



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

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# **Application Notes for Configuring Avtec Scout with Avaya Communication Server 1000 Release 7.5 - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Avtec Scout to interoperate with Avaya Communication Server 1000 7.5 as a SIP endpoint.

Avtec Scout is a Voice over IP Console system. In the compliance testing, Avtec Scout registered as a SIP endpoint to Avaya Communication Server 1000 and thereby able to receive and send calls both internally and over the PSTN.

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 describe the configuration steps required for Avtec Scout to interoperate with Avaya Communication Server 1000 7.5 by registering as a third party SIP endpoint.

Avtec Scout (hereafter referred to as Scout) is a VoIP Console solution. In the compliance testing, Scout registered to Avaya Communication Server 1000 7.5 (hereafter referred to as Communication Server 1000) as a third party SIP endpoint, so as to make and receive calls both internally and over the PSTN.

All applicable SIP endpoint telephony features of Communication Server 1000 were executed on Scout to ensure interoperability with Communication Server 1000.

## 2. General Test Approach and Test Results

The general test approach was to have Scout register to Communication Server 1000 via SIP Line Gateway of Communication Server 1000. Calls were then placed from Communication Server 1000 desktop users to and from Scout. Other telephony features such as Call Forward, Transfer, Conference, Call Waiting, Codec Negotiation etc were also verified.

The feature test cases were performed manually. Calls were manually established among Scout console with Communication Server 1000 IP (UNISTim and SIP) and/or PSTN users. Call controls were performed from both Scout and Communication Server 1000 to verify the call scenarios.

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

### 2.1. Interoperability Compliance Testing

The focus of this testing was to verify that Scout console was able to register as third party SIP endpoints to the Communication Server 1000 SIP Line Gateway. The following areas were tested:

- Registration of Scout lines with SIP Line gateway. During compliance testing 3 lines were registered on one console.
- Call establishment of Scout with Communication Server 1000 Deskphones.
- Telephony features: Basic calls both inbound and outbound, Blind and Consultative transfer and Call Display.
- Scout does not support the following features internally; however these features were tested with Scout as a participant: Call Forward No Answer, Call Forward Busy, Call

Forward All Calls, Call Park and Retrieve, Conference, Voicemail (tested with Scout leaving messages only) and Call Waiting.

- Codec negotiation – G.711 and G.729 at 20 ms.
- Calls from and to PSTN telephone via PRI trunk.

## 2.2. Test Results

The objectives outlined in **Section 2.1** were verified and met. All tests were executed and passed with the following observations:

- Scout does not support Message Waiting Indicator (MWI), Conference, Hold and Do not Disturb (DND) features.
- SIP Line features like Speed Dial, Group Call Pickup, Make Set Busy, Ring Again Busy and Ring Again No Answer were not tested since it is not relevant to Scout.
- Scout does not support Codec negotiation at 30 ms.
- Scout console only supports single call and no multiple calls. If the console has multiple lines and call is presented to all lines, only one line will have an active call and the rest will be put on hold.

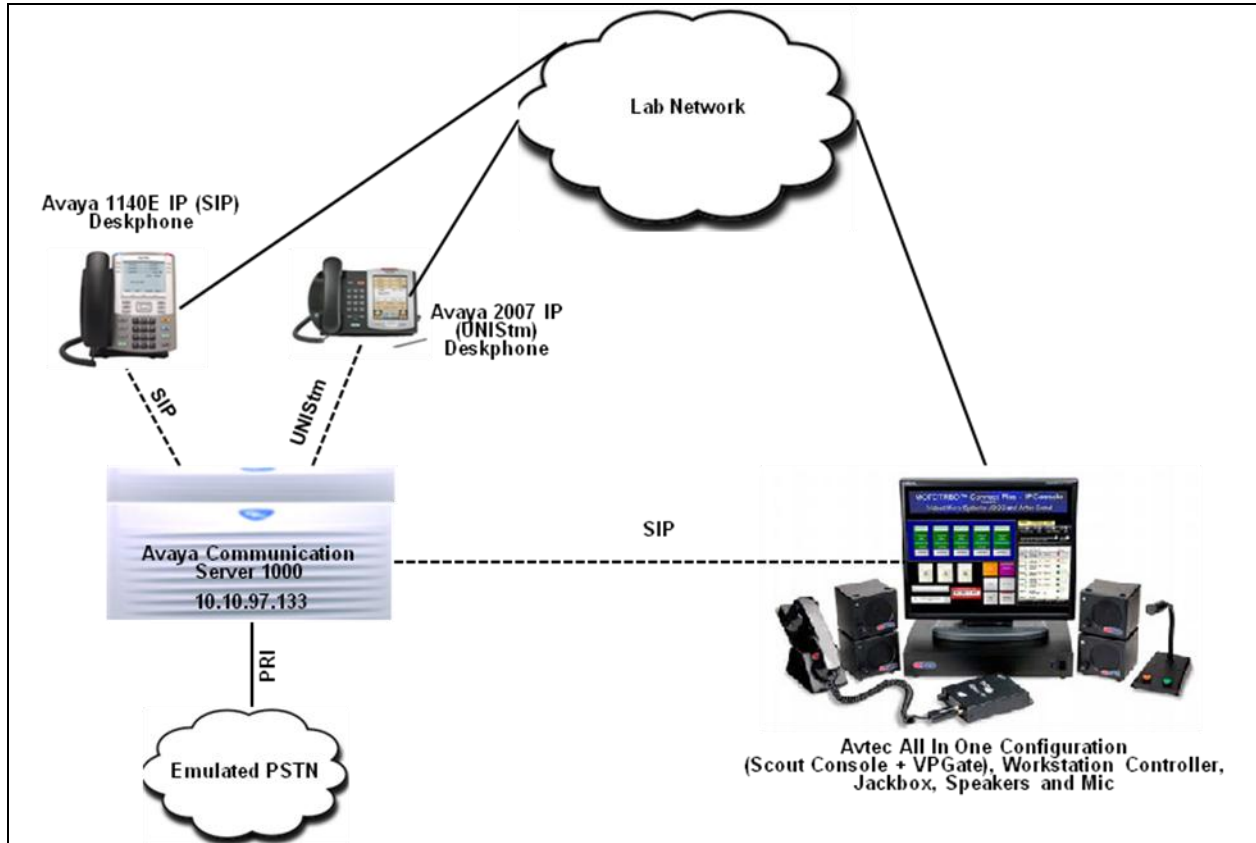
## 2.3. Support

Technical support for Avtec Scout console can be obtained through Avtec technical support:

- Phone: 1-800-545-3034 or 1-803-892-2181 ext. 201
- Web: <http://www.avtecinc.com/services-support/>
- Email: [CustomerSupport@avtecinc.com](mailto:CustomerSupport@avtecinc.com)

### 3. Reference Configuration

**Figure 1** illustrates the reference configuration used during compliance testing.



**Figure 1: Network Configuration Diagram**

## 4. Equipment and Software Validated

The following equipment and software/firmware were used for the reference configuration:

Equipment/Software	Release/Version
Avaya Communication Server 1000	7.50.17
Avaya 1140E IP (SIP) Deskphone	04.03.09.00
Avaya 2007 IP (UNISTim) Deskphone	0621C8A
Avtec Scout Console <ul style="list-style-type: none"><li>VPGate on Windows 7 OS (32 bit)</li><li>Scout on Windows 7 OS (32 bit)</li><li>Workstation Controller</li></ul>	Ver 2.5.4.17 Ver 2.5.4.26 Rev 34.97

## 5. Configure Avaya Communication Server 1000 – SIP Line

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

The following is the summary of steps required for configuring the Communication Server 1000 SIP Line:

- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service in Customer Data Block.
- 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. Prerequisite

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

- Installed with Communication Server 1000 Release 7.5 Linux Base.
- Joined Communication Server 1000 Release 7.5 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 in to Unified Communications Management (UCM) and Element Manager (EM)

Use a web browser to launch the Avaya Communication Server 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 (not shown). For more information, see [2].

On the **Elements** page of Unified Communications Management, under the **Element Name** column, click the server name to navigate to Element Manager for that server.

Avaya Unified Communications Management

Host Name: ucm1.bvwdev.com Software Version: 02.20.0034.00(5666) User Name admin

**Elements**

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Search Reset

Add... Edit... Delete

Element Name	Element Type	Release	Address	Description
EM on cppm1	CS1000	7.5	10.10.97.66	New element

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

On the EM page, navigate to **Customers** on the left column menu; select the customer number to be enabled with SIP Line Service (not shown).

- 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.

CS1000 Element Manager

Managing: 10.10.97.66 Username: admin

Customers » Customer 00 » Customer Details » SIP Line Service

**SIP Line Service**

☒ SIP Line Service

User agent DN prefix: 27

Optional features: ☐ Nortel Multimedia

Save Cancel

## 5.4. Add a new SIP Line Telephony Node

On the EM page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. Click **Add** to add a new SIP Line Node to the IP Telephony Nodes (not shown). The screen below shows the node 550 configured during compliance testing.

- **Node ID:** 550; this is the node ID of SIP Line server.
- **Call Server IP Address:** 10.10.97.66
- **Node IPv4 Address:** 10.10.97.133; this is the IP address that SIP endpoint uses to register to.
- **Subnet Mask:** 255.255.255.192.
- **Gateway IP Address:** 10.10.97.65
- **Subnet Mask:** 255.255.255.192

**AVAYA** CS1000 Element Manager

Managing: 10.10.97.66 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details

**Node Details (ID: 550 - SIP Line)**

Node ID: 550 \* (0-9999)

Call server IP address: 10.10.97.66 \*

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: 10.10.97.133 \*

Subnet mask: 255.255.255.192 \*

Node IPv6 address:

\* Required Value.

Save Cancel



Once the node has been added the SIP Line can be configured. Screen below shows the **SIP Line Configuration Details** page used during compliance testing. Provide a descriptive name for **SIP domain name** and **SLG endpoint name** fields. Enter the node ID for **SLG Group ID** and the port value for the **SLG Local Sip port** fields. Check the box for **Enable gateway service on this node**.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.66 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration

**Node ID: 550 - 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:  \*

SLG endpoint name:

SLG Group ID:

SLG Local Sip port:  (1 - 65535)

SLG Local Tls port:  (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:

Number of byte re-negotiation:

Options: ☐ Client authentication

\* Required Value.

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

Save Cancel

**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, log in as administrator to the command line interface of the SIP Line server and issue command: **appstart restart**.

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

On the EM page, on the left column menu navigate to **Routes and Trunks** → **D-Channels**. Under the **Configuration** section as shown in the screen below, select a D-channel number from the **Choose a D-Channel Number** drop down menu and click on the **to Add** button.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The top header shows the AVAYA logo and the title "CS1000 Element Manager". Below the header, a navigation menu on the left lists various system components, with "Routes and Trunks" and its sub-item "D-Channels" highlighted. The main content area is titled "D-Channels" and is divided into two sections: "Maintenance" and "Configuration". The "Maintenance" section lists several diagnostic links: "D-Channel Diagnostics (LD 96)", "Network and Peripheral Equipment (LD 32, Virtual D-Channels)", "MSDL Diagnostics (LD 96)", "TMDI Diagnostics (LD 96)", and "D-Channel Expansion Diagnostics (LD 48)". The "Configuration" section contains a form with a red border. This form includes a label "Choose a D-Channel Number:" followed by a dropdown menu currently showing "0", and the text "and type:" followed by another dropdown menu showing "DCH" and a "to Add" button.

**AVAYA** **CS1000 Element Manager**

Managing: [10.10.97.66](#) 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:

Screen below shows the D-Channel (DCH 11) that was added for compliance testing. All fields are left at default values except the following,

- **D Channel Card Type:** DCIP
- **Designator:** SIPL; any descriptive name can be used here.
- **Interface type for D-Channel:** Meridian Meridian 1 (SL1)
- **Release ID of the switch at the far end:** 7; since we are dealing here with Release 7.xx.

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	SIPL
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"/> more PRI
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	3700 Range: 0 - 3700

+Basic options (BSCOPT)

Click on the **Basic options (BSCOPT)** link seen in the above screen. The **Basic options (BSCOPT)** list expands (not shown).

Click on **Edit** to configure **Remote Capabilities (RCAP)** (not shown). The **Remote Capabilities Configuration** appears as shown below. Check boxes for the **Message waiting interworking with DMS-100 (MWI)** and **Network name display method 2 (ND2)**. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** (not shown) to return the D-Channel Property Configuration page. **Message waiting interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints. **Network name display method 2 (ND2)** must be enabled to support name display between SIP Line endpoints. Other check boxes are left unchecked.

**AVAYA**  

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

**CS1000 Element Manager**  
Managing: 10.10.97.66 Username: admin  
Routes and Trunks » D-Channels » D-Channels 11 Property Configuration » - Remote Capabilities Configuration  
**- Remote Capabilities Configuration**

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWI)	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>

Click on the **Submit** button (not shown) of the D-Channel Property Configuration page to save changes.

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

On the EM page, navigate to **System** → **Interfaces** → **Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). During compliance testing AML 32 was added and the screen below shows the details of the same.

The screenshot shows the CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, Home, Links, Virtual Terminals, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, and Interfaces. The 'Interfaces' category is expanded, showing 'Application Module Link' as the selected option. The main content area is titled 'Application Module Link Details 32'. It displays the following configuration details: Link Type: ELAN, Description: SIPL, and Maximum octets: 512 (per HDLC frame). At the bottom right, there are 'Save' and 'Cancel' buttons. The top of the interface shows the Avaya logo, the title 'CS1000 Element Manager', and a 'Help | Logout' link. The top status bar indicates the user is logged in as 'admin' with the IP address '10.10.97.66'.

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

On the EM page, navigate to **System** → **Interfaces** → **Value Added Server** and click on the **Add** button to add a new VAS (not shown). The screen below shows the details of VAS 32 configured during compliance testing. Note that in the **Ethernet LAN Link** drop down list; select the AML number of ELAN that was created in the **Section 5.6**. All other values are left at default.

The screenshot shows the CS1000 Element Manager interface. The left sidebar is the same as in the previous screenshot, but the 'Value Added Server' option under the 'Interfaces' category is selected. The main content area is titled 'Edit Value Added Server 032'. It displays the following configuration details: Ethernet LAN Link: 032, ELAN port configured in ADAN, Application security: ☐, Interval: 1 (Time interval for checking the link for overload in five second increments), and Message count threshold: 9999 (10 - 9999). At the bottom left, there is a note: '\* Required value.' At the bottom right, there are 'Save' and 'Cancel' buttons. The top of the interface shows the Avaya logo, the title 'CS1000 Element Manager', and a 'Help | Logout' link. The top status bar indicates the user is logged in as 'admin' with the IP address '10.10.97.66'.

## 5.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System → IP Network → Zones**. The **Zones** page appears on the right, in this page select **Bandwidth Zones** link (not shown).

On the **Bandwidth Zones** page, click on the **Add** button to add a new zone (not shown).

Screen below shows the zone values configured during compliance testing. Enter a zone number in the **Zone Number (ZONE)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**.

All other fields are left at default.

**Note:** Repeat the step above to create another zone for the SIP Line phone; however remember to select **MO**, instead of **VTRK** in the **Zone Intent (ZBRN)** field.

The screenshot displays the AVAYA CS1000 Element Manager interface. The top header shows the AVAYA logo and the title 'CS1000 Element Manager'. Below the header, a navigation sidebar on the left lists various system components, with 'Zones' highlighted. The main content area is titled 'Zone Basic Property and Bandwidth Management'. It features a table with two columns: 'Input Description' and 'Input Value'. The table contains the following entries:

Input Description	Input Value
Zone Number (ZONE):	254 (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	1000000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

At the bottom of the form, there are three buttons: 'Submit', 'Refresh', and 'Cancel'.

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

On the EM page, navigate to the menu **Routes and Trunks** → **Routes and Trunks**; the **Routes and Trunks** page appears (not shown). To add a new route, click on the **Add route** button (not shown) next to the customer number that the route will belong to.

The next two screens shows the values configured for the route used during compliance testing.

- **Route number (ROUT):** 11; any available route number can be used here.
- **Trunk type (TKTP):** TIE
- **Incoming and outgoing trunk (ICOG):** IAO
- **Access code for the trunk route (ACOD):** 1011; any unused DN can be used here.
- **The route is for a virtual trunk route (VTRK):** Checked.
- **Zone for codec selection and bandwidth management (ZONE):** 254; this is the Virtual trunk zone number that created in the **Section 5.8**.
- **Node ID of signaling server of this route (NODE):** 550; this is the node ID of the SIP Line created in **Section 5.4**.
- **Protocol ID for the route (PCID):** SIP Line (SIPL).
- **Integrated services digital network option (ISDN):** Checked.
- **Mode of operation (MODE):** Route uses ISDN Signaling Link (ISLD).
- **D channel number (DCH):** 11; this is the D-channel number that was created in the **Section 5.5**.
- **Interface type for route (IFC):** Meridian M1 (SL1).
- **Network calling name allowed (NCNA):** Checked.
- **Network call redirection (NCRD):** Checked.
- **Channel type (CHTP):** B-channel (BCH).
- **Call type for outgoing direct dialed TIE route (CTYP):** CDP.
- **Calling number dialing plan (CNDP):** CDP.

Leave default values for The **Basic Route Options**, **Network Options**, **General Options**, and **Advanced Configurations** sections.

AVAYA

CS1000 Element Manager

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

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 00

Route number (ROUT): 11

Designator field for trunk (DES): SIPL

Trunk type (TKTP): TIE

Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)

Access code for the trunk route (ACOD): 1011

Trunk type M911P (M911P):

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

- Zone for codec selection and bandwidth management (ZONE): 00254 (0 - 8000)

- Node ID of signaling server of this route (NODE): 550 (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): 11 (0 - 254)

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

- Private network identifier (PNI): 00001 (0 - 32700)

- Network calling name allowed (NCNA):

- Network call redirection (NCRD):

- Trunk route optimization (TRO):

- Recognition of DTI2 ABCD FALT signal for ISL (FALT):

- Channel type (CHTY): B-channel (BCH)

- Call type for outgoing direct dialed TIE route (CTYP): Coordinated Dialing Plan (CDP)

- Insert ESN access code (INAC):

- Integrated service access route (ISAR):

- Display of access prefix on CLID (DAPC):

- Mobile extension route (MBXR):

- 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)

Tools

- + Backup and Restore
- Date and Time
- + Logs and reports

Security

- + Passwords
- Policies
- + Login Options

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## 5.10. Create SIP Line Virtual Trunks

On the EM page, navigate to **Routes and Trunks** → **Routes and Trunks** and select the **Add route** button beside the route that was created in **Section 5.9** to create new trunks.

Screen below shows the values configured for the trunks that were used during compliance testing.

- **Auto increment member number:** checked.
- **Trunk data block:** IP Trunk (IPTI).
- **Terminal number:** 100 0 01 00 → enter the first TN of a range TN.
- **Designator field for trunk:** SIPL; any descriptive name can be used here.
- **Extended trunk:** VTRK
- **Member number:** 1; this is ID of trunk, just enter the first ID for first trunk; next ID will be automatically created and incremented.
- **Start arrangement Incoming:** Immediate (IMM).
- **Start arrangement Outgoing:** Immediate (IMM).
- **Trunk group access restriction:** 1.
- **Channel ID for this trunk:** 1; this ID should be the same with the ID of Member Number.

Click on the **Class of Service Edit** button and assign following class of services (not shown):

- **Media security:** Media Security Never (MSNV).
- **Restriction level:** Unrestricted.

Rest of the values was left at default.

**AVAYA** CS1000 Element Manager

Managing: 10.10.97.66 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, Route 11, Trunk 1 Property Configuration

### Customer 0, Route 11, 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 Signalling:

Card density:

**+ Advanced Trunk Configurations**

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

Channel ID for this trunk:

Class of Service:

## 5.11. Create a SIP Line Phone

To create a SIP Line phone on the Call Server, log in to the Communication Server 1000 CLI and issue the overlay **LD 11** or **20**.

The screen below shows the SIP endpoint configured during compliance testing so that any third party SIP endpoint can register to Communication Server 1000 SIP Line gateway. The values in bold are mandatory and the rest can be left at default. Only values configured during compliance testing are shown below.

```
DES  AVTEC  → Phone Description
TN   100 1 11 13  VIRTUAL
TYPE UEXT  → Universal extension type for SIP line phone
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL  → Universal extension is SIP line type
MCCL YES
SIPN 0
SIP3 1      → Value has to be 1 so that a third party SIP endpoint can register
FMCL 0
TLSV 0
SIPU 58124  → SIP phone user extension DN
NDID 550    → Node ID of SIP line system
.....
CFG_ZONE 00001 → SIP line zone configured
CUR_ZONE 00001
.....
SSU
LNRS 12
XLST
SCPW 1234    Password authentication for SIP user 58124
.....
KEY  00 SCR 58124 0      MARP → Key 00 is SIP phone DN
      CPND
      CPND_LANG ROMAN
      NAME Avtec Line1
      XPLN 13
      DISPLAY_FMT FIRST, LAST
      01 HOT U 2758124 MARP 0
.....
```

## 6. Configure Avtec Scout Console

This section provides the procedures for configuring Scout console so that they can register to the Communication Server 1000 as a third party SIP endpoint. The procedures include the following areas:

- Adding SIP endpoint on VPGate.
- Configuring the SIP driver on VPGate.

The detail installation and configuration of Scout is typically performed by users after attending the training provided by Avtec. The procedural steps are presented in these Application Notes for informational purposes. For further installation, configuration and administration information for Scout, contact the Avtec technical support as mentioned in **Section 2.3**.

## 6.1. Adding SIP endpoint on VPGate

This section shows how to add a SIP endpoint on VPGate.

Open a browser and access the VPGate Configuration page by typing the IP address of VPGate in the URL at port 3082, i.e. <http://<ip-address>:3082>

Navigate to **Endpoints** → **Endpoints Summary** to add a SIP endpoint (not shown).

The screen below shows the information of an already added SIP endpoint. Enter a descriptive name for the field **Endpoint Name**. Select “AVAILABLE” from the drop down for **Service State** field. Select “FULL DUPLEX” from the drop down for **Receive Audio Mode** field. The rest of the fields can be at their default values.

The screenshot displays the VPGate Configuration web interface. At the top, there's a header 'VPGate Configuration' and 'VPGate2'. Below this is a navigation bar with tabs: 'VPGate', 'Endpoints', 'Advanced', 'Diagnostics', and 'Help'. The 'Endpoints' tab is active, and within it, 'Endpoint Summary' and 'Input Alerts' are visible. The main content area shows the 'Endpoint Configuration' page for 'Endpoint Summary > Endpoint Configuration'. The page has a title 'Endpoint Configuration' and buttons for 'Update', 'Defaults', 'Reset', and a green 'Update Successful' message. The configuration is organized into sections: 'Endpoint Configuration' (with fields for 'Endpoint Name' set to 'sip101' and 'Service State' set to 'AVAILABLE'), 'Endpoint Connection' (with 'Endpoint Audio' set to 'VoIP'), 'VoIP Audio Settings' (with 'Receive Audio Mode' set to 'FULL DUPLEX'), 'Inbound Calls' (with 'Emergency Only' set to 'NO'), and 'Miscellaneous' (with 'PTT Time Out' set to '300', 'PTT Override Priority' set to 'FIRST CONSOLE', 'PTT Override Immunity Time' set to '10', 'Secondary Keying Action' set to 'NONE', and 'Redundant Priority' set to '0'). Each field has a small blue circular icon to its right.

Endpoint Configuration	
Endpoint Name	sip101
Service State	AVAILABLE

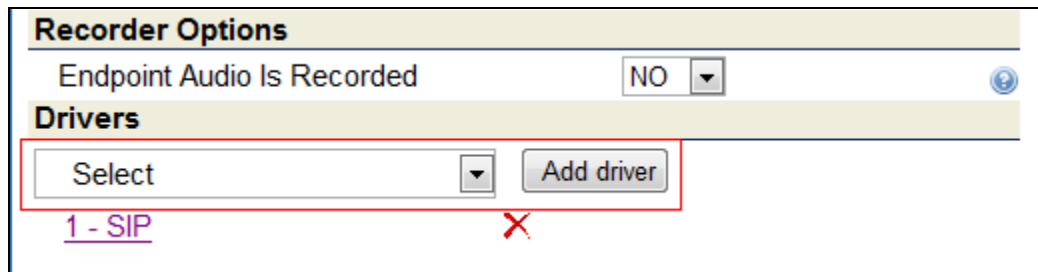
Endpoint Connection	
Endpoint Audio	VoIP

VoIP Audio Settings	
Receive Audio Mode	FULL DUPLEX
Override Receive Audio IP Port	0
Override Transmit Audio IP Port	0
VoIP Audio Jitter Depth (ms)	100
Squelch Tail Time Out	0
Allow Barge-in/Monitor Outbound Audio	YES

Inbound Calls	
Emergency Only	NO
Unanswered Time Out	0
Call Clear Time Out	3600
Duplicate Call Supported	NO

Miscellaneous	
PTT Time Out	300
PTT Override Priority	FIRST CONSOLE
PTT Override Immunity Time	10
Secondary Keying Action	NONE
Redundant Priority	0

From the **Endpoint Configuration** page under the **Drivers** section, select “SIP” from the drop down menu and click on **Add driver** (not shown). Screen below shows the SIP driver added to the endpoint built above.



**Recorder Options**

Endpoint Audio Is Recorded NO

**Drivers**

Select Add driver

1 - SIP

## 6.2. Configuring the SIP driver on VPGate

This section shows how to configure the SIP driver that was added to the SIP endpoint discussed in the section above.

Click on **1-SIP** shown in the above screen to open the SIP driver configuration page as shown in the three screens below.

During compliance testing the following values were selected:

- **Username:** 58124; this is the DN configured as a third party SIP endpoint on Communication Server 100.
- **SIP Server Address/Domain Name:** sipl.com; this is set to a DNS resolvable name for the registrar or an dot-notation IP address (if the registrar will accept registrations to the specific IP address). Communication Server 1000 SIP Line gateway domain name is sipl.com.
- **SIP Server Port:** 0; this is set to 0 which allows the driver to use the default port of 5060.
- **Authentication Username:** Blank; may be left blank if the authentication username is the same as entered into the Username field. If different from the Username field, enter the username required for authentication.
- **Authentication Password:** 1234; this is the same as the password configured on the Communication Server 1000 SIP endpoint configuration.
- **Register with SIP Server?:** Yes; this is set to "Yes" to allow the endpoint to register to the Communication Server 1000.
- **Register Refresh Time:** 70; this is the default value. If a registration fails to the Communication Server 1000, the Scout system will turn the associated extension on the screen black to notify the dispatcher that the line has failed. Extending the duration in this field will lengthen the amount of time it takes to system to determine a failure.
- **Outbound Proxy Address:** 10.10.97.133; this is the node IP address of the SIP Line gateway configured on Communication Server 1000. If a proxy is used between the VPGate and the Communication Server 1000 registrar, populate this field with the address of the proxy. This field may also be used if the SIP Server Address/Domain name field needs to have a name, but is not DNS resolvable. In this case, the name would be

placed in the SIP Server Address/Domain name field, and the actual IP address of the registrar would be placed into this field.

- **Outbound Proxy Port:** 0; this is left at 0 to allow the system to use the default port of 5060.
- **Enable Secondary Identity?:** No; this field is only used for connecting to redundant registrars.
- **CODEC Configuration:** Only G.711 and G.729A CODECS should be enabled when connecting with a Communication Server 1000. Telephone Events may be left on, and all fields for the various CODECS should be left at their default values.
- **Dial Configuration:** All fields may be left at their default value. For Blind Transfer and Attended Transfer, make sure the **Enable Call Transfer** field is set to "YES".
- **Ring Down:** this field should be left to "NONE" to allow standard registration to the Communication Server 1000.
- **Quality of Service Priorities:** Allows the maintainer to override the default DIFFSERV values for both SIP Signaling and RTP/RTCP packets.
- **Advanced Network Configuration:** All fields should be left to their default values.
- **Advanced Configuration:** All fields should be left to their default values except for **RTCP Enabled** and **Enable Inactive Hold of Transfers**. These fields should be set to "YES". The latter field affects Transfer feature in Communication Server 1000 if not enabled.

## VPGate Configuration

### VPGate2

VPGate Endpoints Advanced Diagnostics Help

Endpoint Summary Input Alerts

Endpoint Summary > sip101 > SIP

#### SIP

Update Defaults

##### SIP Configuration

Driver Processing Order 1

##### Primary Identity

Display Name Private caller

Username 58124

SIP Server Address/Domain name sipl.com

SIP Server Port 0

Authentication Username

Authentication Password 1234

Register with SIP Server? YES

Register Refresh Time 70

Outbound Proxy Address 10.10.97.133

Outbound Proxy Port 0

Force Outbound Proxy NO

##### Secondary Identity

Enable Secondary Identity? NO

CODEC Configuration	
G.711 uLaw Enabled	YES
G.711 uLaw SDP Payload Type	0
G.711 uLaw SDP Description	PCMU
G.729A Enabled	YES
G.729A SDP Payload Type	18
G.729A SDP Description	G729
G.726 16kb Enabled	NO
G.726 32kb Enabled	NO
G.723.1 Enabled	NO
Telephone-Event (RFC2833/4733) Enabled	YES
Telephone-Event SDP Payload Type	101
Telephone-Event SDP Description	telephone-event
Dial Configuration	
Hook Flash Enabled	YES
Hook Flash Duration	500
Digit Time	100
Interdigit Time	100
Digit Level	-10
Dial Address/Domain Name	
Dial Port	0
Enable Call Transfer	YES

Enable Call Transfer	YES
Ring Down	
Ring Down Type	NONE
Quality of Service Priorities	
RTP/RTCP Packets	B8
SIP Packets	68
Advanced Network Configuration	
Local RTP Port	0
Local SIP Port	0
SIP Transport	UDP ONLY
UDP Keepalive Interval	0
Advanced Configuration	
RTCP Enabled	YES
Stale RTP Session Time	0
Stale Call Time	180
SIP Session Timer Enabled	NO
Peak Audio Level	0
Enable Auto Answer	NO
Enable Inactive Hold for Transfers	YES
SIP Stack Logging Level	NONE
SIP Stack Log File Size	5
Driver Logging Level	INFO

## 7. Verification Steps

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

### Step1

Verify that the Scout console registers successfully with the Communication Server 1000 SIP Line Gateway server by using the Communication Server 1000 Linux command line.

Log in to the SIP Line Gateway server using an administrator account (not shown).

Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked.

```
slgSetShowByUID 58124

=== VTRK ===
UserID      AuthId      TN      Clients  Calls  SetHandle  Pos
ID      SIPL Type
-----
---
          58124      58124      100-01-11-13      1      0      0x8ba0b00
SIP Lines
  StatusFlags = Registered Controlled KeyMapDwld SSD
  FeatureMask =
  CallProcStatus = 0

  Current Client = 0, Total Clients = 1

  == Client 0 ==
  IPv4:Port:Trans = 10.10.97.225:57967:udp
  Type            = SIP3
  UserAgent       = VPGate - SIP Driver
  x-nt-guid       = 9de0f44751d4ca885af2ecf60fdc383e
  RegDescrip      =
  RegStatus       = 1
  PbxReason       = OK
  SipCode         = 200
  hTransc         = (nil)
  Expire          = 18000
  Nonce           = fc053310c696685b548de8f063b8f80d
  NonceCount      = 2
  hTimer          = 0x8a836e0
  TimeRemain      = 17825
  Stale           = 0
  Outbound        = 0
  ClientGUID      = 0
  MSec CLS        = MSNV (MSEC-Never)
  Contact         =
sip:58124@10.10.97.225:57967;rinstance=c15592063302794b
  KeyNum          = 255
  AutoAnswer      = NO

  Key  Func  Lamp  Label
  0    3     0    58124
  1    126   0    2758124
  2    21    0
```



3	29	0
4	22	0
5	23	0
7	21	0
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

```

== Subscription Info ==
Subscription Event = None
Subscription Handle = (nil)
SubscribeFlag = 0

```

## Step 2

Place a call from and to Scout console 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 Scout to make sure that all SIP request/response messages are correct.

## 8. Conclusion

These Application Notes illustrate the procedures necessary for configuring the Avtec Scout Console to interoperate with the Avaya Communication Server1000. All feature functionality test cases described in **Section 2.1** passed with the observation noted in **Section 2.2**.

## 9. Additional References

This section references the product documentation relevant to these Application Notes. All Avaya documents can be found at <http://support.avaya.com>.

- *Software Input Output Reference — Administration Avaya Communication Server 1000 7.5 NN43001-611, Standard 05.13 September 2012.*
- *SIP Line Fundamentals Avaya Communication Server 1000 7.5 NN43001-508, 03.10 August 2012.*
- *Element Manager System Reference – Administration Avaya Communication Server 1000 7.5 NN43001-632, Standard 05.15 August 2012.*
- *Unified Communications Management Common Services Fundamentals Avaya Communication Server 1000 7.5 NN43001-116, 05.17 January 2012.*
- *ISDN Primary Rate Interface Installation and Commissioning Avaya Communication Server 1000 7.5 NN43001-301, 05.08 February 2012.*

Product documentation for Avtec Scout can be obtained by contacting Avtec technical support. Information for the same is mentioned in **Section 2.3**.

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