



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring PIVOT™ by Spectralink (87-Series) Wireless Telephone Version 1.2.0 with Avaya Communication Server 1000 Release 7.6 - Issue 1.0**

### **Abstract**

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.6 and PIVOT™ by Spectralink 87-Series Wireless Telephones. During the compliance testing, the PIVOT™ 87-Series phone was able to register as a SIP endpoint with the Communication Server 1000 SIP Line Gateway. The PIVOT™ 8741 telephone was able to place and receive calls from Communication Server 1000 Release 7.6 non-SIP and SIP Line clients. The compliance tests focused on basic telephony features.

Readers should pay attention to section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

# 1. Introduction

These application notes provide detailed configuration of Avaya Communication Server 1000 SIP Line Gateway Release 7.6 (hereafter referred to as CS 1000) and the PIVOT™ by Spectralink 87-Series Wireless Telephone Version 1.2.0 (hereafter referred to as PIVOT). The PIVOT™ by Spectralink 87-Series was tested with non-SIP and SIP clients using the CS 1000 SIP Line Gateway. All the applicable telephony feature test cases of release 7.6 SIP line were executed on PIVOT, where applicable, to verify the interoperability with CS 1000.

These Application Notes assume that Avaya CS 1000 is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult the documentation library mentioned in **Section 9**.

## 2. General Test Approach and Test Results

The general test approach was to have the PIVOT telephone register to the CS 1000 SIP line gateway successfully. From CS 1000 telephone clients/users, calls were placed to and from the PIVOT telephone and other telephony features, such as busy, hold, DTMF, MWI and codec negotiation, were exercised.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the PIVOT wireless telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the 8741 wireless telephone to the CS 1000 SIP Line Gateway.
- Telephony features: Basic calls, conference, transfer, DTMF (dual tone multi frequency) RFC2833 transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer, multiple appearances Directory Number.
- PSTN calls over ISDN/PRI trunk.
- Codec negotiation – G.711, G.729, and G.722.

### 2.2. Test Results

The objectives outlined in **Section 2.1** were verified. The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the PIVOT compliance to required industry standards.
- PIVOT telephones are treated by CS 1000 as 3rd party SIP endpoints and use CS 1000 3rd party SIP licenses.
- PIVOT only supports local call forward unconditional and does not support local forward busy and no answer.

## 2.3. Support

Technical support on PIVOT can be obtained through the following:

**North America:**

Phone: 1-800-775-5330

Email: [nolarma@spectralink.com](mailto:nolarma@spectralink.com)

Web: <http://support.spectralink.com>

**EMEA:**

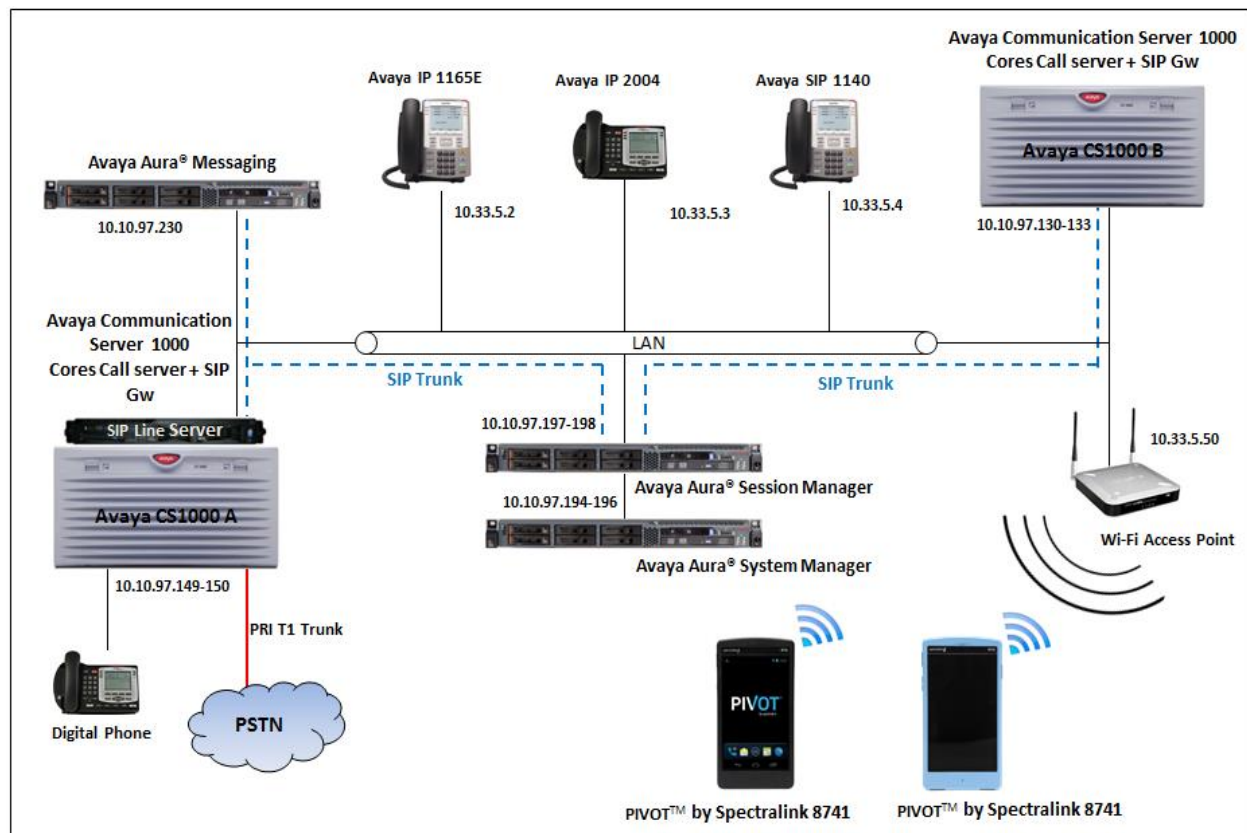
Phone: +33 176774541

Email: [emeaom@spectralink.com](mailto:emeaom@spectralink.com)

Web: <http://support.spectralink.comnumber>

### 3. Reference Configuration

**Figure 1** illustrates the test configuration used during the compliance testing between the Avaya Communication Server 1000 and PIVOT. The PIVOT phone registers to the CS 1000 SIP Line server by going through the Wi-Fi access point that connects to the lab network. Avaya Aura® Session Manager was used for routing SIP calls between the CS 1000 A and CS 1000 B for test cases off-net. The PRI T1 trunk was configured to connect to the PSTN.



**Figure 1: Test Configuration Diagram**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya S8800 server running Avaya Aura® Session Manager Server	6.3.7
Avaya S8800 server running Avaya Aura® System Manager Server	6.3.7 (Build No 6.3.0.8.5682-6.3.8.3204 Software Update Revision No: 6.3.7.7.2275)
Avaya S8800 server running Avaya Aura® Messaging Server	6.3
Avaya Communication Server 1000E/CPPM	Avaya Communication Server Release 7.6 Q+ Deplst 1 (created: 2014-07-23) and Service Pack 5 (Created: 2014-Jul-10)
Avaya IP SIP Phone 1140 <sup>E</sup>	4.3
Avaya IP Unistim Phone 1165 <sup>E</sup>	0x25C8J
Avaya IP Unistim Phone 2004	0604DCN
PIVOT™ by Spectralink 8741	1.2.0.6893

## 5. Configure Avaya Communication Server 1000

This section describes the steps to configure the Avaya CS 1000 SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS 1000 system. For detailed information on how to configure and administer the CS 1000 SIP Line, please refer to the **Section 9 [1]**.

The following is a summary of tasks required for configuring the CS 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service and configure the local SIP Domain.
- Create SIP Line Telephony Node.
- Create D-Channel for SIP Line.
- Create an Application Module Link (AML).
- Create a Value Added Server (VAS).
- Create a Virtual Trunk Zone.
- Create a Route Data Block (RDB).
- Create SIP Line Virtual Trunks.
- Create SIP Line phones.

## 5.1. Prerequisites

This document assumes that the CS 1000 SIP Line server has been:

- Installed with CS 1000 Release 7.6 Linux Base.
- Joined CS 1000 Release 7.6 Security Domain.
- Deployed with SIP Line Application.

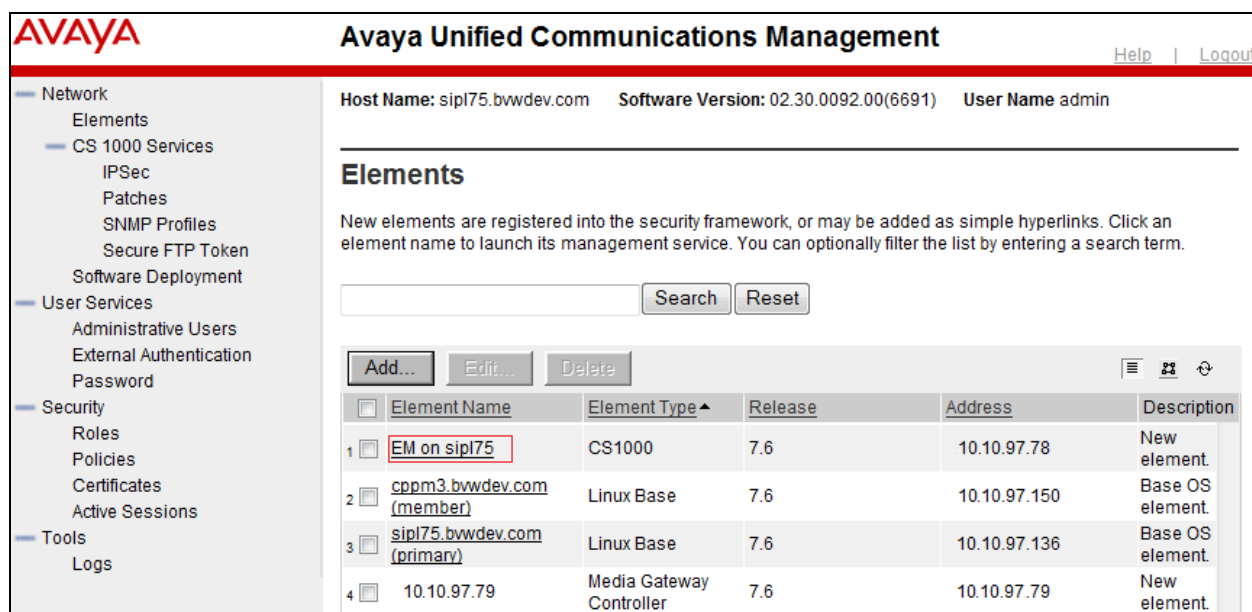
The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at <http://www.avaya.com>.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_AVAYA	415	Avaya SIP Line package	Existing package	Global
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	Global

## 5.2. Log into Unified Communications Management (UCM) and Element Manager (EM)

Use the Microsoft Internet Explorer browser to launch CS 1000 UCM web portal at <http://<IP Address or FQDN>> where <IP address or FQDN> is the UCM Framework IP address or FQDN for UCM server.

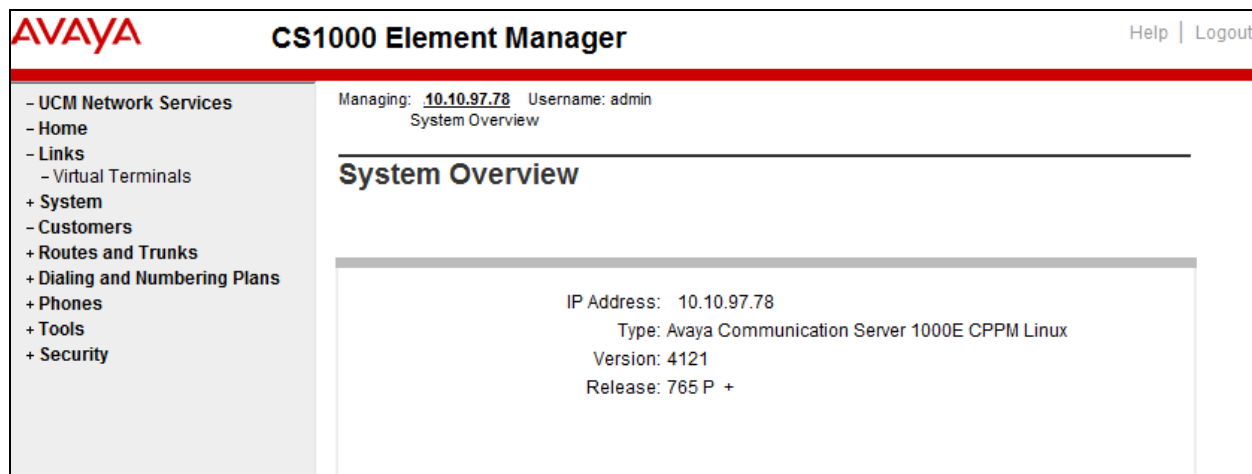
Log in with the username/password which was defined during the primary security server configuration, the UCM home page appears as shown in the screen below. On the UCM home page, under the **Element Name** column, click on the Element Manager name of CS 1000 system that needs to be configured, in this sample that is **EM on sip175**.



The screenshot shows the Avaya Unified Communications Management (UCM) home page. The top header includes the Avaya logo, the title "Avaya Unified Communications Management", and links for "Help" and "Logout". Below the header, the page displays the host name "sip175.bwwdev.com", software version "02.30.0092.00(6691)", and the user name "admin". A left-hand navigation menu lists various categories: Network, CS 1000 Services, User Services, Security, and Tools. The main content area is titled "Elements" and contains a search bar with "Search" and "Reset" buttons. Below the search bar is a table of elements. The table has columns for "Element Name", "Element Type", "Release", "Address", and "Description". The first element, "EM on sip175", is highlighted with a red box. Other elements include "cppm3.bwwdev.com (member)", "sip175.bwwdev.com (primary)", and "10.10.97.79".

	Element Name	Element Type	Release	Address	Description
1	EM on sip175	CS1000	7.6	10.10.97.78	New element.
2	cppm3.bwwdev.com (member)	Linux Base	7.6	10.10.97.150	Base OS element.
3	sip175.bwwdev.com (primary)	Linux Base	7.6	10.10.97.136	Base OS element.
4	10.10.97.79	Media Gateway Controller	7.6	10.10.97.79	New element.

The CS 1000 Element Manager page appears as shown below.



The screenshot shows the CS1000 Element Manager "System Overview" page. The top header includes the Avaya logo, the title "CS1000 Element Manager", and links for "Help" and "Logout". Below the header, the page displays the managing IP address "10.10.97.78", the username "admin", and the "System Overview" title. A left-hand navigation menu lists various categories: UCM Network Services, Home, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, Tools, and Security. The main content area is titled "System Overview" and contains a box with the following information: IP Address: 10.10.97.78, Type: Avaya Communication Server 1000E CPPM Linux, Version: 4121, and Release: 765 P +.

Managing: 10.10.97.78 Username: admin  
System Overview

**System Overview**

IP Address: 10.10.97.78  
Type: Avaya Communication Server 1000E CPPM Linux  
Version: 4121  
Release: 765 P +

### 5.3. Enable SIP Line Service in the Customer Data Block

On the Element Manager page, navigate to **Customers** on the left menu. The list of Customer IDs displays on the right, select the customer number (Customer 0) to be enabled with SIP Line Service (screen not shown). The screen below shows the SIP Line Service page.

- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box, e.g., **26** as shown below. Click the **Save** button to save the changes.

AVAYA CS1000 Element Manager

Help | Logout

Managing: 10.10.97.78 Username: admin  
Customers » Customer 00 » Customer Details » SIP Line Service

### SIP Line Service

☒ SIP Line Service

User agent DN prefix: 26

Optional features: ☒ Nortel Multimedia

\*Required Value

Save Cancel

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### 5.4. Add a New SIP Line Telephony Node

On the Element Manager page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as the screen below. Click **Add** button to add a new SIP Line Node to the IP Telephony Nodes.

CS1000 Element Manager

Help | Logout

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes

### IP Telephony Nodes

Click the Node ID to view or edit its properties.

Add... Import... Export... Delete

Print | Refresh

<input type="checkbox"/> Node ID ▲	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
<input type="checkbox"/> 511	1	LTPS, Gateway ( SIPGw )	-	10.10.97.149		<a href="#">Synchronized</a>
<input type="checkbox"/> 512	1	SIP Line	-	10.10.97.187		<a href="#">Synchronized</a>

Show: ☒ Nodes ☐ Component servers and cards ☒ IPv6 address



The **new IP Telephony Node** page is displayed. Enter the information for each field shown below.

- **Node ID:** Enter **512** which is the node ID of SIP Line server.
- **Telephony LAN (TLAN) Node IP Address:** Enter **10.10.97.187** which is the Node IP address of SIP Line.
- **Embedded LAN (ELAN) Gateway IP Address:** Enter **10.10.97.65** which is the gateway IP of Call server subnet.
- **Applications: SIP Line:** Select the check box to enable SIP Line service for this Node.

**CS1000 Element Manager** Help | Logout

Managing: 10.10.97.78 Username: admin  
System » IP Network » IP Telephony Nodes » New IP Telephony Node

### New IP Telephony Node

Step 1: Define the new Node and its services.  
You will also require pre-configured servers with appropriate application software already deployed to host the selected services.

Node ID: 512 \* (0-9999)

Call server IP address: 10.10.97.78 \*

TLAN address type: ☒ IPv4 only  
☐ IPv4 and IPv6

**Embedded LAN (ELAN)**

Gateway IP address: 10.10.97.65 \*

Subnet mask: 255.255.255.192 \*

**Telephony LAN (TLAN)**

Node IPv4 address: 10.10.97.187 \*

Subnet mask: 255.255.255.192 \*

Node IPv6 address:

Applications: ☒ SIP Line  
☐ UNiStim Line Terminal Proxy Server (LTPS)  
☐ Virtual Trunk Gateway (SIPGw, H323Gw)  
☐ Personal Directory (PD)  
☐ Presence Publisher

\* Required Value.

Next > Cancel

Click on the **Next** button to go to next page. The page, **New IP Telephony Node with Node ID**, is displayed. On this page, in the **Select to Add** drop down menu list, select the desired server to add to the node. Click the **Add** button and select the check box next to the newly added server, and click **Make Leader** (screen not shown).

Click on the **Next** button to go to next page. The **SIP Line Configuration Details** page is displayed as the screen below.

- **SIP Line Gateway Application:** Check on the check box **Enable gateway service on this node**.
- In the **General** section:
  - **SIP domain name:** Enter the SIP domain as “**10.10.97.187**”.

- **SLG Local Sip Port:** Enter port “5060”.
- **SLG Local Tls port:** Enter the port “5061”.
- Keep other sections as default.

Click on the **Save** button to save the changes.

**AVAYA CS1000 Element Manager** Help | Logout

System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

**Node ID: 512 - SIP Line Configuration Details**

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Application: ☒ Enable gateway service on this node

**General**

SIP domain name: 10.10.97.187 \*

SLG endpoint name: sip175

SLG Group ID: 512

SLG Local Sip port: 5060 (1 - 65535)

SLG Local Tls port: 5061 (1 - 65535)

**Virtual Trunk Network Health Monitor**

☐ Monitor IP addresses (listed below)

Information will be captured for the IP addresses listed below.

Monitor IP:  Add

Monitor addresses:  Remove

**SIP Line Gateway Settings**

Security policy: Security Disabled

Number of byte re-negotiation: 0

Options: ☐ Client authentication

Click **Next**. The **Confirm new Node details** page appears (screen not shown). Next click on the **Transfer Now** button in the **Node Saved** page as displayed in the screen below.

**CS1000 Element Manager** Help | Logout

Managing: 10.10.97.78 Username: admin

System » IP Network » IP Telephony Nodes » Node Saved

**Node Saved**

Node ID: 512 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.

Click on the **Transfer Now** button, the **Synchronize Configuration Files (Node ID 512)** page is displayed. Select the SIP Line server that is associated with the changes and then click on the **Start Sync** button to transfer the configuration files to the selected servers as shown below.

CS1000 Element Manager

Help | Logout

Managing: 10.10.97.78 Username: admin  
 System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <512>)

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.

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	sip175	Signaling_Server	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, 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.

**Note:** The first time a new Telephony Node is added and transferred to the call server, the SIP Line services need to be restarted. To restart the SIP Line services, check on the SIP Line server as shown above and click **Restart Application** button.

## 5.5. Create a D-Channel for SIP Line

On the Element Manager page, navigate to **Routes and Trunks → D-Channels**. The **D-Channels** page is displayed on the right. Under the **Configuration** section as shown below, enter an available number in the **Choose a D-Channel Number** drop down menu, e.g., **3** and click on the “to Add” button.

AVAYA

CS1000 Element Manager

Help | Logout

Managing: 10.10.97.78 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: 3 and type: DCH

- Channel: 1	Type: DCH	Card Type: DCIP	Description: SIP	<input type="button" value="Edit"/>
- Channel: 2	Type: DCH	Card Type: TMDI	Description: ToCM	<input type="button" value="Edit"/>

The **D-Channels 3 Property Configuration** page is displayed. In the **Basic Configuration** section:

- **D channel Card Type:** Select **D-Channel is over IP (DCIP)**.
- **Designator:** Enter a descriptive name, e.g., “**SIPLine**”.
- **Interface type for D-channel (IFC):** Select **Meridian Meridian1 (SL1)**.
- Leave the other fields in the section at default values.

**CS1000 Element Manager** Help | Logout

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**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type :	D-Channel is over IP (DCIP) *
Designator:	SIPLine
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User :	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <a href="#">more PRI</a>
Secondary PRI2 loops:	<input type="text"/>
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 <small>Range: 1 - 4000</small>
Signalling server resource capacity:	3700 <small>Range: 0 - 3700</small>

[+ Basic options \(BSCOPT\)](#)

Click on the **Basic options (BSCOPT)** link to expand this section. The **Basic options (BSCOPT)** section is displayed as shown below. Click on **Edit** button to configure **Remote Capabilities (RCAP)**.

**- 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: [Edit](#)

The **Remote Capabilities Configuration** page is displayed. Select the **Message waiting interworking with DMS-100 (MWI)** and **Network name display method 2 (ND2)** check boxes. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** button to return the **D-Channel 3 Property Configuration** page.

Note that the **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints and **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.

Remote D-channel is on a MSDL card (MSL) ☐

Message waiting interworking with DMS-100 (MWI) ☒

Network access data (NAC) ☐

Network call trace supported (NCT) ☐

Network name display method 1 (ND1) ☐

Network name display method 2 (ND2) ☒

Network name display method 3 (ND3) ☐

Name display - integer ID coding (NDI) ☐

Name display - object ID coding (NDO) ☐

Path replacement uses integer values (PRI) ☐

Path replacement uses object identifier (PRO) ☐

Release Link Trunks over IP (RLTI) ☐

Remote virtual queuing (RVQ) ☐

Trunk anti-tromboning operation (TAT) ☐

User to user service 1 (UUS1) ☐

NI-2 name display option. (NDS) ☐

Message waiting indication using integer values (QMWI) ☐

Message waiting indication using object identifier (QMWO) ☐

User to user signalling (UUI) ☐

Return - Remote Capabilities Cancel

Leave the **Advance options (ADVOPT)** section at default.

Click on the **Submit** button at the bottom of the **D-Channel 3 Property Configuration** page to save changes and complete the creation of new D channel.

## 5.6. Create an Application Module Link (AML)

On the Element Manager page, navigate to **System → Interfaces → Application Module Link**. The **Application Module Link** page is displayed on the right (screen not shown). Click on the

**Add** button to add a new Application Module Link. The **New Application Module Link** page is displayed as below.

Enter an AML port number in the **Port number** text box, e.g., **32** and a descriptive name, e.g., “**SIPL**” in the **Description** box. Note that The AML of SIP Line Service can use any port from 32 to 127. In this case, SIP Line Service is configured to use port **32**. Click on the **Save** button to complete the addition of the new AML link.

CS1000 Element Manager Help | Logout

Managing: [10.10.97.78](#) Username: admin  
System » Interfaces » [Application Module Link](#) » New Application Module Link

### New Application Module Link

Port number:  \* (16 - 127)

AML over ELAN

Description:

☐ Link control system parameters

Maximum octets:  (per HDLC frame)

\* Required value. Save Cancel

## 5.7. Create a Value Added Server (VAS)

On the Element Manager home page, navigate to **System → Interfaces → Value Added Server**. The **Value Added Server** page is displayed on the right, click on the **Add** button. The **Add Value Added Server** page is displayed; select the link **Ethernet LAN Link**.

The **Ethernet Link** page is displayed as shown below. Enter a number in the **Value added server ID** field, e.g., **32** and in the **Ethernet LAN Link** drop down list, select the AML number of ELAN that was created in **Section 5.6**. Leave the other fields as default values and click on the **Save** button to complete the addition of the new **VAS**.

CS1000 Element Manager Help | Logout

Managing: [10.10.97.78](#) Username: admin  
System » Interfaces » [Value Added Server](#) » [Add Value Added Server](#) » Ethernet Link

### Ethernet Link

Value added server ID:  \* (16 - 127)

Ethernet LAN Link:

ELAN port configured in ADAN

Application security: ☐

Interval:

Time interval for checking the link for overload in five second increments

Message count threshold:  \* (10 - 9999)

\* Required value. Save Cancel

## 5.8. Create a Virtual Trunk Zone

On the Element Manager home page, navigate to menu **System → IP Network → Zones**. The **Zones** page is displayed on the right, in this page select **Bandwidth Zones** link. On the **Bandwidth Zones** page, click on the **Add** button, the **Zone Basic Property and Bandwidth Management** page is displayed as shown the screen below.

Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**. Leave other fields as default values and click on the **Save** button to complete adding the Zone.

Repeat the procedure above to create another zone for the SIP Line phone; however remember to select **MO**, instead of **VTRK** in the **Zone Intent** field.

**CS1000 Element Manager**Help | Logout

Managing: [135.10.97.78](#) Username: admin  
System » IP Network » [Zones](#) » [Bandwidth Zones](#) » Bandwidth Zones 2 » [Edit Bandwidth Zone](#) » Zone Basic Property and Bandwidth Management

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**Zone Basic Property and Bandwidth Management**

Input Description	Input Value
Zone Number (ZONE):	<input type="text" value="2"/> * ( 1 - 8000 )
Intrazone Bandwidth (INTRA_BW):	<input type="text" value="1000000"/> ( 0 - 10000000 )
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) ▼
Interzone Bandwidth (INTER_BW):	<input type="text" value="1000000"/> ( 0 - 10000000 )
Interzone Strategy (INTER_STGY):	Best Quality (BQ) ▼
Resource Type (RES_TYPE):	Shared (SHARED) ▼
Zone Intent (ZBRN):	VTRK (VTRK) ▼
Description (ZDES):	<input type="text"/>

## 5.9. Create a SIP Line Route Data Block (RDB)

On the Element Manager home page, navigate to the menu **Routes and Trunks** → **Routes and Trunks**. The **Routes and Trunks** page is displayed on the right. In this page, click on the **Add route** button next to the customer number that the route will belong to.

**CS1000 Element Manager**  
Managing: **10.10.97.78** Username: admin  
Routes and Trunks » Routes and Trunks

**Routes and Trunks**

- Customer: 0	Total routes: 8	Total trunks: 151	<b>Add route</b>
+ Route: 1	Type: TIE	Description: SIP	<b>Edit</b> <b>Add trunk</b>
+ Route: 2	Type: TIE	Description: TOCM	<b>Edit</b> <b>Add trunk</b>
+ Route: 3	Type: TIE	Description: SIPLINE	<b>Edit</b> <b>Add trunk</b>
- Route: 4	Type: DID	Description: CONV	<b>Edit</b> <b>Add trunk</b>
- Route: 5	Type: TIE	Description: SIP_UDP	<b>Edit</b> <b>Add trunk</b>
+ Route: 6	Type: TIE	Description: SIPG729	<b>Edit</b> <b>Add trunk</b>
+ Route: 7	Type: IMUS	Description: IPMUS	<b>Edit</b> <b>Add trunk</b>
- Route: 10	Type: TIE	Description: PROGNOSIS	<b>Edit</b> <b>Add trunk</b>

The **Customer ID, New Route Configuration** page is displayed. There are 5 sections in the new route configuration page.

**CS1000 Element Manager** Help | Logout  
Managing: **10.10.97.78** Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, New Route Configuration

**Customer 0, New Route Configuration**

- + **Basic Configuration**
- + **Basic Route Options**
- + **Network Options**
- + **General Options**
- + **Advanced Configurations**

\* Required value.

**Save** **Cancel**



Expand the **Basic Configuration** section, and enter values as shown in the two screens below.

- **Route Number (ROUT):** Select an available number in the list, e.g., **8**.
- **Designator field for trunk (DES):** Enter a descriptive name, e.g. **SIPL**.
- **Trunk type (TKTP):** Select **TIE trunk data block (TIE)**.
- **Incoming and Outgoing trunk (ICOG):** Select **Incoming and Outgoing (IAO)**.
- **Access Code for Trunk group (ACOD):** Enter a number for ACOD, for example 8008.  
Note that this number has to follow the dialing plan rule.
- **The route is for a virtual trunk route (VTRK):** Select the checkbox.
- **Zone for codec selection and bandwidth management (ZONE):** Enter **2** which is the Virtual trunk zone number created in **Section 5.8**.
- **Node ID of signaling server of this route (NODE):** Enter **512** which is the node ID of the SIP Line configured in **Section 5.4**.
- **Protocol ID for the route (PCID):** Select **SIP Line (SIPL)** in the list.
- **Integrated services digital network option (ISDN):** Select the check box.

CS1000 Element Manager
Help | Logout

Managing: 10.10.97.78 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, New Route Configuration

### Customer 0, New Route Configuration

**- 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):   (0 - 8000)

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

- Protocol ID for the route (PCID):

Integrated services digital network option (ISDN): ☒

- **Mode of operation (MODE):** Select **Route uses ISDN Signaling Link (ISLD)**.
- **D channel number (DCH):** Enter **3** which is the D-channel number created in the **Section 5.5**.
- **Interface type for route (IFC):** Select **Meridian M1 (SL1)**.
- **Network calling name allowed (NCNA):** Select the check box.

- **Channel type (CHTY):** B-channel (BCH).
- **Trunk route optimization (TRO):** Select the check box.
- **Call type for outgoing direct dialed TIE route (CTYP):** Select **Unknown Call type (UKWN)**.
- **Calling Number dialing plan (CNDP):** Select **Coordinated dialing plan (CDP)**.

Leave default values for The **Basic Route Options, Network Options, General Options, and Advanced Configurations** sections. Click the **Submit** button to complete the addition of new route and save configuration.

Integrated services digital network option (ISDN):	<input checked="" type="checkbox"/>	
- 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):	1	(0 - 32700)
- Network calling name allowed (NCNA):	<input checked="" type="checkbox"/>	<==
- Network call redirection (NCRD):	<input checked="" type="checkbox"/>	<==
-- Trunk route optimization (TRO):	<input checked="" type="checkbox"/>	<==
- Recognition of DTI2 ABCD FALT signal for ISL (FALT):	<input type="checkbox"/>	
- Channel type (CHTY):	B-channel (BCH)	<==
- Call type for outgoing direct dialed TIE route (CTYP):	Unknown Call type (UKWN)	<==
- Insert ESN access code (INAC):	<input type="checkbox"/>	
- Integrated service access route (ISAR):	<input type="checkbox"/>	
- Display of access prefix on CLID (DAPC):	<input type="checkbox"/>	
- Mobile extension route (MBXR):	<input type="checkbox"/>	
- Mobile extension outgoing type (MBXOT):	National number (NPA)	
- Mobile extension timer (MBXT):	0	(0 - 8000 milliseconds)
Calling number dialing plan (CNDP):	Coordinated dialing plan (CDP)	

## 5.10. Create Virtual Trunks for SIP Line Route

On the Element Manager home page, navigate to **Routes and Trunks** → **Routes and Trunks**. The **Routes and Trunks** page is displayed on the right. Select the **Add trunk** button beside the route **8** that was created in the **Section 5.9** above to create new trunks.

**CS1000 Element Manager**

Managing: [10.10.97.78](#) Username: admin  
Routes and Trunks » Routes and Trunks

**Routes and Trunks**

- Customer: 0	Total routes: 9	Total trunks: 151	Add route	
+ Route: 1	Type: TIE	Description: SIP	Edit	Add trunk
+ Route: 2	Type: TIE	Description: TOCM	Edit	Add trunk
+ Route: 3	Type: TIE	Description: SIPLINE	Edit	Add trunk
- Route: 4	Type: DID	Description: CONV	Edit	Add trunk
- Route: 5	Type: TIE	Description: SIP_UDP	Edit	Add trunk
+ Route: 6	Type: TIE	Description: SIPG729	Edit	Add trunk
+ Route: 7	Type: IMUS	Description: IPMUS	Edit	Add trunk
- Route: 10	Type: TIE	Description: PROGNOSIS	Edit	Add trunk
- Route: 8	Type: TIE	Description: SIPL	Edit	Add trunk

The **Customer 0, Route 8, Trunk type TIE trunk data block** page is displayed. Enter values for fields as shown below:

- **Multiple trunk input number (MTINPUT):** Enter **32** to create 32 trunks.
- **Auto increment member number:** Select the check box. The trunks are created incrementally.
- **Trunk data block (TYPE):** Select **IP Trunk (IPTI)**.
- **Terminal Number (TN): 100 0 8 0.** Enter the first Terminal Number in a range of Terminal number.
- **Designator field for trunk:** Enter a descriptive name, e.g. “**SIPL Trk**”.
- **Member number:** enter **97**. This is the ID of the trunk, just enter the first ID for the first trunk, next ID will be automatically created and incremented.
- **Start arrangement Incoming:** Select **Immediate (IMM)**.
- **Start arrangement Outgoing:** Select **Immediate (IMM)**.
- **Channel ID for this trunk:** **97**, this channel ID should be the same as the ID of Member Number and it has to be a unique number in the same type of trunk.

CS1000 Element Manager

Help | Logout

Customer 0, Route 8, Trunk type TIE trunk data block

- Basic Configuration

Multiple trunk input number: 32 Range: 2 - 3700

Auto increment member number: ☒ <==

Trunk data block: IP Trunk (IPTI) <== ▼

Terminal number: 100 0 8 0 <== \*

Designator field for trunk: SIPL Trk <==

Extended trunk: VTRK

Member number: 97 <== \*

Level 3 Signaling: ▼

Card density: ▼

Start arrangement Incoming: Immediate (IMM) <== ▼

Start arrangement Outgoing: Immediate (IMM) <== ▼

Trunk group access restriction:

Channel ID for this trunk: 97 <==

Class of Service: Edit <==

+ Advanced Trunk Configurations

\* Required value.

Save Cancel

Click on the **Class of Service** button and assign following class of services as shown the screen below:

- **Dial Pulse:** Select **Digitone (DTN)**.
- **Media security:** Select **Media Security Never (MSNV)**.
- **Restriction level:** Select **Unrestricted (UNR)**.
- Leave other class of services at default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page.

- Centrex Switchhook Flash:

- Dial Pulse: Digitone (DTN)

- DTR PAD value:

- Echo Canceling:

- Hong Kong DTI:

- Loop Break Supervised COT:

- Make-break ratio for dial pulse:

- Manual Incoming:

- Media Security: Media Security Never (MSNV)

- Network Hook Flash Over M911P:

- Polarity:

- Priority:

- Restriction level: Unrestricted (UNR)

- Reversed Ear Piece:

- Short or long line:

- Transmission Class of Service:

- Warning Tone:

- Reversed Ear Piece:

- ARF Supervised COT:

Leave the **Advance Trunk Configurations** section at default values and click on the **Save** button to complete the addition of new virtual trunks for SIP Line.

## 5.11. Create a SIP Line Phone

To create a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 11/20 as shown below.

The bold fields must be properly inputted as they are configured on the Call server, for other fields hit enter to leave it at default values.

```
LD 20
Req prt
TYPE: uext
TN 104 0 0 2
DES SL8741
TN 104 0 00 02 → Terminal number of Universal Extension of SIP Line phone
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL → Type of UXTY is SIP Line
MCCL YES
SIPN 0
SIP3 1 → 3rd SIP endpoint is enabled
FMCL 0
TLSV 0
SIPU 54009 → SIP user which is used in the SIP endpoint for registration
NDID 512 → The node ID of SIP Line.
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001 → Zone for SIP endpoint configured as MO
MRT
ERL
ECL 0
VSIT NO
FDN
TGAR 1
LDN NO
NCOS 0
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW 1234 → The password used to register to SIP Line server
SFLT NO
CAC_MFC 0
CLS CTD FBA WTA LPR MTD FNA HTD TDD HFD CRPD → Depend on feature cls enabled
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCB
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXD ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTB AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCB
RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
```

```

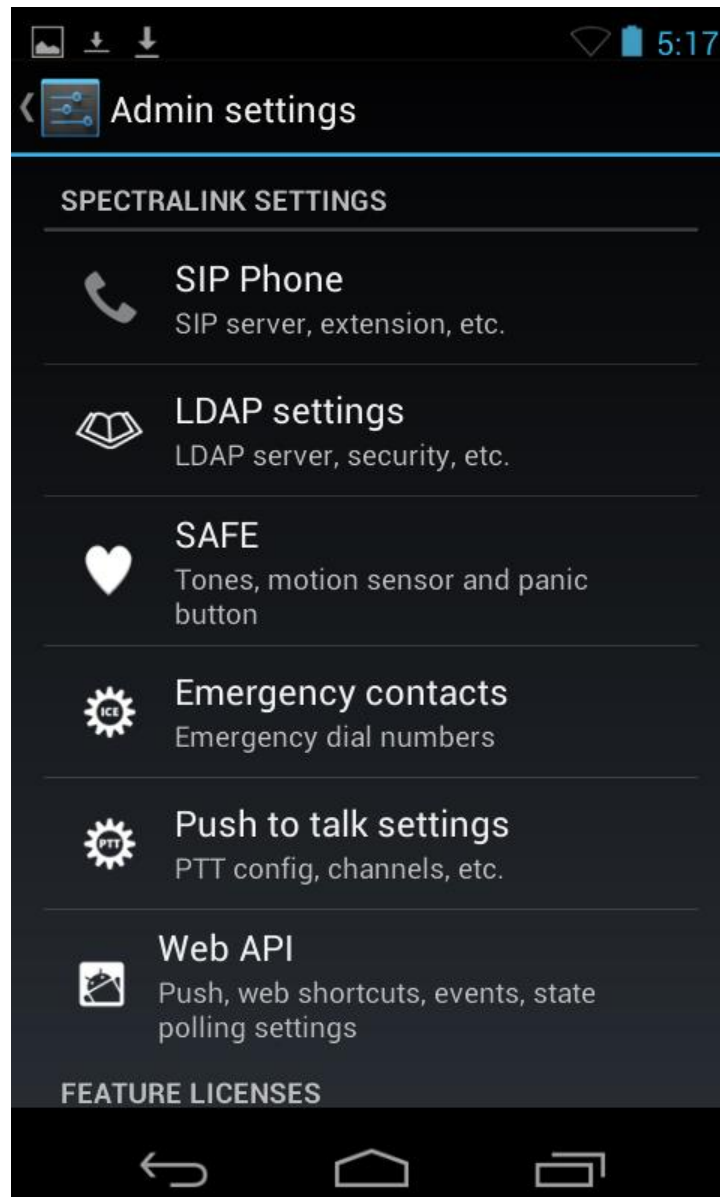
    MSNV FRA  PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO 0
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 54009 0      MARP → The main directory number of SIP endpoint
    CPND
        CPND_LANG ROMAN
        NAME Poly1 54502
        XPLN 13
        DISPLAY_FMT FIRST, LAST
01 HOT U 2654009 MARP 0 → The Hot U with the prefix 26 configured in
adding SIP Line server.
02 MSB → MSB key is used for Make Set busy feature on SIP endpoint
03 CWT → CWT key is used for Call Waiting feature on SIP endpoint
04
05
06
07
08
09
10
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27

```

## 6. Configure Spectralink PIVOT™ 8741

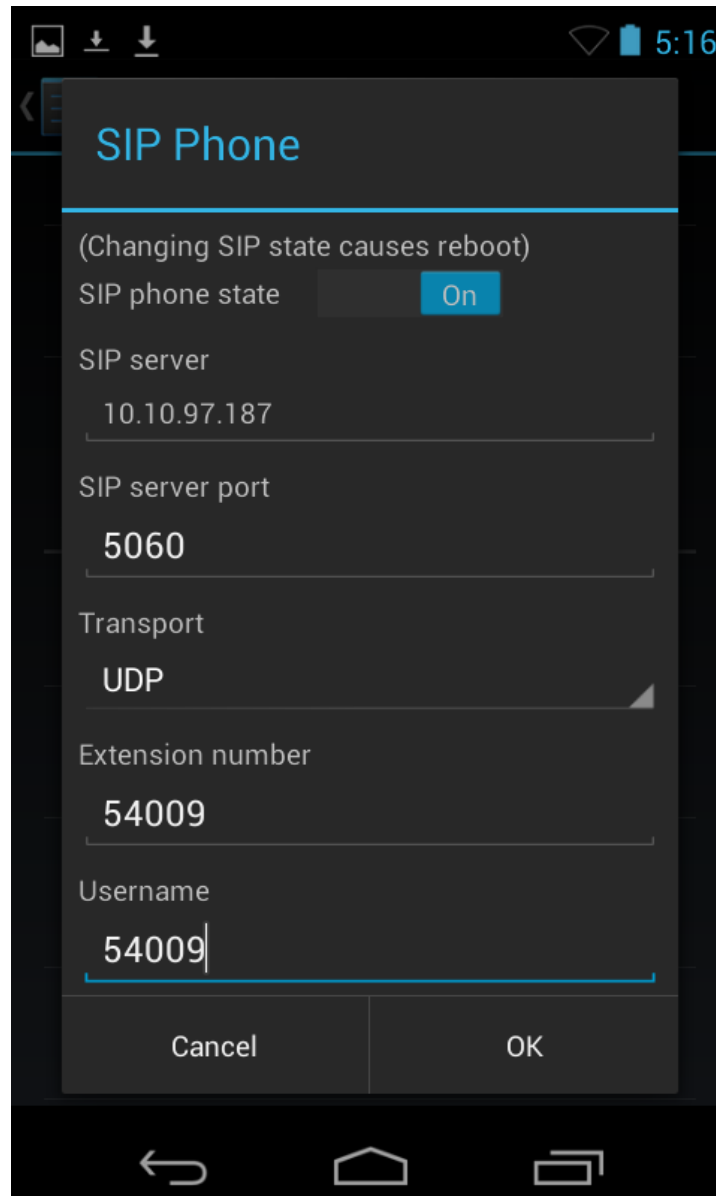
The configuration for the PIVOT phone is done by using the Settings menu on the phone itself and not using the Spectralink Communication Manager System (CMS).

Navigate to **APPS → Settings → Admin settings**. The **Admin password** window is displayed (not shown), enter appropriate password to access to the **Admin settings** menu. The **Admin settings** menu is displayed as shown below.





In the **SPECTRALINK SETTINGS**, select **SIP PHONE**. The **SIP Phone** window is displayed as below. Enter the IP address 10.10.97.187 of SIP Line server in the **SIP server** field, port 5060 in the **SIP server port** field, UDP in the **Transport** field, 54009 in the **Extension number** field, and 54009 in the **Username** field.



The screenshot shows a mobile application interface for configuring a SIP phone. The window is titled "SIP Phone" in blue text. Below the title, a note states "(Changing SIP state causes reboot)". The "SIP phone state" is set to "On" with a blue toggle switch. The "SIP server" field contains the IP address "10.10.97.187". The "SIP server port" field contains the number "5060". The "Transport" field is set to "UDP". The "Extension number" field contains the number "54009". The "Username" field contains the number "54009". At the bottom of the window are "Cancel" and "OK" buttons. The background of the application shows a dark interface with a back arrow and a home button at the bottom.

Field	Value
SIP phone state	On
SIP server	10.10.97.187
SIP server port	5060
Transport	UDP
Extension number	54009
Username	54009

Continue scrolling down to lower sections, enter password for SIP user 54009 as configured in **Section 5.11** in the **Password** field, leave blank for the **Voice mail retrieval address** field and remain default value for the other fields.

Click **OK** button to complete.

**SIP Phone**

Username  
54009

Password  
.....

Voice mail retrieval address  
(Blank=auto or VM pilot #)

Audio DSCP  
0x2e

Call control DSCP  
0x28

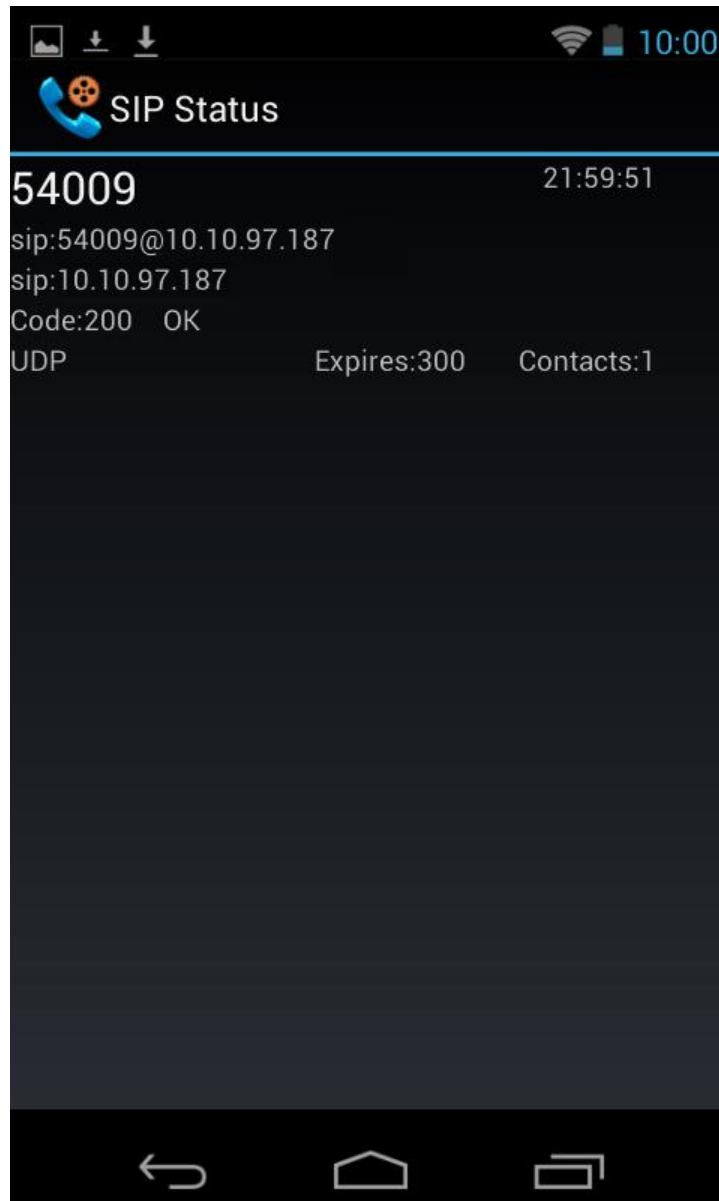
G.711u codec priority  
1

Cancel OK

## 7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- From the PIVOT phone, verify that the phone successfully registers to the CS 1000 SIP Line server by navigating to **APPS → SIP Status**. The SIP Status window should indicate **Code: 200 OK** as shown below.



- Verify that the PIVOT phone registers successfully with the CS 1000 SIP Line Gateway server by using the CS 1000 Linux command.

- Log in to the SIP Line server as an administrator by using the Avaya account. Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked.

```
[admin@sip175 ~]$ slgSetShowByUID 54009
```

```
=== VTRK ===
```

UserID	AuthId	TN	Clients	Calls	SetHandle	Pos ID	SIPL Type
54009	54009	104-00-00-02	1	0	0x8d35ee0		SIP Lines

```
StatusFlags = Registered Controlled KeyMapDwld SSD
FeatureMask =
CallProcStatus = -1

Current Client = 0, Total Clients = 1

== Client 0 ==
IPv4:Port:Trans = 10.33.5.31:5060:udp
Type = Unknown
UserAgent = Spectralink-UA_0_4_4
x-nt-guid = c6051e2a3b2c60535066cd838f5eaced
RegDescrip =
RegStatus = 1
PbxReason = OK
SipCode = 200
hTransc = (nil)
Expire = 300
Nonce = 7cc155f3db5a2fae3cdd10e4a0ee5644
NonceCount = 2
hTimer = 0x8cbf138
TimeRemain = 223
Stale = 0
Outbound = 0
ClientGUID = 0
MSec CLS = MSNV (MSEC-Never)
Contact = sip:54009@10.33.5.31:5060;ob
KeyNum = 255
AutoAnswer = NO
```

Key	Func	Lamp	Label
0	2	0	54009
1	126	0	2654009
2	29	0	
3	48	0	
4	3	0	54358
5	2	0	54359
17	16	0	
18	18	0	
19	27	0	
20	19	0	
21	52	0	
22	25	0	
24	11	0	
25	30	0	
26	31	0	

- Place a call from and to the PIVOT telephone and verify that the call is established with 2-way speech path.
- During the call, use a pcap tool (ethereal/wireshark) at the SIP Line Gateway and clients to make sure that all SIP request/response messages are correct.

## 8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**. The PIVOT™ by Spectralink 87 Series Wireless Telephone Version 1.2.0.6893 is considered to be in compliance with Avaya Communication Server 1000 SIP Line Gateway Release 7.6.

## 9. Additional References

Product documentation for the Avaya CS 1000 products may be found at:

<https://support.avaya.com/css/Products/>

Product documentation for the PIVOT™ by SpectraLink 87 Series products may be found at:

<http://partneraccess.spectralink.com/products/wi-fi/spectralink-8000-portfolio/pivot-87-series>

[1] Avaya CS 1000 Documents:

[Avaya Communication Server 1000E Installation and Commissioning](#)

[Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6](#)

[Avaya Communication Server 1000 Element Manager System Reference – Administration](#)

[Avaya Communication Server 1000 Co-resident Call Server and Signaling Server](#)

[Fundamentals](#)

[Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals.](#)

[Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning](#)

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