



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Windstream SIP Trunking with the Avaya Communication Server 1000 release 7.0 and ACME Packet Net-Net 3800 Session Border Controller Release 6.2 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of the Avaya Communication Server 1000 release 7.0 and the Windstream SIP Trunking. The ACME Net-Net 3800 Session Border Controller is used as IP-IP network border between service providers to control security and interactive communications among multimedia and data services.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

This document provides a typical network configuration deployment of the Avaya Communication Server 1000 and the Windstream SIP Trunking service Voice & Data bundle (hereafter referred to as Windstream system or Metaswitch). The ACME Net-Net 3800 Session Border Controller is used as IP-IP network border between Windstream Metaswitch and Avaya Communication Server 1000.

2. General Test Approach and Test Results

The Avaya Communication Server 1000 system was connected to the ACME Net-Net 3800 via the Network Routing Service/SIP Proxy Server (NRS/SPS). Then the Net-Net 3800 was connected to the Windstream system via SIP. Various call types were made from the Communication Server 1000 to the Windstream system and vice versa to verify the interoperability.

2.1. Interoperability Compliance Testing

The focus of this testing is to verify that Communication Server 1000 can interoperate with the Windstream system. The following interoperability areas were covered:

- General call processing between Communication Server 1000 and Windstream systems including:
 - Codec/ptime(G.711 u-law / 20ms)
 - Hold/Retrieve on both ends
 - CLID displayed
 - Ring-back tone
 - Speech path
 - Dialing plan support
 - Advanced features (Call on Mute, Call Park, Call Waiting)
 - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- Fax/Modem Pass Through is supported only with G.711
- DTMF in both directions
- SIP Transport UDP
- Thru dialing via the Communication Server 1000 Call Pilot
- Voice Mail Server Call Pilot (hosted on Avaya system)
- Early Media Transmission

The following assumptions were made for this lab test configuration:

1. Communication Server 1000 R7.0 software and implementation of latest patches

2. Windstream provides support to setup, configure and troubleshoot on carrier switch during testing execution.

During testing, the following activities were made to each test scenario:

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window were open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.
8. Speech path was checked before and after calls were put on/off hold from each end.
9. Applicable files were screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Avaya Communication Server files.
10. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends.

2.2. Test Results

The objectives outlined in the **Section 2.1** were verified. All the applicable test cases were executed. However, the following observations were noted during the compliance testing:

1. Call is made from Communication Server 1000 phone to a PSTN phone with CLID (Caller Identification) hidden. The call is being rejected with SIP error code 403 (URI not recognized) by the Windstream system (namely Metaswitch). Windstream team is investigating and providing the resolution.
2. Incoming calls from PSTN to Communication Server 1000, CLID number works intermittently. Windstream team is investigating and providing the resolution.
3. Call is made from Communication Server 1000 phone to a PSTN phone with CPND (call party name display) hidden. The call is established with 2 way speech path but the PSTN phone did not display the correct CPND of the caller. SIP Field Privacy is send ID, Metaswitch interprets as CPND private and sends. This is a design intended from Metaswitch.
4. Call is made from a PSTN phone to a Communication Server 1000 phone, which is set Call Forward No Answer to Voicemail. There is about 20-40% of early media greeting from Voicemail system being lost. The Windstream team is investigating.
5. Toll free number was not tested due to the Windstream lab environment does not provide this service.
6. The directory search number 411 service is tested with Windstream emulated 411 number where Communication Server 1000 sends and Windstream terminated as an assign mailbox number.

7. 911 emergency service is tested with Windstream emulated 911 number where Communication Server 1000 sends and Windstream terminated as an assign mailbox number.
8. Call from Communication Server 1000 phone that is programmed to reach PSTN Operator 0. This is not tested since Windstream lab environment does not have this service available.
9. Call from Communication Server 1000 phone that is programmed to reach PSTN Operator 0+10-digits. This is not applicable for Windstream, operator services are reached via Long Distance number 1-xxx-555-1212 to the area code you are wishing to lookup.
10. Communication Server 1000 phone holds/ retrieves a call will cause CLID not to work properly. This is a Communication Server 1000 known issue and has no plan to implement this feature.
11. PSTN1 phone calls to Communication Server 1000 phone, then phone does blind transfer to PSTN2 phone. PSTN 1 phone could not hear ringback tone from PSTN2 phone when Communication Server 1000 phone completed blind transfer. This is a limitation on Windstream because the system does not support UPDATE SIP message.

It was agreed that with Windstream that the above observations were not severe enough to fail the testing.

2.3. Support

For technical support on Windstream system, please contact Windstream technical support at:

- Toll Free: 1-800-843-9214
- <http://www.windstreambusiness.com/support-center.html>

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing event between the Communication Server 1000 and Windstream System.

For confidentiality and privacy purposes, actual public IP addresses used in this testing had been masked out and replaced with imaginary IP addresses throughout the document.

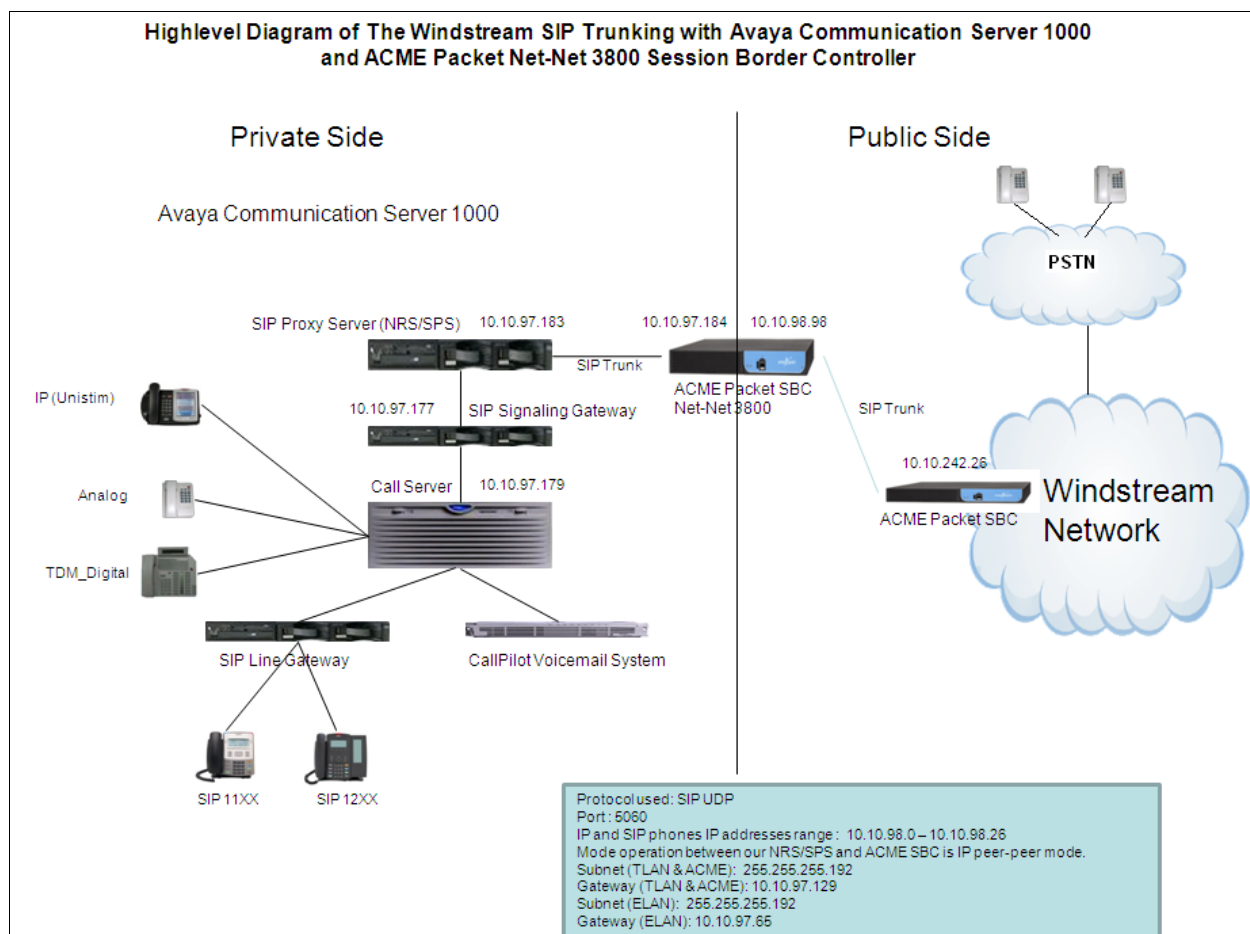


Figure 1- Network diagram for Avaya Communication Server 1000 and Windstream System

4. Equipment and Software Validated

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

Avaya system:

System	Software/Loadware version
Avaya Communication Server 1000 7.0 (CPPM)	<ul style="list-style-type: none"> ● Call Server: 700 Q+ GA ● Signaling Server: 7.00.20 GA ● SIP Line Server: 7.00.20 GA ● NRS/SPS Server: 7.00.20 GA
Avaya phones	<ul style="list-style-type: none"> ● 2002 p2: 0604DCN (Unistim) ● 1140: 0625C8D (Unistim) ● 1120: 0624C8D (Unistim) ● 2007: 0621C8D (Unistim) ● 1120: 4 1 13 0 (SIPLine)
ACME Net-Net 3800	<ul style="list-style-type: none"> ● Firmware SCX6.2.0 MR-4 Patch 3 (Build 754)

Windstream system:

System	Software/Loadware version
Metaswitch	<ul style="list-style-type: none">● Call Feature Server: 7.1.01-B48 P90.41● Universal Media Gateway: 7.1.01-SU64 P86.00● Element Management System: 7.3.00-SU16 P86.00

Additional software and patch lineup for the configuration and active patch list on the SIP Signalling Gateway are listed as below:

Call Server: 7.00 Q+GA plus latest DEPLIST – Deplists_CPL_X21_07_00Q.zip

SSG Server: 7.00.20 GA plus latest DEPLIST –Service_Pack_Linux_7.00.20_20110503.ntl

SLG Server: 7.00.20 GA plus latest DEPLIST –Service_Pack_Linux_7.00.20_20110503.ntl

NRS Server: 7.00.20 GA plus latest DEPLIST –Service_Pack_Linux_7.00.20_20110503.ntl

5. Avaya Communication Server 1000 Configuration

These Application Notes used the Incoming Digit Translation feature to receive the calls and used the Numbering Plan Area Code (NPA), Special Number (SPN) features to route calls from the Avaya Communication Server 1000, over the Windstream SIP trunk to PSTN.

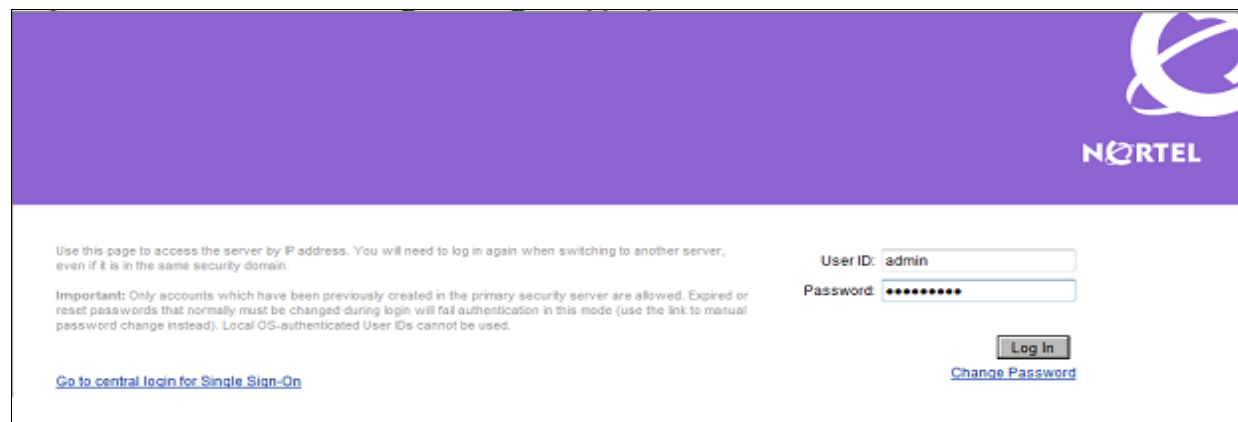
These Application Notes assume that the basic configuration has already been administered. For further information on Avaya Communications Server 1000, please consult references in **Section 9**

The below procedures describe the configuration details of Communication Server 1000 with a SIP trunk to the Windstream system.

5.1. Login to Communication Server 1000 System

5.1.1. Login to Unified Communications Management (UCM) and Element Manager (EM)

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



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

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

User ID:

Password:

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

Figure 2 – Login Unified Communications Management

b) The **Unified Communications Management** screen is displayed. Click on the **Element Name** of the Communication Server 1000 Element as highlighted in red box as shown in **Figure 3**.

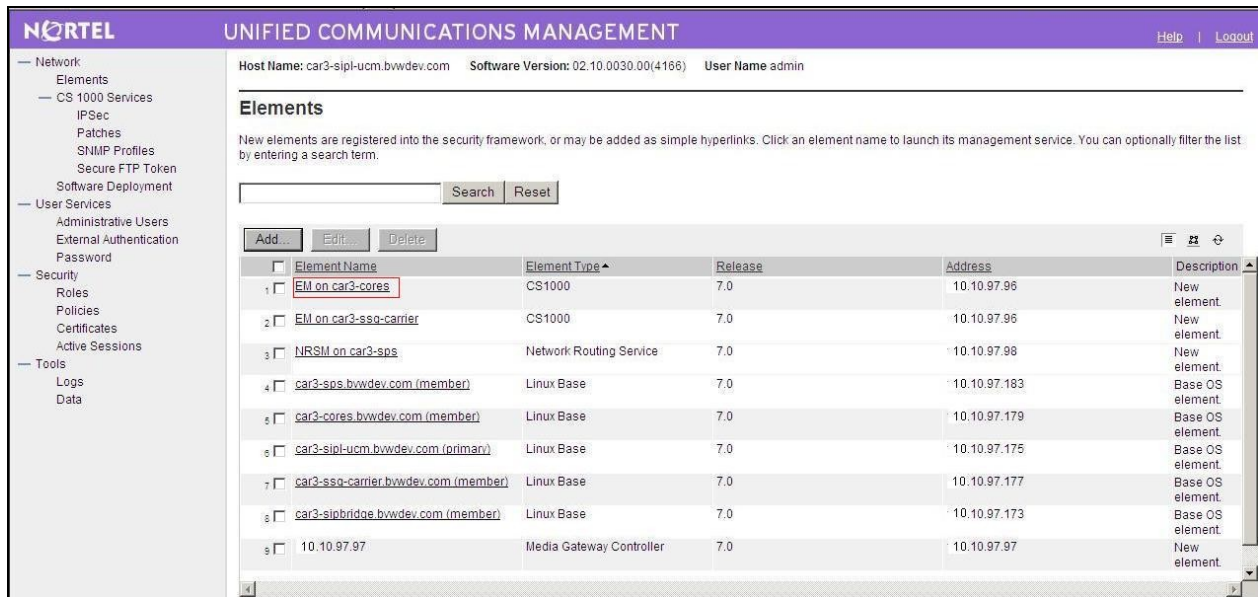


Figure 3 – Unified Communications Management

c) The Communication Server 1000 Element Manager **System Overview** page is displayed as shown in **Figure 4**.

IP Address: 10.10.97.96
Type: Communication Server 1000E CPPM Linux
Version: 4121
Release: 7.00 Q+



Figure 4 – Element Manager System Overview

5.1.2. Login to Call Server by using the Overlay Command Line Interface (CLI)

- a) Use Putty, SSH to connect to IP address of SSG Server with the admin account.
- b) Run the command “cslogin” and login with the appropriate admin account and password.
- c) Here are the logs.

login as: **admin**

Nortel Networks Linux Base 7.00

The software and data stored on this system are the property of, or licensed to, Nortel Networks and are lawfully available only to authorized users for approved purposes. Unauthorized access to any software or data on this system is strictly prohibited and punishable under appropriate laws. If you are not an authorized user then do not try to login. This system may be monitored for operational purposes at any time.

admin@10.10.97.177's password: <----**enter your password**

Last login: Fri Jun 10 14:20:05 2011 from 10.10.98.78

[admin@car3-ssg-carrier ~]\$ cslogin

SEC054 A device has connected to, or disconnected from, a pseudo tty without authenticating
>login

USERID? admin

PASS? <----**enter your password**

.

TTY #08 LOGGED IN

The software and data stored on this system are the property of, or licensed to, Nortel Networks and are lawfully available only to authorized users for approved purposes. Unauthorized access to any software or data on this system is strictly prohibited and punishable under appropriate laws. If you are not an authorized user then logout immediately. This system may be monitored for operational purposes at any time.

ADMIN 12:56 16/6/2011

>

5.2. Administer a Node IP Telephony

This section describes how to configure a Node IP Telephony on the Communication Server 1000.

5.2.1. Obtain Node IP address

These Application Notes assume that the basic configuration has already been administered and that Node has already been created. This Section describes the steps for configuring a Node (Node ID 3000) in Communication Server 1000 IP network to work with Windstream system. For further information on Avaya Communications Server 1000, please consult reference in **Section 9**.

a) Select **System ->IP Network ->Nodes: Servers, Media Cards** and then click on the Node ID of your **CS 1000 Element Manager** as shown in **Figure 5**.

The screenshot shows the Nortel CS 1000 Element Manager web interface. The left sidebar contains a navigation tree with the following items: UCM Network Services, Home, Links, Virtual Terminals, System (expanded), Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network (expanded), Nodes: Servers, Media Cards (highlighted), Maintenance and Reports, Media Gateways, Zones, Host and Route Tables, Network Address Translation (NAT), and QoS Thresholds. The main content area is titled 'IP Telephony Nodes' and shows a table of nodes. The table has columns for Node ID, Components, Enabled Applications, ELAN IP, Node/TLAN IPv4, Node/TLAN IPv6, and Status. Two nodes are listed: Node ID 3000 (Components: 1, Enabled Applications: LTPS, Gateway (SIPGw), ELAN IP: -, Node/TLAN IPv4: 10.10.97.178, Node/TLAN IPv6: -, Status: Synchronized) and Node ID 3002 (Components: 1, Enabled Applications: SIP Line, ELAN IP: -, Node/TLAN IPv4: 10.10.97.176, Node/TLAN IPv6: -, Status: Synchronized). The Node ID 3000 is highlighted with a red box. Below the table, there are checkboxes for 'Show: Nodes' (checked), 'Component servers and cards' (unchecked), and 'IPv6 address' (unchecked). The top of the interface shows the Nortel logo and the title 'CS 1000 ELEMENT MANAGER'. The top right shows the managing IP address '10.10.97.96' and the username 'admin'. The breadcrumb trail is 'System » IP Network » IP Telephony Nodes'.

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
3000	1	LTPS, Gateway (SIPGw)	-	10.10.97.178	-	Synchronized
3002	1	SIP Line	-	10.10.97.176	-	Synchronized

Figure 5 – IP Telephony Nodes

b) The **Node Details** screen is displayed in **Figure 6a**, **Figure 6b** with the IP address of the Communication Server 1000 node. The **Node IP Address** is a virtual address which corresponds to the TLAN IP address of the Signaling Server, SIP Signaling Gateway. The SIP Signaling Gateway uses this **Node IP Address** to communicate with other components to process the SIP call.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.10.97.96** Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 3000 - LTPS, Gateway (SIPGw))

Node ID: * (0-9999)

Call server IP address: * TLAN address type: ☒ IPv4 only ☐ IPv4 and IPv6

Embedded LAN (ELAN) Gateway IP address: * Telephony LAN (TLAN) Node IPv4 address: *

Subnet mask: * Subnet mask: *

Node IPv6 address:

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> car3-ssg-carrier	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	10.10.97.95	<input type="text" value="10.10.97.177"/>	Leader

Show: ☐ IPv6 address

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 6a –Node Details

CS 1000 ELEMENT MANAGER

Managing: **10.10.97.96** Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 3000 - LTPS, Gateway (SIPGw))

Subnet mask: *
Subnet mask: *
Node IPv6 address:

IP Telephony Node Properties

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

Applications (click to edit configuration)

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

* Required Value.

Associated Signaling Servers & Cards

<input type="checkbox"/> Hostname ^	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> car3-ssg-carrier	Signaling_Server	LTPS, Gateway, PD, Presence Publisher, IP Media Services	10.10.97.95	10.10.97.177	Leader

Show: ☐ IPv6 address

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 6b –Node Details

5.2.2. Administer Terminal Proxy Server (TPS)

c) Continue from **Section 5.2.1**. On the **Node Details** page, select the **Terminal Proxy Server (TPS)** link as shown in **Figure 6b**.

d) Check the **UNISlim Line Terminal Proxy Server** check box and then click the **Save** button as shown in **Figure 7**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.10.97.96** Username: admin
System » IP Network » IP Telephony Nodes » Node Details » UNISlim Line Terminal Proxy Server (LTPS) Configuration

Node ID: 3000 - UNISlim Line Terminal Proxy Server (LTPS) Configuration Details

UNISlim Line Terminal Proxy Server: ☒ Enable proxy service on this node

Firmware

IP address:
Full file path:
Server Account/User ID:
Password:

DTLS

DTLS policy:

Options: ☐ Client authentication
☐ Periodic re-keying

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

Save **Cancel**

Figure 7 – TPS Configuration Details

5.2.3. Administer Quality of Service (QoS)

e) Continue from **Section 5.2.1**. On the **Node Details** page, select the **Quality of Service (QoS)** link as shown in **Figure 6b**.

f) The default Diffserv values are as shown in **Figure 8**. Click on the **Save** button.

The screenshot shows the Nortel CS 1000 Element Manager interface. The top header is purple with 'NORTEL' and 'CS 1000 ELEMENT MANAGER'. Below the header, a navigation tree on the left lists various system components. The main content area shows the 'Node ID: 3000 - Quality of Service (QoS)' configuration page. It includes a breadcrumb trail: 'System » IP Network » IP Telephony Nodes » Node Details » Quality of Service (QoS)'. The configuration section is titled 'Diffserv Codepoint (DSCP)' and contains several settings: 'Enable Nortel automatic QoS' (unchecked), 'Control packets' (40), 'Voice packets' (46), 'VLAN tagging' (unchecked), and '802.1Q bits value (802.1P)' (6). At the bottom, there is a note: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' and two buttons: 'Save' and 'Cancel'.

Figure 8 – QoS Configuration Details

5.2.4. Synchronize the New Configuration

g) Continue from **Section 5.2.3**, return to the **Node Details** page (**Figure 6a**) and click on the **Save** button.

h) The **Node Saved** screen is displayed. Click on the **Transfer Now** (not shown).

i) The **Synchronize Configuration Files** screen is displayed. Check the **Signaling Server** check box and click on the **Start Sync** (not shown).

j) When the synchronization completes, check the **Signaling Server** check box and click on the **Restart Applications** (not shown).

5.3. Administer Voice Codec

5.3.1. Enable Voice Codec G711, Node IP Telephony.

- a) Select **IP Network** -> **Nodes: Servers, Media Cards** -> **Configuration** from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this Communication Server 1000 system. The **Node Details** screen is displayed. (See **Section 5.2.1** for more detail).
- b) On the **Node Details** page as shown in **Figure 6b**, click on **Voice Gateway (VGW) and Codec**.
- c) The Windstream system only supports **G711,ptime 20ms with VAD disabled**. The Windstream system does not support G729 therefore the system ensures that the **Codec G729** and **Voice Activity Detection (VAD)** checkboxes are unchecked as shown in **Figure 9**. Then click on the **Save** button

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.10.97.96** Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs

Node ID: 3000 - Voice Gateway (VGW) and Codecs

General | Voice Codes | Fax

Voice Codes

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

Figure 9 –Voice Gateway and Codec Configuration Details

- d) Synchronize the new configuration (please refer to **Section 5.2.4**)

5.3.2. Enable Voice Codecon Media Gateways.

- From the left menu of the Element Manager page in **Figure 9**, select **IP Network ->Media Gateways** menu item. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page.
- In the following screen scroll down to the **Codec G711** and uncheck **VAD**, ensure to uncheck Codec G729A as shown in **Figure 10**.

NORTEL CS 1000 ELEMENT MANAGER

Help | Logout

UCM Network Services

- Home
- Links
- Virtual Terminals
- System
 - Alarms
 - Maintenance
 - Core Equipment
 - Peripheral Equipment
 - IP Network
 - Nodes: Servers, Media Cards
 - Maintenance and Reports
 - Media Gateways**
 - Zones
 - Host and Route Tables
 - Network Address Translation (NAT)
 - QoS Thresholds
 - Personal Directories
 - Unicode Name Directory
- Interfaces
 - Engineered Values
 - Emergency Services
 - Geographic Redundancy
 - Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Views
 - Lists
 - Properties
 - Migration
- Tools
 - Backup and Restore
 - Date and Time
 - Logs and reports
- Security
 - Passwords
 - Policies
 - Login Options

VGW and IP phone codec profile

Enable echo canceller ☒

Echo canceller tail delay 128 (milliseconds)

Enable dynamic attenuation ☒

Voice activity detection threshold 1 (0 - 4 DBM)

Idle noise level 0 (0 - 1 DBM)

R factor calculation ☐

DTMF tone detection ☒

Enable low latency mode ☐

Remove DTMF delay (squelch DTMF from TDM to IP) ☒

Enable modem/fax pass through mode ☒

Enable V.21 FAX tone detection ☒

Fax TCF method 2

FAX maximum rate 14400 (bps)

FAX playback nominal delay 100 (0 - 300 milliseconds)

FAX no activity timeout 20 (10 - 32000 milliseconds)

FAX packet size 30

Codec G711 Select ☒

Codec name G711

Voice payload size 20 (ms/frame)

Voice playback (jitter buffer) nominal delay 40

Modifications may cause changes to dependent settings

Voice playback (jitter buffer) maximum delay 80

Modifications may cause changes to dependent settings

VAD ☐

+ Codec G729A Select ☐

+ Codec G723.1 Select ☐

+ Codec T38 FAX Select ☒

+ QoS

+ Media Based CLID

Figure 10 – Media Gateways Configuration Details

- Then scroll down to the bottom of the page and click on the **Save** button.

5.4. Zones and Bandwidth Management

This section describes the steps to create 2 zones: zone 10 for VGW and IP sets, and zone 255 for SIP Trunk.

5.4.1. Create a zone for IP phones (zone 10)

The following figures show how to configure a zone for VGW and IP sets for bandwidth management purposes. The bandwidth strategy can be adjusted to preference.

a) Select **IP Network** -> **Zones** configuration from the left pane, click on the **Bandwidth Zones** as shown in **Figure 11**.

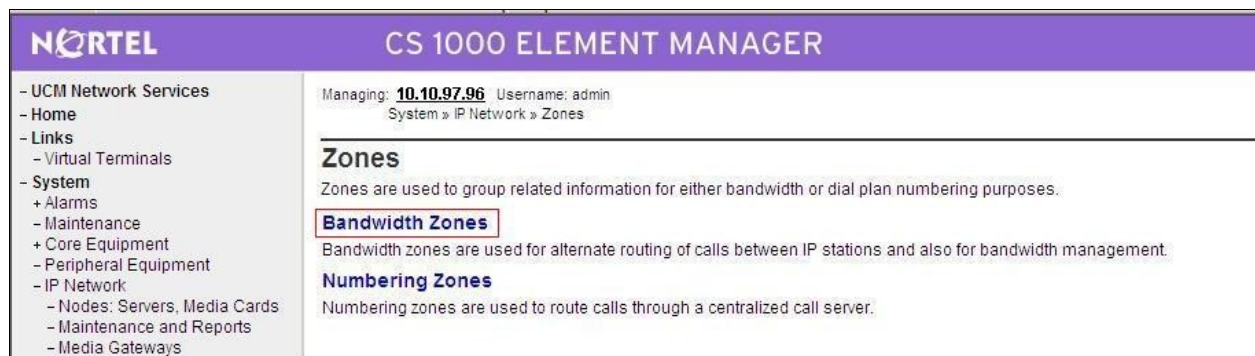


Figure 11 –Zones Page

b) The **Bandwidth Zones** screen is displayed as shown in **Figure 12**. Click **ADD** to create new zone for IP Phones.

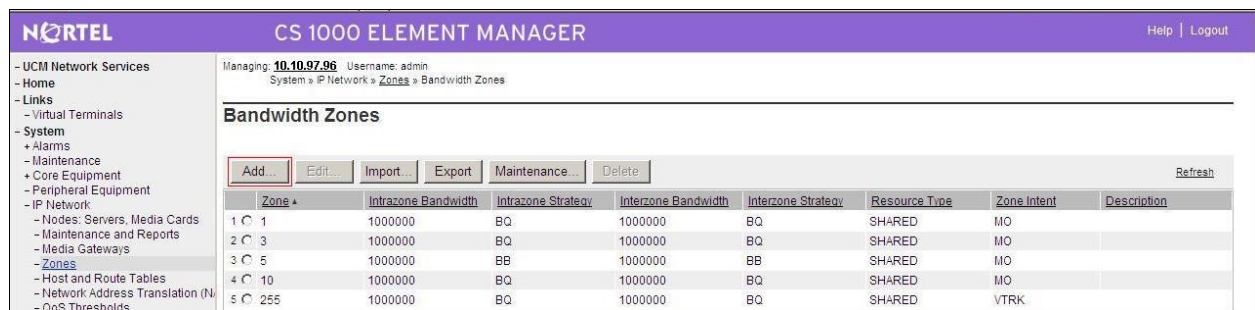


Figure 12 –Bandwidth Zones

c) Select the values as shown (in red box) in **Figure 13** and click on the **Submit** button.

- INTRA_STGY: Codec configuration for local calls.
- INTER_STGY: Codec configuration for the calls over trunk.
- BQ: G711 is first choice and G729 is second choice.
- BB: G729 is first choice and G711 is second choice.
- MO: is used for IP phones, VGWetc
- VTRK: is used for virtual trunk.

Input Description	Input Value
Zone Number (ZONE):	10 (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

Figure 13 –Bandwidth Management ConfigurationDetails – IP phone

5.4.2. Create a zone for virtual SIP trunk (zone 255)

Follow **Section 5.4.1** to create a zone for the virtual trunk. The difference is in **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in **Figure 14** and then click on the **Submit** button

Input Description	Input Value
Zone Number (ZONE):	255 (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

Figure 14 –Bandwidth Management ConfigurationDetails –virtual SIP trunk

5.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP connection between SIP Signaling Gateway (SSG) to NRS/SPS.

5.5.1. Integrated Services Digital Network (ISDN)

- a) Select **Customers** in the left pane. The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. The **Customer 00Edit** page will appear (not shown). Select the **Feature Packages** option from this page.
- b) The screen is updated with a listing of feature packages populated below **Feature Packages** (not all features shown in **Figure 15**below). Select **Integrated Services Digital Network** to edit its parameters. The screen is updated with parameters populated below **Integrated Services Digital Network**. Click on **Integrated Services Digital Network (ISDN)**, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button at the bottom of the page (not shown).

NORTEL CS 1000 ELEMENT MANAGER

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
+ Alarms
- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways

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

Integrated Services Digital Network: ☒

- Virtual Private Network Identifier: 1 (1 - 10383)
- Private Network Identifier: 1 (1 - 10383)
- Node DN:
- Multi-location Business Group: 0 (0 - 05535)

Figure 15 –Customer – ISDN Configuration

5.5.2. Administer SIP Trunk Gateway to NRS/SPS

- Select **IP Network** -> **Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this Communication Server 1000 system. The **Node Details** screen is displayed as shown in **Figure 6b, Section 5.2.1**.
- On the **Node Details** screen, select **Gateway (SIPGw)**.
- Under **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 16**. The parameters (highlighted in red boxes) are filled in, and were obtained when a user created a SIP profile on the NRS/SPS (these are shown in **Section 5.8.2**).

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.10.97.96 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 3000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw)
SIP domain name: bwdev7.com
Local SIP port: 5060
Gateway endpoint name: car3-ssg-carrier
Gateway password:
Application node ID: 3000
Enable failsafe NRS: ☐
SIP ANAT: ☒ IPv4 ☐ IPv6

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below.
Monitor IP:
Add
Monitor addresses:
Remove

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

Save Cancel

Figure 16 – Virtual Trunk Gateway Configuration Details

NORTEL
CS 1000 ELEMENT MANAGER

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

Managing: 10.10.97.96 Username: admin
 System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 3000 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

SIP Gateway Settings

TLS Security: Security Disabled

Port: 5061 (1 - 65535)

Number of byte re-negotiation: 0

Options: ☐ Client authentication
☐ X509 certificate authority

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.97.183
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: UDP

Options: ☒ Support registration
☐ Primary CDS proxy

* Required Value.

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

Save
Cancel

e) On the same page as shown in **Figure 17**, scroll down the parameters box to the **SIP URI Map** section.

- **National:** leave this SIP URI field as blank
- **Subscriber:** leave this SIP URI field as blank
- **Special Number:** leave this SIP URI field as blank
- **Unknown:** leave this SIP URI field as blank

- **UDP**: leave this SIP URI field as blank
- **CDP**: leave this SIP URI field as blank
- **Special Number**: leave this SIP URI field as blank
- **Vacant number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank

The remaining fields can be left at their default values as shown in **Figure 18**. Then click on the **Save** button.

Figure 18 – Virtual Trunk Gateway Configuration Details

f) **Synchronize** the new configuration (please refer to **Section 5.2.4**).

g) After the configuration is completed on the NRS/SPS, the Gateway endpoint entry for the SSG (in this case is car3-ssg-carrier) shows the IP address of the SSG which is successfully registered to the NRS/SPS) as shown in **Figure 19**. Please refer to **Section 5.8** for more details on the NRS/SPS configuration.

Gateway Endpoints (2)							
ID	Supported Protocols	SIP mode	Call Signaling IP	Description	# of Routing Entries	Context	
1 acme	Static SIP endpoint	Proxy Mode	10.10.97.184		5	bwwdev7.com / udp / cdp	
2 car3-ssg-carrier	Dynamic SIP endpoint	Proxy Mode	10.10.97.178		1	bwwdev7.com / udp / cdp	

Figure 19–SSG is registered successfully to the NRS/SPS

5.5.3. Administer Virtual D-Channel

a) 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 as shown in **Figure 20**. Click **Add** button.

NORTEL CS 1000 ELEMENT MANAGER

Managing: **10.10.97.96** 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)
- [TMDI Diagnostics](#) (LD 96)
- [D-Channel Expansion Diagnostics](#) (LD 48)

Configuration

Choose a D-Channel Number: and type:

- Channel: 11	Type: DCH	Card Type: DCIP	Description: sipi	<input type="button" value="Edit"/>
- Channel: 100	Type: DCH	Card Type: DCIP	Description: VoIP	<input type="button" value="Edit"/>

Figure 20 – D-Channels

b) The D-Channels 100 Property Configuration screen is displayed next as shown in **Figure 21**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **D channel Card Type (CTYP):** D-Channel is over IP (DCIP)
- **Designator (DES):** A descriptive name
- **Interface type for D-channel (IFC):** Meridian Meridian1 (SL1)
- **Release ID of the switch at the far end (RLS):**25

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.10.97.96 Username: admin
Routes and Trunks » D-Channels » D-Channels 100 Property Configuration

D-Channels 100 Property Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	VoIP
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="button" value="more PRI"/>
Secondary PRI2 loops:	
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	25
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	1800 Range: 0 - 3700
+ Basic options (BSOOPT)	
- Advanced options (ADVOPT)	
- Layer 3 call control message count per 5 second time interval:	300 Range: 60 - 350
- Number of Status Enquiry Messages sent within 128 ms:	1
- Map channel number to timeslots on a PRI2 loop:	<input checked="" type="checkbox"/>
+ H323 Overlap Signaling Settings (H323)	
- Overlap Timer:	
- Multilocation Business Group Allowed:	<input type="checkbox"/>
- Network Attendant Service Allowed:	<input checked="" type="checkbox"/>
+ Link Access Protocol for D-channel (LAPD)	

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Figure 21 – D-Channels Configuration Details

- c) Click on the **Advanced options (ADVOPT)**, check on the **Network Attendant Service Allowed** check box as shown in **Figure 21**. Other fields are left as default.
- d) Click on the **Basic Options** and click on the **Edit** button at the **Remote Capabilities (RCAP)** attribute. The **Remote Capabilities Configuration** page will appear. Then check on the **ND2** and the **MWI** checkboxes as shown in **Figures 22 and 23**.

NORTEL CS 1000 ELEMENT MANAGER

D-Channel Configuration

Action Device And Number (ADAN): DCH

D channel Card Type: DCIP

Designator: VoIP

Recovery to Primary: ☐

PRI loop number for Backup D-channel:

User: Integrated Services Signaling Link Dedicated (ISLD)

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

Country: ETS 300 = 102 basic protocol (ETSI)

D-Channel PRI loop number:

Primary Rate Interface: [more PRI](#)

Secondary PRI2 loops:

Meridian 1 node type: Slave to the controller (USR)

Release ID of the switch at the far end: 7

Central Office switch type: 100% compatible with Bellcore standard (STD)

Integrated Services Signaling Link Maximum: 4000 Range: 1 - 4000

Signalling server resource capacity: 1800 Range: 0 - 3700

Primary D-channel for a backup DCH: Range: 0 - 254

- PINX customer number:

- Progress signal:

- Calling Line Identification:

- Output request Buffers: 32

- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)

- Channel Negotiation option: No alternative acceptable, exclusive, (1)

- Remote Capabilities: **Edit**

- B channel Service messaging: ☐

[+ - Change protocol timer value \(TIMR\)](#)

[+ Advanced options \(ADVOPT\)](#)

[+ Feature Packages](#)

[Submit](#) [Refresh](#) [Delete](#) [Cancel](#)

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Figure 22 – D-Channel Configuration Details

NORTEL CS 1000 ELEMENT MANAGER	
<ul style="list-style-type: none"> - UCM Network Services - Home - Links <ul style="list-style-type: none"> - Virtual Terminals - System <ul style="list-style-type: none"> + Alarms - Maintenance + Core Equipment - Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks <ul style="list-style-type: none"> - Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans <ul style="list-style-type: none"> - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones <ul style="list-style-type: none"> - Templates - Reports - Views - Lists - Properties - Migration - Tools <ul style="list-style-type: none"> + Backup and Restore - Date and Time + Logs and reports - Security <ul style="list-style-type: none"> + Passwords 	<ul style="list-style-type: none"> Call completion on busy using object identifier (CCBO) <input type="checkbox"/> Call completion on busy for QSIG and EuroSDN BRI (CCBS) <input type="checkbox"/> Call completion on no response using integer value (CCNI) <input type="checkbox"/> Call completion on no response using object identifier (CCNO) <input type="checkbox"/> Call completion to no reply for QSIG and EuroSDN BRI (CCNR) <input type="checkbox"/> Network call park (CPK) <input type="checkbox"/> Connected line identification presentation (COLP) <input type="checkbox"/> Call transfer integer (CTI) <input type="checkbox"/> Call transfer object (CTO) <input type="checkbox"/> Diversion info. is sent using integer value (DV1I) <input type="checkbox"/> Diversion info. is sent using object identifier (DV1O) <input type="checkbox"/> Rerouting requests processed using integer value (DV2I) <input type="checkbox"/> Rerouting requests processed using object identifier (DV2O) <input type="checkbox"/> Diversion info. sent. rerouting requests processed (DV3I) <input type="checkbox"/> EuroSDN - div. info sent. rerouting req. processed (DV3O) <input type="checkbox"/> Call transfer notification and invocation to EuroSDN (ECTO) <input type="checkbox"/> Malicious call identification (MCID) <input type="checkbox"/> MCDN QSIG conversion (MQC) <input type="checkbox"/> Remote D-channel is on a MSDL card (MSL) <input type="checkbox"/> Message waiting interworking with DMS-100 (MWI) <input checked="" type="checkbox"/> Network access data (NAC) <input type="checkbox"/> Network call trace supported (NCT) <input type="checkbox"/> Network name display method 1 (ND1) <input type="checkbox"/> Network name display method 2 (ND2) <input checked="" type="checkbox"/> Network name display method 3 (ND3) <input type="checkbox"/>

Figure 23 – Remote Capabilities Configuration Details

- e) Click on the **Return – Remote Capabilities** button (not shown).
 f) Click on the **Submit** button (not shown).

5.5.4. Administer Virtual Super-Loop

Select **System** -> **Core Equipments** -> **Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click “**Add**” button to create a new one as shown in **Figure 24**. In this example, superloop4, 96, 100 and 124 has been added and is being used.



Figure 24 – Administer Virtual Super-Loop Page

5.5.5. Administer Virtual SIP Routes

a) Select **Routes and Trunks** -> **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 0** is being used. Click on the **Add route** button as shown in **Figure 25**.



Figure 25 – Add route

b) The **Customer 0**, New **Route Configuration** screen is displayed next. Scroll down until the **Basic Configuration** Section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields as shown in **Figures 26**.

- **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.
- 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 255 (created in **Section 5.4.2**).
- For the **Node ID of signaling server of this route (NODE)** field, enter the node number 3000 (created in **Section 5.2.1**).

- Select **SIP** (SIP) from the drop-down list for the **Protocol ID for the route (PCID)** field.
- 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.
 - o **Mode of operation (MODE)**: Route uses ISDN Signalling Link (ISLD)
 - o **D channel number (DCH)**: D-Channel number 100 (created in **Section 5.5.3**)
 - o **Network calling name allowed (NCNA)**: Check the field.
 - o **Network call redirection (NCRD)**: Check the field.
 - o **Insert ESN access code (INAC)**: Check the field.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 10.10.97.96 Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 100 Property Configuration

Customer 0, Route 100 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE): RDB
 Customer number (CUST): 00
 Route number (ROUT): 100
 Designator field for trunk (DES): 100
 Trunk type (TKTP): TIE
 Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)
 Access code for the trunk route (ACOD): 8100
 Trunk type M911P (M911P): ☐
 The route is for a virtual trunk route (VTRK): ☒
 Zone for codec selection and bandwidth management (ZONE): 00255 (0 - 8000)
 Node ID of signaling server of this route (NODE): 3000 (0 - 9999)
 Protocol ID for the route (PCID): SIP (SIP)
 Print correlation ID in CDR for the route (CRID): ☐
 Integrated services digital network option (ISDN): ☒
 Mode of operation (MODE): Route uses ISDN Signaling Link (ISLD)
 D channel number (DCH): 100 (0 - 254)
 Interface type for route (IFC): Meridian M1 (SL1)
 Private network identifier (PNI): 00001 (0 - 32700)
 Network calling name allowed (NCNA): ☒
 Network call redirection (NCRD): ☒
 Trunk route optimization (TRO): ☐
 Recognition of DT12 ABCD FALT signal for ISL (FALT): ☐
 Channel type (CHTY): B-channel (BCH)
 Call type for outgoing direct dialed TIE route (CTYP): Unknown Call type (UKWN)
 Insert ESN access code (INAC): ☒

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Figure 26 – Route Configuration Details

-Click on **Basic Route Options**, check the **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)**, input **DCNO 1** for both **Day IDC Tree Number** and **Night IDC Tree Number** as shown in **Figure 27**.

NORTEL CS 1000 ELEMENT MANAGER

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
+ Alarms
- Maintenance
- Core Equipment
- Loops
- Superloops
- MSDLMISP Cards
- Conference/TDS/Multifrequen
- Tone Senders and Detectors
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Geographic Redundancy
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
- Views
- Lists
- Properties
- Migration
- Tools
+ Backup and Restore
- Date and Time
- Logs and reports
- Security
+ Passwords
+ Policies

- Mobile extension timer (MBXT): 0 (0 - 8000 milliseconds)
Calling number dialing plan (CNDP): Unknown (UKWN)

- Basic Route Options

Attendant announcement (ATAN): No Attendant Announcement (NO)
Billing number required (BILN): ☐
Call detail recording (CDR): ☒
- CDR records generated on incoming calls (INC): ☒
- CDR record printing content option for redirected calls (LAST): ☒
- Time to answer output in CDR (TTA): ☐
- CDR ACD Q initial connection records to be generated (QREC): ☒
- CDR on outgoing calls (OAL): ☒
- CDR on outgoing toll calls (OTL): ☐
- Answered call identification allowed (AIA): ☒
- CDR timing starts on answer supervision of outgoing calls (OAN): ☒
- outpulsed digits in CDR (OPD): ☒
- Number of digits printed (NDP): EXC 0
North American toll scheme (NATL): ☒
Controls or timers (CNTL): ☐
Conventional (Tie trunk only) (CNVT): ☐
Incoming DID digit conversion on this route (IDC): ☒
- Day IDC tree number (DCNO): 1 (0 - 254)
- Night IDC tree number (NDNO): 1 (0 - 254)
- Display external dialed digits (DEXT): ☐
Multifrequency compelled or MFC signaling (MFC): No MFC (NO)
Process notification networked calls (PNNC): ☐

+ Network Options
+ General Options
+ Advanced Configurations

Submit Refresh Delete Cancel

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Figure 27 – Route Configuration Details

c) Click on the **Submit** button.

5.5.6. Administer Virtual Trunks

a) From the EM, select **Routes and Trunks->Route and Trunks**, the Route list is now updated with the newly added route. In the example, the Route 100 was being added. Click on the **Add trunk** button next to the newly added route 100 as shown in **Figure 28**.

The screenshot displays the 'Routes and Trunks' management interface. On the left is a navigation menu with options like 'UCM Network Services', 'Home', 'Links', 'Virtual Terminals', 'System', 'Alarms', 'Maintenance', 'Core Equipment', 'Loops', 'Superloops', 'MSDL/MISP Cards', and 'Conference/TDS/Multifrequen'. The main area shows the current management session for IP 10.10.97.96 as 'admin'. Below this, a table lists routes for 'Customer: 0'. The table has columns for route identification, type, and description. Two routes are listed: Route 11 (Type: TIE, Description: SIPL) and Route 100 (Type: TIE, Description: 100). Each route has an 'Edit' button and an 'Add trunk' button. The 'Add trunk' button for Route 100 is highlighted with a red rectangular box.

Customer: 0	Total routes: 5	Total trunks: 100		
+ Route: 11	Type: TIE	Description: SIPL	Edit	Add trunk
+ Route: 100	Type: TIE	Description: 100	Edit	Add trunk

Figure 28 – Route and Trunks Page

b) The **Customer 0, Route 100, Trunk 1 Property Configuration** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields. The Media Security (sRTP) needs to be disabled at the trunk level by editing the **Class of Service** (CLS) at the bottom basic trunk configuration page. Click on the **Edit** button as shown in **Figure 29**.

- The Multiple trunk input number (**MTINPUT**) field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, 32 trunks were created.
- Trunk data block (**TYPE**): IP Trunk (IPTI)
- Terminal Number (**TN**): Available terminal number (created in **Section 5.5.4**)
- Designator field for trunk (**DES**): A descriptive text
- Extended Trunk (**XTRK**): Virtual trunk (VTRK)
- Route number, Member number (**RTMB**): Current route number and starting member
- Card Density: 8D
- Start arrangement Incoming (**STRI**): IMM
- Start arrangement Outgoing (**STRO**): IMM
- Trunk Group Access Restriction (**TGAR**): Desired trunk group access restriction level
- Channel ID for this trunk (**CHID**): An available starting channel ID

Figure 29 – New Trunk Configuration Details

c) For **Media Security**, select **Media Security Never (MSNV)**. Enter the remaining values for the specified fields as shown in **Figure 30**. Scroll down to the bottom of the screen and click **Return Class of Service** and then click on the **Save** button (not shown)

NORTEL CS 1000 ELEMENT MANAGER

- Class of Service

Input Description	Input Value
- ACD Priority:	ACD Priority not required (APN)
- Analog Semi-Permanent Connections:	Analog Semi-Permanent Connections Denied (SPCD)
- ARF Supervised COT:	
- Barring:	
- Battery Supervised COT:	
- Busy Tone Supervised COT:	
- Calling Line Identification:	
- Calling party:	Calling party Denied (CND)
- Central Office Ringback:	
- Centrex Switchhook Flash:	Centrex Switchhook Flash Denied (THFD)
- Dial Pulse:	Digitone (DTN)
- DTR PAD value:	
- Echo Canceling:	Echo Canceling Denied (ECD)
- Hong Kong DTI:	
- Loop Break Supervised COT:	
- Make-break ratio for dial pulse:	10 pulses per second (P10)
- Manual Incoming:	Manual Incoming Denied (MID)
- Media Security:	Media Security Never (MSNV)
- Network Hook Flash Over M911P:	
- Polarity:	
- Priority:	Low Priority (LPR)
- Restriction level:	Unrestricted (UNR)
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP)
- Short or long line:	
- Transmission Class of Service:	Non-Transmission Compensated (NTC)
- Warning Tone:	Warning Tone Allowed (WTA)
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP)
- ARF Supervised COT:	

Return Class of Service Cancel

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Figure 30 – Class of Service ConfigurationDetailsPage

5.5.7. Administer Calling Line Identification Entries

a) Select **Customers->00->ISDN and ESN Networking**. Click on **Calling Line Identification Entries** as shown in Figure 31.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: 10.10.97.96 Username: admin
Customers » Customer 00 » Customer Details » ISDN and ESN Networking

ISDN and ESN Networking

General Properties

Flexible trunk to trunk connection option:

Flexible orbiting prevention timer:

Country code:
(0 - 9999)

Code for processing the called number

National access code:

International access code:

Options: ☒ Transfer on ringing of supervised external trunks
☒ Connection of supervised external trunks

Network option: ☒ Coordinated dialing plan routing

Integrated services digital network: ☒

Microsoft converged office dialing plan:

Private dialing plan for non-DID users: ☐ Coordinated dialing plan
☐ Uniform dialing plan

Calling Line Identification

Information for incoming/outgoing calls:

Size:
(0 - 4000)

Country code:
(0 - 9999)

Code displayed as part of calling number

[Calling Line Identification Entries](#)

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Figure 31 – ISDN and ESN Networking

b) Click on **Add** as shown in **Figure 32**.

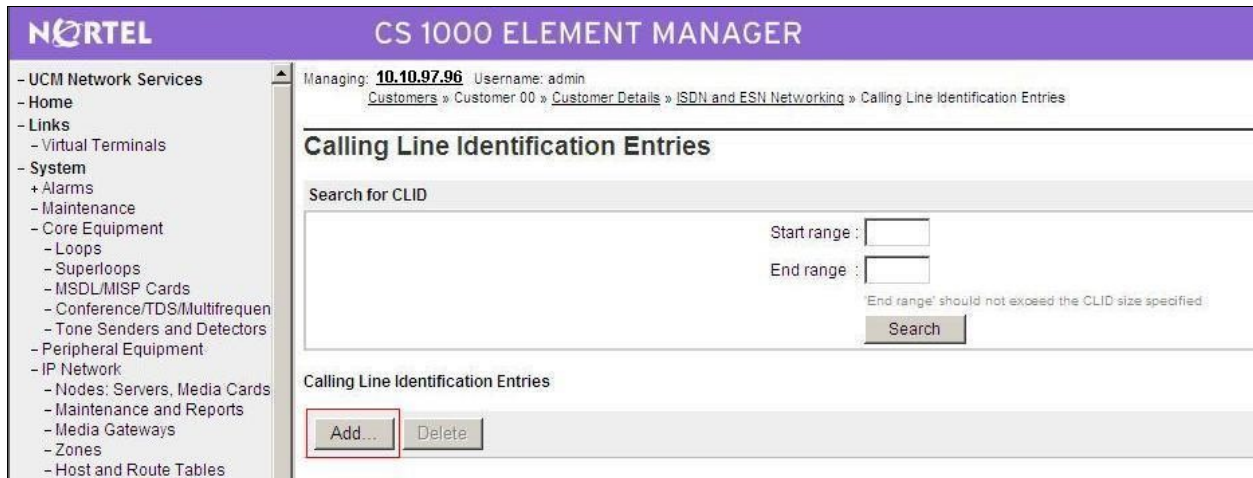


Figure 32 – Calling Line Identification Entries

c) Add entry 0 as shown in **Figure 33**:

- **National Code**: leave as blank
- **Local Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits – 501287. This **Local Code** will be used for call display purpose of outbound international call configuration in **Section 5.6.6** in which the **Special Number 011** is associated with Call Type = Unknown.
- **Home Location Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 501287. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 501287. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Calling Party Name Display**: Uncheck for **Roman characters**.

Click on the **Save** button as shown in **Figure 33**.

The screenshot shows the 'Edit Calling Line Identification 0' configuration page in the CS 1000 ELEMENT MANAGER. The left sidebar contains a navigation menu with categories like UCM Network Services, System, Interfaces, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, and Tools. The main content area is titled 'Edit Calling Line Identification 0' and contains three sections: 'General Properties', 'Emergency Services Access', and 'Calling Party Name Display'. The 'General Properties' section includes fields for National Code, Local Code, Home Location Code, and Local Steering Code, all with dropdown menus. The 'Emergency Services Access' section includes a checkbox for 'Emergency Options' and a checkbox for 'Append the originating directory number for emergency services access calls'. The 'Calling Party Name Display' section includes a checkbox for 'Roman characters' and a dropdown for 'Display Format'. The 'Save' button is highlighted at the bottom right of the page.

Figure 33 – Edit Calling Line Identification 0

5.5.8. Enable External Trunk to Trunk Transferring

This section shows how to enable External Trunk to Trunk Transferring feature which is a mandatory configuration to make call transfer and conference work properly over SIP trunk.

- Login Call Server Overlay CLI (please refer to **Section 5.1.2** for more detail)
- Allow External Trunk to Trunk Transferring for Customer Data Block by using **LD 15**

```
>ld 15
CDB000
MEM AVAIL: (U/P): 33600126  USED UP: 8345621 954062  TOT: 45579868
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
...
TRNX YES
EXTT YES
...
```

5.6. Administer Dialing Plans

5.6.1. Define ESN Access Codes and Parameters (ESN)

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 34**

The screenshot displays the Nortel CS 1000 Element Manager interface. The top header bar is purple with the Nortel logo on the left and 'CS 1000 ELEMENT MANAGER' on the right. Below the header, a status bar shows 'Managing: 10.10.97.96 Username: admin' and 'Dialing and Numbering Plans » Electronic Switched Network (ESN)'. The main content area is titled 'Electronic Switched Network (ESN)'. On the left, a navigation pane lists various system components, with 'Dialing and Numbering Plans' and 'Electronic Switched Network' highlighted. The main pane shows a tree structure for 'Customer 00'. Under 'Network Control & Services', several items are listed, with 'ESN Access Codes and Parameters (ESN)' highlighted. Under 'Coordinated Dialing Plan (CDP)', 'Local Steering Code (LSC)', 'Distant Steering Code (DSC)', and 'Trunk Steering Code (TSC)' are listed. Under 'Numbering Plan (NET)', 'Access Code 1' and 'Access Code 2' are listed, each with a sub-tree of parameters. Several parameters are highlighted with red boxes: 'ESN Access Codes and Parameters (ESN)', 'Digit Manipulation Block (DGT)', 'Home Area Code (HNPA)', 'Flexible CLID Manipulation Block (CMDB)', 'Free Calling Area Screening (FCAS)', 'Free Special Number Screening (FSNS)', 'Route List Block (RLB)', 'Incoming Trunk Group Exclusion (ITGE)', 'Network Attendant Services (NAS)', 'Home Location Code (HLOC)', 'Location Code (LOC)', 'Numbering Plan Area Code (NPA)', 'Exchange (Central Office) Code (NXX)', 'Special Number (SPN)', and 'Network Speed Call Access Code (NSCL)'.

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

Managing: 10.10.97.96 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)
 - Home Area Code (HNPA)
 - Flexible CLID Manipulation Block (CMDB)
 - Free Calling Area Screening (FCAS)
 - Free Special Number Screening (FSNS)
 - 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)
 - Numbering Plan (NET)
 - Access Code 1
 - Home Location Code (HLOC)
 - Location Code (LOC)
 - Numbering Plan Area Code (NPA)
 - Exchange (Central Office) Code (NXX)
 - Special Number (SPN)
 - Network Speed Call Access Code (NSCL)
 - Access Code 2
 - Home Location Code (HLOC)
 - Location Code (LOC)
 - **Numbering Plan Area Code (NPA)**
 - Exchange (Central Office) Code (NXX)
 - **Special Number (SPN)**
 - Network Speed Call Access Code (NSCL)

Figure 34 –ESN Configuration Details

b) In the **ESN Access Codes and Basic Parameters** page, define **NARS Access Code 2** as shown in **Figure 35**.

c) Click Submit button (not shown).

Figure 35 – ESN Access Codes and Basic Parameters

5.6.2. Associate NPA and SPN call to ESN Access Code 2

a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail), change Customer Net Data block by using **LD 15**.

```
>ld 15
CDB000
MEM AVAIL: (U/P): 35600086   USED U P: 8325631 954152   TOT: 44879869
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
AC1xNPAxSPN   ----- > (Set NPA, SPN not to associate to ESN Access Code 1)
FNP
CLID
...
```

b) Verify Customer Net Data block by using LD 21

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1
AC2INTL NPA SPN NXX LOC ----- > (NPA, SPN are associated to ESN Access Code 2)
FNP YES
...
```

5.6.3. Digit Manipulation Block (DMI)

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Digit Manipulation Block (DGT)** as shown in **Figure 34**.

b) In the Choose a DMI Number field, select an available DMI from the drop-down list and click to **Add** as shown in **Figure 36**.

c) Enter the **Number of leading digits to be Deleted (Del)** field and select the **Call Type to be used by the manipulated digits (CTYP)** and then click **Submit** (see **Section 5.6.4**).

5.6.4. Digit Manipulation Block (DMI) for Outbound Call

In the following steps show how to add DMI for the outbound call, there are 4 indexes, which were added to the Digit Manipulation Block List (14 and 15).

a) Select **Dialing and Numbering Plans** ---> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Digit Manipulation Block (DGT)** as above.

b) In the Choose a DMI Number field, select an available DMI from the drop-down list and click onto **Add** button as shown in **Figure 36**.

The screenshot displays the Nortel CS 1000 Element Manager web interface. The top header is purple with 'NORTEL' and 'CS 1000 ELEMENT MANAGER' in white. A left sidebar contains a tree view with categories like 'UCM Network Services', 'Home', 'Links', 'Virtual Terminals', 'System', 'Alarms', 'Maintenance', 'Core Equipment', 'Peripheral Equipment', 'IP Network', 'Interfaces', 'Engineered Values', and 'Emergency Services'. The main content area has a breadcrumb trail: 'Managing: 10.10.97.96 Username: admin > Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Digit Manipulation Block List'. Below this, the title 'Digit Manipulation Block List' is shown. A form prompts the user to 'Please choose the' followed by a dropdown menu currently showing 'Digit Manipulation Block Index 1' and a 'to Add' button. Below the form, there is a table with two entries: 'Digit Manipulation Block Index -- 14' and 'Digit Manipulation Block Index -- 15', each with an 'Edit' button next to it.

Figure 36 – Add a DMI

c) Add DMI_14: Enter 0 for the **Number of leading digits to be Deleted (Del)** field and select **NPA** for the **Call Type to be used by the manipulated digits (CTYP)** and then click on **Submit** button as shown in **Figure 37a**

Figure 37a – DMI_14 Configuration Details

d) Add DMI_15: Enter 1 for the **Number of leading digits to be Deleted (Del)** field and select **NPA** for the **Call Type to be used by the manipulated digits (CTYP)** and then click on **Submit** button as shown in **Figure 37b**

Figure 37b – DMI_15 Configuration Details

5.6.5. Route List Block (RLB) (RLB 14)

This session shows how to add a RLB associated with the DMI created in **Section 5.6.4**.

a) 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)** as shown in **Figure 34**.

b) Select an available value in the textbox for the **route list index** (in this case is 14) and click onto **Add** button as shown in **Figure 38**.

Figure 38 – Add a Route List Block.

c) Enter the following values for the specified fields, and retain the default values for the remaining fields (**Figure 39a**). Scroll down to the bottom of the screen, and click on the **Submit** button.

- **Route number (ROUT):** 100 (created in **Section 5.5.5**)
- **Digit Manipulation Index (DMI):** 14 (created in **Section 5.6.4**)
- **Incoming CLID Table:** 0 (created in **Section 5.5.7**)

Figure 39a – RLB_14 Route List Block Configuration Details

5.6.6. Route List Block (RLB) (RLB 15)

a) 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)** as shown in **Figure 34**.

b) Select an available value in the textbox for the **route list block index** (in this case is 15) and click on the “**to Add**” button as shown in **Figure 38**.

c) Enter the following values for the specified fields, and retain the default values for the remaining fields (**Figure 39b**). Scroll down to the bottom of the screen, and click on the **Submit** button.

- **Route number (ROUT)** : 100 (created in **Section 5.5.5**)
- **Digit Manipulation Index (DMI)**: 15 (created in **Section 5.6.4**)
- **Incoming CLID Table**: 0 (created in **Section 5.5.7**)

The screenshot displays the 'CS 1000 ELEMENT MANAGER' interface. The left sidebar contains a navigation tree with categories like 'UCM Network Services', 'Links', 'System', 'Customers', 'Routes and Trunks', 'Dialing and Numbering Plans', 'Phones', 'Tools', and 'Security'. The 'Dialing and Numbering Plans' category is expanded, showing 'Electronic Switched Network' as the selected option. The main content area is titled 'Data Entry of a Route List Block' and shows the configuration for 'Route List Block Index: 15'. The 'General Properties' section includes 'Entry Number for the Route List' set to 0. The 'Indexes' section contains several dropdown menus and checkboxes: 'Time of Day Schedule' (0), 'Facility Restriction Level' (0), 'Digit Manipulation Index' (15), 'ISL D-Channel Down Digit Manipulation Index' (0), 'Free Calling Area Screening Index' (0), 'Free Special Number Screening Index' (0), 'Business Network Extension Route' (unchecked), and 'Incoming CLID Table' (0). The 'Options' section includes 'Local Termination entry' (unchecked), 'Route Number' (100), 'Skip Conventional Signaling' (unchecked), 'Use Tone Detector' (unchecked), 'Conversion to LDN' (unchecked), and 'Expensive Route' (unchecked).

Figure 39b – RLB_15 Route List Block Configuration Details

5.6.7. Inbound Call – Incoming Digit Translation Configuration

This section describes the steps for receiving the calls from PSTN via the Windstream system.

- a) Select **Dialing and Numbering Plans** ->**Incoming Digit Translation** from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button as shown in **Figure 40**.



Figure 40 – Incoming Digit Translation

- b) Click on the **New DCNO** to create the digit translation mechanism. In this example, Digit Conversion Tree Number 1 has been created as shown in **Figure 41**.



Figure 41 – Incoming Digit Conversion Property

c) Detail configuration of the Digit Conversion Tree Configuration is shown in **Figure 42**. The **Incoming Digits** can be added to map to the Converted Digits which would be the Communication Server 1000 system phones DN. This **DCN0** has been assigned to route 100 as shown in **Figure 26** and **27**.

In the following configuration, the incoming call from PSTN with the prefix 501287xxxx will be translated to DN xxxx. The DID number 5012871072 is translated to 1700 for Voicemail accessing purpose.

NORTEL		CS 1000 ELEMENT MANAGER		Help Logout																																																																											
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones - Templates - Reports - Views - Lists - Properties - Migration - Tools		Managing 10.10.97.96 Username: admin Dialing and Numbering Plans » Incoming Digit Translation » Customer 00 » Digit Conversion Tree 1 Configuration																																																																													
		Digit Conversion Tree 1 Configuration Regular IDC tree Send calling party DID disabled																																																																													
		<div> <input type="button" value="Add..."/> <input type="button" value="Delete IDC"/> <input type="button" value="Delete IDC tree"/> <input type="button" value="Refresh"/> </div>																																																																													
		<table> <thead> <tr> <th></th><th>Incoming Digits ▲</th><th>Converted Digits</th><th>CPND Name</th><th>CPND language</th></tr> </thead> <tbody> <tr><td>1</td><td>5012871070</td><td>1070</td><td></td><td>Roman characters</td></tr> <tr><td>2</td><td>5012871071</td><td>1071</td><td></td><td>Roman characters</td></tr> <tr><td>3</td><td>5012871072</td><td>1700</td><td></td><td>Roman characters</td></tr> <tr><td>4</td><td>5012871073</td><td>1073</td><td></td><td>Roman characters</td></tr> <tr><td>5</td><td>5012871074</td><td>1074</td><td></td><td>Roman characters</td></tr> <tr><td>6</td><td>5012871490</td><td>1490</td><td></td><td>Roman characters</td></tr> <tr><td>7</td><td>5012871492</td><td>1492</td><td></td><td>Roman characters</td></tr> <tr><td>8</td><td>5012871493</td><td>1493</td><td></td><td>Roman characters</td></tr> <tr><td>9</td><td>5012871494</td><td>1494</td><td></td><td>Roman characters</td></tr> <tr><td>10</td><td>5012871495</td><td>1495</td><td></td><td>Roman characters</td></tr> <tr><td>11</td><td>5012871496</td><td>1496</td><td></td><td>Roman characters</td></tr> <tr><td>12</td><td>5012871497</td><td>1497</td><td></td><td>Roman characters</td></tr> <tr><td>13</td><td>5012871498</td><td>1498</td><td></td><td>Roman characters</td></tr> <tr><td>14</td><td>5012871499</td><td>1499</td><td></td><td>Roman characters</td></tr> </tbody> </table>				Incoming Digits ▲	Converted Digits	CPND Name	CPND language	1	5012871070	1070		Roman characters	2	5012871071	1071		Roman characters	3	5012871072	1700		Roman characters	4	5012871073	1073		Roman characters	5	5012871074	1074		Roman characters	6	5012871490	1490		Roman characters	7	5012871492	1492		Roman characters	8	5012871493	1493		Roman characters	9	5012871494	1494		Roman characters	10	5012871495	1495		Roman characters	11	5012871496	1496		Roman characters	12	5012871497	1497		Roman characters	13	5012871498	1498		Roman characters	14	5012871499	1499		Roman characters
	Incoming Digits ▲	Converted Digits	CPND Name	CPND language																																																																											
1	5012871070	1070		Roman characters																																																																											
2	5012871071	1071		Roman characters																																																																											
3	5012871072	1700		Roman characters																																																																											
4	5012871073	1073		Roman characters																																																																											
5	5012871074	1074		Roman characters																																																																											
6	5012871490	1490		Roman characters																																																																											
7	5012871492	1492		Roman characters																																																																											
8	5012871493	1493		Roman characters																																																																											
9	5012871494	1494		Roman characters																																																																											
10	5012871495	1495		Roman characters																																																																											
11	5012871496	1496		Roman characters																																																																											
12	5012871497	1497		Roman characters																																																																											
13	5012871498	1498		Roman characters																																																																											
14	5012871499	1499		Roman characters																																																																											

Figure 42 – Digit Conversion Tree

5.6.8. Outbound Call - Special Number Configuration

There are special numbers which have been configured to be used for this testing such as: 011, 1800, 411, 911 and so on.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Special Number (SPN)** as shown in **Figure 34**.

b) Enter SPN number and then click on **to Add** button. **Figure 43** shows all the special number used for this testing.

The screenshot displays the 'Special Number List' configuration page in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation tree with categories like UCM Network Services, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans (with 'Electronic Switched Network' selected), Phones, Tools, and Security. The main content area shows a list of configured special numbers. At the top, there is a form to 'Please enter a Special Number' with an input field and a 'to Add' button. Below this, four special numbers are listed: 011, 1800, 411, and 911. Each entry includes its flexible length, whether it inhibits the time-out handler, the type of call it defines (INTL or NATL), and its route list index. Each entry also has an 'Edit' button.

Special Number	Flexible length	Inhibit time-out handler	Type of call	Route list index
Special Number -- 011	14	NO	INTL	14
Special Number -- 1800	11	NO	NATL	14
Special Number -- 411	3	NO	NATL	14
Special Number -- 911	3	NO	NATL	14

Figure 43 – Add a SPN.

5.6.9. Outbound Call - Numbering Plan Area (NPA)

This section describes the creation of NPA used in this testing configuration.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen. Select **Numbering Plan Area Code (NPA)** as shown in **Figure 34**.

b) Enter the area code desired in the text box and click on the “to Add” button. The 1501,1613 and 1647 area codes were used in this configuration as shown in **Figure 44**.

The screenshot displays the Nortel CS 1000 Element Manager web interface. The top header bar is purple with the Nortel logo and the text 'CS 1000 ELEMENT MANAGER'. Below the header, a navigation pane on the left lists various system components, with 'Dialing and Numbering Plans' and 'Electronic Switched Network' highlighted. The main content area shows the 'Numbering Plan Area Code List' page. At the top of this page, it indicates the user is managing IP 10.10.97.96 as 'admin'. The breadcrumb trail shows the path: 'Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Numbering Plan (NET) > Access Code 2 > Numbering Plan Area Code List'. The page title is 'Numbering Plan Area Code List'. Below the title, there is a form with the label 'Please enter an area code' followed by a text input field and a 'to Add' button. The list contains three entries, each with a minus sign, the area code, and an 'Edit' button: 'Numbering Plan Area Code -- 1501', 'Numbering Plan Area Code -- 1613', and 'Numbering Plan Area Code -- 1647'. Each entry also displays 'Route List Index: 15' and 'Incoming Trunk group Exclusion Index: NONE'.

Numbering Plan Area Code	Route List Index	Incoming Trunk group Exclusion Index
1501	15	NONE
1613	15	NONE
1647	15	NONE

Figure 44 – Numbering Plan Area Code List

5.7. Administer Phone

This section describes the creation of Communication Server 1000 clients used in this configuration.

5.7.1. Phone creation

- a) Refer to **Section 5.5.4** to create a virtual super-loop - **96** used for IP phone.
- b) Refer to **Section 5.4.1** to create a bandwidth zone - **10** for IP phone.
- c) Login Call ServerCommand Line Interface (please refer to **Section 5.1.2** for more detail).
- d) Create an IP phone by using **LD 11**.

```
REQ: prt
TYPE: 2002p2
TN 96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2002P2
TN 96 0 00 02 VIRTUAL
TYPE 2002P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
MRT
ERL 12345
ECL 0
FDN
TGAR 0
LDN NO
NCOS 7
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBD WTA LPR MTD FND HTD TDD CRPD
MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMD LLCN MCTD CLBD AUTU
```



```

GPUD DPUD DNDD CFXD ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBP RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
MSNV FRA PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 1492 0 MARP
    CPND
    CPND_LANG ROMAN
NAME Carrier1
    XPLN 13
    DISPLAY_FMT FIRST, LAST
    01
    02
<Text removed for brevity>

```

5.7.2. Enable Privacy for Phone

In this section, it shows how to enable Privacy for a phone by changing its class of service (CLS). By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately.

a) To hide the display name, set CLS to **namd**. Communication Server 1000 will include “Privacy:user” in the SIP message header before sending it to the Service Provider.

```

>ld 11
REQ: chg
TYPE: 2002p2
TN 96 0 0 2
ECHG yes
ITEM clsnamd
...

```

b) To hide the display number, set CLS to **ddgd**. Communication Server 1000 will include “Privacy:id” in the SIP message header before sending it to the Service Provider.

```

>ld 11
REQ: chg
TYPE: 2002p2
TN 96 0 0 2

```

```
ECHG yes  
ITEM clsddgd  
...
```

c) To hide display name and number, set CLS to **namd, ddgd**. Communication Server 1000 will include “Privacy:id, user” in the SIP message header before sending to the Service Provider.

```
>ld 11  
REQ: chg  
TYPE: 2002p2  
TN 96 0 0 2  
ECHG yes  
ITEM clsnamddgd  
...
```

d) To allow display name and number, set CLS to **nama, ddga**. Communication Server 1000 will not send the Privacy header to the Service Provider.

```
>ld 11  
REQ: chg  
TYPE: 2002p2  
TN 96 0 0 2  
ECHG yes  
ITEM clsnamaddga  
...
```

5.7.3. Enable Call Forward for Phone

In this section, it shows how to configure the Call Forward feature at the system and phone level.

a) Select **Customer->00->CallRedirection**. The Call Redirection page is shown in **Figure 45**.

- **Total redirection count limit: 0** (unlimited)
- **Call Forward: Originating**
- **Number of normal ring cycle of CFNA: 4**

The screenshot displays the 'Call Redirection' configuration page. On the left is a navigation menu with categories like 'UCM Network Services', 'Links', 'System', 'Customers', 'Routes and Trunks', 'Dialing and Numbering Plans', 'Phones', 'Tools', and 'Security'. The 'Customers' section is expanded, showing 'Routes and Trunks' and 'D-Channels'. The main content area is titled 'Call Redirection' and contains several sections:

- Days for day option 1:** (empty field)
- Days for day option 2:** (empty field)
- Days for day option 3:** (empty field)
- Redirection Holidays:** ☐ Do not disturb hunting:
- Total redirection count limit:** 0 (dropdown menu)
- Options:**
 - ☐ Call forward reminder tone for 500/2500 sets
 - ☐ CFNA treatment for call waiting calls on a DN
 - ☐ DID call to second degree busy treatment
 - ☒ Message center
 - ☒ Prevention of reciprocal call forward
- Call forward:** ☒ Originating, ☐ Forwarding
- Number of normal ringing cycles for CFNA:**
 - Option 0: 4 (dropdown menu)
 - Option 1: 4 (dropdown menu)
 - Option 2: 4 (dropdown menu)
- Number of distinctive ringing cycles for CFNA:**
 - Option 0: 4 (dropdown menu)
 - Option 1: 4 (dropdown menu)
 - Option 2: 4 (dropdown menu)
- Calls routed to message center:**
 - ☐ No answer DID calls:
 - ☐ No answer non-DID calls:
 - ☐ DID calls to busy telephones:

At the bottom right, there are 'Save' and 'Cancel' buttons.

Figure 45 – Call Redirection

b) To enable **Call Forward All Call (CFAC)** for a phone over a trunk, use **LD 11**, change its CLS to **CXFA**, **SFA** then program the forward number on the phone set. Following is the configuration of a phone that has **CFAC** enabled with forwarding number 916139675205

```
REQ: prt
TYPE: 2007
TN 96 0 0 4
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 96 0 00 04 VIRTUAL
TYPE 2007
...
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDA CDMA LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXA ARHD CLTD ASCD
...
19 CFW 16 916139675205
...
```

d) To enable **Call Forward Busy (CFB)** for phone over trunk by using **LD 11**, change its **CLS** to **FBA**, **HTA**, **SFA** then program the forward number as is **HUNT**. Following is the configuration of a phone has **CFB** enabled with forward number is 916139675205

```
REQ: prt
TYPE: 2007
TN 96 0 0 4
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 96 0 00 04 VIRTUAL
TYPE 2007
...
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBF
...
FDN 916139675205
HUNT916139675205
...
```

c) To enable **Call Forward No Answer (CFNA)** for a phone over a trunk by using **LD 11**, change its **CLS** to **FNA**, **SFA** then program the forward number as **FDN**. Following is the configuration of a phone that has CFNA enabled with forward number 916139675205

```
REQ: prt
TYPE: 2007
TN 96 0 0 4
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES 2007
TN 96 0 00 04 VIRTUAL
TYPE 2007
...
FDN 916139675205
HUNT 916139675205
...
CLS UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBF
...
```


5.7.4. Enable Call Waiting for Phone

In this section, it shows how to configure Call Waiting feature at phone level.

Login Call Server CLI (please refer to **Section 5.1.2** for more detail), configure Call Waiting feature for phone by using **LD 11** to change **CLS** to **HTD**, **SWA** and adding a **CWT** key.

```
REQ: prt
TYPE: 2002p2

TN 96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES 2002P2
TN 96 0 00 02 VIRTUAL
TYPE 2002P2
...
CLS UNR FBD WTA LPR MTD FNA HTD TDD HFD CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWA LNA CNDA
...
KEY 00 SCR 1492 0 MARP
CPND
CPND_LANG ROMAN
NAME Carrier1
XPLN 13
DISPLAY_FMT FIRST, LAST
01 CWT
...
```

5.8. NRS/SPS Configuration

In this section, it shows how to configure a NRS/SPS on Communication Server 1000. Follow the steps bellow to setup the NRS/SPS server. It is assumed that the NRS/SPS has been deployed on the Communication Server 1000 UCM environment with all latest Service Pack applied.

5.8.1. Create a New Domain Name on NRS/SPS

In this section, it shows how to create a new domain name for this test configuration.

- Under the **Element Names**, select the **NRSM on car3-sps**
- The **Network Routing Service Manager** page will appear, click on the radio button of the **Standby database**. Then select **Numbering Plan -> Domains**. Click on the **Service Domain (1)** tab then click on the **Add** button to add a new domain as shown in **Figure 46**.

Service Domains (1)					
Add... Delete Refresh					
	Domain Name	Description	# of L1 Domains	# of L0 Domains	# of Gateway Endpoints
1	bvwdev7.com		1	1	2

Figure 46 – Add Domain Name

- Enter the domain name to be added; in this case it is **bvwdev7.com**. Then click on the **Save** button.
- Select the **L1 Domains (UDP) (1)**, from the **Filter by Domain**, select the newly created domain **bvwdev7.com**. Click on the **Add** button and enter **udp** as L1 Domain name. Then click on the **Save** button (not shown).
- Select the **L0 Domains (CDP) (1)**, from the **Filter by Domain**, select the newly created domain **bvwdev7.com**. Click on the **All L1 Domain** pull down menu to choose the **udp**. Click on the **Add** button and enter **cdp** as L0 Domain name. Then click on the **Save** button (not shown).
- From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save it to the **Active Database**. Click on the **Commit** button (not shown).

5.8.2. Configure NRS/SPS to create Gateway Endpoints for the SIP Signaling Gateway

This section shows how to add the SIP Signaling Gateway as a dynamic gateway endpoint on the NRS/SPS.

a) Click on the radio button of the **Standby database**, select the **Numbering Plan -> Endpoints**. Enter the endpoint name and choose the values which are highlighted in red boxes as shown in **Figure 47**. Click on the **Add** button to create new gateway endpoint.

The screenshot shows the Nortel Network Routing Service Manager (NRS/SPS) interface. The left sidebar contains a navigation menu with categories like «UCM Network Services», «System», «Numbering Plans», «Tools», and «Routing Tests». The main content area is titled 'Managing: 10.10.97.99' and shows 'Standby database' selected. Below this is a 'Search for Endpoints' section with a text input field containing 'car3-ssg-carrier' and three dropdown menus for 'Limit results to Domain' (bwwdev7.com), 'udp', and 'cdp'. The 'Add' button is highlighted in red. Below the search section is a table with two tabs: 'Gateway Endpoints (2)' and 'User Endpoints (0)'. The table has columns for ID, Supported Protocols, SIP mode, Call Signaling IP, Description, # of Routing Entries, and Context. The first row shows 'acme' as a Static SIP endpoint in Proxy Mode with IP 135.10.97.184. The second row shows 'car3-ssg-carrier' as a Dynamic SIP endpoint in Proxy Mode with IP Not available. The 'Add' button is also highlighted in red.

ID	Supported Protocols	SIP mode	Call Signaling IP	Description	# of Routing Entries	Context
1 acme	Static SIP endpoint	Proxy Mode	135.10.97.184		5	bwwdev7.com / udp / cdp
2 car3-ssg-carrier	Dynamic SIP endpoint	Proxy Mode	Not available		1	bwwdev7.com / udp / cdp

Figure 47 – Adding New Gateway Endpoints

b) The detail gateway endpoint configuration page will appear. Using the values for the highlighted attributes in red boxes as shown in **Figures 48 and 49** and click on the **Save** button.

Figure 48 – SSG Gateway Endpoint Details Configuration

Figure 49 – SSG Gateway Endpoint Details Configuration

c) From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save in to the **Active Database**. Click on the **Commit** button (not shown).

5.8.3. Create a Static Gateway Endpoint for the ACME Session Border Controller

This section shows how to add the Session Border Controller as a static gateway endpoint on the NRS/SPS.

- Click on the radio button of the **Standby database**, select the **Numbering Plan -> Endpoints**. Enter the endpoint name for the ACME session border controller as **acme** and choose the values “Limit results Domain: **bwvdev7.com/udp/cdp**”. Click on the **Add** button.
- The detail gateway endpoint configuration page will appear. For the Static Endpoint IP Address, enter the ACME internal (private side) IP address and use the values for the highlighted attributes in red boxes as shown in **Figure 51**, **Figure 52**. Click on the **Save** button.
- From the left menu column, select **System -> Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save it to the **Active Database**. Click on the **Commit** button (not shown).

The screenshot displays the NORTTEL NETWORK ROUTING SERVICE MANAGER interface. The left sidebar shows a navigation menu with categories like «UCM Network Services», «System», «Numbering Plans», and «Tools». The main content area is titled 'Edit Gateway Endpoint bwvdev7.com / udp / cdp'. It features a 'Managing:' section with radio buttons for 'Active database' and 'Standby database'. Below this, the endpoint name 'acme' is entered in a text field. Other fields include 'Description', 'Trust Node' (checked), 'Tandem gateway endpoint name' (Not Applicable), 'Endpoint authentication enabled' (Authentication off), 'Authentication password', and several E.164 fields: 'E.164 country code' (613), '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), and 'E.164 local (subscriber) dialing access code'. A 'Save' button is at the bottom right.

Figure 50 – ACME Gateway Endpoint Details Configuration

Static endpoint address type: IP version 4

Static endpoint address: 10.10.97.184

H.323 support: H.323 not supported

SIP support: Static SIP endpoint

SIP mode: Proxy Mode

SIP TCP transport enabled: ☐

SIP TCP port: 5060

SIP UDP transport enabled: ☒

SIP UDP port: 5060

SIP TLS transport enabled: ☐

SIP TLS port: 5061

Persistent TCP support enabled: ☒

End to end security support: ☐

Network Connection Server enabled: ☐

★ Required value

Save Cancel

Figure 51 – ACME Gateway Endpoint Details Configuration

Note: After adding the SIP Signaling Gateway (SSG) and the ACME session border controller to the NRS/SPS, Active database shows IP addresses of the gateways(highlighted redbox) and both SSG and **acme** gateways are successfully registered to NRS/SPS (as shown in **Figure 52**)

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: All service domains / All L1 domains / All L0 domains

Results per page: 50 Search

Gateway Endpoints (2)		User Endpoints (0)	
ID	Supported Protocols	SIP mode	Call Signaling IP
1 acme	Static SIP endpoint	Proxy Mode	10.10.97.184
2 car3-ssg-carrier	Dynamic SIP endpoint	Proxy Mode	10.10.97.178

1 - 2 of 2 Gateway Endpoint(s) Page 1 of 1 First Previous Next Last

Figure 52 – SSG and ACME successfully register to NRS/SPS

5.8.4. Creating Routing Entry for the SSG on the NRS/SPS

In this section, it describes how to create the routing entry on the NRS/SPS to route the inbound calls. In the test configuration, we have used the routing entry 501 which is sent from Windstream as the first 3 digits of the DIDs range, 501 - XXX - XXXX.

- a) Click on radio button **Standby database**, then on the Network Routing Service Manager page as shown in **Figure 46**, on the left menu column, select **Numbering Plan ->Routes**. On the Routing Entries page, choose and change the attributes highlighted in red boxes to the values as shown in **Figure 53**. Click the **Add** button.

Network Routing Service Manager

Managing: ☐ Active database **10.10.97.98**
☒ Standby database [Numbering Plans > Routes](#)

Search for Routing Entries [Hide](#)

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

DN Prefix: * DN Type: All DN Types

Limit results to Domain: bwdev7.com / udp / cdp

Endpoint Name: car3-ssg-carrier

Results per page: 50 [Search](#)

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

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

DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
501	Private level 0 regional (CDP steering code)	1	cdp.udp	bwdev7.com / udp / cdp / car3-ssg-carrier

1 - 1 of 1 Routing Entry(ies) Page 1 of 1 First Previous Next Last

Figure 53 – Routing Entries Page

- b) The Routing Entries page will appear as shown in **Figure 54**. Fill in the textboxes with the values highlighted in red boxes. Then click on the **Save** button.
- DN Type: **Private level 0 regional (CDP steering code)**
 - DN Prefix: **501**
 - Route cost: **1**

Network Routing Service Manager

Managing: ☐ Active database **10.10.97.98**
☒ Standby database [Numbering Plans > Routes > Routing Entry](#)

Edit Routing Entry (bwdev7.com / udp / cdp / car3-ssg-carrier)

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

DN prefix: 501 *

Route cost: 1 * (1-255)

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

Figure 54 – Routing Entries Configuration Details Page

c) From the left menu column, select **System ->Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save in to the **Active Database**. Click on the **Commit** button (not shown).

5.8.5. Creating Routing Entry for the ACME on the NRS/SPS

This section describes how to create the routing entry on the NRS/SPS to route the call from NRS/SPS to the ACME session border controller. In the test configuration, we have used the routing entries 011, 411, 911, 1800, 613, 647, 1613 and 501. In the following example, it shows only one entry 613. Others entries can be done the same way.

a) Click on radio button **Standby database**, then on the Network Routing Service Manager page as shown in **Figure 46**, select **Numbering Plan ->Routes**. On the Routing Entries page, choose and change the attributes highlighted in red boxes to the values as shown in **Figure 55**. Click the **Add** button.

NETWORK ROUTING SERVICE MANAGER

Managing: ☐ Active database: **10.10.97.98**
☒ 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: All DN Types

Limit results to Domain: **bvwddev7.com** / **udp** / **cdp**

Endpoint Name: **acme**

Results per page: 50 Search

Routing Entries (8) Default Routes (0) Emergency Fallback Routes (0)

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

	DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1	011	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
2	1613	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
3	1800	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
4	411	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
5	501	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
6	613	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
7	647	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme
8	911	Private level 0 regional (CDP steering code)	1	cdp.udp	bvwddev7.com / udp / cdp / acme

1 - 8 of 8 Routing Entry(ies) Page 1 of 1 First Previous Next Last

Figure 55 – Add a Routing Entry for ACME

b) The Routing Entries page will appear as shown in **Figure 56**. Fill in the textboxes with the values highlighted in red boxes. Then click on the **Save** button.

DN Type:Private level 0 regional (CDP steering code)

- DN Prefix:613

- Route cost:1

The screenshot shows the 'Nortel NETWORK ROUTING SERVICE MANAGER' interface. On the left is a sidebar with a tree view containing 'System', 'Numbering Plans', and 'Tools'. The main content area is titled 'Edit Routing Entry (bvwddev7.com / udp / cdp / acme)'. It features a 'Managing:' section with 'Active database' and 'Standby database' options. Below this, there are three input fields: 'DN type' (a dropdown menu showing 'Private level 0 regional (CDP steering code)'), 'DN prefix' (a text box with '813'), and 'Route cost' (a text box with '1'). A legend at the bottom left indicates that an asterisk (*) denotes a 'Required value'. At the bottom right are 'Save' and 'Cancel' buttons.

Figure 56 – Routing Entry Configuration Details for ACME

c) From the left menu column, select **System ->Database**. Then click on the **Cut Over** button to transfer the configured data of the domain name to save in to the **Active Database**. Click on the **Commit** button (not shown).

6. Configure Acme Packet Net-Net 3800

This section describes the configuration of the Acme Packet Net-Net 3800 necessary for interoperability with the Communication Server 1000 and Windstream systems. The Net-Net 3800 was configured via the Acme Packet Command Line Interface (ACLI). This section assumes the reader is familiar with accessing and configuring the Acme Packet products.

This section will not attempt to describe each component in its entirety but instead will highlight critical fields in each component which relates to the functionality in these Application Notes and the direct connection to Communication Server 1000. The remaining fields are generally the default/standard value used by the Net-Net 3800 for that field.

In this testing, according to the configuration reference in **Figure 1**, the Avaya elements reside on the Private side and the Windstream elements reside on the Public side of the network.

6.1. Acme Packet Command Line Interface Summary

The Net-Net 3800 is configured using the Acme Packet Command Line Interface (ACLI). The following are the generic ACLI steps for configuring various elements.

1. Access the console port of the Net-Net 3800 using a PC and a terminal emulation program such as HyperTerminal (use the RJ-45 to DB9 adapter as packaged with the Net-Net 3800 server for cable connection). Use the following settings for the serial port on the PC.
 - Bits per second: 115200
 - Data bits: 8
 - Parity: None
 - Stop bits: 1
 - Flow control: None
2. Log in to the Net-Net 3800 with the user password.
3. Enable the Super-user mode by entering the **enable** command and then the superuser password. The command prompt will change to include a “#” instead of a “>” while in Superuser mode. This level of system access (i.e. at the “acmesystem#” prompt) will be

referred to as the *main* level of the ACLI. Specific sub-levels of the ACLI will then be accessed to configure specific *elements* and specific *parameters* of those elements.

4. In Super-user mode, enter the **configure terminal** command. The **configure terminal** command is used to access the system level where all operating and system elements may be configured. This level of system access will be referred to as the *configuration* level.
5. Enter the name of an element to be configured (e.g., **system**).
6. Enter the name of a sub-element, if any (e.g., **phy-interface**).
7. Enter the name of an element parameter followed by its value (e.g., **name INSIDE**).
8. Enter **done** to save changes to the element. Use of the **done** command causes the system to save and display the settings for the current element.
9. Enter **exit** as many times as necessary to return to the configuration level.
10. Repeat **Steps 5 - 9** to configure all the elements.
11. Enter **exit** to return to the main level.
12. Type **save-config** to save the entire configuration.
13. Type **activate-config** to activate the entire configuration.

After accessing different levels of the ACLI to configure elements and parameters, it is necessary to return to the main level in order to run certain tasks such as saving the configuration, activating the configuration, and rebooting the system.

Note – Net-Net 3800 provisioning applicable to the reference configuration is shown in **boldtext**. Other parameters and setting are shown for informational purposes.

6.2. Physical and Network Interfaces

As part of the compliance test, the Ethernet slot 0/port 0 was connected to the internal corporate LAN. The Ethernet interface slot 1/port 0 was connected to the external un-trusted network. A network interface was defined for each physical interface to assign it a routable IP address.

The physical interface below defines the ports on the interface connected to the network on which the Avaya elements reside.

phy-interface		
name	INSIDE	
operation-type	Media	
port	0	
slot	0	
virtual-mac		
admin-state	enabled	
auto-negotiation	enabled	
duplex-mode	FULL	
speed	100	
overload-protection	disabled	
last-modified-by	admin@console	
last-modified-date	2011-06-07 10:11:20	

The physical interface below defines the ports on the interface connected to the network on which the Windstream elements reside.

phy-interface		
name	OUTSIDE	
operation-type	Media	
port	0	
slot	1	
virtual-mac		
admin-state	enabled	
auto-negotiation	enabled	
duplex-mode	FULL	
speed	100	
overload-protection	disabled	
last-modified-by	admin@console	
last-modified-date	2011-06-07 10:11:30	

The network interface below defines the IP addresses on the interface connected to the network on which the Avaya elements reside.

network-interface	
name	INSIDE
sub-port-id	0
description	
hostname	
ip-address	10.10.97.184
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.192
gateway	10.10.97.129
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	10.10.97.184
ftp-address	
icmp-address	10.10.97.184
snmp-address	
telnet-address	
ssh-address	
last-modified-by	admin@console
last-modified-date	2011-06-07 10:20:11

The network interface below defines the IP addresses on the interface connected to the network on which the Windstream elements reside.

network-interface	
name	OUTSIDE
sub-port-id	0
description	
hostname	
ip-address	10.10.98.98
pri-utility-addr	
sec-utility-addr	
netmask	255.255.255.224
gateway	10.10.98.97
sec-gateway	
gw-heartbeat	
state	disabled
heartbeat	0
retry-count	0
retry-timeout	1
health-score	0
dns-ip-primary	
dns-ip-backup1	
dns-ip-backup2	
dns-domain	
dns-timeout	11
hip-ip-list	10.10.98.98
ftp-address	
icmp-address	10.10.98.98
snmp-address	
telnet-address	
ssh-address	
last-modified-by	admin@console
last-modified-date	2011-06-07 15:22:28

6.3. Realm

A realm represents a group of related Net-Net 3800 components. Two realms were defined for the compliance test.

The realm configuration “INSIDE” below represents the internal network on which the Avaya elements reside.

realm-config	
identifier	INSIDE
description	
addr-prefix	0.0.0.0
network-interfaces	
INSIDE:0	
mm-in-realm	disabled
<Text removed for brevity>	

The realm configuration “OUTSIDE” below represents the external network on which the Windstream system resides.

realm-config	
identifier	OUTSIDE
description	
addr-prefix	0.0.0.0
network-interfaces	
OUTSIDE:0	
mm-in-realm	disabled
<Text removed for brevity>	

6.4. Session Agent

A session agent defines the characteristics of a signaling peer to the Net-Net 3800.

The **session agent** below represents the Windstream border element. The Acme will attempt to send calls to the border element. The **in-manipulationid** and **out-manipulationid** define the SIP header manipulation applying to the OUTSIDE realm.

```
session-agent
hostname                20.20.242.26
ip-address             20.20.242.26
  port                   5060
  state                   enabled
app-protocol           SIP
  app-type
transport-method       UDP
  realm-id               OUTSIDE
  egress-realm-id
description            Windstream_CS1K 7.0
  carriers
  allow-next-hop-lp       enabled
  constraints              disabled

  <Text removed for brevity>

  ping-interval           0
ping-send-mode         keep-alive

  <Text removed for brevity>

  ping-from-user-part
  li-trust-me              disabled
in-manipulationid      WS_TO_CS1K7_NAT_IP
out-manipulationid     CS1K7_TO_WS_NAT_IP
  manipulation-string
```

The **session agent** below represents the configuration for inside interface to connect to NRS mentioned in **Section 5.8**

```
session-agent
hostname                10.10.97.183
ip-address              10.10.97.183
  port                  5060
  state                 enabled
app-protocol            SIP
  app-type
transport-method        UDP
realm-id                INSIDE
  egress-realm-id
description              Windstream_CS1K7.0
  carriers
  allow-next-hop-lp      enabled
  constraints             disabled
  <Text removed for brevity>
```

6.5. SIP Configuration

The SIP configuration (*sip-config*) defines the global system-wide SIP parameters.

The key SIP configuration (*sip-config*) field is:

- **home-realm-id**: The name of the realm on the private side of the Net-Net 3800.
- **egress-realm-id**: The name of the realm on the private side of the Net-Net 3800.

```
sip-config
  state                 enabled
  operation-mode         dialog
  dialog-transparency    enabled
home-realm-id           INSIDE
  egress-realm-id        INSIDE
nat-mode                None

  <Text removed for brevity>
```


6.6. SIP Interface

The SIP interface (*sip-interface*) defines the receiving characteristics of the SIP interfaces on the Net-Net 3800. Two SIP interfaces were defined; one for each realm.

The SIP interface below is used to communicate with the Communication Server 1000 system.

```
sip-interface
  state                enabled
realm-id              INSIDE
  description
  sip-port
address              10.10.97.184
                        port          5060
                        transport-protocol UDP
tls-profile
  allow-anonymous      all

<Text removed for brevity>
```

The SIP interface below is used to communicate with the Windstream system.

```
sip-interface
  state                enabled
realm-id              OUTSIDE
  description
  sip-port
address              10.10.98.98
                        port          5060
                        transport-protocol UDP
tls-profile
  allow-anonymous      all

<Text removed for brevity>
```

6.7. SIP Manipulation

SIP manipulations are rules used to modify the SIP messages (if necessary) for interoperability. The following sip-manipulation **CS1K7_TO_WS_NAT_IP** is applied to **OUTSIDE** realm *out-manipulationid*. These rules perform the following:

- The header rule **manipRURI** changes Avaya Domain Name/IP address to 20.20.242.26 (Windstream border element) in the Request URI headers sent to Windstream.
- The header rule **manipTo** performs address translation and topology hiding for SIP messages between the Windstream system and the Avaya elements.

sip-manipulation			
nameCS1K7_TO_WS_NAT_IP			
description			
split-headers			
join-headers			
header-rule			
name		manipRURI	
header-name		request-uri	
action		manipulate	
comparison-type		case-sensitive	
msg-type		any	
methods		INVITE	
match-value			
new-value			
element-rule			
name		modRURI	
parameter-name			
type		uri-host	
action		replace	
match-val-type		any	
comparison-type		case-sensitive	
match-value			
new-value		20.20.242.26	
header-rule			
name		manipTo	
header-name		To	
action		manipulate	
comparison-type		case-sensitive	
msg-type		any	
methods			
match-value			
new-value			
element-rule			

name		To	
	parameter-name		
	type		uri-host
	action		replace
	match-val-type		any
	comparison-type		case-sensitive
	match-value		
new-value		\$REMOTE_IP	
	header-rule		
name		HistRegex	
	header-name		History-Info
	action		store
	comparison-type		pattern-rule
msg-type		request	
methods		INVITE	
	match-value		()
	new-value		
	element-rule		
name		GetUser	
	parameter-name		
type		uri-user	
	action		store
	match-val-type		any
	comparison-type		pattern-rule
	match-value		
	new-value		
	element-rule		
name		GetHost	
	parameter-name		
type		uri-host	
	action		store
	match-val-type		any
	comparison-type		pattern-rule
	match-value		
	new-value		
	element-rule		
name		GetUserReason1	
	parameter-name		
type		header-value	
	action		store
	match-val-type		any
	comparison-type		pattern-rule
	match-value		(.*)(Moved)(.*)
	new-value		
	element-rule		

name		GetUserReason2
	parameter-name	
type		header-value
	action	store
	match-val-type	any
	comparison-type	pattern-rule
	match-value	(.*)(Busy)(.*)
	new-value	
	element-rule	
name		GetUserReason3
	parameter-name	
type		header-value
	action	store
	match-val-type	any
comparison-type		pattern-rule
	match-value	(.*)(Unavailable)(.*)
	new-value	
	header-rule	
name		AddDiversion1
	header-name	Diversion
	action	add
	comparison-type	boolean
msg-type		request
	methods	INVITE
	match-value	\$HistRegex[0].\$GetUserReason1
	new-value	<sip:+\$HistRegex[0].\$GetUser.\$0+@+\$HistRegex[0].\$GetHost.\$0+>;privacy=off;reason=unconditional;screen=no
	header-rule	
name		AddDiversion2
	header-name	Diversion
	action	add
	comparison-type	boolean
msg-type		request
	methods	INVITE
	match-value	\$HistRegex[0].\$GetUserReason2
	new-value	<sip:+\$HistRegex[0].\$GetUser.\$0+@+\$HistRegex[0].\$GetHost.\$0+>;privacy=off;reason=user\'-busy;screen=no
	header-rule	
name		AddDiversion3
	header-name	Diversion
	action	add
	comparison-type	boolean

msg-type	request
methods	INVITE
match-value	\$HistRegex[0].\$GetUserReason3
new-value	<sip:+\$HistRegex[0].\$GetUser.\$0+@+\$HistRegex[0].\$GetHost.\$0+>;privacy=off;reason=no\-answer;screen=no
header-rule	
name	delHistInfo
header-name	History-Info
action	delete
comparison-type	case-sensitive
msg-type	any
methods	INVITE
match-value	
new-value	
header-rule	
name	manipFrom
header-name	From
action	manipulate
comparison-type	case-sensitive
msg-type	any
methods	
match-value	
new-value	
element-rule	
name	From
parameter-name	
type	uri-host
action	replace
match-val-type	any
comparison-type	case-sensitive
match-value	
new-value	10.10.98.98
last-modified-by	admin@console
last-modified-date	2011-07-06 21:42:22

The following sip-manipulation **WS_TO_CS1K7_NAT_IP**, *in-manipulationid*, is applied to **OUTSIDE**realm and translates the SIP header information for Communication Server 1000 to understand. These rules perform the following:

- The header rules**manipRURI**changesIP address to the Avaya Communication Server 1000 Domain Name in the Request URI headers sent to the Communication Server 1000 elements.

sip-manipulation
name WS_TO_CS1K7_NAT_IP

description		
split-headers		
join-headers		
header-rule		
name		manipRURI
header-name		request-uri
action		manipulate
comparison-type		case-sensitive
msg-type		any
methods		INVITE
match-value		
new-value		
element-rule		
name		modRURI
parameter-name		
type		uri-host
action		replace
match-val-type		any
comparison-type		case-sensitive
match-value		
new-value		bvwdev7.com
header-rule		
name		manipTo
header-name		To
action		manipulate
comparison-type		case-sensitive
msg-type		any
methods		
match-value		
new-value		
element-rule		
name		To
parameter-name		
type		uri-host
action		replace
match-val-type		any
comparison-type		case-sensitive
match-value		
new-value		bvwdev7.com
last-modified-by		admin@console
last-modified-date		2011-06-07 12:52:23

6.8. Steering Pools

Steering pools define the range of ports to be used for the RTP voice stream. Two steering pools were defined; one for each realm.

The key steering pool (*steering-pool*) fields are:

- **ip-address:** The address of the interface on the Net-Net 3800.
- **start-port:** An even number of the port that begins the range.
- **end-port:** An odd number of the port that ends the range.
- **realm-id:** The realm to which this steering pool is assigned.

steering-pool	
ip-address	10.10.98.98
start-port	20000
end-port	40000
realm-id	OUTSIDE
network-interface	
last-modified-by	admin@console
last-modified-date	2011-06-07 22:20:07
steering-pool	
ip-address	10.10.97.184
start-port	20000
end-port	40000
realm-id	INSIDE
network-interface	
last-modified-by	admin@console
last-modified-date	2011-06-07 22:20:22

6.9. Local Policy

The local policies below govern the routing of SIP messages from elements on the network on which the Avaya elements, reside to the Windstream system and vice versa.

local-policy	
from-address	
20.20.242.26	
to-address	
	5012871070
5012871071	
5012871072	
5012871073	
	5012871074
5012871490	
5012871491	
5012871492	

```

5012871493
5012871494
5012871495
5012871496
5012871497
5012871498
5012871499
source-realm
  OUTSIDE
description      WS_TO_CS1K7
activate-time    N/A
deactivate-time  N/A
state            enabled
policy-priority  none
last-modified-by admin@console
last-modified-date 2011-06-07 14:44:50
policy-attribute
  next-hop      10.10.97.183
  realm         INSIDE
  action        none
  terminate-recursion disabled
  carrier
  start-time    0000
  end-time      2400
  days-of-week  U-S
  cost          0
  app-protocol  SIP
  state         enabled
  methods
  media-profiles
  lookup        single
  next-key
eloc-str-lkup disabled
eloc-str-match

```

```

local-policy
  from-address
    anonymous.invalid
    bvwddev7.com
  to-address
    *
  source-realm
    INSIDE
  description      CS1K_TO_WS

```

activate-time	N/A
deactivate-time	N/A
state	enabled
policy-priority	none
last-modified-by	admin@console
last-modified-date	2011-06-07 20:25:30
policy-attribute	
next-hop	20.20.242.26
realm	OUTSIDE
action	none
terminate-recursion	disabled
carrier	
start-time	0000
end-time	2400
days-of-week	U-S
cost	0
app-protocol	SIP
state	enabled
methods	
media-profiles	
lookup	single
next-key	
eloc-str-lkup	disabled
eloc-str-match	

7. Verification Steps

The following steps may be used to verify the configuration.

7.1. General

Place an inbound call from a PSTN phone to an internal Avaya phone, answer the call, and verify that two-way speech path exists. Verify that the call remains stable for several minutes and disconnects properly.

7.2. Verification an Active Call on Call Server

a) Active Call Trace (LD 80)

The following is an example of one of the commands available on the Communication Server 1000 to trace the DN which the call is in progress and or idle. The call scenario involved PSTN phone number 6139675205 calling 5012871492.

- Login on to Signaling Server 10.10.97.177 with admin account and password.
- Issue a command “cslogin” call to login on to the Call Server.
- Login to the Overlay command prompt, issue the command **LD80** and then **trace 0 1492**.
- After the call is released, issue command **trac 0 1492** again to see if the DN is released back to idle state.

Bellow is the actual output of the Call Server Command Line mode when the 1492 is in call state:

```
USERID? admin
PASS?....
.
TTY #09 LOGGED IN admin 16:22 08/6/2011
.
>ld 80
.trac 0 1492

ACTIVE VTN 96 0 00 02

ORIG VTN 100 0 00 00 VTRK IPTI RMBR 100 1 INCOMING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 10.10.97.184
FAR-END MEDIA ENDPOINT IP: 10.10.97.184 PORT: 21638
FAR-END VendorID: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.00.20
TERM VTN 96 0 00 02KEY 0 SCR MARP CUST 0 DN 1492 TYPE 2002P2
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 10.10.98.36 PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW PAYLOAD 20 ms VAD OFF
RFC2833: RXPT 101 TXPT 101 DIAL DN 1492
MAIN PM ESTD
TALKSLOT ORIG 17 TERM 81
QUEUE NONE
CALL ID 501 84

---- ISDN ISL CALL (ORIG) ----
CALL REF # = 484
BEARER CAP = VOICE
HLC =
CALL STATE = 10 ACTIVE
CALLING NO = NUM_PLAN:UNKNOWN TON:UNKNOWN ESN:UNKNOWN
CALLED NO = 5012871492 NUM_PLAN:UNKNOWN TON:UNKNOWN ESN:UNKNOWN
```

And this is the example after the call on 1492 is finished.

```
.trac 0 1492
IDLE VTN 96 0 00 02 MARP
```

b) SIP Trunk monitoring (LD 32)

Place a call inbound from PSTN (6139675205) to an internal device (5012871492). Then check the SIP trunk status by using LD 32, one trunk is BUSY

```
>ld 32
NPR000
.stat 100 0
031 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
.
```

After the call is released, check all SIP trunk status changed to IDLE state.

```
.stat 100 0
032 UNIT(S) IDLE
000 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

7.3. Protocol Trace

Below is a wireshark trace of the same call scenario described in **Section 7.2**. It is shown in text format bellow. Note that only detail of the INVITE message is being shown here.

No.	Time	Source	Destination	Protocol Info
41	36.977060	20.20.242.26	10.10.98.98	SIP/SDP Request: INVITE sip:5012871492@10.10.98.98:5060, with session description

Frame 41: 841 bytes on wire (6728 bits), 841 bytes captured (6728 bits)
Ethernet II, Src: Nortel_01:b4:49 (00:17:65:01:b4:49), Dst: AcmePack_a1:8c:a5 (00:08:25:a1:8c:a5)
Internet Protocol, Src: 20.20.242.26 (20.20.242.26), Dst: 10.10.98.98 (10.10.98.98)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
Request-Line: INVITE sip:5012871492@10.10.98.98:5060 SIP/2.0
Message Header
Via: SIP/2.0/UDP 20.20.242.26:5060;branch=z9hG4bK1n34a920dgnhces1o081.1
Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info
Max-Forwards: 69
Call-ID: E3F6469F@75.89.98.228
From: "Anonymous"
<sip:6139675205@20.20.242.26:5060;transport=udp>;tag=75.89.98.228+1+23db11+c56cdbdc;i
sup-oli=00
To: <sip:5012871492@10.10.98.98>
CSeq: 1006829771 INVITE
Expires: 180
Organization:
Supported: 100rel
Content-Length: 170
Content-Type: application/sdp
Contact: "Anonymous" <sip:61396715205@20.20.242.26:5060;transport=udp>;isup-oli=00
Privacy: id
Message Body

No.	Time	Source	Destination	Protocol Info
42	36.978738	10.10.98.98	20.20.242.26	SIP Status: 100 Trying

No.	Time	Source	Destination	Protocol Info
43	37.026764	10.10.98.98	20.20.242.26	SIP Status: 180 Ringing
44	37.096280	20.20.242.26	10.10.98.98	SIP Request: PRACK sip:5012871491@10.10.98.98:5060;user=phone;transport=udp
45	37.105426	10.10.98.98	20.20.242.26	SIP Status: 200 OK
51	40.692824	10.10.98.98	20.20.242.26	SIP/SDP Status: 200 OK, with session description
70	41.191712	10.10.98.98	20.20.242.26	SIP/SDP Status: 200 OK, with session description
121	42.192600	10.10.98.98	20.20.242.26	SIP/SDP Status: 200 OK, with session description
126	42.265984	20.20.242.26	10.10.98.98	SIP Request: ACK sip:5012871491@10.10.98.98:5060;user=phone;transport=udp
2444	65.105429	10.10.98.98	20.20.242.26	SIP Request: BYE sip:anonymous@20.20.242.26:5060;transport=udp
2452	65.175428	20.20.242.26	10.10.98.98	SIP Status: 200 OK

8. Conclusion

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test result met the objectives outlined in **Section 2.1**. The Winstream system is considered **compliant** with the Avaya Communication Server 1000 Release 7.0.

9. Additional References

Product documentation for ACME packet and Avaya products may be found at:

<http://www.acmepacket.com/support.htm>
<http://support.avaya.com/css/appmanager/public/support>

[1]*Communication Server 1000 Network Routing Service Fundamentals, Release 7.0, Revision 02.01, June 2010, Document Number NN43001-130*

[2]*IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000, Release 7.0, Revision: 04.01, June 2010, Document Number NN43001-313*

[3] *Communication Server 1000E Overview Release 7.0, Revision 04.02, April 2011, Document Number NN43041-110*

[4]*Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision 04.01, June 2010, Document Number NN43001-116*

[5]*Communication Server 1000 Dialing Plans Reference, Release 7.0, Revision 04.01, June 2010, Document Number NN43001-283*

[6]*Product Compatibility Reference, Avaya Communication Server 1000, Release 7.0, Revision 04.01, June 2010, Document Number NN43001-256*

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