



Application Notes for Avaya B179 SIP Conference Phone with Avaya Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference Telephone that registers as a 3rd Party SIP Line client with Communication Server 1000 Release 6.0. The solution supports calling among the B179 and other Communication Server 1000-supported non-SIP and SIP Line clients.

Testing was conducted by the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a 3rd Party SIP Line client with Communication Server 1000 Release 6.0. This solution supports calling among the B179 and other Communication Server 1000-supported non-SIP and SIP Line clients.

Figure 1 illustrates the network configuration of equipment that was used for testing. All telephones, including the B179, are registered to Avaya Communication Server 1000 release 6.0. The telephones were configured in the 55xxx extension range.

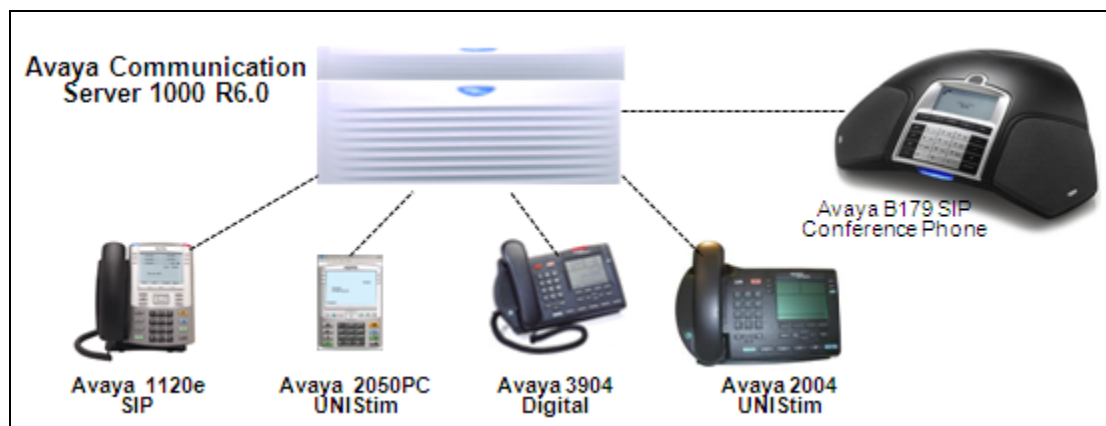


Figure 1: Network Configuration

2. Equipment and Software Validated

Provider	Hardware Component	Software Version
Avaya	Avaya Communication Server 1000E CPPM	VERSION 4021 RELEASE 6 ISSUE 00 R +
Avaya	Avaya 1140e IP Desk phone	SIP: 04.01.13.00
Avaya	Avaya 2004 IP Desk phone	UNiStim: 0622B76
Avaya	Avaya IP Softphone 2050PC	UNiStim: 4.01.041
Avaya	Avaya M3904 Digital Phone	N/A
Avaya	Avaya B179 SIP Conference Phone	2.2 and 2.2.1

Table 1: Hardware Components and Software Versions

Update Type	Update Components																																						
DepList	DepList 1: core Issue: 03 (created: 2011-04-26 15:23:48 (est))																																						
Service Packs	<p>In system patches: 6</p> <table> <tr> <td>NAME</td> <td>RPM</td> </tr> <tr> <td>p28774_1</td> <td>nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386</td> </tr> <tr> <td>p28797_1</td> <td>nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386</td> </tr> <tr> <td>p29703_1</td> <td>nortel-cs1000-shared-ssSubagent-6.00.18.00.i386</td> </tr> <tr> <td>p28961_1</td> <td>nortel-cs1000-pi-control-1.00.00.00-00.noarch</td> </tr> <tr> <td>p30043_1</td> <td>nortel-cs1000-OS-1.00.00.00-00.noarch</td> </tr> <tr> <td>p30274_1</td> <td>nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386</td> </tr> </table> <p>In System service updates: 11</p> <table> <tr> <td>NAME</td> <td></td> </tr> <tr> <td>nortel-cs1000-shared-pbx-6.00.18.065-01.i386.002</td> <td></td> </tr> <tr> <td>nortel-cs1000-patchWeb-6.00.18.65-01.i386.001</td> <td></td> </tr> <tr> <td>nortel-cs1000-vtrk-6.00.18.65-TMP410.i386.000</td> <td></td> </tr> <tr> <td>nortel-cs1000-dmWeb-6.00.18.62-00.i386.001</td> <td></td> </tr> <tr> <td>nortel-cs1000-ISECSH-6.00.18.62-00.i386.000</td> <td></td> </tr> <tr> <td>ntp-4.2.4p8-1.el5.pp.i386.000</td> <td></td> </tr> <tr> <td>tzdata-2009u-1.el5.noarch.000</td> <td></td> </tr> <tr> <td>nortel-cs1000-csv-6.00.18.65-04.i386.000</td> <td></td> </tr> <tr> <td>nortel-cs1000-linuxbase-6.00.18.65-06.i386.000</td> <td></td> </tr> <tr> <td>nortel-cs1000-auth-6.00.18.65-01.i386.000</td> <td></td> </tr> <tr> <td>nortel-cs1000-shared-general-6.00.18.62-01.i386.000</td> <td></td> </tr> </table>	NAME	RPM	p28774_1	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386	p28797_1	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386	p29703_1	nortel-cs1000-shared-ssSubagent-6.00.18.00.i386	p28961_1	nortel-cs1000-pi-control-1.00.00.00-00.noarch	p30043_1	nortel-cs1000-OS-1.00.00.00-00.noarch	p30274_1	nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386	NAME		nortel-cs1000-shared-pbx-6.00.18.065-01.i386.002		nortel-cs1000-patchWeb-6.00.18.65-01.i386.001		nortel-cs1000-vtrk-6.00.18.65-TMP410.i386.000		nortel-cs1000-dmWeb-6.00.18.62-00.i386.001		nortel-cs1000-ISECSH-6.00.18.62-00.i386.000		ntp-4.2.4p8-1.el5.pp.i386.000		tzdata-2009u-1.el5.noarch.000		nortel-cs1000-csv-6.00.18.65-04.i386.000		nortel-cs1000-linuxbase-6.00.18.65-06.i386.000		nortel-cs1000-auth-6.00.18.65-01.i386.000		nortel-cs1000-shared-general-6.00.18.62-01.i386.000	
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nortel-cs1000-vtrk-6.00.18.65-TMP410.i386.000																																							
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3. Configure Avaya Communication Server 1000

This section describes the steps to configure the following, using CS 1000 Element Manager:

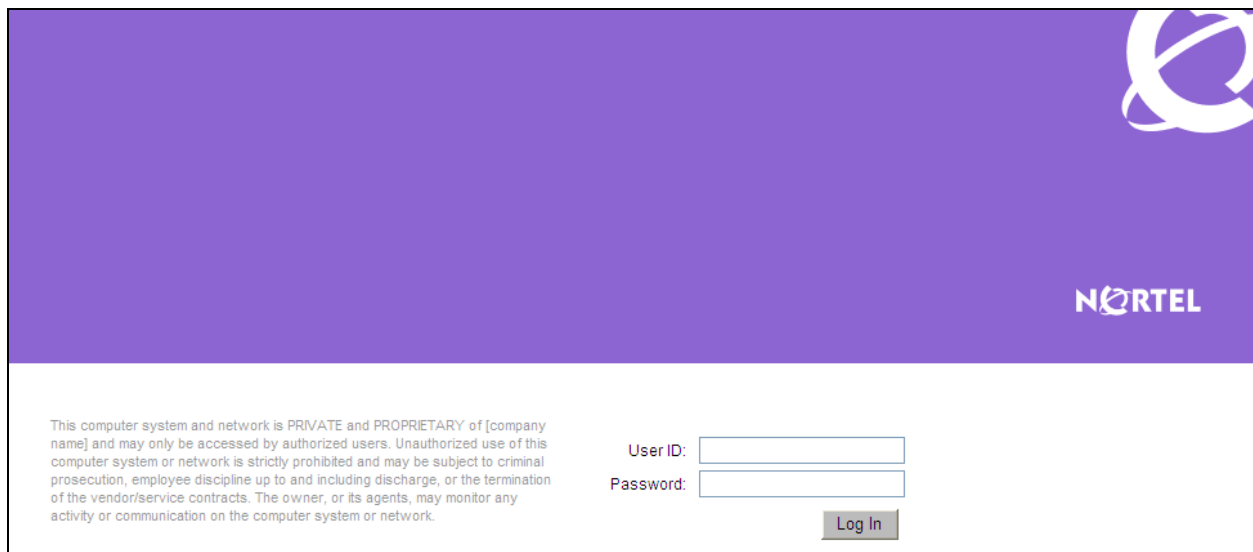
- SIP Line service
- SIP Line D-Channel
- Application Module Link (AML)
- Value Added Server (VAS)
- Zone for SIP phones
- SIP Line Route Data Block (RDB)
- SIP Line Virtual Trunk
- Media Gateway Controller
- SIP Line telephone corresponding to the B179 SIP Conference Phone

It is assumed that basic installation and configuration of the CS 1000 call server, signaling server, and node have been completed. Additional configuration details are provided in [1, 2].

3.1. Log in to Element Manager (EM)

Access the Unified Communications Management (UCM) web based interface by using the URL “http://<ip-address>” in an Internet browser window, where “<ip-address>” is the IP address of the UCM server.

Log in with the appropriate user name and password.



This computer system and network is PRIVATE and PROPRIETARY of [company name] and may only be accessed by authorized users. Unauthorized use of this computer system or network is strictly prohibited and may be subject to criminal prosecution, employee discipline up to and including discharge, or the termination of the vendor/service contracts. The owner, or its agents, may monitor any activity or communication on the computer system or network.

User ID:

Password:

The following **Unified Communications Management** screen will be displayed. Click on the **Element Name** corresponding to the **Element Manager (EM)**.

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

Host Name: ucm2.bvwdev.com Software Version: 02.00.0055.00(3266) User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service.

[Add...](#) [Edit...](#) [Delete](#)

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input checked="" type="checkbox"/>	EM on ss2	CS1000	6.0		New element.

The CS 1000 Element Manager page appears as shown below.

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

Managing: [redacted] Username: admin
System Overview

System Overview

IP Address: [redacted]

Type: Nortel Communication Server 1000E CPPM

Version: 4021

Release: 600 R +

[Active Sessions](#)

3.2. Enable SIP Line Service

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

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar contains a tree view with categories like UCM Network Services, Links, System, and Routes and Trunks. The 'Customers' link under 'Routes and Trunks' is highlighted. The main content area is titled 'Customers' and displays a table with columns for Customer Number, Total Routes, and Total Trunks. Two customers are listed: Customer 00 with 14 routes and 301 trunks, and Customer 03 with 0 routes and 0 trunks. The '00' in the first row is circled in red.

Customer Number	Total Routes	Total Trunks
1 00	14	301
2 03	0	0

The **Customer Details** screen is displayed next. Select **SIP Line Service** to edit its parameters.

The screenshot shows the 'Edit' screen for Customer 00. The left sidebar is the same as the previous screen, but the 'Customers' link is now expanded, showing a list of configuration options. The 'SIP Line Service' option is highlighted with a red circle. The main content area is titled 'Edit' and lists various configuration options like Basic Configuration, Application Module Link, Call Detail Recording, etc.

- Basic Configuration
- Application Module Link
- Call Detail Recording
- Call Party Name Display
- Call Redirection
- Centralized Attendant Service
- Controlled Class of Service
- Feature Options
- Feature Packages
- Flexible Feature Codes
- Intercept Treatments
- ISDN and ESN Networking
- Listed Directory Numbers
- Mobile Service Directory Numbers
- Multi-Party Operations
- Night Service
- Options
- Recorded Overflow Announcement
- SIP Line Service**
- Timers

Check the **SIP Line Service** checkbox, enter an appropriate **Root domain**, and click **Save**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: Username: admin
Customers » Customer 00 » Edit » SIP Line Service

SIP Line Service

☒ SIP Line Service

Root domain: SIPL60.COM

User agent DN prefix:

Optional features: ☐ Nortel Multimedia

*Required Value

Save Cancel

3.3. Enable SIP Line Service on Telephony Node

On the Element Manager page, navigate to **System → IP Network → Nodes: Servers, Media Cards**. Note the IP address of the SIP Line Node, as it will be used in configuring the B179 later. It would be displayed where X.X.X.X is indicated below. Select the **Node ID** on which SIP Line service is to be enabled.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

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

IP Telephony Nodes

Click the Node ID to view or edit its properties.

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

Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
555	2	LTPS, Gateway (SIPGw, H323Gw	-	X.X.X.X	Synchronized
556	1	SIP Li	-	X.X.X.X	Synchronized

Show: ☒ Nodes ☐ Component Servers and Cards

Scroll down the top section to display the **Applications** section on the right, and click on **SIP Line**.

The screenshot shows the 'Node Details (ID: 556 - SIP Li)' page in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation menu with categories like UCM Network Services, System, and Interfaces. The main content area shows configuration fields for the node, including Node ID (556), Call Server IP Address, and IP Telephony Node Properties. The Applications section on the right is expanded, showing options like SIP Line, Terminal Proxy Server (TPS), and Gateway. The SIP Line option is circled in red.

The SIP Line Configuration Details page is displayed. Check **Enable gateway service on this node** next to **SIP Line Gateway Application**: Then click **Save**.

The screenshot shows the 'SIP Line Configuration Details' page in the Nortel CS 1000 Element Manager. The page displays configuration fields for the SIP Line Gateway Application, including SIP Domain name, SLG endpoint name, and SLG Group ID. The 'Enable gateway service on this Node' checkbox is checked and circled in red. The 'Save' button is also circled in red.

The **Node Details** screen then returns. Click the **Save** button on this screen.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

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

Node Details (ID: 556 - SIP Li)

Node ID: * (0-9999)

Call Server IP Address:

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

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

IP Telephony Node Properties

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

Applications (click to edit configuration)

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

* Required Value.

Select **Transfer Now** on the **Node Saved** page as shown below.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

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

Node Saved

Node ID: 556 has been saved on the call server.

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

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

You may initiate a transfer manually at a later time.

Once the transfer completes, the **Synchronize Configuration Files (Node ID <id>)** page is displayed as shown below. Check the appropriate SIP Line Server and click **Start Sync**. The screen will automatically refresh until the synchronization is finished. The **Synchronization Status** field will update from **Sync required** (as shown) to **Synchronized** (not shown). After synchronization completes, click **Restart Applications** to use the new SIP Gateway settings.

Nortel CS 1000 ELEMENT MANAGER

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

Synchronize Configuration Files (Node ID <556>)

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

[Print](#) | [Refresh](#)

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	ucm2	Signaling Server	SIP Line	Sync required

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

3.4. Configure SIP Line D-Channel

On the left column menu of the main Element Manager page, navigate to **Routes and Trunks** → **D-Channels**. Under the **Configuration** section, select a D-Channel number from the **Choose a D-Channel Number** list (channel 33 in the sample configuration), and select **DCH** for the **type**. Click to **Add**.

Nortel CS 1000 ELEMENT MANAGER

Managing: 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	Type	Card Type	Description	Edit
Channel: 3	Type: DCH	Card Type: TMDI	Description: QSIG_M3K	<input type="button" value="Edit"/>
Channel: 4	Type: DCH	Card Type: TMDI	Description: core7	<input type="button" value="Edit"/>
Channel: 11	Type: DCH	Card Type: MSDI	Description: PRI2_QSIG	<input type="button" value="Edit"/>
Channel: 18	Type: DCH	Card Type: TMDI	Description: Rls75	<input type="button" value="Edit"/>
Channel: 20	Type: DCH	Card Type: DCH	Description: VTRK	<input type="button" value="Edit"/>

The **D-Channels Property Configuration** screens below show the parameter values after configuring the D-channel. **DCIP** is selected for **D channel Card Type**, **Meridian Meridian1 (SL1)** is selected for **Interface type for D-channel**, and an appropriate **Designator** is entered. The remaining parameters have their default values.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: [] Username: admin
Routes and Trunks » **D-Channels** » D-Channels 30 Property Configuration

D-Channels 30 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SIPLine
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CNTY)	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	<input type="text"/>
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	6
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000

[+ Basic options \(BSCOPT\)](#)
[+ Advanced options \(ADVOPT\)](#)
[+ Feature Packages](#)

Click the **Basic options (BSCOPT)** link to expand that section. Click **Edit** to configure **Remote Capabilities**.

Signaling Server Resource Capacity (SSRC) 1800 Range: 0 - 4000

- Basic options (BSCOPT)

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

- PINX customer number (PINX_CUST)

- Progress signal (PROG)

- Calling Line Identification (CLID)

- Output request Buffers (OTBF) 32

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

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

- Remote Capabilities (RCAP) [Edit](#)

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

- B channel Service messaging. (BSRV) ☐

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

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

The **Remote Capabilities Configuration** page is displayed as shown below. Select the **Message waiting interworking with DMS-100 (MWI)** check box,¹ and the **Network name display method 2 (ND2)** check box. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** (not shown), and the **D-Channel Property Configuration** page reappears as shown in the previous screen. Click on **Submit**.

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

Managing: Username: admin
Routes and Trunks » [D-Channels](#) » [D-Channels 30 Property Configuration](#) » - Remote Capabilities Configuration

- Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN 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 EuroISDN 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"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (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"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>

¹ Note that although the Avaya B179 SIP Conference Telephone does not support Message Waiting Indicator, this D channel can also be used for other SIP Line IP telephones that do support it, so it is enabled here for that purpose.

3.5. Configure Application Module Link (AML)

On the left column menu of the main Element Manager page, navigate to **System → Interfaces → Application Module Link**, and click **Add** (not shown). The **New Application Module Link** page is displayed. Enter the AML port number in the **Port number** text box. The SIP Line Service can use ports 32 through 127. In the sample configuration, the SIP Line Service is configured to use port 32. Enter an appropriate **Description**. Click **Save** to save the configuration.

The screenshot shows the 'New Application Module Link' configuration page in the CS 1000 ELEMENT MANAGER. The left sidebar contains a navigation menu with 'Application Module Link' selected. The main content area has the following fields: 'Port number' set to 32 (range 16-127), 'Description' set to 'ForSIPLine', and 'Link control system parameters' unchecked. The 'Maximum octets' is set to 512 (per HDLC frame). At the bottom right, there are 'Save' and 'Cancel' buttons, with 'Save' circled in red.

3.6. Configure Value Added Server (VAS)

On the left column menu of the main Element Manager page, navigate to **System → Interfaces → Value Added Server**. Click **Add** and then click **Ethernet LAN Link** on the **Add Value Added Server** page that is displayed next (not shown). On the **Ethernet Link** page that is displayed next, enter a **Value added server ID** (32 in the sample configuration), and select the AML number created in the previous section for **Ethernet LAN Link**. Ensure that the **Application Security** check box is unchecked. Click **Save** (not shown). The screen below shows the result of adding the value added server.

The screenshot shows the 'Edit Value Added Server 032' configuration page in the CS 1000 ELEMENT MANAGER. The left sidebar contains a navigation menu with 'Value Added Server' selected. The main content area has the following fields: 'Ethernet LAN Link' set to 032, 'Application Security' unchecked, 'Interval' set to 1 (Time interval for checking the link for overload in five second increments), and 'Message Count Threshold' set to 9999 (range 10-9999). At the bottom right, there are 'Save' and 'Cancel' buttons, with 'Save' circled in red.

3.7. Configure Zone for SIP Phones

On the left column menu of the main Element Manager page, navigate to **System → IP Network → Zones**. On the **Zones** page, select **Bandwidth Zones** (not shown). On the **Bandwidth Zones** page, choose a new Bandwidth Zone from the drop-down box and click on **Add**.

The screenshot shows the 'Bandwidth Zones' page in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation menu with 'Zones' highlighted. The main content area has a breadcrumb trail: 'System » IP Network » Zones » Bandwidth Zones'. Below this, there are sections for 'Maintenance' (Maintenance Commands for Zones (LD 117)) and 'Configuration' (Configuration Spreadsheet). A 'Please Choose the' dropdown menu is set to 'Bandwidth Zones 5', and a red box highlights the 'Add' button next to it.

On the **Zone Basic Property and Bandwidth Management** page, enter an appropriate **Description**. Defaults can be used for the remaining fields. Click **Submit**.

The screenshot shows the 'Zone Basic Property and Bandwidth Management' page. The left sidebar is the same as the previous screenshot. The main content area has a breadcrumb trail: 'System » IP Network » Zones » Bandwidth Zones » Bandwidth Zones 5 » Zone Basic Property and Bandwidth Management'. Below this, there is a table with 'Input Description' and 'Input Value' columns. The table contains the following fields: Zone Number (ZONE), Intrazone Bandwidth (INTRA_BW), Intrazone Strategy (INTRA_STGY), Interzone Bandwidth (INTER_BW), Interzone Strategy (INTER_STGY), Resource Type (RES_TYPE), Zone Intent (ZBRN), and Description (ZDES). The 'Description (ZDES)' field is highlighted with a red box and contains the text 'IPPHONES'. At the bottom left, there are 'Submit' and 'Cancel' buttons, with the 'Submit' button highlighted by a red box.

3.8. Configure SIP Line Route Data Block (RDB)

On the left column menu of the main Element Manager page, navigate to **Routes and Trunks** → **Routes and Trunks**. Click **Add route** for the appropriate customer number.

The screenshot shows the 'Routes and Trunks' configuration page in the Nortel CS 1000 Element Manager. The left sidebar contains a navigation menu with options like 'UCM Network Services', 'Home', 'Links', 'Virtual Terminals', 'System', 'Alarms', 'Maintenance', 'Core Equipment', 'Peripheral Equipment', 'IP Network', 'Nodes: Servers, Media Cards', 'Maintenance and Reports', and 'Media Gateways'. The main content area displays a table with the following data:

Customer	Total routes	Total trunks	Action
+ Customer: 0	14	301	Add route
- Customer: 3	0	0	Add route

The following screen shows the parameter settings after the route has been added. Set the following parameters and leave default values for the remaining parameters. The **Basic Route Options**, **Network Options**, **General Options**, and **Advanced Configurations** sections (not shown) can be left at the defaults. Click **Submit** (not shown) to save the configuration changes.

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

The route is for a virtual trunk route (VTRK)

Zone for codec selection and bandwidth management (ZONE)

Node ID of signaling server of this route (NODE)

Protocol ID for the route (PCID)

Integrated services digital network option (ISDN)

Mode of operation (MODE)

D channel number (DCH)

Interface type for route (IFC)

Network calling name allowed (NCNA)

Select the route number

Enter an appropriate name

Select **TIE trunk data block (TIE)**

Select **Incoming and Outgoing (IAO)**

Enter the access code

Check the box

Enter a zone²

Enter the node ID of the SIP Line Gateway

Select **SIP Line (SIPL)**

Check the box

Select **Route uses ISDN Signaling Link (ISLD)**

Enter the D-channel number

Select **Meridian M1 (SL1)**

Check the box

- Basic Configuration

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P) ☐

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

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

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

- Protocol ID for the route (PCID)

Integrated services digital network option (ISDN) ☒

- Mode of operation (MODE)

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

- Interface type for route (IFC)

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

- Network calling name allowed (NCNA) ☒

² Note that this must be a zone of type VTRK and must be different than the zone created for the SIP phones in **Section 3.7**. In the sample configuration, the VTRK zone was 254.

3.9. Configure SIP Line Virtual Trunk

When the **Routes and Trunks** screen is displayed after adding the route in **Section 3.8**, click **Add trunk** corresponding to the newly added route to add new trunk members. The following screen shows the parameter settings for one of the trunks after they have been added. Set the following parameters and leave default values for the remaining parameters. Click **Save** to save the configuration changes.

Multiple trunk input number

Enter the number of trunks (only shown when adding trunks)

Trunk data block

Select **IP Trunk (IPTI)**

Terminal Number

An available terminal number.

Designator field for trunk

A descriptive text.

Extended Trunk

Select **Virtual trunk (VTRK)**

Route number, Member number

Current route number and starting member. (only shown when adding trunks)

Card Density

Select **Octal Density (8D)**

Start arrangement Incoming

Select **Immediate (IMM)**

Start arrangement Outgoing

Select **Immediate (IMM)**

Trunk Group Access Restriction

Desired trunk group access restriction level.

Channel ID for this trunk

An available starting channel ID.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: [redacted] Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 30, Trunk 1 Property Configuration

Customer 0, Route 30, Trunk 1 Property Configuration

- Basic Configuration

Input Description	Input Value
Trunk data block (TYPE)	IPTI
Terminal Number (TN)	100 0 01 00
Designator field for trunk (DES)	SIPLINE
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	30 1 *
Level 3 Signaling (SIGL)	[dropdown]
Card Density (CDEN)	8D
Start arrangement Incoming (STRI)	Immediate (IMM) [dropdown]
Start arrangement Outgoing (STRO)	Immediate (IMM) [dropdown]
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk. (CHID)	32
Increase or decrease the member numbers (INC)	Increase channel and member number (YES) [dropdown]
Class of Service (CLS)	Edit

+ Advanced Trunk Configurations

Save Delete Cancel

3.10. Configure Media Gateway Controller

This section describes configuration of the G.729 audio codec for the Media Gateway Controller (MGC) to support calls between the B179 and non-IP telephones. On the left column menu of the main Element Manager page, navigate to **IP Network** → **Media Gateways**. Click on the **IPMG** that supports the digital and analog phones in the system.

CS 1000 ELEMENT MANAGER

Managing: [] Username: admin
System » IP Network » Media Gateways

Media Gateways

Buttons: Add... Digital Trunking... Reboot Delete Virtual Terminal More Actions Refresh

IPMG	IP Address	Zone	Type
004.00	[]	001	MGC
012.00	[]	001	MGC

On the **IPMG Property Configuration** screen, click **Next** (not Shown). Expand the **VGW and IP phone codec profile** section. In that section, check the **Select** checkbox next to and expand the **Codec G729A** section.

CS 1000 ELEMENT MANAGER

Telephony LAN (TLAN) gateway IP address []
Telephony LAN (TLAN) subnet mask 255.255.255.192
Hostname DB2

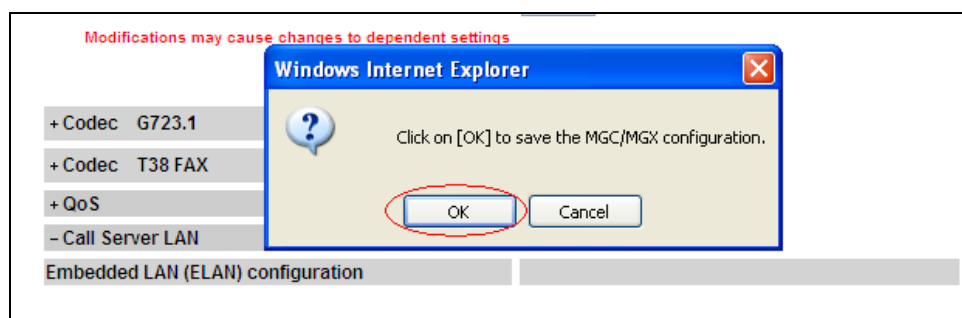
- VGW and IP phone codec profile

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)
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 playout nominal delay 100 (0 - 300 milliseconds)
FAX no activity timeout 20 (10 - 32000 milliseconds)
FAX packet size 30

+ Codec G711 Select ☒
+ Codec G729A Select ☒
+ Codec G723.1 Select ☐

If Annex B support is desired as in the sample configuration, check the **VAD** checkbox. Note that the VAD setting should be consistent with the VAD setting in the B179 configuration. Click **Save**.

Click on **OK** to save the configuration.



When the Media Gateway screen returns, select the radio button for the **IPMG** and click **Reboot**.

CS 1000 ELEMENT MANAGER

Managing: 135.10.97.71 Username: admin
System » IP Network » Media Gateways

Media Gateways

Buttons: Add..., Digital Trunking..., **Reboot**, Delete, Virtual Terminal, More Actions, Refresh

	IPMG	IP Address	Zone	Type
<input checked="" type="radio"/>	004 00		001	MGC
<input type="radio"/>	012 00		001	MGC

3.11. Configure SIP Line Telephone

This section describes the screens for configuring a SIP Line telephone to support the Avaya B179 SIP Conference Telephone. On the left column menu of the main Element Manager page, navigate to **Phones**. On the **Search For Phones** page, click **Add....**

CS 1000 ELEMENT MANAGER

Managing: EM on ss2/ Search for Phone

Search For Phones

Criteria: Prime DN Value: Search

Results Per Page: 10 Search

Phones Found (0)

Buttons: **Add...**, Import..., Retrieve..., Delete, <More Actions>, Refresh

	Customer	TN	Prime DN	Designation	Phone Type	Template	UUID
No records to display!							

On the **New Phones** page, select the **Customer**, select the **Phone Type** radio button, and then select **UEXT-SIPL – Universal Extension SIPL**. Click **Preview**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: EM on ss2i
[Phones](#) » New Phones

New Phones

Number of phones : 1 (1-100)

Customer : 0

☒ Phone Type **UEXT-SIPL - Universal Extension SIPL**

Type : ☐ Template ☐ Copy From TN

Options :

- ☐ Default value for DES
- ☐ Default value for ZONE
Only applicable to IP phone types
- ☐ Default value for Node Id
Only applicable to UEXT-SIPL phone types
- ☐ Automatically assign DN starting DN
- ☐ Automatically assign TN starting TN

* Required value. **Preview** Cancel

The following screens show the parameter values after the phone has been added. In the **General Properties** section, fill in the following fields, and leave the remaining fields at their default values:

Customer Number

Select the customer number

Terminal Number

Enter a free TN number

Designation

Enter a reference name

Zone

Enter the zone from **Section 3.7**

SIP User Name

The phone extension number used to log in at the phone

Node Id

The ID of this node

Optional Features: Max Client Count

Select the check box

SIPN

Set to **0**

SIP3

Set to **1**

The screenshot displays the Nortel CS 1000 Element Manager web interface. The top navigation bar includes the Nortel logo, the title "CS 1000 ELEMENT MANAGER", and links for "Help" and "Logout". A left-hand menu lists various system management categories such as "UCM Network Services", "System", "Customers", "Routes and Trunks", "Dialing and Numbering Plans", "Phones", "Tools", and "Security". The main content area is titled "Managing: EM on ss2(.....)" and "Phones»Phone Details". Below this, the "Phone Details" section shows a phone icon and system information: "System: EM on ss2", "Phone Type: UEXT-SIPL", and "Sync Status: TRN". A tabbed interface below shows "General Properties" as the active tab. This section contains two red-bordered boxes. The first box contains fields for "Customer Number" (a dropdown menu), "Terminal Number" (text input "096 0 01 29"), "Designation" (text input "B179"), "Zone" (dropdown menu "001"), "SIP User Name" (text input "55575"), and "Node Id" (text input "556"). Below these is a "Super User" checkbox. The second box, titled "Optional Features", shows the "Max Client Count" checkbox checked, with input fields for "SIPN" (0), "SIP3" (1), "FMCL" (0), and "TLSV" (0).

In the **Features** section, fill in the following fields, leaving the remaining fields at their defaults.

Call Party Name Display (CNDA)	Allowed
Call Number Information (CNIA)	Allowed
Restricted Conference or Transfer (FTTC))	Unrestricted Conf. or Transfer
Media Security Encryption (MSEC)	Media Security Never (MSNV)
Station Control Password (SCPW)	Enter password used to log in at the phone
Trunk Group Access Restriction (TGAR)	Set appropriately
Instrument Type (TYPE)	UEXT
Universal Extension User (UTXY)	SIPL

In the **Keys** section, fill in the following:

Key No. 0	SCR – Single Call Ringing
Directory Number	Phone extension number
Multiple Appearance Redirection Prime (MARP)	Select the Checkbox
First Name	Enter a name
Last Name	Enter a name
Key No. 1	HOT_U – Hotline(Universal)
UADN	The phone extension prefixed by the UADN Prefix ³

The screenshot displays the Avaya configuration interface. On the left is a navigation tree with categories like UCM Network Services, Home, Links, 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, and Unicast Name Directory. The main area is divided into two sections: Features and Keys.

Features Section: A table with columns Feature, Description, and Value. It lists features like CLTA (Network Call Trace, Denied), CNDA (Call Party Name Display, Allowed), CNIA (Call Number Information, Allowed), CNTA (Network ACD Countdown, Denied), and CPFA (Forced Camp-On From This Set, Allowed).

Keys Section: A table with columns Key No., Key Type, and Key Value. It shows two keys:

- Key No. 0:** Key Type is "SCR - Single Call Ringing". Key Value includes Directory Number "55575", a checked checkbox for "Multiple Appearance Redirection Prime(MARP)", First Name "Avaya", Last Name "B179", Display Format "First, Last", and Language "Roman".
- Key No. 1:** Key Type is "HOT_U - Hotline(Universal)". Key Value includes UADN "2655575".

Click **Save** (not shown) to save the configuration for this phone.

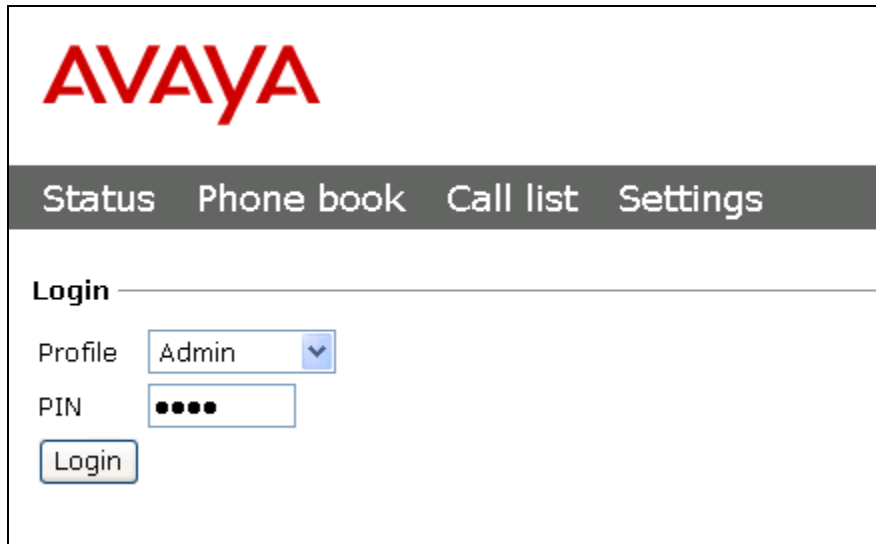
³ The UADN is used to make and receive calls between the SIP Line Gateway and the Universal Extensions. However, this key is used only by the SIP Line Gateway (SLG) application. The UADN is not dialed by end users. It is only used internally between the Call Server and the SIP Line Gateway application. See **Section 3.2**.

4. Configure Avaya B179 IP Conference Phone

This section describes how to access the B179 web interface and configure the phone to register to Avaya Communication Server 1000. It assumes that the telephone has been administered an IP address either through DHCP or static configuration. Additional configuration details are provided in [3].

4.1. SIP Registration

In the web browser address field, enter the B179 IP address. The login page will appear as shown below. Enter **admin** for the user name and the appropriate password.

The image shows the login page of the Avaya B179 IP Conference Phone web interface. At the top is the red 'AVAYA' logo. Below it is a dark grey navigation bar with four white links: 'Status', 'Phone book', 'Call list', and 'Settings'. Underneath the navigation bar is a white box with the title 'Login'. Inside this box, there is a 'Profile' label next to a dropdown menu showing 'Admin' with a downward arrow. Below that is a 'PIN' label next to a text input field containing five black dots. At the bottom of the login box is a 'Login' button.

Click **Login**, and the main configuration screen appears as shown on the next page, where **Settings** → **Network** has been selected and shows the DHCP network configuration that was configured on the B179 in the sample configuration.

Network

DHCP ☒ On ☐ Off

IP address Hostname

Netmask Domain

Gateway

Primary DNS

Secondary DNS

Quality of Service

SIP DiffServ (0-63)

Media DiffServ (0-63)

VLAN ☐ On ☒ Off

VLAN ID

VLAN map enable ☐ On ☒ Off

VLAN prio SIP

VLAN prio media

802.1x

Enable 802.1x ☐ On ☒ Off

EAP method ☐ MD5 ☐ TLS

Username

4.2. Configure SIP Signalling Settings

To configure the SIP signalling settings, navigate to **Settings → SIP**, and fill in the following:

Under **Account 1**:

Enable account	Select the Yes radio button
Account name	Meaningful name for account status display on phone screen
User	Extension (SIP User Name) of the SIP Line telephone configured in Section 3.11
Realm	Use the default of “*”
Registrar and Proxy Authentication name	SIP domain configured in the CS 1000, with Port number Extension (SIP User Name) of the SIP Line telephone configured in Section 3.11
Password	The Station Control Password of the SIP Line telephone configured in Section 3.11
Registration interval	Enter a value (300 was used in the sample configuration)

Under **Advanced**:

Enable blind transfer	Select the No radio button ⁴
Outbound proxy	No input was used in this configuration

Under **Transport**:

Protocol	Select the TCP or UDP radio button (UDP shown)
Local Port	Enter 5070

Click **Save**.

The SIP configuration screen is shown on the next page.

⁴ This feature is not yet supported in this configuration



Status Phone book Call list **Settings**

Basic SIP Network Media LDAP Web interface Time & Region Provisioning System

Account 1

Enable account	<input checked="" type="radio"/> Yes <input type="radio"/> No		
Account name ⓘ	<input type="text" value="55575"/>	Realm ⓘ	<input type="text" value="*"/>
User ⓘ	<input type="text" value="55575@sipl60.com"/>	Authentication name ⓘ	<input type="text" value="55575"/>
Registrar ⓘ	<input type="text" value=":5070"/>	Password	<input type="password" value="••••"/>
Proxy ⓘ	<input type="text" value=":5070"/>	Registration interval ⓘ	<input type="text" value="300"/>

Account 2

Enable account	<input type="radio"/> Yes <input checked="" type="radio"/> No		
Account name	<input type="text"/>	Realm	<input type="text" value="*"/>
User	<input type="text"/>	Authentication name	<input type="text"/>
Registrar	<input type="text"/>	Password	<input type="password"/>
Proxy	<input type="text"/>	Registration interval	<input type="text"/>

NAT Traversal

STUN ⓘ	<input type="radio"/> On <input checked="" type="radio"/> Off	STUN host	<input type="text"/>
Offer ICE	<input type="radio"/> Yes <input checked="" type="radio"/> No		
TURN ⓘ	<input type="radio"/> On <input checked="" type="radio"/> Off	TURN user	<input type="text"/>
TURN host	<input type="text"/>	Password	<input type="password"/>

Advanced

Enable SIP Replaces	<input checked="" type="radio"/> Yes <input type="radio"/> No
Enable Blind Transfer	<input type="radio"/> Yes <input checked="" type="radio"/> No
Allow contact rewrite	<input type="radio"/> Yes <input checked="" type="radio"/> No
Outbound proxy	<input type="text"/>

Transport

Protocol	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS <input type="radio"/> SIPs	<i>Please check corresponding media signalling setting</i>
Local UDP port	<input type="text" value="5070"/>	

Save Cancel

4.3. Media Configuration

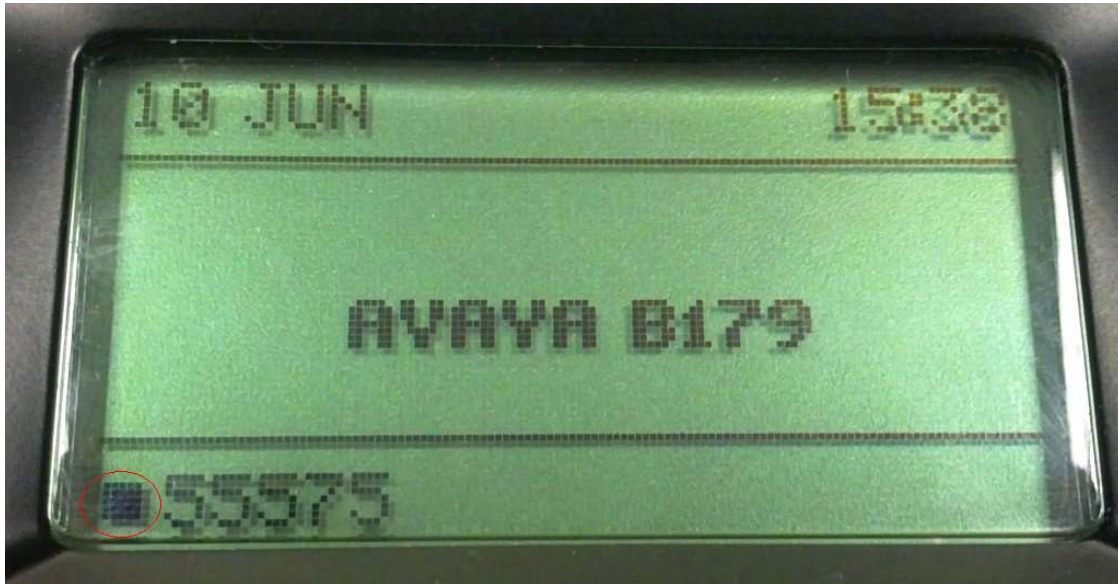
To configure the audio codec settings, navigate to **Settings** → **Media**, and select the priority for codec selection. CS1000 6.0 does not support the G7.22 Codec, so this selection should be set to **0 – Disabled**. Because there were some interoperability issues with G.729 it is recommended that this option should also be set to **0 – Disabled**. Set the G711 Ulaw codec to **4 – High**. Defaults can be used for the remaining fields. Click on **Save** when done.

The screenshot shows the Avaya web interface for Media Configuration. The top navigation bar includes Status, Phone book, Call list, and Settings. The Settings tab is active, and the Media sub-tab is selected. The interface is divided into sections: Codec, Security, VAD, DTMF, and Advanced. The Codec section contains a table with the following data:

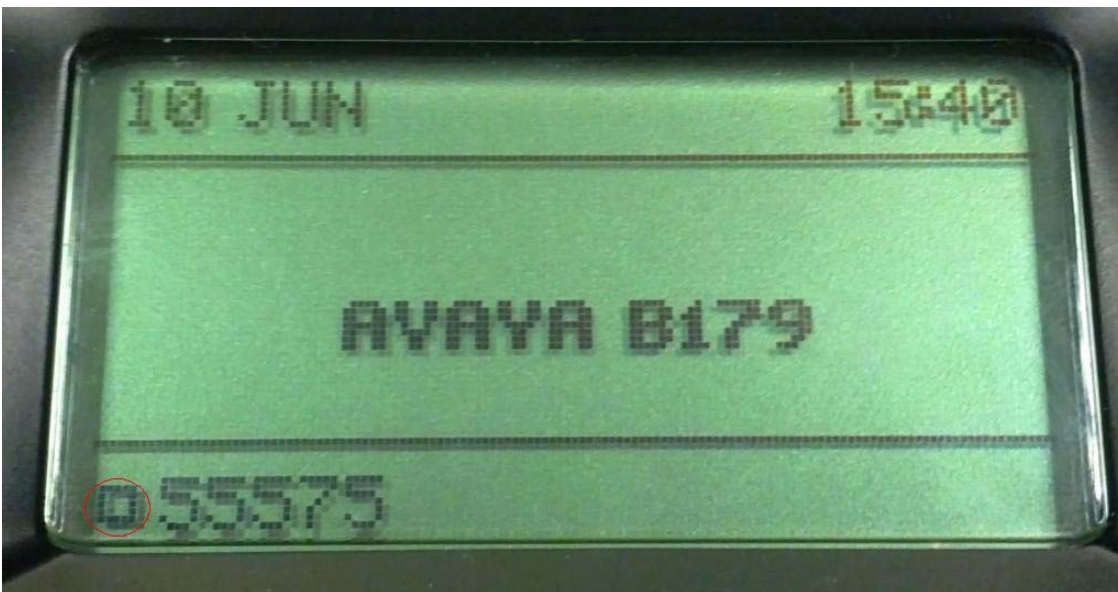
Codec	Priority
G722	0 - Disabled
G711 Alaw	0 - Disabled
G711 Ulaw	4 - High
G729	0 - Disabled

The Security section includes options for SRTP (Disabled, Optional, Mandatory) and Secure signalling (No, TLS, SIPs). The VAD section has an option for Enable VAD (Yes, No). The DTMF section has an option for DTMF Signalling (RFC 2833, SIP Info, Inband). The Advanced section has a text input for First RTP port (4000). The Save and Cancel buttons are at the bottom left.

After the configuration has been saved, the B179 will register with the CS 1000, and a display similar to those shown in the figures below will appear on the telephone. The **Hostname** is displayed at the center, and in the lower left corner is the **Account name**. To the left of the **Account name** is a square icon that indicates the SIP registration status of the B179. If the square is filled in as shown below, the B179 has successfully registered.



If the square is not filled in, registration was unsuccessful.



5. Observations

During testing with this configuration the following observations were made:

- Calls from the B179 via the inbound call log are not supported.
- Group conference by the B179 is not supported.
- The use of the G.729 audio codec is not recommended because of possible interoperability issues.
- It is recommended to disable Blind Transfer as indicated in section 4.2 because of possible interoperability issues.

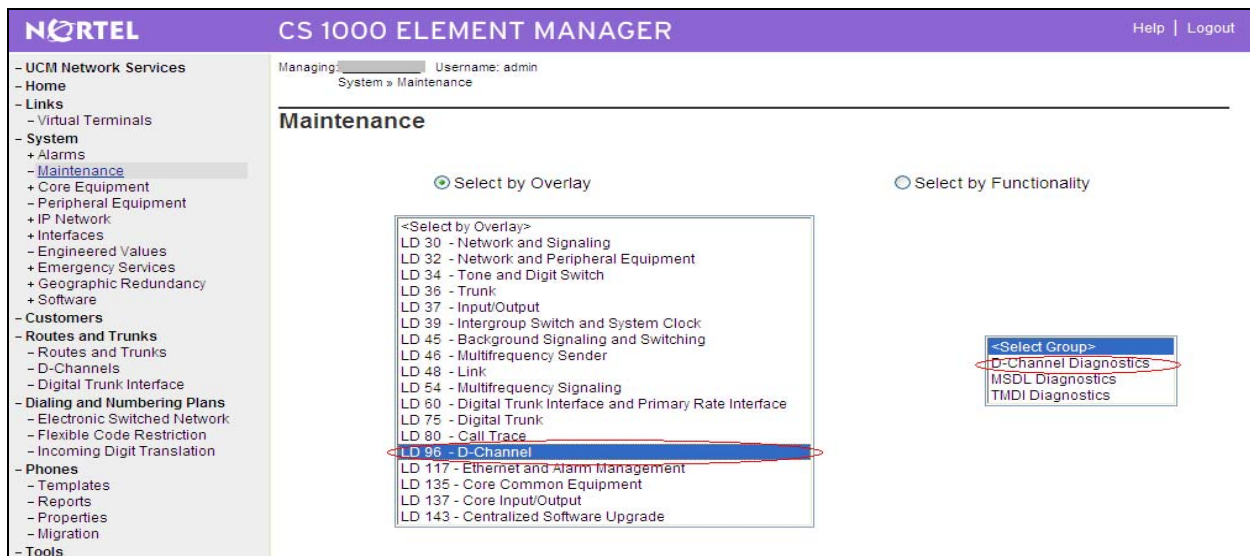
6. Verification Steps

This section provides tests that can be performed to verify proper configuration of the CS 1000 and B179.

6.1. Verify Avaya Communication Server 1000

6.1.1. Verify D-Channel Status

Verify status of the SIP Line D-Channels by navigating to **System → Maintenance**, selecting **Select by Overlay**, **LD 96 – D-Channel**, and **D-Channel Diagnostics**.



The screen below shows the **APPL_STATUS** of the SIP Line D-Channel as “OPER” and the **LINK_STATUS** as “EST ACTV”. This is normal.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing: Username: admin
System » Maintenance » D-Channel Diagnostics

D-Channel Diagnostics

Diagnostic Commands	Command Parameters	Action
Status for D-Channel (STAT DCH)		<input type="button" value="Submit"/>
Disable Automatic Recovery (DIS AUTO)	<input type="checkbox"/> ALL	<input type="button" value="Submit"/>
Enable Automatic Recovery (ENL AUTO)	<input type="checkbox"/> FDL	<input type="button" value="Submit"/>
Test Interrupt Generation (TEST 100)		<input type="button" value="Submit"/>
Establish D-Channel (EST DCH)		<input type="button" value="Submit"/>

DCH	DES	APPL_STATUS	LINK_STATUS	AUTO_REC	PDCH	BDCH
<input type="radio"/> 003	QSIG_M3K	DSBL	RST	AUTO		
<input type="radio"/> 004	core7	OPER	EST ACTV	AUTO		
<input type="radio"/> 011	PRI2_QSIG	OPER	AEST	AUTO		
<input type="radio"/> 018	Rls75	OPER	EST ACTV	AUTO		
<input type="radio"/> 020	VTRK	OPER	EST ACTV	AUTO		
<input type="radio"/> 030	SIPLine	OPER	EST ACTV	AUTO		

6.1.2. Verify SIP Registration Status

In the Element Manager Web interface, navigate to **System → IP Network → Maintenance and Reports** on the left pane. If there are multiple Nodes, select the node of the SIP Line. Click **GEN CMD**.

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

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

Node Maintenance and Reports

Index	ELAN IP	Type	TN	ELAN
ucm2		Signaling Server-HP	NO TN	

DL320G4

The **General Commands** page is displayed. From the **Group** drop-down menu select **SipLine**, from the **Command** drop-down menu select **slgSetShowByUID**, enter the B179 extension in **UserID**, and click on **RUN**. The output shown indicates successful registration and displays details of the registration parameters. Note that if the B179 has not registered, the error message “Invalid userId 55575” will be returned instead of the detailed registration information.

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

Managing: Username: admin

System » IP Network » Node Maintenance and Reports » General Commands

General Commands

Element IP : 135.10.97.74 Element Type : Signaling Server-HP DL320G4

Group SipLine

Command slgSetShowByUID

UserID 55575

RUN

IP address

Number of pings 3

PING

UserID	TN	Clients	Calls	SetHandle
55575	096-00-01-29	1	0	0x95dbec8

```

StatusFlags = Registered Controlled KeyMapDwid SSD
FeatureMask = MWI
CallProcStatus = -1
Current Client = 0, Total Clients = 1
== Client 0 ==
IP:Port:Trans = :5070:udp
Type = SIP3
UserAgent = Avaya B179 2.2.1
x-nt-guild = 399eeabbe596a3e30a1c2ae7a8c960d6
RegDescrip =
RegStatus = 1
PbxReason = OK

```

6.2. Verify Avaya B179 SIP Conference Phone

Successful registration of the phone can be verified by inspecting the status icon to the left of the **Account name**, shown at the lower left of the telephone display as described in Section 4.3. Registration and call tracing can be performed on the B179 by navigating to **Status → Log**. Select **SIP Trace** on the left and click **Change**. Ensure that the **SIP logging** radio button is selected. After attempting registration, click **Refresh** to see the result. The log can be cleared at any time by clicking **Clear Log**. The screen below shows the REGISTER message sent by the B179 for a successful registration to the CS 1000.

The screenshot displays the Avaya B179 configuration web interface. At the top, the Avaya logo is on the left, and the user is logged in as ADMIN with a Logout link on the right. A navigation bar contains tabs for Status, Phone book, Call list, and Settings. Below this, a secondary bar highlights Device, Network, Time & Region, SIP, Media, Log, and Licenses. The SIP Trace section is active, showing a dropdown menu set to 'SIP Trace' with a 'Change' button and a 'Refresh' button. Underneath, the 'SIP Trace' heading is followed by 'SIP logging' with 'On' selected (radio button) and 'Off' as an option, along with 'Set' and 'Clear Log' buttons. A scrollable text area displays a SIP REGISTER message log entry from Jun 10 16:15:05.

```
Jun 10 16:15:05: TX 523 bytes Request msg REGISTER/cseq=41162 (tdta0x1f1a40) to UDP [redacted]:5070:
REGISTER sip:[redacted]:5070;transport=udp SIP/2.0
Via: SIP/2.0/UDP [redacted]:5070;rport;branch=z9hG4bKPjdctB911tSb4A2DVTNhFzE7ruUxHysL4
Max-Forwards: 70
From: <sip:55575@sipl60.com>;tag=RbXoi4TYE6awKj7sMb1cQW-RmmgI1M3Y
To: <sip:55575@sipl60.com>
Call-ID: 2wLHXUeq56Tm-.yaY15NLBhQkF6HJY5U
CSeq: 41162 REGISTER
User-Agent: Avaya B179 2.2.1
Contact: <sip:55575@[redacted]:5070;ob>
Expires: 0
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Content-Length: 0
```

7. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone can be used together in an integrated solution.

8. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] *Communication Server 1000 - Element Manager System Reference – Administration*, Release: 6.0, Document Revision: 03.20, Document #NN43001-632.
- [2] *Communication Server 1000 SIP Line Fundamentals*, Release 6.0, Document #NN43001-508, 01.08, 9 February 2010.
- [3] *Avaya B179 SIP Conference Phone Installation and Administration Guide*

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