods of routing may be used. Use the “change uniform-dialplan 0” command, and add an entry to specify use of AAR for routing of digits 777xxxx. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Matching Pattern:** Dialed prefix digits to match on, in this case “777”.
- **Len:** Length of the full dialed number.
- **Del:** Number of digits to delete.
- **Net:** “aar”

change uniform-dialplan 0						Page 1 of 2	
UNIFORM DIAL PLAN TABLE							
						Percent Full: 0	
Matching				Insert		Node	
Pattern	Len	Del		Digits	Net	Conv	Num
777	7	0			aar	n	

Use the “change aar analysis 0” command, and add an entry to specify how to route the calls to 777xxxx. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case “53”.
- **Total Min:** Minimum number of digits.
- **Total Max:** Maximum number of digits.
- **Route Pattern:** The route pattern number from **Section 3.7**.
- **Call Type:** “aar”

change aar analysis 0						Page 1 of 2	
AAR DIGIT ANALYSIS TABLE							
Location: all						Percent Full: 1	
Dialed	Total		Route	Call	Node	ANI	
String	Min	Max	Pattern	Type	Num	Reqd	
777	7	7	32	aar		n	

3.10 Save Translations


Configuration of Communication Manager is complete. Use the save Translations command to save these changes.

4 Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to the SIP telephony systems and Session Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager.
- Local host name resolution entries corresponding to fully qualified domain names (FQDN's) referenced in the previous steps.

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “http://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The menu shown below is displayed. Expand the **Network Routing Policy** Link on the left side as shown. The sub-menus displayed in the left column below will be used to configure all but the last two of the above items (**Sections 4.1** through **4.6**).

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 9:19 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▼ 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

Introduction to Network Routing Policy (NRP)

Network Routing Policy consists of several NRP applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the NRP applications (that means the overall NRP workflow) to configure your network configuration is as follows:

Step 1: Create "Domains" of type SIP (other NRP applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Step 6: Create "Time Ranges"

4.1 Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Select **SIP Domains** on the left and click the **New** button (not shown) on the right. Fill in the following:

- **Name:** The authoritative domain name (e.g., “avaya.com”)
- **Notes:** Descriptive text (optional).

Click **Commit**.

The screenshot shows the Avaya Aura System Manager 5.2 web interface. The top navigation bar includes the Avaya logo, the product name "Avaya Aura™ System Manager 5.2", and a user status message: "Welcome, admin Last Logged on at Dec. 01, 2009 10:10 AM" with links for "Help" and "Log off". Below this is a red breadcrumb trail: "Home / Network Routing Policy / SIP Domains". On the left is a sidebar menu with categories: "Asset Management", "Communication System Management", "Monitoring", "User Management", and "Network Routing Policy". Under "Network Routing Policy", several sub-items are listed: "Adaptations", "Dial Patterns", "Entity Links", "Locations", "Regular Expressions", "Routing Policies", and "SIP Domains", which is highlighted with a red circle. The main content area is titled "Domain Management" and contains a table with one item, "avaya.com". The table has columns for "Name", "Type", "Default", and "Notes". The "Name" column contains "avaya.com" with a red asterisk indicating it is required. The "Type" column shows "sip" with a dropdown arrow. The "Default" column has an unchecked checkbox. The "Notes" column is empty. Above the table, there is a "Filter: Enable" link. Below the table, there is a red asterisk and the text "Input Required". At the top right of the main content area, there are "Commit" and "Cancel" buttons. At the bottom right, there are also "Commit" and "Cancel" buttons.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Dec. 01, 2009 10:10 AM [Help](#) | [Log off](#)

Home / Network Routing Policy / SIP Domains

Domain Management [Commit](#) [Cancel](#)

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

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

* Input Required [Commit](#) [Cancel](#)

4.2 Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. Under *General*, enter:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

The remaining fields under *General* can be filled in to specify bandwidth management parameters between Session Manager and this location. These were not used in the sample configuration, and reflect default values. Note also that although not implemented in the sample configuration, routing policies can be defined based on location.

Under *Location Pattern*:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the Basking Ridge location, which includes Communication Manager and Session Manager in the 10.1.2 subnet. Click **Commit** to save the Location definition.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM [Help](#) | [Log off](#)

Home / Network Routing Policy / Locations / **Location Details**

Location Details Commit Cancel

General

* **Name:**

Notes:

Managed Bandwidth:

* **Average Bandwidth per Call:**

* **Time to Live (secs):**

Location Pattern

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

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.1.2.*	<input type="text"/>

Select : [All](#), [None](#) (0 of 1 Selected)

Shortcuts * Input Required Commit Cancel

The following screen shows the addition of a second location based on the subnet used by Nortel Communication Server 1000.

AVAYA Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy / Locations / Location Details

Location Details Commit Cancel

General

* Name:

Notes:

Managed Bandwidth:

* Average Bandwidth per Call: kbit/sec

* Time to Live (secs):

Location Pattern

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.80.50.*	Nortel CS1000e

Select : All, None (0 of 1 Selected)

Session Manager Commit Cancel

* Input Required

The fields under *General* can be filled in to specify bandwidth management parameters between Session Manager and this location. These were not used in the sample configuration, and reflect default values. Note also that although not implemented in the sample configuration, routing policies can be defined based on location.

4.3 Add SIP Entities

A SIP Entity must be added for Avaya Aura™ Session Manager and for each SIP telephony system supported by it using SIP trunks: the C-LAN board in the Avaya G650 Media Gateway, and the Nortel Communication Server 1000. Select **SIP Entities** on the left and click on the **New** button (not shown) on the right. Under *General*, fill in:

- **Name:** A descriptive name.
- **FQDN or IP Address:** FQDN or IP address of the Session Manager or the signaling interface on the telephony system.
- **Type:** “Session Manager” for Session Manager, “CM” for Communication Manager, “Other” for the Nortel Communication Server 1000.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise (e.g., “avaya.com”).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The following screen shows addition of Session Manager. The IP address of the SM-100 Security Module is entered for **FQDN or IP Address**. Two *Port* entries are added. TCP port 5060 is used for communicating with the Nortel Communication Server 1000, and TLS port 5061 is used for communication with Communication Manager.

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

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

Asset Management
Communication System Management
Monitoring
User Management
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

Shortcuts
Change Password
Help for SIP Entity Details fields
Help for Committing configuration changes

SIP Entity Details

Commit Cancel

General

* Name:
SM1

* FQDN or IP Address:
10.1.2.170

Type:
Session Manager

Notes:

Location:
BaskingRidge

Outbound Proxy:

Time Zone:
America/New_York

Credential name:

SIP Link Monitoring

SIP Link Monitoring:
Use Session Manager Configuration

Entity Links

Entity Links can be modified after SIP Entity is committed.

Port
Add Remove

2 Items | Refresh
Filter: Enable

	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5061	TLS	avaya.com	
<input type="checkbox"/>	5060	TCP	avaya.com	

Select : All, None (0 of 2 Selected)

* Input Required

Commit Cancel

The following screen shows the results of adding Communication Manager. In this case, **FQDN or IP Address** is the Fully Qualified Domain Name (FQDN) of the C-LAN board in the Avaya G650 Media gateway. Note that although not shown in the sample configuration, definition of multiple IP addresses (e.g., C-LANs) for the same FQDN (see **Section 4.8**) will cause Session Manager to load balance call traffic among those addresses.

AVAYAAvaya Aura™ System Manager 5.2Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

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

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▼ 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

SIP Entity Details

CommitCancel

General

* Name: CallCenter

* FQDN or IP Address: callcenter.avaya.com

Type: CM

Notes:

Adaptation:

Location: BaskingRidge

Time Zone: America/New_York

Override Port & Transport with DNS SRV:

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

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

The following screen shows addition of Nortel Communication Server 1000. The IP address used is that of the “Voice LAN (TLAN) Node IP address” (See **Figure 2** in **Section 5.2**).

AVAYAAvaya Aura™ System Manager 5.2Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

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

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▼ 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

SIP Entity Details

CommitCancel

General

* Name: Denver Nortel CS1000e

* FQDN or IP Address: 10.80.50.50

Type: Other

Notes:

Adaptation:

Location: Westminster

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: Use Session Manager Configuration

SIP Link Monitoring

4.4 Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 4.3** will be denied.*

Click **Commit** to save each Entity Link definition. The following screens illustrate adding the Entity Links for Communication Manager and the Nortel Communication Server 1000.

The screenshot shows the Avaya Aura System Manager 5.2 interface. The left sidebar has a menu with 'Entity Links' highlighted. The main area is titled 'Entity Links' and contains a table with one row. The table columns are: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The row contains: * Call Center, * SM1, TLS, * 5061, * CallCenter, * 5061, ☒, and CLAN .233. There are 'Commit' and 'Cancel' buttons at the top right and bottom right. A 'Filter: Enable' link is also present.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* Call Center	* SM1	TLS	* 5061	* CallCenter	* 5061	<input checked="" type="checkbox"/>	CLAN .233

The screenshot shows the Avaya Aura System Manager 5.2 interface. The left sidebar has a menu with 'Entity Links' highlighted. The main area is titled 'Entity Links' and contains a table with one row. The table columns are: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The row contains: * Nortel CS1000e, * SM1, TCP, * 5060, * Denver Nortel CS1000e, * 5060, ☒, and an empty field. There are 'Commit' and 'Cancel' buttons at the top right and bottom right. A 'Filter: Enable' link is also present.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* Nortel CS1000e	* SM1	TCP	* 5060	* Denver Nortel CS1000e	* 5060	<input checked="" type="checkbox"/>	

4.5 Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 4.3**. Two routing policies must be added for Communication Manager and the Nortel Communication Server 1000. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity 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** to save each Routing Policy definition. The following screens show the Routing Policies for Communication Manager and the Nortel Communication Server 1000.

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy / Routing Policies / Routing Policy Details

Asset Management
Communication System Management
Monitoring
User Management
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

Routing Policy Details

Commit Cancel

General

* Name: Call Center

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CallCenter	callcenter.avaya.com	CM	

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 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)

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy / Routing Policies / Routing Policy Details

Asset Management
Communication System Management
Monitoring
User Management
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

Routing Policy Details

Commit Cancel

General

* Name: Denver CS1000e

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Denver Nortel CS1000e	10.80.50.50	Other	

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 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)

4.6 Add Dial Patterns

Define dial patterns to direct calls to the appropriate SIP Entity. Calls to 5-digit extensions beginning with “3” should be routed to Communication Manager. Calls to 7-digit extensions beginning with “777” should be routed to the Nortel Communication Server 1000. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following, as shown in the screens below:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min:** Minimum length of dialed number.
- **Max:** Maximum length of dialed number.
- **SIP Domain:** SIP domain specified in **Section 4.1**
- **Notes:** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location (or “ALL”) and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save each dial pattern. The following screens show the resulting two dial pattern definitions.

AVAYA

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy / Dial Patterns / Dial Pattern Details

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▼ 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

Dial Pattern Details

Commit

Cancel

General

* Pattern:

3

* Min:

5

* Max:

5

Emergency Call:

☐

SIP Domain:

avaya.com

Notes:

Call Center ACM CLAN1

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	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	Call Center	0	<input type="checkbox"/>	CallCenter	

Select : [All](#), [None](#) (0 of 1 Selected)

The following screenshot shows the dial pattern for Nortel Communication Server 1000.

AVAYA

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) | [Log off](#)

Home / Network Routing Policy / Dial Patterns / **Dial Pattern Details**

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▼ 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

Dial Pattern Details

Commit

Cancel

General

* Pattern:

777

* Min:

7

* Max:

7

Emergency Call:

☐

SIP Domain:

-ALL-

Notes:

IP phones on Denver CS1000e

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	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	Denver CS1000e	0	<input type="checkbox"/>	Denver Nortel CS1000e	

Select : [All](#), [None](#) (0 of 1 Selected)

Shortcuts

4.7 Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:


Under *General*:

- **SIP Entity Name:** Select the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:**
Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Save** (not shown) to add this Session Manager. The screen below shows the resulting Session Manager definition.

 Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM
[Help](#) [Log off](#)

Home / Session Manager / Session Manager Administration / **View Session Manager**

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▶ Network Routing Policy

▶ Security

▶ Applications

▶ Settings

▼ **Session Manager**

▶ Session Manager Administration

▶ Network Configuration

▶ Device and Location Configuration

▶ Application Configuration

▶ System Status

▶ System Tools

Shortcuts

[Change Password](#)

[Help for Session Manager Administration](#)

View Session Manager

[Return](#)

General | Security Module | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
[Expand All](#) | [Collapse All](#)

General

SIP Entity Name | SM1

Description | Session Mgr 1

Management Access Point Host Name/IP | 10.1.2.171

Direct Routing to Endpoints | Enable

Security Module

SIP Entity IP Address | 10.1.2.170

Network Mask | 255.255.255.0

Default Gateway | 10.1.2.1

Call Control PHB | 46

QOS Priority | 6

Speed & Duplex | Auto

VLAN ID

4.8 Define Local Host Names

The host name (FQDN) used for Communication Manager in **Section 4.3** must be defined. To do so, Select **Session Manager → Network Configuration → Local Host Name Resolution** on the left. Click **New** and enter the following:

- **Host Name:** The FQDN used for the host
- **IP Address:** IP address of the host's network interface
- **Port:** Port number to which SIP requests are sent
- **Transport:** Transport to be used for SIP requests

Defaults can be used for the remaining fields. The **Priority** and **Weight** fields are used when multiple IP addresses are defined for the same host. The following screen shows the host name resolution entry used in the sample configuration (circled).

AVAYA Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 01, 2009 10:10 AM [Help](#) [Log off](#)

Home / Session Manager / Network Configuration / Local Host Name Resolution

Local Host Name Resolution

This page allows you to add, edit, or remove local host name entries. Host name entries on this page will override information provided by DNS.

Local Host Name Entries

[New](#) [Edit](#) [Delete](#) [More Actions](#)

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

	Host Name	IP Address	Port	Priority	Weight	Transport
<input type="checkbox"/>	allanc-s8300-g350	10.32.2.80	5060	100	100	TCP
<input type="checkbox"/>	callcenter.avaya.com	10.1.2.233	5060	100	100	TCP
<input type="checkbox"/>	m1000.avaya.com	10.1.2.100	5060	100	100	TCP
<input type="checkbox"/>	retailmmBR2.avaya.com	30.1.1.73	5060	100	100	TCP
<input type="checkbox"/>	retailmmHQ.avaya.com	30.1.1.71	5060	100	100	TCP

Select : [All](#), [None](#) (0 of 5 Selected)

5 Configure Nortel Communication Server 1000

Nortel Communication Server 1000 uses the Signaling Server to provide SIP and H.323 signaling interfaces to IP networks. The Signaling Server communicates with a Call Server over a private Ethernet interface. There can be one or more Signaling Servers supported per Nortel Communication Server 1000 system. The applications that can run on the Signaling Server include the following:

- **SIP Gateway** Provides SIP signaling for IP networks.
- **Network Routing Service (NRS)** Provides SIP Redirect & Registrar service components.
- **NRS Manager** Provides web interface for NRS management.
- **Element Manager** Provides web interface for system administrative tasks.

The Nortel Communication Server 1000 in the interoperability test configuration contained one Signaling Server and Call Server co-resident on the same CS1000e server. The Element Manager was used to configure system resources such as SIP virtual routes and trunks, and the NRS Manager was used to configure the routing for SIP devices. These Application Notes assume that the basic configuration of the Signaling Server with the Call Server is in place and the configuration will not be described.

Furthermore, these Application Notes used the Coordinated Dial Plan (CDP) feature to route calls from the Nortel Communication Server 1000, over the SIP trunks to Session Manager to reach endpoints on Communication Manager. The CDP feature is assumed to be already enabled on Nortel Communication Server 1000, and therefore will not be described in detail.

The procedures below describe the details of configuring Nortel Communication Server 1000 for SIP trunks:

- Launch Unified Communications Manager
- Obtain node and IP addresses
- Configure ISDN
- Configure D-Channel
- Configure zones
- Configure virtual SIP routes and trunks
- Configure route list block and distant steering code
- Configure node SIP parameters
- Launch NRS Manager
- Configure service domain
- Configure SIP gateway endpoints
- Configure routing entries
- Cut over and commit changes

5.1 Launch Unified Communications Manager

Access the Nortel Communication Server 1000 web based interface by using the URL “http://<ip-address>” in an Internet browser window, where “<ip-address>” is the IP address of the Call Server. Note that the IP address for the Call Server may vary, and in this case “10.80.51.10” is used. Log in with the appropriate user name and password. The following Unified Communications Management screen will be displayed. Click on the **Element Name** corresponding to the element of type *CS1000*.

NORTEL UNIFIED COMMUNICATIONS MANAGEMENT [Help](#)

Host Name: interop-cs1000e.interop.avaya.com Software Version: 02.00.0055.00(3266) 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.

Buttons: Add... Edit... Delete

Element Name	Element Type	Release	Address	Description
1 EM on interop-cs1000e	CS1000	6.0	10.80.51.10	New element
2 interop-cs1000e.interop.avaya.com (primary)	Linux Base	6.0	10.80.50.10	Base element
3 10.80.51.13	Media Gateway Controller	6.0	10.80.51.13	New element
4 10.80.51.12	Media Gateway Controller	6.0	10.80.51.12	New element
5 NRSM on interop-cs1000e	Network Routing Service	6.0	10.80.51.10	New element

5.2 Obtain Node and IP Addresses

The Element Manager System Overview screen is displayed. Expand the **IP Network** menu on the left pane and select **Nodes: Servers, Media Cards**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
System Overview

System Overview

IP Address: 10.80.51.10
Type: Nortel Communication Server Linux
Version: 4121
Release: 600 R +

[Active Sessions](#)

The Node Configuration screen is displayed. Click **Node ID 1** to expand it. Note that the node number and IP address may vary.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

IP Telephony Nodes
Click the Node ID to view or edit its properties.

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

Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
1	1	LTPS, PD, Gateway (SIPGw, H323Gw)	-	10.80.50.50	Synchronized

Show: ☒ Nodes ☐ Component Servers and Cards

The Node Details screen is updated with additional details as shown below. Make a note of the **Node IP Address** “10.80.50.50”, and Signaling Server **TLAN IP** address of “10.80.50.10”. These values are used to configure other sections.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1 - LTPS, PD, Gateway (SIPGw, H323Gw))

Node ID: 1 * (0-9999)
Call Server IP Address: 10.80.51.10 *

Telephone LAN (TLAN)
Node IP Address: 10.80.50.50 *
Subnet Mask: 255.255.255.0 *

Embedded LAN (ELAN)
Gateway IP address: 10.80.51.1 *
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)

- Terminal Proxy Server (TPS)
- Gateway (SIPGw & H323Gw)
- Personal Directories (PD)

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Add Remove Make Leader Print | Refresh

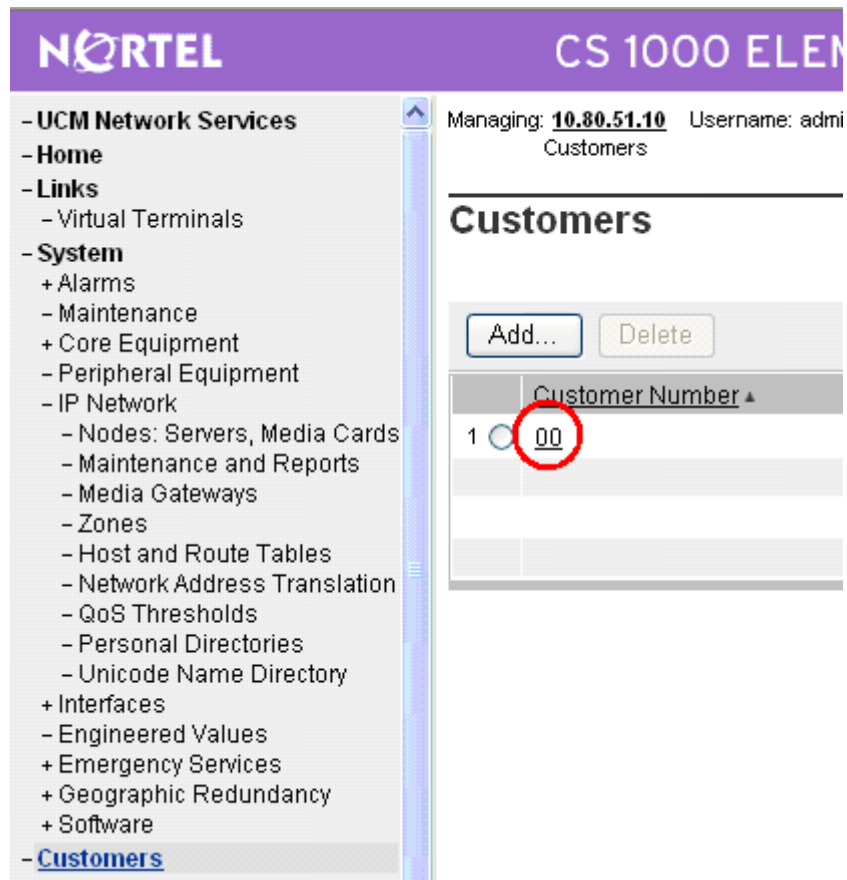
Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
interop-cs1000e	Signaling Server	LTPS, Gateway, PD	10.80.51.10	10.80.50.10	Leader

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.

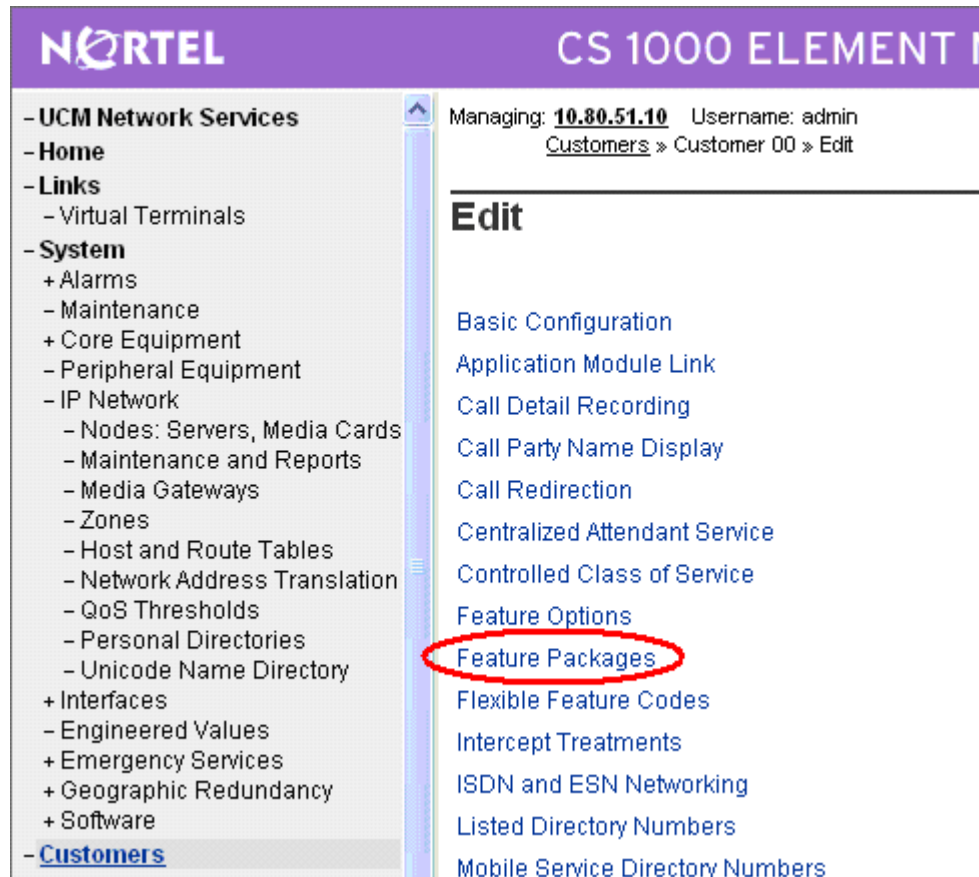
Figure 2: Node Configuration

5.3 Configure 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. In the sample configuration, only one customer was configured on the system.



The Customer 00 Property Configuration 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

Managing: **10.80.51.10** Username: admin
[Customers](#) » Customer 00 » [Edit](#) » Feature Packages

Feature Packages

+ Do Not Disturb Individual	Package: 9
+ End-to-End Signaling	Package: 10
+ Message Waiting Center	Package: 46
+ New Flexible Code Restriction	Package: 49
+ Set Relocation	Package: 53
+ Network Alternate Route Selection	Package: 58
+ Distinctive Ringing	Package: 74
+ Departmental Listed Directory Number	Package: 76
+ 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
+ Enhanced Night Service	Package: 133
+ Integrated Services Digital Network	Package: 145

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

+ Flexible Tones and Cadences Package: 125
 + Enhanced Night Service Package: 133
- Integrated Services Digital Network Package: 145
 + Dial Access Prefix on CLID table entry option

Integrated Services Digital Network: ☒ (1 - 16383)

- Virtual Private Network Identifier: 0

5.4 Configure 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 (in this case “1”). 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, the left sidebar contains a navigation menu with the following items: - UCM Network Services, - Home, - Links (with sub-item - Virtual Terminals), - System (with sub-items + Alarms, - Maintenance, + Core Equipment, - Peripheral Equipment, + IP Network, + Interfaces, - Engineered Values, + Emergency Services, + Geographic Redundancy, + Software), - Customers, - Routes and Trunks (with sub-items - Routes and Trunks, and - D-Channels, which is highlighted), and - D-Channels. The main content area has a status bar indicating "Managing: 10.80.51.10 Username: admin" and "Routes and Trunks » D-Channels". Below this, the "D-Channels" section is titled. Under the "Maintenance" heading, there are four 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)". Under the "Configuration" heading, there is a form with the text "Choose a D-Channel Number:" followed by a dropdown menu showing "1", and "and type:" followed by a dropdown menu showing "DCH" and a "to Add" button.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** 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 1 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.

Click **Submit**.

CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Routes and Trunks » **D-Channels** » D-Channels 10 Property Configuration

D-Channels 1 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	D-Channel is over IP (DCIP)
Designator (DES)	SIPToASM
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 DMS-100 (D100)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	<input type="text"/>
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
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000

+ Basic options (BSCOPT)
+ Advanced options (ADVOPT)

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

5.5 Configure Zones

Select **IP Network** → **Zones** from the left pane to display the Zones screen, and then select **Bandwidth Zones** (not shown). For the **Please Choose the** field, select an available zone number from the drop-down list (in this case “3”). Click to **Add**.

The screenshot shows the Nortel CS 1000 Element Manager interface. The left navigation pane is expanded to 'IP Network' > 'Zones'. The main content area is titled 'Bandwidth Zones' and includes sections for 'Maintenance' (with a link to 'Maintenance Commands for Zones (LD 117)') and 'Configuration' (with a link to 'Configuration Spreadsheet'). Below these links are two buttons: 'Browse...' and 'Import'. At the bottom, there is a form with the label 'Please Choose the' followed by a dropdown menu showing 'Bandwidth Zones 3' and a 'to Add' button.

The Zone Basic Property and Bandwidth Management screen is displayed next. For the **Zone Intent (ZBRN)** field, select “VTRK (VTRK)” from the drop-down list. For the Description (ZDES) field, enter descriptive text. Retain the default values for all remaining fields, and click **Submit**.

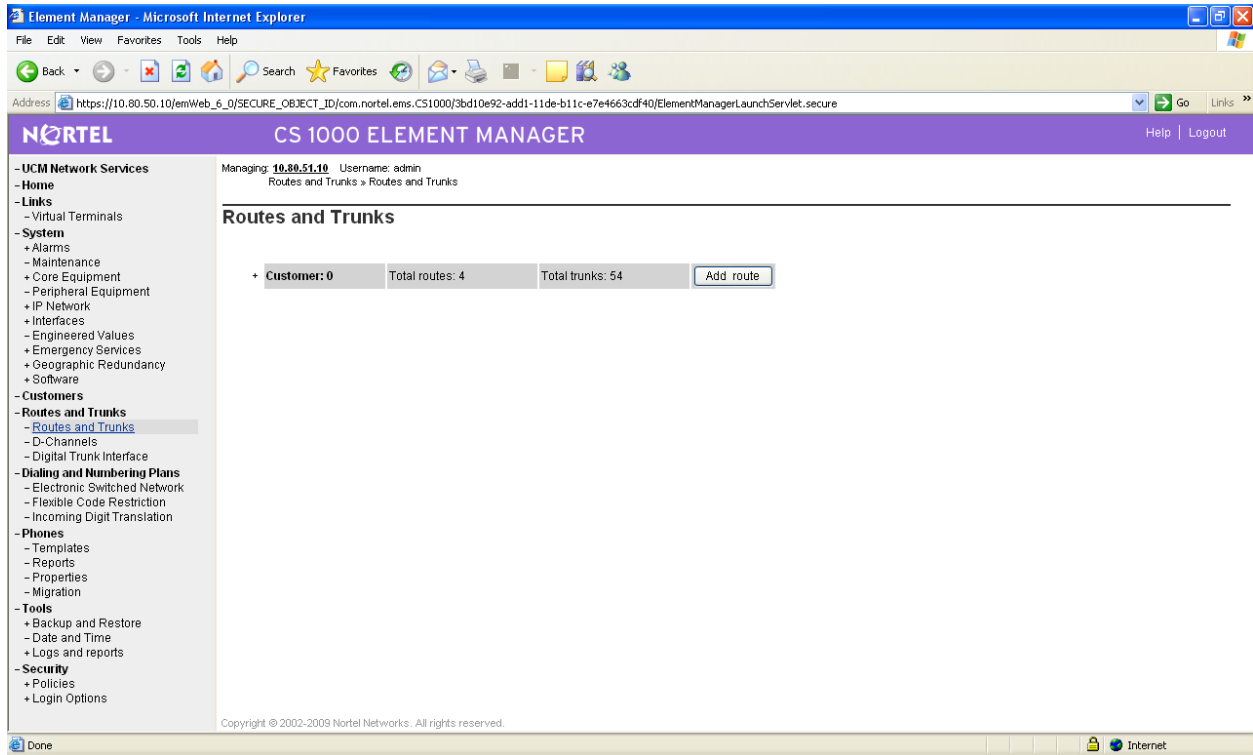
The screenshot shows the Nortel CS 1000 Element Manager interface with the 'Zone Basic Property and Bandwidth Management' screen. The left navigation pane is expanded to 'IP Network' > 'Zones'. The main content area has a title bar 'Zone Basic Property and Bandwidth Management' and a table with two columns: 'Input Description' and 'Input Value'. The table contains the following rows:

Input Description	Input Value
Zone Number (ZONE):	3
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	ASMSIPZONE

At the bottom of the screen are four buttons: 'Submit', 'Refresh', 'Delete', and 'Cancel'.

5.6 Configure 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, Route 1 Property Configuration screen is displayed next. 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

Routes and Trunks » [Routes and Trunks](#) » Customer 0, Route 1 Property Configuration

- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers
- Routes and Trunks
 - [Routes and Trunks](#)
 - D-Channels

Customer 0, Route 1 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE)

RDB

Customer number (CUST)

00

Route number (ROUT)

1

Designator field for trunk (DES)

ASMSIP

Trunk type (TKTP)

TIE

Incoming and outgoing trunk (ICOG)

Incoming and Outgoing (IAO) ▼

Access code for the trunk route (ACOD)

7770001

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 zone number from **Section 5.5**. For the **Node ID of signaling server of this route (NODE)** field, enter the node number from **Section 5.2**. Select “SIP (SIP)” from the drop-down list for the **Protocol ID for the route (PCID)** field.

- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Properties
 - Migration
- Tools

The route is for a virtual trunk route (VTRK)

☒

- Zone for codec selection and bandwidth management (ZONE)

003

Range: 0 - 255

- Node ID of signaling server of this route (NODE)

1

Range: 0 - 9999

- Protocol ID for the route (PCID)

SIP (SIP) ▼

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

☐

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 5.4**.
- **Network Calling Name Allowed (NCNA):** Check the field.
- **Network Call Redirection (NCRD):** Check the field.

- + Interfaces
- Engineered Values
- + Emergency Services
- + Geographic Redundancy
- + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans

Integrated services digital network option (ISDN) ☒

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

- D channel number (DCH) 1 Range: 0 - 254

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

- Private network identifier (PNI) 00000 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.

NORTEL
CS 1000 ELEMENT MANAGER

- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + Geographic Redundancy
 - + Software
- Customers

Managing: 10.80.51.10 Username: admin
Routes and Trunks » Routes and Trunks

Routes and Trunks

- Customer: 0	Total routes: 4	Total trunks: 54	
+ Route: 1	Type: TIE	Description: ASMSIP	<div style="display: flex; justify-content: flex-end; gap: 5px;"> <div style="border: 1px solid #ccc; padding: 2px 5px;">Edit</div> <div style="border: 1px solid #ccc; padding: 2px 5px;">Add trunk</div> </div>
+ Route: 3	Type: TIE	Description: QSIG TO CM	<div style="display: flex; justify-content: flex-end; gap: 5px;"> <div style="border: 1px solid #ccc; padding: 2px 5px;">Edit</div> <div style="border: 1px solid #ccc; padding: 2px 5px;">Add trunk</div> </div>
+ Route: 4	Type: TIE	Description: QSIGTOM1K	<div style="display: flex; justify-content: flex-end; gap: 5px;"> <div style="border: 1px solid #ccc; padding: 2px 5px;">Edit</div> <div style="border: 1px solid #ccc; padding: 2px 5px;">Add trunk</div> </div>
+ Route: 10	Type: TIE	Description: H323	<div style="display: flex; justify-content: flex-end; gap: 5px;"> <div style="border: 1px solid #ccc; padding: 2px 5px;">Edit</div> <div style="border: 1px solid #ccc; padding: 2px 5px;">Add trunk</div> </div>

The Customer 0, Route 1, New Trunk 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. The total number of trunks should match the number of trunk group members provisioned in the SIP trunk from Communication Manager 5.2.1 to Nortel in **Section 3.7**. 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.
- **Start arrangement Incoming (STRI):** “Wink or Fast Flash (WNK)”
- **Start arrangement Outgoing (STRO):** “Wink or Fast Flash (WNK)”
- **Card Density:** Select Octal Density (8D)
- **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 |

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

Customer 0, Route 1, New Trunk Configuration

- Basic Configuration

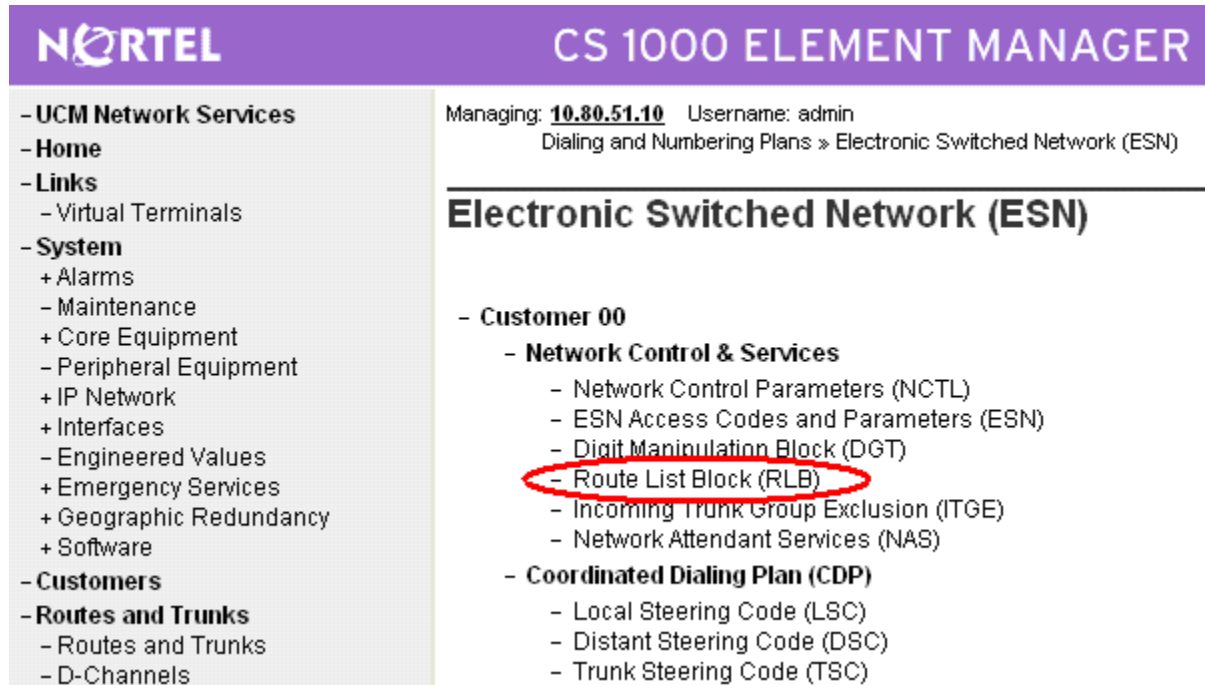
Input Description	Input Value
Multiple trunk input number (MTINPUT)	4
Trunk data block (TYPE)	IP Trunk (IPTI)
Terminal Number (TN)	096 0 00 00
Designator field for trunk (DES)	ASMSIPTRK
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	1 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	Octal 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)	1
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

+ Advanced Trunk Configurations

Save Cancel

5.7 Configure 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)**.



NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN)

Electronic Switched Network (ESN)

- Customer 00
 - Network Control & Services
 - Network Control Parameters (NCTL)
 - ESN Access Codes and Parameters (ESN)
 - Digit Manipulation Block (DGT)
 - **Route List Block (RLB)**
 - Incoming Trunk Group Exclusion (ITGE)
 - Network Attendant Services (NAS)
 - Coordinated Dialing Plan (CDP)
 - Local Steering Code (LSC)
 - Distant Steering Code (DSC)
 - Trunk Steering Code (TSC)

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 “1”). Click **to Add**.



NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Route List Blocks

Route List Blocks

Please enter a route list index (0 - 999)

Figure 3: Route List Blocks

The Route List Block screen is displayed with a listing of parameters. For the **Route Number (ROUT)** field, select the route number from **Section 5.6**. 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

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) » Customer 00 » Network Control & Services » [Route List Blocks](#) » Route List Block

Route List Block

Input Description	Input Value
Route List Index (RLI):	1
Entry Number for the Route List (ENTR):	0 (0 - 63)
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	1
Skip Conventional Signaling (SCNV):	<input type="checkbox"/>
Display Originator's Information (DORG):	<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"/>

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 3xxxx calls to Communication Manager.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#)

Electronic Switched Network (ESN)

- Customer 00
 - Network Control & Services
 - Network Control Parameters (NCTL)
 - ESN Access Codes and Parameters (ESN)
 - Digit Manipulation Block (DGT)
 - Route List Block (RLB)
 - Incoming Trunk Group Exclusion (ITGE)
 - Network Attendant Services (NAS)
 - Coordinated Dialing Plan (CDP)
 - Local Steering Code (LSC)
 - **Distant Steering Code (DSC)**
 - Trunk Steering Code (TSC)

The Distant Steering Code List screen is displayed next. In the **Please enter a distant steering code** field, enter the dialed prefix digits to match on (in this case “30”). Click to **Add**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) »

Distant Steering Code List

Add

Please enter a distant steering code

The Distant Steering Code screen is displayed. For the **Route List to be accessed for trunk steering code (RLI)** field, select the route list index in **Figure 3** of **Section 5.7** from the drop-down list. Retain the default values in all remaining fields and click on **Submit**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.80.51.10** Username: admin
Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) » Customer 00 » Coordinated Dialing Plan (CDP) » [Distant Steering Code List](#) » D

Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	<input type="text" value="3"/>
Flexible Length number of digits (FLEN):	<input type="text" value="5"/> (0 - 10)
Display (DSP):	<input type="text" value="Local Steering Code (LSC)"/>
Remote Radio Paging Access (RRPA):	<input type="checkbox"/>
Route List to be accessed for trunk steering code (RLI):	<input type="text" value="1"/>
Collect Call Blocking (CCBA):	<input type="checkbox"/>
maximum 7 digit NPA code allowed (NPA):	<input type="text"/>
maximum 7 digit NXX code allowed (NXX):	<input type="text"/>

5.8 Configure Node SIP 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 5.2**). The Node Details screen is displayed. It is assumed that the TLAN and ELAN IP addresses have already been configured as a result of basic configuration of the Signaling Server. Click on **Voice Gateway (VGW) and Codecs**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1 - LTPS, PD, 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)

- Terminal Proxy Server (TPS)
- Gateway (SIPGw & H323Gw)
- Personal Directories (PD)

* 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"/> interop-cs1000e	Signaling Server	LTPS, Gateway, PD	10.80.51.10	10.80.50.10	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, verify the default options shown under *General*.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

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

General | Voice Codecs | Fax

General

Echo Cancellation: ☒ Use canceller, with tail delay: 128
☒ Dynamic attenuation

Voice Activity Detection Threshold: -17 (-20 - +10 DBM)
Idle Noise Level: -65 (-327 - +327 DBM)

Signaling Options: ☒ DTMF Tone Detection
☐ Low latency mode
☒ Remove DTMF delay (squelch DTMF from TDM to IP)
☒ Modem/Fax pass-through
☒ V.21 Fax Tone Detection

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. Click on **Save**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1 - 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)

Codec G723.1: ☐ Enabled

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

Save Cancel

When the Node Details screen is displayed, click on **Gateway (SIPGw and H.323Gw)**. Under *General* on 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 “SIP Gateway (SIPGw)”; or select “SIPGw and H.323Gw” if both protocols will be supported on this system.
- **SIP Domain Name** Domain name used in **Section 4.1**
- **Local SIP Port** “5060”
- **Gateway endpoint name** An appropriate name
- **Gateway password** Enter a password

The screenshot shows the Nortel CS 1000 Element Manager interface. The top bar displays the Nortel logo and the title 'CS 1000 ELEMENT MANAGER'. Below the bar, the left sidebar contains a navigation menu with categories like 'UCM Network Services', 'Home', 'Links', 'System', 'Interfaces', and 'Customers'. The main content area shows the 'Node ID: 1 - Virtual Trunk Gateway Configuration Details' screen. The 'General' tab is selected, and the 'Vtrk Gateway Application' is set to 'SIPGw and H.323Gw'. Other fields include 'SIP Domain name' (avaya.com), 'Local SIP Port' (5060), 'Gateway endpoint name' (CS1KGateway), 'Gateway password' (empty), and 'H.323 ID' (CS1KGateway). The 'Virtual Trunk Network Health Monitor' section on the right includes a checkbox for 'Monitor IP Addresses' and a list of monitor addresses.

Figure 4: Node Details

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 **Figure 2** in **Section 5.2**.
- **Port** “5060”
- **Transport Protocol** “TCP”
- **Options** Check “Support registration”

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, Home, Links, System, Interfaces, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area is titled 'Node ID: 1 - Virtual Trunk Gateway Configuration Details'. It features a tabbed interface with 'SIP Gateway Settings' selected. The 'SIP Gateway Settings' section includes 'TLS Security' (set to 'Security Disabled'), 'SIP Gateway Settings' (with fields for Port: 5061, Number of Byte Re-negotiation: 0, and Options: Client Authentication, X509 certificate authority), and 'Proxy Or Redirect Server' (with fields for Primary TLAN IP Address: 10.80.50.10, Secondary TLAN IP Address: 0.0.0.0, Port: 5060, Transport protocol: TCP, and Options: Support registration, Primary CDS Proxy). A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' Buttons for 'Save' and 'Cancel' are visible.

Scroll down the parameters box to the *SIP URI Map* section. Under *Public E.164 Domain Names*, enter the appropriate **National** and **Subscriber** values for the network configuration. In the test configuration, “1” is the country code and “732” is the area code. The remaining fields can be left at their default values. Click on **Save**.

CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1 - Virtual Trunk Gateway Configuration Details

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

Country code (CCC):
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: +1
Subscriber: +1732
Special number: PublicSpecial
Unknown: PublicUnknown

Private Domain Names

UDP: udp
CDP: cdp.udp
Special number: PrivateSpecial
Vacant number: PrivateUnknown
Unknown: UnknownUnknown

* 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.

CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1 - LTPS, PD, Gateway (SIPGw, H323Gw))

Node ID: 1 * (0-9999)
Call Server IP Address: 10.80.51.10 *

Telephony LAN (TLAN)

Node IP Address: 10.80.50.50 *
Subnet Mask: 255.255.255.0 *

Embedded LAN (ELAN)

Gateway IP address: 10.80.51.1 *
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)

- Terminal Proxy Server (TPS)
- Gateway (SIPGw & H323Gw)
- Personal Directories (PD)

* Required Value.

Save **Cancel**

Associated Signaling Servers & Cards

Select to add

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input type="checkbox"/> interop-cs1000e	Signaling Server	LTPS, Gateway, PD	10.80.51.10	10.80.50.10	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.

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

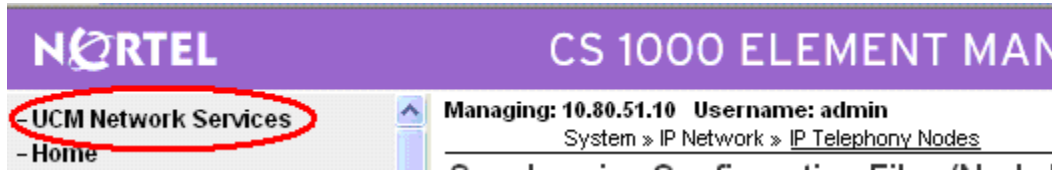
The screenshot shows the 'Node Saved' screen in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation menu with categories like UCM Network Services, Home, Links, and System. The main content area has a header with 'Managing: 10.80.51.10 Username: admin' and a breadcrumb trail 'System > IP Network > IP Telephony Nodes'. Below this, the title 'Node Saved' is displayed. The main text area contains a message: 'Node ID: 1 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards.' At the bottom, there are two buttons: 'Transfer Now...' and 'Show Nodes'. The 'Transfer Now...' button has a tooltip that says 'You will be given an option to select individual servers, or transfer to all.' The 'Show Nodes' button has a tooltip that says '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**. When the synchronization completes, click on **Restart Applications**.

The screenshot shows the 'Synchronize Configuration Files (Node ID <1>)' screen in the Nortel CS 1000 Element Manager. The left sidebar is the same as the previous screenshot. The main content area has a header with 'Managing: 10.80.51.10 Username: admin' and a breadcrumb trail 'System > IP Network > IP Telephony Nodes'. Below this, the title 'Synchronize Configuration Files (Node ID <1>)' is displayed. The main text area contains a message: '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.' At the top of the main content area, there are three buttons: 'Start Sync', 'Cancel', and 'Restart Applications'. To the right of these buttons are links for 'Print' and 'Refresh'. Below the buttons is a table with the following columns: 'Hostname', 'Type', 'Applications', and 'Synchronization Status'. The table has one row with the following data: 'Interop-cs1000e', 'Signaling Server', 'LTPS, Gateway, PD', and 'Sync required'. At the bottom of the table, there is a footnote: '* 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.'

5.9 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”

	Element Name	Element Type	Release	Address	Description
1	EM on interop-cs1000e	CS1000	6.0	10.80.51.10	New element
2	interop-cs1000e.interop.avaya.com (primary)	Linux Base	6.0	10.80.50.10	Base element
3	10.80.51.13	Media Gateway Controller	6.0	10.80.51.13	New element
4	10.80.51.12	Media Gateway Controller	6.0	10.80.51.12	New element
5	NRS on interop-cs1000e	Network Routing Service	6.0	10.80.51.10	New element

The Network Routing Service Manager screen is displayed. Click **EDIT**.

	Service Name	Service Status
1	SIP Proxy Server (SPS)	In service
2	Gatekeeper (GK)	In service
3	Network Connection Server (NCS)	In service

Server Configuration

NRS Setting

Host name SS_Node
Primary TLAN IP address 10.80.50.10
Secondary TLAN IP address 0.0.0.0

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).

- **UDP transport enabled:** Check the checkbox
- **Primary Server UDP IP:** The Call Server IP address from **Section 5.2**
- **Primary Server UDP port:** “5060”
- **TCP transport enabled:** Check the checkbox
- **Primary Server TCP IP:** The Call Server IP address from **Section 5.2**
- **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 "Logout".

On the left is a navigation menu under "«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

The main content area is titled "Managing: 10.80.51.10" and "System » NRS Server » Edit". Below this is the "Edit Server Configuration" section.

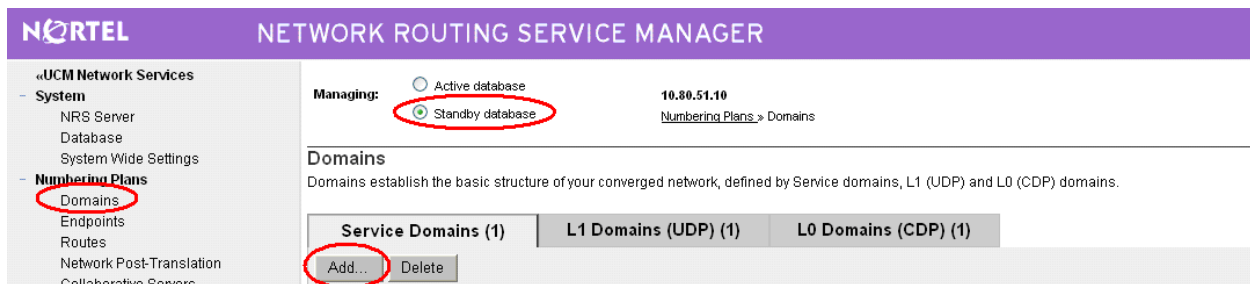
The "SIP Server Settings" section contains the following fields:

- Public name for non-trusted networks: unknown
- Public number for non-trusted networks: 000-000
- UDP Transport enabled: ☒
- Primary server UDP IP: 10.80.50.10
- Primary server UDP port: 5060
- Secondary server UDP IP: 0.0.0.0
- Secondary server UDP port: 5060
- TCP Transport enabled: ☒
- Primary server TCP IP: 10.80.50.10
- Primary server TCP port: 5060
- Secondary server TCP IP: 0.0.0.0
- Secondary server TCP port: 5060

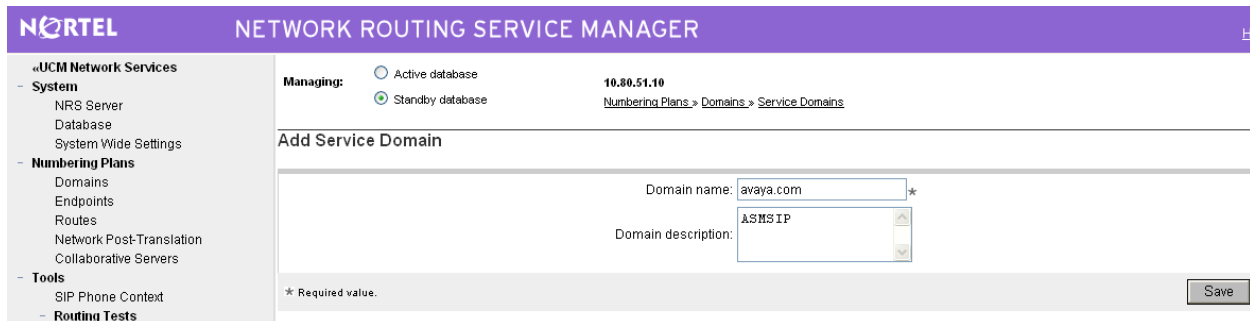
At the bottom of the configuration area, there is a note: "(Note: Any modification of NRS Server configuration would not take effect until you restart all the services.)". Below the note is a small text: "* Required value." and two buttons: "Save" and "Cancel".

5.10 Configure 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 **Figure 4 of Section 5.8** into the **Domain name** field, and a descriptive text for the **Domain description** field. Click **Save**.



Select the **L1 Domains (UDP)** tab to display the L1 Domains (UDP) screen. 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**.

The Add L1 Domain (avaya.com) screen is displayed next, as shown below. Enter a descriptive **Domain name** and **Domain description**, and applicable **E.164 country code** and **E.164 area code** for the network configuration. Retain the default value in the remaining fields, and on **Save**.

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

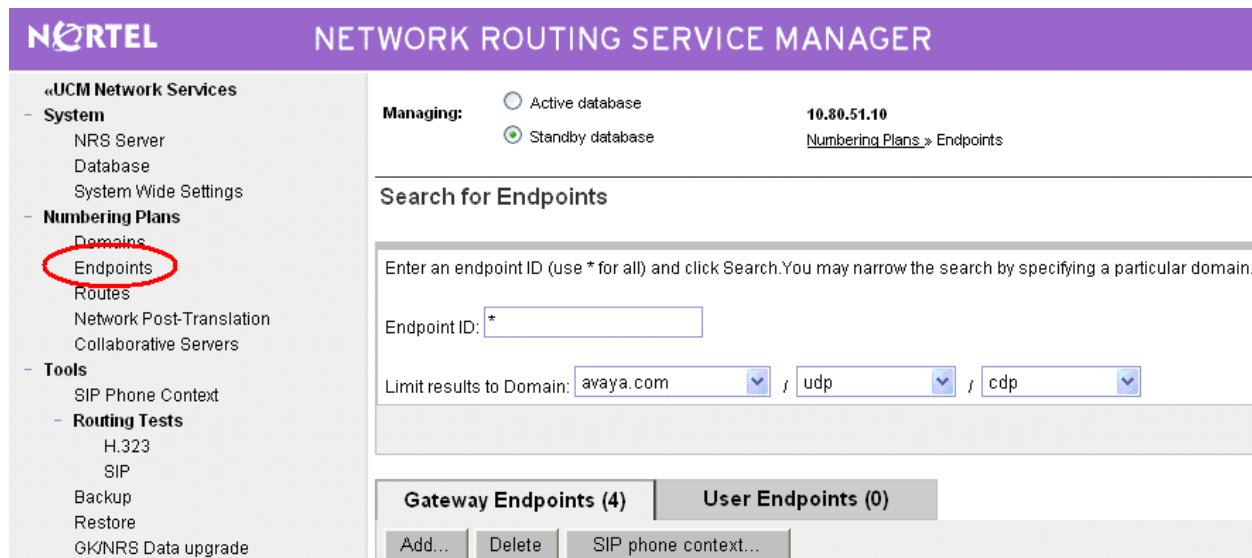
The screenshot shows the Nortel Network Routing Service Manager interface. On the left is a navigation menu with categories: «UCM Network Services», - System (NRS Server, Database, System Wide Settings), - Numbering Plans (Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers), and - Tools. The main area has a header with the Nortel logo and 'NETWORK ROUTING SERVICE MANAGER'. Below this, it says 'Managing: Active database' and 'Standby database' with the IP '10.80.51.10' and a link 'Numbering Plans » Domains'. The 'Domains' section title is followed by the text: 'Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.' Below this are three tabs: 'Service Domains (1)', 'L1 Domains (UDP) (1)', and 'L0 Domains (CDP) (1)'. The 'Filter by Domain' dropdown is set to 'avaya.com' and the second dropdown is set to 'udp'. There are 'Add...' and 'Delete' buttons at the bottom.

The Add L0 Domain (avaya.com /udp) screen is displayed next, as shown below. Enter a descriptive **Domain name** and **Domain description**. Retain the default values in the remaining fields and click **Save**.

The screenshot shows the 'Add L0 Domain (avaya.com / udp)' screen. The left navigation menu is the same as the previous screenshot. The main area has the same header. Below the header, it says 'Managing: Active database' and 'Standby database' with the IP '10.80.51.10' and a link 'Numbering Plans » Domains » L0 Domain'. The title 'Add L0 Domain (avaya.com / udp)' is displayed. The form contains the following fields: 'Domain name' (required, value: cdp), 'Domain description' (value: Nortel L0 Domain), 'Endpoint authentication enabled' (dropdown: Not configured), 'Authentication password' (text field), 'E.164 country code' (text field), 'E.164 area code' (text field), 'Private unqualified number label' (value: PrivateUnknown), 'E.164 international dialing access code' (text field), 'E.164 international dialing code length' (text field, range: 0-99), 'E.164 national dialing access code' (text field), 'E.164 national dialing code length' (text field, range: 0-99), 'E.164 local (subscriber) dialing access code' (text field), and 'E.164 local (subscriber) dialing code length' (text field, range: 0-99). A note at the bottom left says '* Required value.' and a 'Save' button is at the bottom right.

5.11 Configure SIP Gateway Endpoints

Next, configure two SIP gateway endpoints - one for the Session Manager server, and the other for the Nortel SIP Redirect Server. 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, “udp” and “cdp”. Click **Add** to add a new gateway endpoint for Session Manager.



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 **10.80.51.10**
☒ Standby database [Numbering Plans » Endpoints](#)

Search for Endpoints

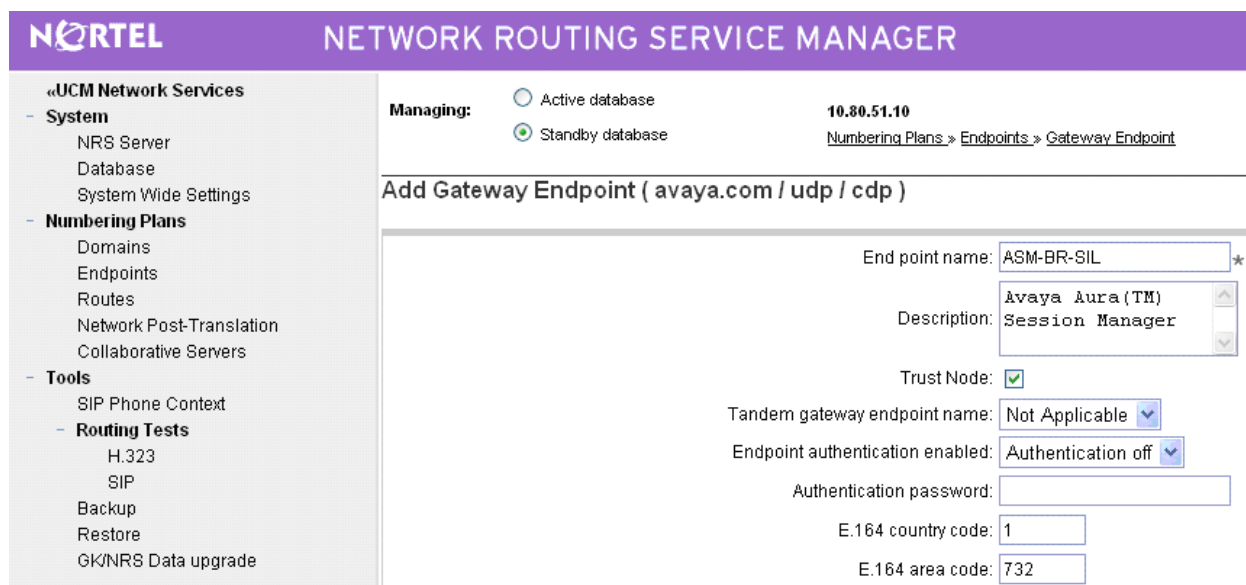
Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID: *

Limit results to Domain: / /

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

Enter a descriptive **End point name** and **Description**, as shown below and applicable **E.164 country code** and **E.164 area code** for the network configuration.



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 **10.80.51.10**
☒ Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Add Gateway Endpoint (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:

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

- **Static endpoint address:** IP address of Avaya Aura™ Session Manager SM-100 Module interface
- **H.323 Support:** “Not RAS H.323 endpoint”
- **SIP support:** “Static SIP endpoint”
- **SIP TCP transport enabled:** “TCP”

The screenshot displays the configuration interface for Avaya Aura Session Manager. On the left, a navigation pane shows the hierarchy: **Numbering Plans** (Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers) and **Tools** (SIP Phone Context, **Routing Tests** (H.323, SIP, Backup, Restore, GK/NRS Data upgrade)). The main area shows the configuration for the selected item. The fields are as follows:

- Private Special number 2: [Empty text box]
- Private Special number 2 dialing code length: [Empty text box] (0-31)
- Static endpoint address type: IP version 4 (dropdown)
- Static endpoint address: 10.1.2.170
- H.323 support: Not RAS H.323 endpoint (dropdown)
- SIP support: Static SIP endpoint (dropdown)
- 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: ☒

At the bottom left, there is a note: * Required value. At the bottom right, there is a **Save** button.

Repeat the procedures to add a gateway endpoint for the Nortel SIP Redirect Server as shown below. Select the desired value for **Endpoint authentication enabled**. If the authentication is turned on, then the value entered in the **Authentication password** field must match the **Gateway password** value from **Figure 4 of Section 5.8**.

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 **10.80.51.10**
☒ Standby database [Numbering Plans » Endpoints » Gateway Endpoint](#)

Add Gateway Endpoint (avaya.com / udp / cdp)

End point name: CS1KGateway *

Description: NortelRedirectServer

Trust Node: ☒

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Authentication on

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. 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 **Figure 4 of Section 5.8**. Maintain the default values in the remaining fields, and click **Save**.

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

Private Special number 2:

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

Static endpoint address type: IP version 4

Static endpoint address:

H.323 support: 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: ☒

★ Required value

Save

5.12 Configure Routing Entries

Configure two routing entries. The first entry uses the Session Manager gateway endpoint to reach Avaya endpoints with extension digits 3xxxx. The second entry uses the Nortel Redirect Server gateway endpoint to reach Nortel endpoints with extension digits 777xxx.

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**.

The screenshot displays the 'Nortel Network Routing Service Manager' web interface. On the left is a navigation menu under 'UCM Network Services' with categories: System, Numbering Plans, and Tools. The 'Routes' option under 'Numbering Plans' is highlighted with a red circle. The main content area shows the 'Search for Routing Entries' screen. At the top, it indicates the managing database (Active/Standby) and IP (10.80.51.10). Below is a search form with fields for 'DN Prefix' (set to '*'), 'DN Type' (set to 'Private level 0 regional (CDP steering code)'), and 'Limit results to Domain' (set to 'avaya.com', 'udp', and 'cdp'). The 'Endpoint Name' is set to 'ASM-BR-SIL'. At the bottom, there are tabs for 'Routing Entries (1)' and 'Default Routes (0)', with a row of action buttons: Add..., Copy..., Move..., Import..., Export..., Routing test..., and 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 “3”.
- **Route cost (1 – 255):** An appropriate cost value with 1 being least cost.

NORTEL NETWORK ROUTING SERVICE MANAGER

Managing: ☐ Active database 10.80.51.10
☒ Standby database [Numbering Plans > Routes > Routing Entry](#)

Add Routing Entry (avaya.com / udp / cdp / ASM-BR-SIL)

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

DN prefix: 3 *

Route cost: 1 * (1-255)

★ Required value. Save

Repeat the same procedures to add a routing entry to reach the Nortel Communication Server 1000 endpoints with extension digits 777xxxx behind the Nortel SIP Redirect Server gateway endpoint.

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 **10.80.51.10**
☒ 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 particular domain.

Dn Prefix: Dn Type:

Limit results to Domain: / /

Endpoint Name:

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

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

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

Managing: ☐ Active database **10.80.51.10**
☒ Standby database [Numbering Plans > Routes > Routing Entry](#)

Add Routing Entry (avaya.com / udp / cdp / CS1KGateway)

DN type:

DN prefix: *

Route cost: * (1-255)

* Required value.

Save

5.13 Cut Over and Commit Changes

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

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 left, a navigation menu shows "System" selected, with "Database" highlighted. The main content area displays "Managing: 10.80.51.10" and "System > Database". Below this, the "Database" section explains that NRS uses a redundant database with Active and Standby copies. The "Database status" is shown as "Changed". At the bottom right, there are four buttons: "Cut over", "Revert", "Commit", and "Roll back".

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

This screenshot shows the same Nortel NRS interface as the previous one, but after the "Cut over" action. The "Database status" is now "Switched over". The "Commit" button is now enabled and highlighted in grey. The other buttons ("Cut over", "Revert", "Roll back") are still present but disabled.

6 Verification Steps

This section provides the tests that can be performed on Communication Manager 5.2.1 and Session Manager to verify proper configuration of Communication Manager, Session Manager, and Nortel Communication Server 1000.

6.1 Verify Avaya Aura™ Communication Manager

Verify the status of the SIP trunk group by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 3.6**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 32
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0032/001	T00226	in-service/idle	no
0032/002	T00227	in-service/idle	no
0032/003	T00228	in-service/idle	no
0032/004	T00229	in-service/idle	no

Verify the status of the SIP signaling groups by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 3.6**. Verify the signaling group is “in-service” as indicated in the **Group State** field shown below.

```
status signaling-group 32
```

STATUS SIGNALING GROUP	
Group ID: 32	Active NCA-TSC Count: 0
Group Type: sip	Active CA-TSC Count: 0
Signaling Type: facility associated signaling	
Group State: in-service	

Make a call between the Avaya 9600 Series IP Telephone and the Nortel i2004 H.323 Telephone. Verify the status of connected SIP trunks by using the “status trunk x/y”, where “x” is the number of the SIP trunk group from **Section 3.6.2** to reach Session Manager, and “y” is the member number of a connected trunk. Verify on Page 1 that the **Service State** is “in-service/active”. On Page 2, verify that the IP addresses of the C-LAN and Session Manager are shown in the **Signaling** section. In addition, the **Audio** section shows the G.729 codec and the IP addresses of the Avaya H.323 and Nortel H.323 endpoints. The **Audio Connection Type** displays “ip-direct”, indicating direct media between the two endpoints.

status trunk 32/1	Page 1 of 3
TRUNK STATUS	
Trunk Group/Member: 0032/001	Service State: in-service/active
Port: T00226	Maintenance Busy? no
Signaling Group ID: 32	
IGAR Connection? no	
Connected Ports: S00504	

status trunk 32/1	Page 2 of 3
CALL CONTROL SIGNALING	
Near-end Signaling Loc: 01A0217	
Signaling	IP Address Port
Near-end:	10.1.2.233 : 5060
Far-end:	10.1.2.170 : 5060
H.245 Near:	
H.245 Far:	
H.245 Signaling Loc:	H.245 Tunneled in Q.931? no
Audio Connection Type: ip-direct	Authentication Type: None
Near-end Audio Loc:	Codec Type: G.711MU
Audio	IP Address Port
Near-end:	10.1.2.253 : 6646
Far-end:	10.80.50.253 : 5200
Video Near:	
Video Far:	
Video Port:	
Video Near-end Codec:	Video Far-end Codec:

6.2 Verify Avaya Aura™ Session Manager

Navigate to *Session Manager* → *System Status* → *SIP Entity Monitoring* on the left to verify that none of the links to the defined SIP entities is down.

Home / Session Manager / System Status / SIP Entity Monitoring

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▶ 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

Security Module Status

Data Replication Status

RegistrationSummary

User Registrations

▶ System Tools

SIP Entity Link Monitoring Status Summary

This page provides a summary of Session Manager SIP entity link monitoring status.

Entity Link Status for All Session Manager Instances

Refresh

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Monitoring Status
SM1	3/16	0	0	1
SM2	0/0	0	0	0

All Monitored SIP Entities

Refresh

15 Items Filter: Enable

SIP Entity Name
AllanC-S8300-G350
alpinemas1
AudioCodes M1000
Avaya_MAS-Br2
Avaya_MAS-HQ
CallCenter
Cisco-UCM6
CiscoUCME
Denver Nortel CS1000e

Under *All Monitored SIP entities*, select the appropriate SIP entities and verify that the connection status is “Up”, as shown below for the Nortel Communication Server 1000.

AVAYA

Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Dec. 03, 2009 4:01 PM
[Help](#) [Log off](#)

Home / Session Manager / System Status / SIP Entity Monitoring / SIP Entity Link Status

▶ Asset Management

▶ Communication System Management

▶ Monitoring

▶ User Management

▶ 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

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: Denver Nortel CS1000e

Refresh Summary View

1 Item Filter: Enable

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
<input type="checkbox"/> Show	SM1	10.80.50.50	5060	TCP	Up	200 OK	Up

FS; Reviewed:
SPOC 01/31/2010

Solution & Interoperability Test Lab Application Notes
©2010 Avaya Inc. All Rights Reserved.

63 of 66
NrtlR6_ASM52

6.3 Verify Nortel Communication Server 1000

Select **IP Network→Nodes: Servers, Media Cards→Maintenance and Reports** on the left. Click **Status** for the Signaling Server node to verify that it is enabled and operational.

CS 1000 ELEMENT MANAGER

Managing: 10.80.51.10 Username: admin
System > IP Network > Node Maintenance and Reports

Node Maintenance and Reports

Node ID: 1		Node IP: 10.80.50.50		Total elements: 1
Index	ELAN IP	Type	TN	ELAN
interop-cs1000e	10.80.51.10	Signaling Server-Nortel CPPMv1	NO TN	

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

10.80.51.10 : Enabled

6.4 Verification Scenarios

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

- Basic calls between various telephones on the Communication Manager and Nortel Communication Server 1000 can be made in both directions using G.711MU, and G.729.
- Proper display of the calling and called party name and number information was verified for all telephones with the basic call scenario.
- Supplementary calling features were verified. The feature scenarios involved additional endpoints on the respective systems, such as performing an unattended transfer of the SIP trunk call to a local endpoint on the same site, and then repeating the scenario to transfer the SIP trunk call to a remote endpoint on the other site. The supplementary calling features verified are shown below. Note that calling/called party name and number display may not be consistent in some cases.
 - Unattended transfer
 - Attended transfer
 - Hold/Unhold
 - Consultation hold
 - Call forwarding
 - Conference
 - Calling number block

7 Conclusion

As illustrated in these Application Notes, Avaya Aura™ Communication Manager can interoperate with Nortel Communication Server 1000 using SIP trunks via Avaya Aura™ Session Manager. The following is a list of interoperability items to note:

- For G.729 interoperability, “G.729” must be included in the codec set in Communication Manager.
- Audio shuffling between the H.323 IP telephones is supported.
- Calling/called party name and number display may not be consistent for some supplementary calling features.

8 Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Aura™ Session Manager:

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

Avaya Aura™ Communication Manager 5.2.1:

- [5] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Doc ID 555-245-206, May, 2009, available at <http://support.avaya.com>.
- [6] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009, available at <http://support.avaya.com>.

Avaya Application Notes:

- [7] *Front-Ending Nortel Communication Server 1000 with an AudioCodes Mediant 1000 Modular Media Gateway to Support SIP Trunks to Avaya Aura™ Session Manager with Avaya Aura™ Communication Manager 5.2 as a Feature Server – Issue 1.0*, available at <http://www.avaya.com>.
- [8] *Configuring SIP Trunks among Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager 5.2, and Nortel Communication Server 1000 – Issue 1.1*, available at <http://www.avaya.com>.

Nortel Communication Server 1000:

- [9] *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.

©2010 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com