



Avaya Solution & Interoperability Test Lab

Application Notes for Avaya B179 SIP Conference Phone with Avaya Communication Server 1000 Release 7.5 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 7.5 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a standard SIP Line client with Communication Server 1000. 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 at the request of Product Management.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 7.5 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a standard SIP Line client with Communication Server 1000. 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 at the request of Product Management.

As shown in **Figure 1**, all telephones, including the B179, are registered to Avaya Communication Server 1000, which is configured as a co-resident single server system. The telephones are configured in the 57xxx extension range.



Figure 1: Network Configuration

2. Equipment and Software Validated

Provider	Hardware Component	Software Version
Avaya	Avaya Communication Server 1000E	7.50Q 7.50.17 (see Table 2 for applied updates)
Avaya	Avaya 1120E IP Deskphone	UNIStim: 0624C8A
Avaya	Avaya 1165e IP Deskphone	SIP: 04.00.04.00
Avaya	Avaya 1230 IP Deskphone	SIP: 04.00.04.00
Avaya	Avaya 2007 IP Deskphone	UNIStim: 0621C8A
Avaya	Avaya IP Softphone 2050PC	UNIStim: 4.01.041
Avaya	Avaya M3903 Digital Phone	N/A
Avaya	B179 SIP Conference Phone	2..2

Table 1: Hardware Components and Software Versions

Update Type	Update Components
Patch	None
Service Pack	cs1000-baseWeb-7.50.17.01-1.i386.000
	cs1000-dbcom-7.50.17-01.i386.000
	cs1000-sps-7.50.17-01.i386.000
	cs1000-linuxbase-7.50.17.04-00.i386.000
	cs1000-Jboss-Quantum-7.50.17.01-1.i386.000
	cs1000-bcc-7.50.17.03-00.i386.000
	cs1000-dmWeb-7.50.17.04-00.i386.001
	cs1000-shared-pbx-7.50.17-01.i386.000
	cs1000-vtrk-7.50.17-11.i386.000
Deplist	p30588_1, p30550_1, p30613_1, p30618_1, p30621_1, p30565_1, p30597_1, p30595_1, p30591_1, p30560_1, p30594_1, p30619_1
Loadware	IPMG TYPE CSP/SW MSP APP FPGA BOOT DBL1 DBL2 4 0 MGC BD01 AB01 BA07 AA18 BA07 DSP1AB03 N/A

Table 2: CS1000E Applied Updates

3. Configure Avaya Communication Server 1000

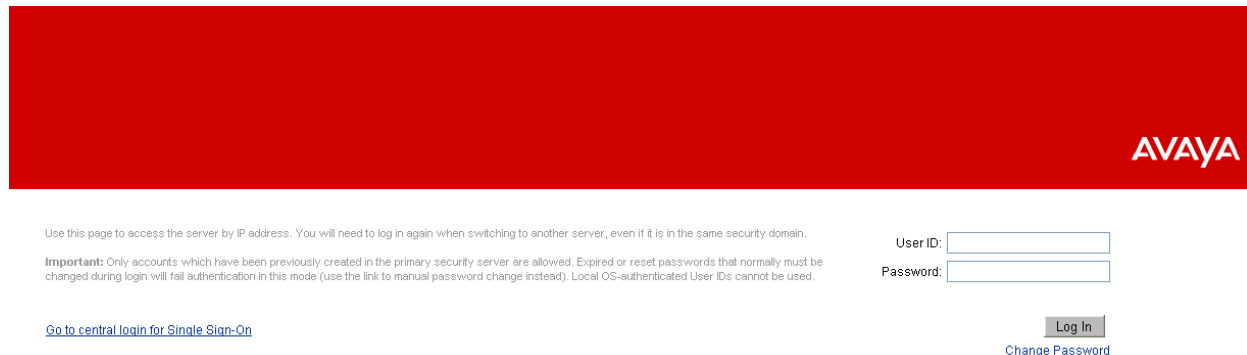
This section describes the steps to configure the following, using CS 1000 Element Manager:

- 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 call server. Note that the IP address for the Call Server may vary, and in this case “10.7.7.61” is used. Log in with the appropriate user ID and password.



The following **Unified Communications Management** screen will be displayed. Click on the **Element Name** corresponding to the **Element Type** of “CS1000”.

AVAYA Avaya Unified Communications Management

Host Name: 10.7.7.61 Software Version: 02.20-SNAPSHOT(0000) 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 manager by entering a search term.

Search [] [Search] [Reset]

<input type="checkbox"/>	Element Name	Element Type	Release	Address
<input type="checkbox"/>	EM on cs1k75	CS1000	7.5	10.7.8.61
<input type="checkbox"/>	cs1k75.avaya.com (primary)	Linux Base	7.5	10.7.7.61

The CS 1000 Element Manager page appears as shown below.

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
System Overview

System Overview

IP Address: 10.7.8.61
Type: Avaya Communication Server 1000E CPPM Linu:
Version: 4121
Release: 750 Q +

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

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 displays the AVAYA CS1000 Element Manager interface. On the left is a navigation menu with categories: UCM Network Services, Links, System, Customers (highlighted), and Routes and Trunks. The main content area shows the 'Customers' page with a status bar indicating 'Managing: 10.7.8.61 Username: admin Customers'. Below this is a table with a header 'Customer Number ▲' and one row containing '1' and '00'. A red circle highlights the '1' and '00' in the table row. There are 'Add...' and 'Delete' buttons above the table.

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

The screenshot displays the Avaya CS1000 Element Manager interface. On the left is a navigation menu with categories like UCM Network Services, Home, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, Tools, and Security. The 'Customers' category is selected. The main content area shows the 'Customer Details' page for 'Customer 00'. At the top right, it indicates 'Managing: 10.7.8.61' and 'Username: admin'. Below this is a breadcrumb trail: 'Customers » Customer 00 » Customer Details'. A list of configuration options is shown, with 'SIP Line Service' circled in red. Other options include Basic Configuration, Application Module Link, Attendant, Call Detail Recording, Call Party Name Display, Call Redirection, Centralized Attendant Service, Controlled Class of Service, Features, Feature Packages, Flexible Feature Codes, Intercept Treatments, ISDN and ESN Networking, Listed Directory Numbers, Media Services Properties, Mobile Service Directory Numbers, Multi-Party Operations, Night Service, Recorded Overflow Announcement, and Timers.

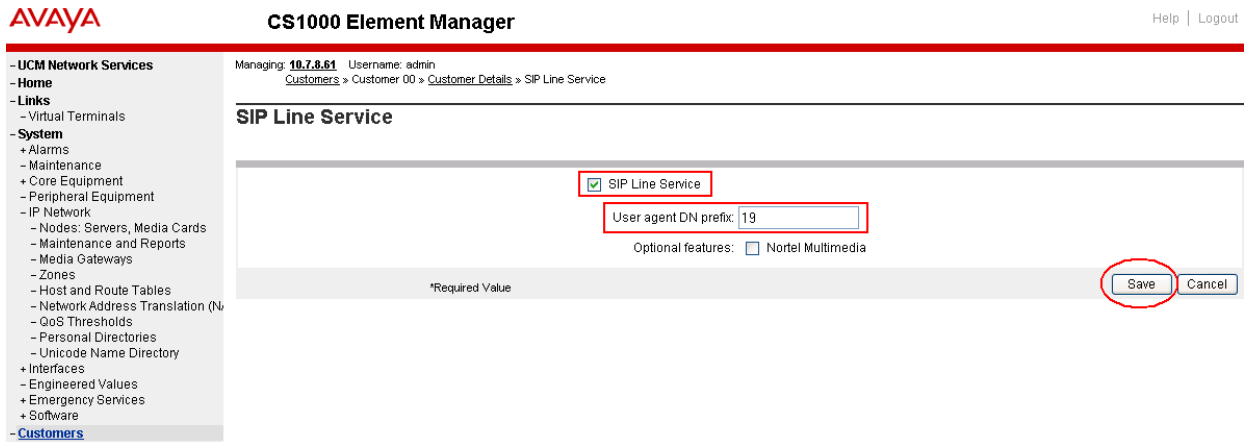
AVAYA **CS1000 Element Manager**

Managing: **10.7.8.61** Username: admin
[Customers](#) » Customer 00 » Customer Details

Customer Details

- [Basic Configuration](#)
- [Application Module Link](#)
- [Attendant](#)
- [Call Detail Recording](#)
- [Call Party Name Display](#)
- [Call Redirection](#)
- [Centralized Attendant Service](#)
- [Controlled Class of Service](#)
- [Features](#)
- [Feature Packages](#)
- [Flexible Feature Codes](#)
- [Intercept Treatments](#)
- [ISDN and ESN Networking](#)
- [Listed Directory Numbers](#)
- [Media Services Properties](#)
- [Mobile Service Directory Numbers](#)
- [Multi-Party Operations](#)
- [Night Service](#)
- [Recorded Overflow Announcement](#)
- [SIP Line Service](#)
- [Timers](#)

Check the **SIP Line Service** checkbox, enter an appropriate **User Agent DN prefix**¹, and click **Save**.



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 Node, as it will be used in configuring the B179 later. Select the **Node ID** on which SIP Line service is to be enabled.

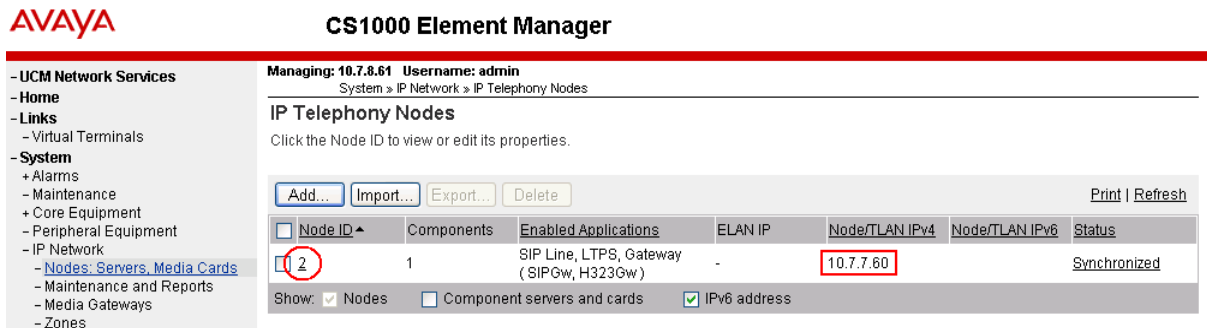


Figure 2: CS 1000 Node Screen

¹ The User Agent DN Prefix is used to form the User Agent DN. See **Section 3.11**.

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

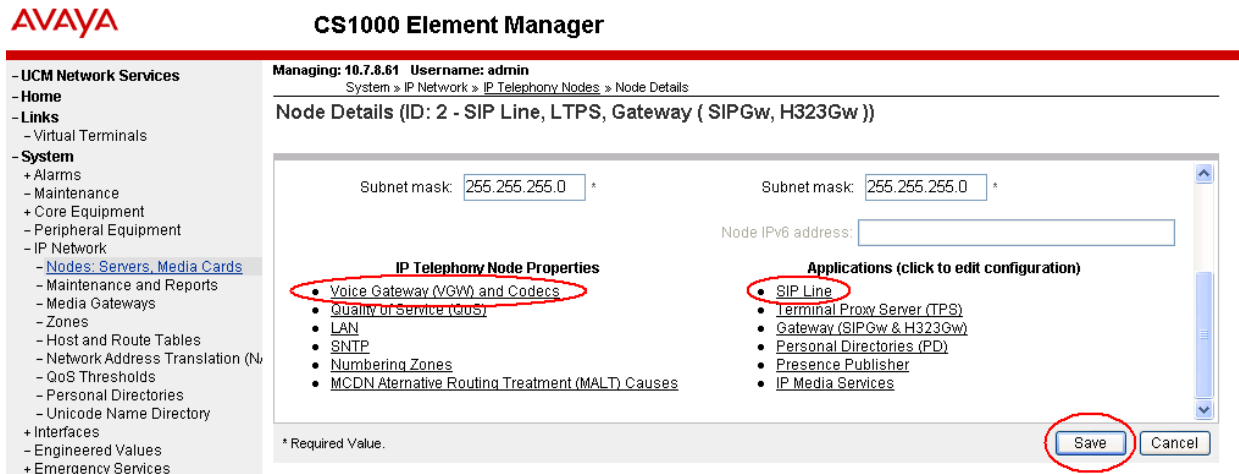
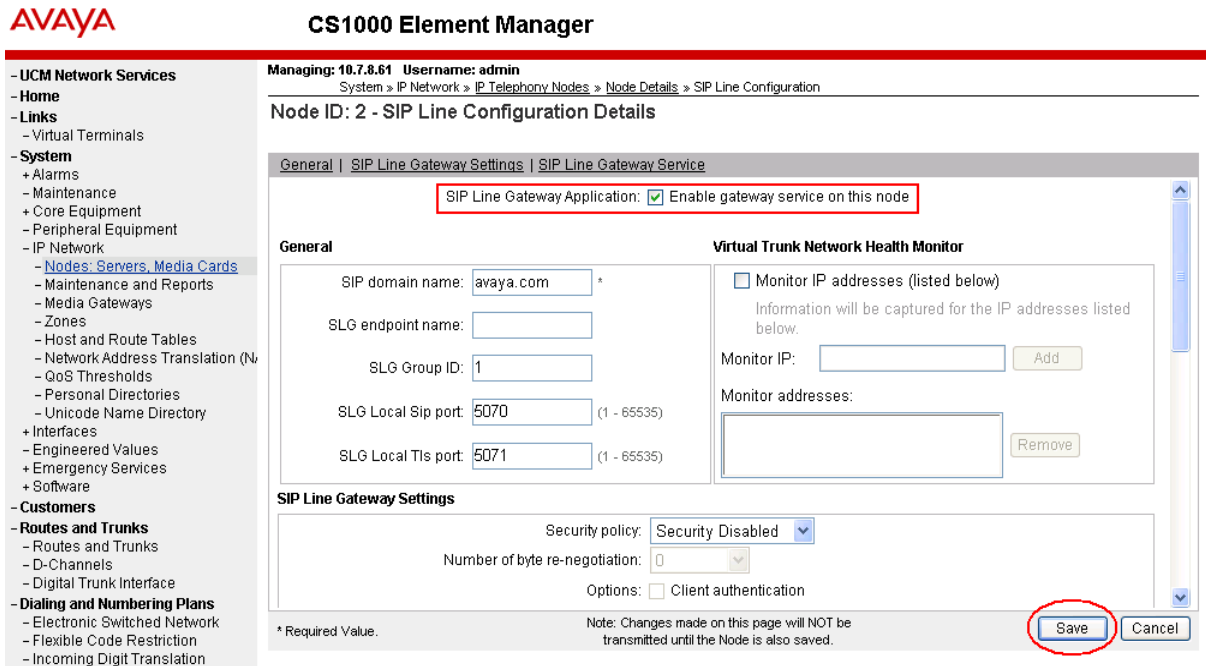


Figure 3: Node Details Screen

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



Return to the Node Details Screen (**Figure 3**) and click on **Voice Gateway (VGW) and Codecs**. For G.722 and G.729 support, check **Enabled** next to **Codec G.722:** and **Codec G.729:**. If G.729 Annex B (silence suppression) is desired as in the sample configuration, check the **Voice Activity Detection (VAD)** checkbox. Note that the VAD setting should be consistent with the VAD setting in the B179 configuration (see **Section 4.2 Figure 9**). Click **Save**. Then click **Save** on the **Node Details** screen (**Figure 3**).

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
System > IP Network > IP Telephony Nodes > Node Details > VGW and Codecs

Node ID: 2 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Codec G.722: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playback (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Codec G.729: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playback (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.

Voice Activity Detection (VAD)

Codec G.723.1: Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playback (jitter buffer) delay: 60 120 (milliseconds)

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

Save Cancel

Figure 4 – Node Codec Selection

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

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
System > IP Network > IP Telephony Nodes > Node Saved

Node Saved

Node ID: 2 has been saved on the call server.

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

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

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

Once the transfer completes, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <2>)

Note: 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"/>	cs1k75	Signaling_Server	SIP Line, LTPS, Gateway, PD, Presence Publisher, IP Media Services	Sync required

* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNMP 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.

Check the appropriate Call 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.

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 3 in the sample configuration), and select **DCH** for the **type**. Click to **Add**.

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 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:

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.

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
Routes and Trunks > D-Channels > D-Channels 3 Property Configuration

D-Channels 3 Property Configuration

- Basic Configuration

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	ForSIPLineGW
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 (ETS)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <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:	3700 Range: 0 - 3700

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

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

Basic options (BSCOPT)

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:

- B channel Service messaging.:

[- Change protocol timer value \(TIMR\)](#)
[+ Advanced options \(ADVOPT\)](#)
[+ Feature Packages](#)

Figure 5 – D-Channel Basic Options

The **Remote Capabilities Configuration** page is displayed. Select the **Message waiting interworking with DMS-100 (MWI)** check box,² 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. Click on **Submit** (see lower left in **Figure 5**).

- Virtual Terminals
- System**
- + Alarms
- Maintenance
- Core Equipment
 - Loops
 - Superloops
 - MSDL/MISP 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
 - Application Module Link
 - Value Added Server
 - Property Management System
- Engineered Values
- + Emergency Services
- + 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

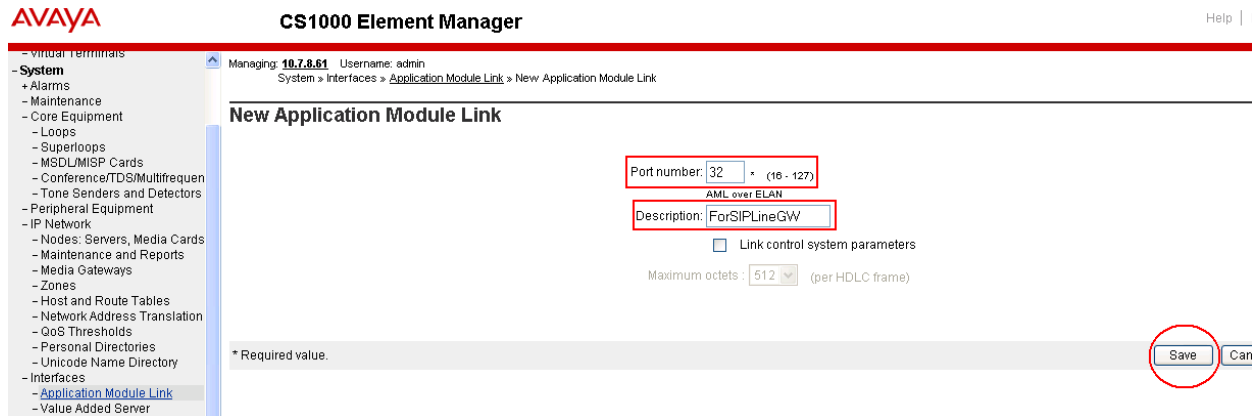
- Remote Capabilities Configuration

Input Description	
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"/>

² Note that although the Avaya B179 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.

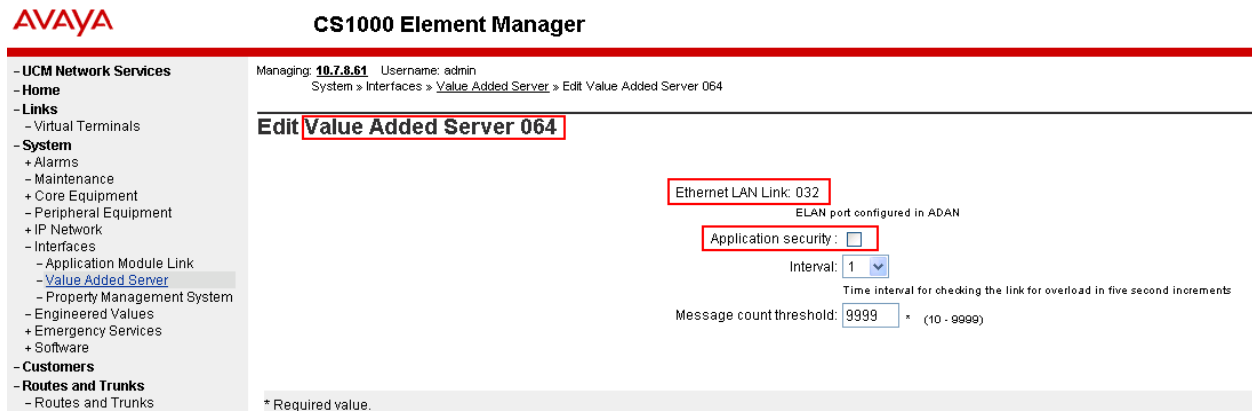
3.5. Configure Application Module Link (AML)

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.



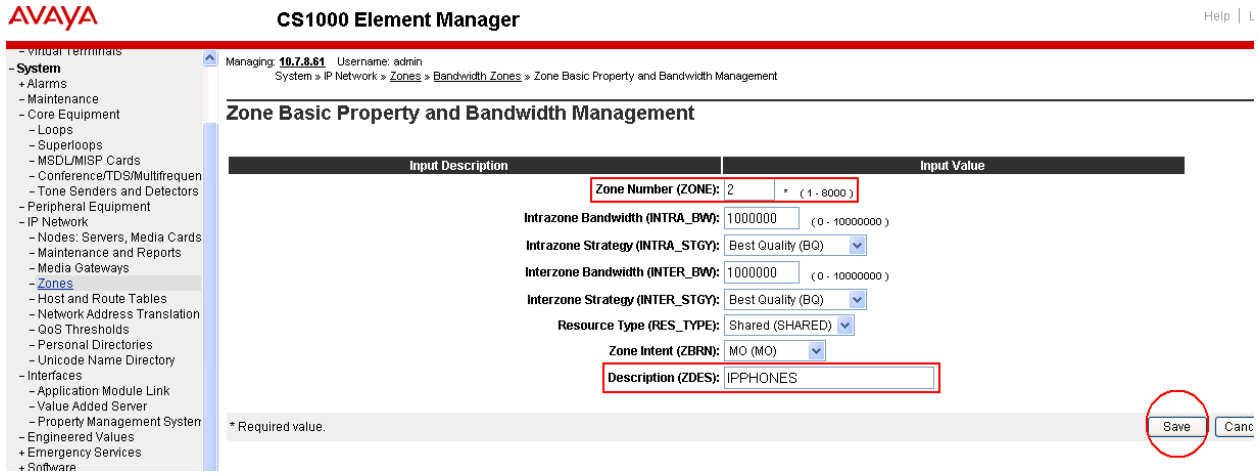
3.6. Configure Value Added Server (VAS)

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 (not shown), enter a **Value added server ID** (64 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.



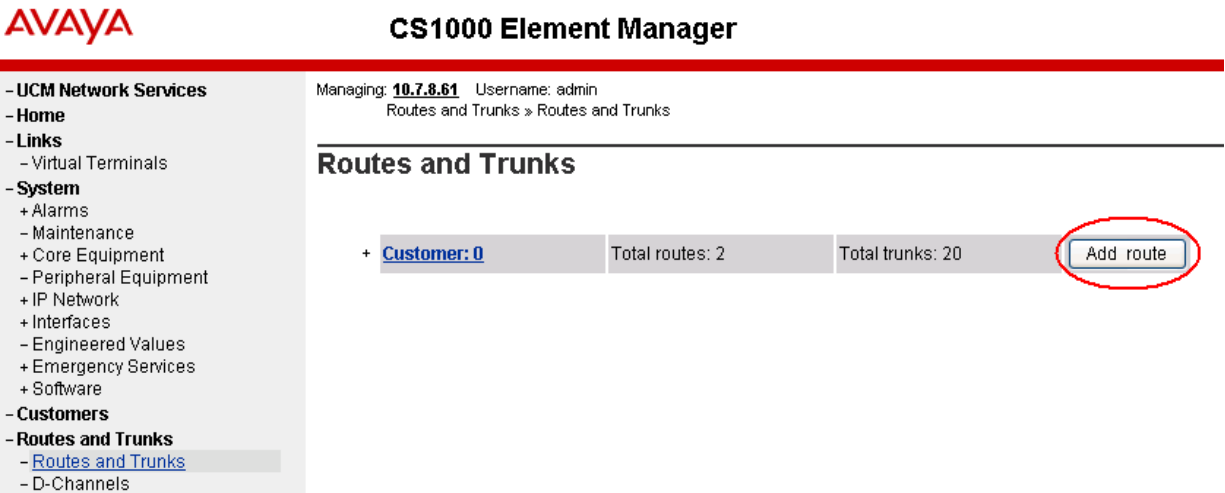
3.7. Configure Zone for SIP Phones

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), and on the **Zone Basic Property and Bandwidth Management** page, enter a **Zone number (ZONE)** and an appropriate **Description**. Defaults can be used for the remaining fields. Click **Save**.



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 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)
Network call redirection (NCRD)
Trunk route optimization (TRO)

Select a 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 zone³

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

Check the box

Check the box

- Maintenance
- + Core Equipment
 - Peripheral Equipment
 - + IP Network
- Interfaces
 - Application Module Link
 - Value Added Server
 - Property Management System
 - Engineered Values
- + Emergency Services
- + 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

- Basic Configuration

Route data block (RDB) (TYPE) : RDB
 Customer number (CUST) : 00
 Route number (ROUT) : 2
 Designator field for trunk (DES) : SIPLINE
 Trunk type (TKTP) : TIE
 Incoming and outgoing trunk (ICOG) : Incoming and Outgoing (IAO)
 Access code for the trunk route (ACOD) : 5570002 *
 Trunk type M911P (M911P) :

The route is for a virtual trunk route (VTRK) :
 - Zone for codec selection and bandwidth management (ZONE) : 00001 (0 - 8000)
 - Node ID of signaling server of this route (NODE) : 2 (0 - 9999)
 - Protocol ID for the route (PCID) : SIP Line (SIPL)
 Integrated services digital network option (ISDN) :
 - Mode of operation (MODE) : Route uses ISDN Signaling Link (ISLD)
 - D channel number (DCH) : 3 (0 - 254)
 - Interface type for route (IFC) : Meridian M1 (SL1)
 - Private network identifier (PNI) : 00000 (0 - 32700)

- Network calling name allowed (NCNA) :
 - Network call redirection (NCRD) :
 - Trunk route optimization (TRO) :

- Recognition of DTI2 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) :
 - Integrated service access route (ISAR) :

³ 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 1.

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 **Wink or Fast Flash (WNK)**

Start arrangement Outgoing

Select **Wink or Fast Flash (WNK)**

Trunk group access restriction

Desired trunk group access restriction level.

Channel ID for this trunk

An available starting channel ID.

AVAYA CS1000 Element Manager Help

Managing **10.7.8.61** Username: admin
Routes and Trunks > Routes and Trunks > Customer 0, Route 2, Trunk 1 Property Configuration

Customer 0, Route 2, Trunk 1 Property Configuration

- Basic Configuration

Auto increment member number:

Trunk data block:

Terminal number:

Designator field for trunk:

Extended trunk:

Member number:

Level 3 Signaling:

Card density:

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

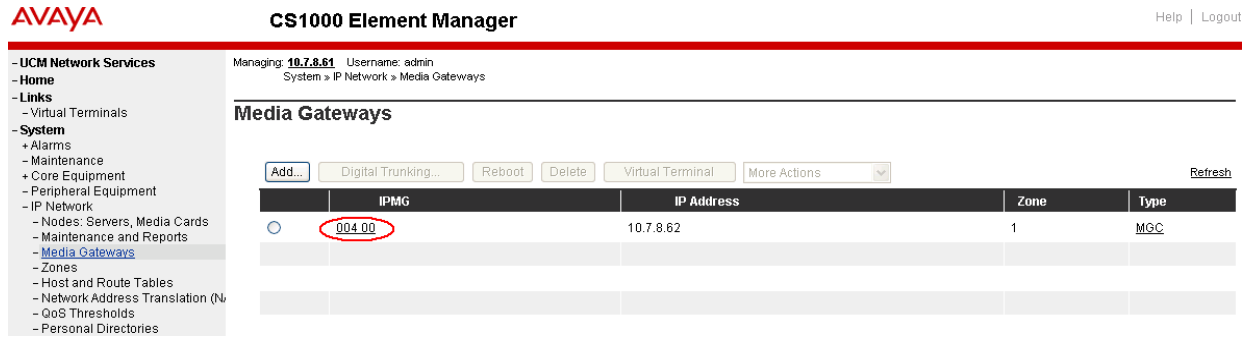
Channel ID for this trunk:

Class of Service:

+ Advanced Trunk Configurations

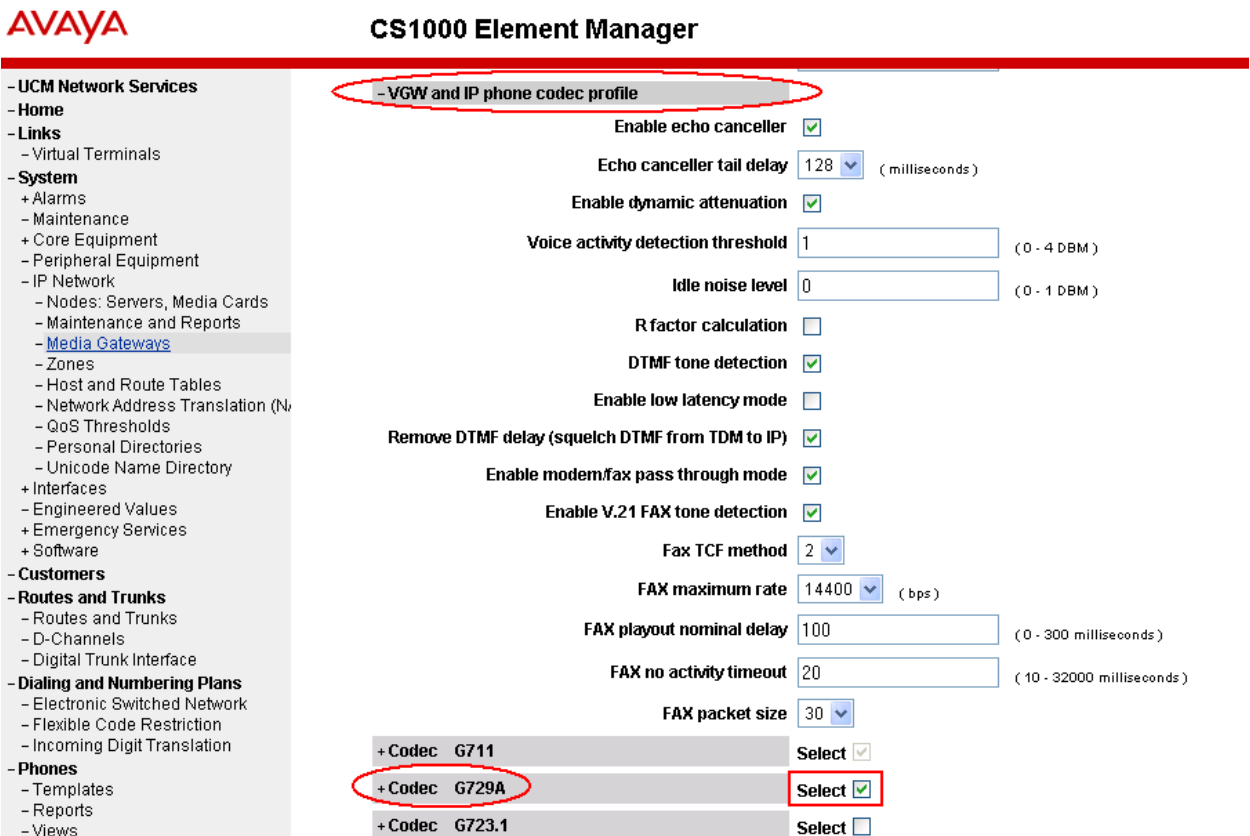
3.10. Configure Media Gateway Controller

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.



	IPMG	IP Address	Zone	Type
<input type="radio"/>	004.00	10.7.8.62	1	MGC
<input type="radio"/>				
<input type="radio"/>				

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.



- 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 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 (see **Section 4.2 Figure 9**). Click **Save**.

The screenshot displays the configuration interface for MGC Codec Selection. On the left is a navigation tree with categories like Core Equipment, Interfaces, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, Tools, and Security. The 'Media Gateways' option is selected. The main configuration area is divided into several sections:

- Codec G729A**: Includes a 'Select' checkbox (checked), 'Codec name' G729A, 'Voice payload size' 20 (ms/frame), 'Voice playback (jitter buffer) nominal delay' 40, and 'Voice playback (jitter buffer) maximum delay' 80. A red warning message states 'Modifications may cause changes to dependent settings'. The 'VAD' checkbox is checked and highlighted with a red box.
- + Codec G723.1**: Includes a 'Select' checkbox (unchecked).
- + Codec T38 FAX**: Includes a 'Select' checkbox (checked).
- + QoS**: A section header.
- + Media Based CLID**: A section header.
- Call Server LAN**: A section header.
- Embedded LAN (ELAN) configuration**: Includes 'Primary call server IP address' (10.7.8.61), 'Primary call server hostname' (Primary_CS), 'Signaling port' (15000), and 'Broadcast port' (15001).
- Telephony LAN (TLAN) configuration**: Includes 'Signaling port' (5000) and 'Voice port' (5200).
- Routes**: Includes 'Add' and 'Remove' buttons, a note 'Click 'Add' to add routes to the IPMG', and a 'Save' button circled in red.

Figure 6 – MGC Codec Selection

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

Managing: **10.7.8.61** Username: admin
System » IP Network » Media Gateways

Media Gateways

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

	IPMG	IP Address
<input checked="" type="radio"/>	004 00	10.7.8.62

3.11. Configure SIP Line Telephone

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

Managing: **EM on cs1k75(10.7.8.61)**
Search for Phone

Search For Phones

Criteria: Prime DN Value:

Phones

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

Select the search criteria, enter or select the desired value and click Search.
New Phones may also be added or retrieved.

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

- UCM Network Services
- Home
- Links
 - Virtual Terminals
- System
 - + Alarms
 - Maintenance
 - + Core Equipment
 - + Peripheral Equipment
 - + IP Network
 - + Interfaces
 - Engineered Values
 - + Emergency Services
 - + 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: **EM on cs1k75(10.7.8.61)**
Phones>New Phones

New Phones

Number of phones: * (1-100).
Maximum value for Attendant consoles is 63.

Customer:

Phone Type

Type: Template *

Copy From TN *

Options:

Default value for DES * (1-6 characters)

Default value for ZONE
Only applicable to IP phone types

Default value for Node Id
Only applicable to UEXT-SIPL phone types

Automatically assign TN starting TN

Automatically assign DN starting DN *

* Required value

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 **1**

SIP3

Set to **0**

- + 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
- + 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



System: EM on cs1k75
 Phone Type: UEXT-SIPL
 Sync Status: TRN

[General Properties](#) |
 [Features](#) |
 [Keys](#) |
 [User Fields](#)

General Properties

Customer Number: *

Terminal Number:

Designation: * (1-6 characters)

Zone: *

SIP User Name: * (1-16 characters)

Node Id: *

Super User:

Optional Features: Max Client Count

SIPN:

SIP3:

FMCL:

TLSV:

In the **Features** section, fill in the following fields, leaving the remaining fields at their defaults. Note that only the first two feature settings are shown in the screen below; the scroll bar must be used to display and set the remaining features, which are not shown here.

- | | |
|--|--|
| 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 ⁴ |

The screenshot displays a configuration page with a left-hand navigation menu and two main sections: Features and Keys.

Features Section:

Feature	Description	Value
CLTA	Network Call Trace	Denied
CNDA	Call Party Name Display	Allowed
CNIA	Call Number Information	Allowed
CNTA	Network ACD Countdown	Denied
CPFA	Forced Camp-On From This Set	Allowed

Keys Section:

Key No.	Key Type	Key Value
0	SCR - Single Call Ringing	Directory Number: 57010 <input checked="" type="checkbox"/> Multiple Appearance Redirection Prime(MARP) First Name: Avaya, Last Name: B179, Display Format: First, Last, Language: Roman CLID Entry (Numeric or D): 0 ANIE Entry:
1	HOT_U - Hotline(Universal)	UADN: 1957010
2	NUL - Unassigned	

⁴ 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**.

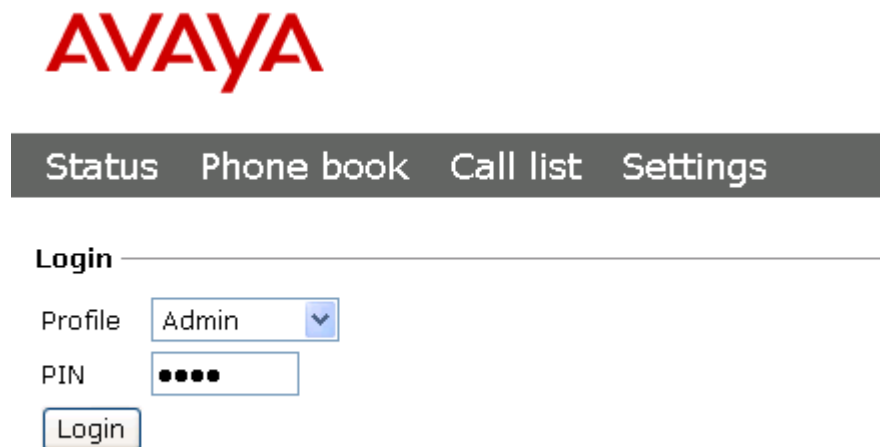
Click **Save** (not shown) to save the configuration for this phone.

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. Select **Admin** in the **Profile** dropdown list and enter the appropriate password.



The screenshot shows the Avaya logo at the top. Below it is a navigation bar with four tabs: Status, Phone book, Call list, and Settings. Underneath the navigation bar is a 'Login' section. It contains a 'Profile' dropdown menu with 'Admin' selected, a 'PIN' input field with four dots, and a 'Login' button.

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

Status Phone book Call list **Settings**

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

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

Figure 7 – B179 Network Configuration Status

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 “*”
Authentication name	Extension (SIP User Name) of the SIP Line telephone configured in Section 3.11
Registrar and Proxy Password	SIP domain configured in the CS 1000 The Station Control Password of the SIP Line telephone configured in Section 3.11
Registration interval	Enter a value (1800 was used in the sample configuration)

Under Advanced:

Enable Blind Transfer	Select the No radio button ⁵
Outbound proxy	Enter the IP address of the CS 1000 Node (see Figure 2), port 5070 , and lr (loose routing), as shown

Under Transport:

Protocol	Select the TCP or UDP radio button (UDP shown)
Local UDP Port	Enter 5060

Click **Save**.

⁵ This feature is not yet supported in this configuration

Account 1

Enable account	<input checked="" type="radio"/> Yes <input type="radio"/> No		
Account name ⓘ	<input type="text" value="CS1KR7.5"/>	Realm ⓘ	<input type="text" value="*"/>
User ⓘ	<input type="text" value="57010"/>	Authentication name ⓘ	<input type="text" value="57010"/>
Registrar ⓘ	<input type="text" value="avaya.com"/>	Password	<input type="password" value="*****"/>
Proxy ⓘ	<input type="text" value="avaya.com"/>	Registration interval ⓘ	<input type="text" value="1800"/>

Account 2

Enable account	<input type="radio"/> Yes <input checked="" type="radio"/> No		
Account name	<input type="text" value="200"/>	Realm	<input type="text" value="*"/>
User	<input type="text" value="200"/>	Authentication name	<input type="text" value="200"/>
Registrar	<input type="text" value="192.168.0.1"/>	Password	<input type="password" value="*****"/>
Proxy	<input type="text"/>	Registration interval	<input type="text" value="300"/>

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
Outbound proxy	<input type="text" value="10.7.7.60:5070;lr"/>

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="5060"/>	

ine

Figure 8 – B179 SIP Configuration

To configure the audio codec settings, navigate to **Settings** → **Media**, and select the priority for codec selection. One combination shown below prefers the high fidelity G.722 codec if the other party's telephone can support it, with a fallback to G.711. Defaults can be used for the remaining fields. Note that call transfer by CS 1000 telephones is not supported for G.722.



Status Phone book Call list **Settings**

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

Codec

	Priority
G722	4 - High
G711 Alaw	0 - Disabled
G711 Ulaw	3
G729	0 - Disabled

Security

S RTP Disabled Optional Mandatory

Secure signalling No TLS SIPS *Please check corresponding SIP transport setting*

VAD

Enable VAD Yes No

DTMF

DTMF Signalling RFC 2833 SIP Info Inband

Advanced

First RTP port

Another combination that could be used in low bandwidth environments would be to give high preference to the compressed G.729A codec, with fallback to G.711, as shown below. In the case of G.729A, ensure that the **VAD** setting matches that configured in the CS 1000. In this case, checking **Enable VAD** results in G.729AB. After choosing an appropriate codec preference list, click **Save**. Note also that for this release of the Avaya B179, G.711 is required to be in the codec list with G.729 in order for call hold by CS 1000 telephones to operate correctly.



Status Phone book Call list **Settings**

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

Codec

	Priority
G722	0 - Disabled
G711 Alaw	0 - Disabled
G711 Ulaw	3
G729	4 - High

Security

SRTP Disabled Optional Mandatory

Secure signalling No TLS SIPS *Please check corresponding SIP transport setting*

VAD

Enable VAD Yes No

DTMF

DTMF Signalling RFC 2833 SIP Info Inband

Advanced

First RTP port

Figure 9 – B179 Codec Selection

After the configuration has been saved, the B179 will register with the CS 1000, and a display similar to those shown in **Figures 10** and **11** below will appear on the telephone. The **Hostname** (see **Figure 7**) is displayed at the center, and in the lower left corner is the **Account name** (see **Figure 8**). 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 (**Figure 10**), the B179 has successfully registered. If the square is not filled in (**Figure 11**), registration was unsuccessful.



Figure 10 – Successful B179 Registration



Figure 11 – Unsuccessful B179 Registration

5. Verification Scenarios

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

- Registration and recovery, including power cycling and network disruption.
- SIP signaling using UDP and TCP transport
- Basic calling among the B179 and the following CS 1000 supported telephones:
 - 1120e UNISlim
 - 1165e SIP
 - 2007 UNISlim
 - 2050PC UNISlim (soft phone)
 - M3903 Digital
- RFC 2833 DTMF support
- G.711mu-law, G.722, G.729A, and G.729AB audio codec support.
- Hold, consultative hold.
- Manual conference by the B179.
- Unattended transfer.
- Placement of calls via the outbound call log.

The following restrictions to the above features apply:

- Attended call transfer of the B179 by CS 1000 supported telephones is supported, except for M3900 series digital telephones and when G.722 codec is used.. Attended call transfer by the B179 is not supported.
- Call hold by CS 1000 supported telephones is supported for G.729 only if G.711 is included in the B179 codec list.
- Calls from the B179 via the inbound call log are not supported.
- Group conference by the B179 is not supported.

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

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with 'Maintenance' selected. The main content area shows the 'Maintenance' page with 'Select by Overlay' selected. A dropdown menu is open, listing various LDs, with 'LD 96 - D-Channel' highlighted in red. To the right, there is a 'Select by Functionality' option and a 'Select Group' dropdown menu with 'D-Channel Diagnostics' selected.

The screen below shows the **APPL_STATUS** of the SIP trunk D-Channel as “OPER” and the **LINK_STATUS** as “EST ACTV”. Note that for the SIP line D-Channel, the **APPL_STATUS** is “DSBL” and the **LINK_STATUS** is “RST”. This is normal.

The screenshot shows the AVAYA CS1000 Element Manager interface with the 'D-Channel Diagnostics' page. The left sidebar shows 'Maintenance' selected. The main content area displays a table of diagnostic commands and their status for two D-Channel entries. The table has columns for DCH, DES, APPL_STATUS, LINK_STATUS, AUTO_REC, PDCH, and BDCH. The first entry (001) has APPL_STATUS 'OPER' and LINK_STATUS 'EST ACTV'. The second entry (003) has APPL_STATUS 'DSBL' and LINK_STATUS 'RST'. A red box highlights the second entry.

DCH	DES	APPL_STATUS	LINK_STATUS	AUTO_REC	PDCH	BDCH
001	VirDchToSS	OPER	EST ACTV	AUTO		
003	ForSIPLineGW	DSBL	RST	AUTO		

Instruction: Select a command, add value and click on [Submit].

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. Click **GEN CMD**.

AVAYA CS1000 Element Manager

Managing: 10.7.8.61 Username: admin
System > IP Network > Node Maintenance and Reports

Node Maintenance and Reports

- Node ID: 2 Node IP: 10.7.7.60

Hostname	ELAN IP	Type	TN
cs1k75	10.7.8.61	Signaling Server-Avaya CPPMv1	NO TN

Buttons: GEN CMD, SYS LOG, OMRPT, Reset

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 57010” will be returned instead of the detailed registration information.

CS1000 Element Manager Help |

Managing: 10.7.8.61 Username: admin
System > IP Network > Node Maintenance and Reports > General Commands

General Commands

Element IP: 10.7.8.61 Element Type: Signaling Server-Avaya CPPMv1

Group: SipLine Command: slgSetShowByUID UserID: 57010 RUN

IP address: 10.7.8.61 Number of pings: 3 PING

User ID	AuthId	TN	Clients	Calls	SetHandle	Pos ID	SIPL Type
57010	57010	096-00-00-19	1	0	0x8ca3fa0		SIP Lines

```

StatusFlags = Registered Controlled KeyMapDwld SSD
FeatureMask =
CallProcStatus = -1
Current Client = 0, Total Clients = 1
== Client 0 ==
IPv4:Port:Trans = 10.7.7.130:5060:udp
Type = SIPN
UserAgent = Avaya B179 2.1.0
x-nt-guid = 85b5869a169f22a3cf2f1fe83a4dd6ff
RegDescrip =
RegStatus = 1
PhxReason = 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. See **Figures 10** and **11** for examples of successful and unsuccessful registration. 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 **On** radio button of the **SIP logging** field 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 shows the Avaya B179 SIP Conference Phone interface. At the top, there is a navigation bar with tabs for Status, Phone book, Call list, and Settings. Below this is a red bar with tabs for Device, Network, Time & Region, SIP, Media, Log, and Licenses. The Log tab is selected. In the SIP Trace section, there is a dropdown menu set to 'SIP Trace', a 'Change' button, and a 'Refresh' button. Below this, the SIP logging section has a radio button set to 'On', a 'Set' button, and a 'Clear Log' button. The main area displays a log entry for a successful REGISTER message to the CS 1000.

```
Mar 10 04:18:42: TX 445 bytes Request msg REGISTER/cseq=50620 (tdta0x1f1f00) to UDP 10.7.7.60:5070:
REGISTER sip:avaya.com;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.7.7.130:5060;rport;branch=z9hG4bKPjTgaP7qyCLc1p4wbQa4X8B39gdFCYniJ1
Route: <sip:10.7.7.60:5070;lr>
Max-Forwards: 70
From: <sip:57010@avaya.com>;tag=Ba-De.t5DhKRdtgRmoB5NjgHhILYbRKU
To: <sip:57010@avaya.com>
Call-ID: sdI1qg69zzf8wZM2ujTmcsHoUQP888x6
CSeq: 50620 REGISTER
User-Agent: Avaya B179 2.1.0
Contact: <sip:57010@10.7.7.130:5060>
Expires: 0
Content-Length: 0

-----
Mar 10 04:18:42: RX 454 bytes Response msg 100/REGISTER/cseq=50620 (rdata0x1e1a6c) from UDP 10.7.7.60:5070:
SIP/2.0 100 Trying
From: <sip:57010@avaya.com>;tag=Ba-De.t5DhKRdtgRmoB5NjgHhILYbRKU
To: <sip:57010@avaya.com>
Call-ID: sdI1qg69zzf8wZM2ujTmcsHoUQP888x6
CSeq: 50620 REGISTER
Via: SIP/2.0/UDP 10.7.7.130:5060;rport=5060;branch=z9hG4bKPjTgaP7qyCLc1p4wbQa4X8B39gdFCYniJ1
Supported: 100rel,x-nortel-sipvc,replaces,timer,outbound,x-nortel-sca,join,answermode
User-Agent: Nortel CS1000 SIPLine GW release_7.0 version_ssLinux-7.50.17
Content-Length: 0
```

7. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 and the Avaya B179 SIP Conference Phone can be used together in an integrated solution supporting the features described in **Section 5**.

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: 7.5, Document Revision: 05.04, Document #NN43001-632.
- [2] *Communication Server 1000 SIP Line Fundamentals*, Release 7.0, Document #NN43001-508, 02.03, August 2010.
- [3] *Installation and Administration of B179*, 110047-61-001, Rev 3d.

©2011 Avaya Inc. All Rights Reserved.

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

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