



Application Notes for G-Tek/AEi Communications SSP-2110-S Hospitality SIP Phone with Avaya Communication Server 1000E – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 and G-Tek/AEi Communications SSP-2110-S Hospitality SIP phone. G-Tek/AEi Communications SSP-2110-S Hospitality SIP phone registered as a SIP endpoint to Avaya Communication Server 1000 SIP Line gateway, and was able to place and receive calls from Avaya Communication Server 1000 non-SIP and SIP Line telephones.

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 detail configuration of Avaya Communication Server 1000 SIP Line Release 7.6 (hereafter referred to as CS1000) and G-Tek/AEi Communications SSP- 2110-S Hospitality SIP phone (hereafter referred as SSP-2110-S). SSP-2x10-S is part of a series of hospitality IP phones. All the applicable telephony and hospitality feature test cases of release 7.6 SIP Line were executed on SSP- 2110-S, where applicable, to verify the interoperability with CS 1000.

2 General Test Approach and Test Results

The general test approach was to have the SSP-2110-S register to the CS1000 SIP Line gateway successfully. From the CS1000 telephone calls were placed to and from SSP-2110-S as well as with simulated PSTN using various codec settings and exercising common PBX features such as busy, hold, DTMF, MWI and code negotiation. The telephony features were activated and deactivated using speed-dial buttons. Testing also included serviceability where the SSP-2110-S telephone Ethernet cable was disconnected.

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 SSP-2110-S was able to interoperate with CS1000, which included a SIP Line gateway. The following areas were covered:

- Registration of the SSP-2110-S to the CS1000 SIP Line gateway and session refresh interval.
- Voice call establishment of SSP-2110-S with Avaya CS1000 SIP and Avaya UNSTim telephones, and PSTN telephones, to ensure two way speech paths.
- Telephony features: basic calls inbound and outbound, call display, mute/un-mute, music on hold, DTMF transmission, message waiting indicator (MWI), speed dial, group call pickup, call waiting, ring again busy/no answer and feature access code dialing.
- Hospitality features including automatic wakeup, housekeeping status update, make set busy, voicemail, and programmable buttons.
- Telephony features including drop, abort, call waiting, hold/un-hold, busy, forward, transfer, attended conference, redial, and authorization code.
- Codec negotiation – G.711a-law and G.711u-law.
- Proper system recovery after a SSP-2110-S telephone restart and loss of IP connection.

2.2 Test Results

The objectives outlined in **Section 2.1** were verified and passed successfully. However the following observations were noted during the compliance testing:

- G.729 codec is not supported.
- Blind/attended transfer is not supported.
- Blind conference is not supported.
- Call park is not supported.
- Outgoing call screening (authorization code) test case is not tested with the following reason: Able to verify this feature on UNStim phone only, it does not work on SIP devices (Avaya and G-Tek/AEi Communications SIP endpoints). A CS1000 support ticket was opened to further investigate.

2.3 Support

For technical support on SSP-2110-S, please contact G-Tek/AEi Communications technical support team:

- **Telephone:** +1-650-552-9416
- **Email:** techsupport@aeicomcommunications.com
- **Website:** www.aeicomcommunications.com

3 Reference Configuration

The diagram below illustrates an enterprise site with an Avaya SIP-based network, including System Manager, Session Manager, Messaging, CS1000 and Avaya SIP and UNISlim endpoints. The enterprise site also contains two SSP-2110-S Phones used in the compliance testing. The SSP-2110-S phones are registered with SIP Line gateway and are configured as SIP endpoint users.

In the compliance testing, two SSP-2110-S phones were used for testing shown in the table below.

Device Type	Extension
SSP-2110-S	54504 (TN: 104 0 1 4)
	54332 (TN: 104 0 0 4)

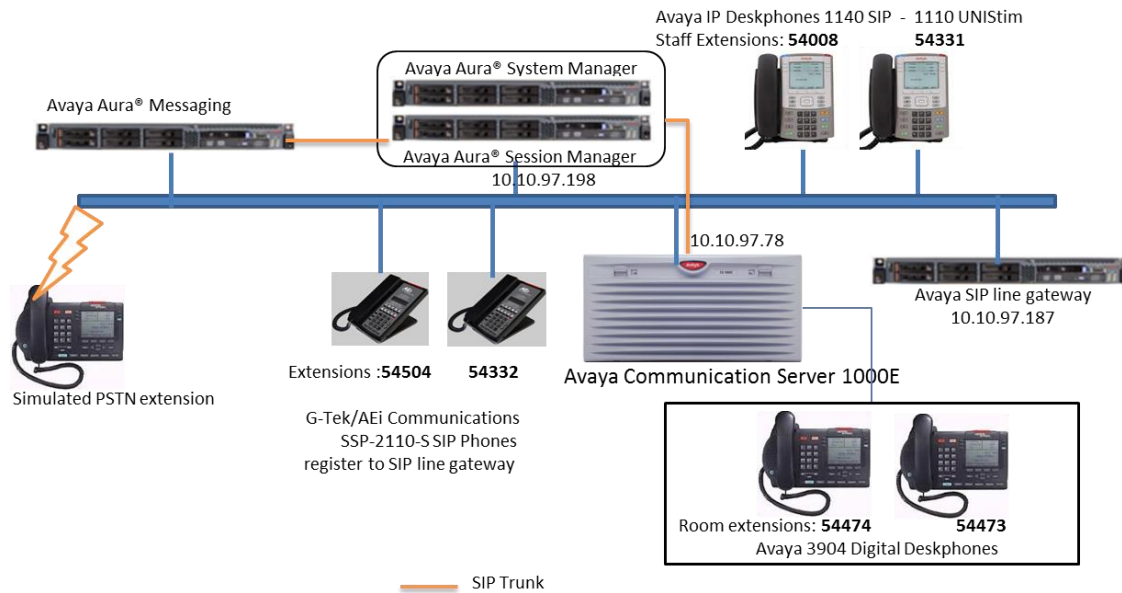


Figure 1: Network Configuration Diagram

4 Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment/Software	Release/Version
Avaya Communication Server 1000E	Call Server (CPPM): 7.65.16.22 SP5 Signaling Server (CPPM): 7.65.16.22 SP5 SIP Line gateway: 7.65.16.22 SP5
Avaya Aura® System Manager	R6.3.10
Avaya Aura® Session Manager	R6.3.10
Avaya Aura® Messaging	06.3-03.0.124.0-315
G-Tek/AEi Communications SSP-2110-S SIP Phone	190220.16.2.01D8

5 Configure Avaya Communication Server 1000E

This section shown information of SIP Line Node used during compliance test on CS1000 Element Manager and describes the steps to configure an Avaya CS1000 SIP Line user using the CS1000 command line interface (CLI). For detailed information on how to configure and administer the CS1000 SIP Line, please refer to **Section 9**.

This document assumes that CS1000 SIP Line configuration had been setup and operational.

5.1 Administer SIP Line Node

It is assumed that SIP Line Node has been setup and in operational state. This section shows step to view detail of SIP Line Node 512 on CS1000 Element Manager used during compliance test.

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 appropriate username/password (not shown). On the UCM home page, under the **Element Name** column, click on the Element Manager name of CS 1000 system in this case **EM on sip175**.

The screenshot shows the Avaya Unified Communications Management interface. The left sidebar contains a navigation menu with categories like Network, CS 1000 Services, User Services, Security, and Tools. The main content area displays the 'Elements' page. At the top, it shows 'Host Name: sip175.bvwdev.com', 'Software Version: 02.30.0093.00(6695)', and 'User Name admin'. Below this is a search bar with 'Search' and 'Reset' buttons. A table lists several elements:

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on sip175	CS1000	7.6	10.97.78	New element.
<input type="checkbox"/>	cppm3.bvwdev.com (member)	Linux Base	7.6	10.97.150	Base OS element.
<input type="checkbox"/>	sip175.bvwdev.com (primary)	Linux Base	7.6	10.97.136	Base OS element.
<input type="checkbox"/>	10.97.79	Media Gateway Controller	7.6	10.97.79	New element.

On CS1000 Element Manager page, navigate to **System → IP Network → Nodes: Server, Media Cards**. Click on selected Node, **512**.

The screenshot shows the CS1000 Element Manager interface. The left sidebar contains a navigation menu with categories like UCM Network Services, Home, Links, System, and IP Network. The main content area displays the 'IP Telephony Nodes' page. At the top, it shows 'Managing: 10.97.78' and 'Username: admin'. Below this is a breadcrumb trail: 'System → IP Network → IP Telephony Nodes'. The page title is 'IP Telephony Nodes' and it includes a sub-header: 'Click the Node ID to view or edit its properties.' Below this is a search bar with 'Add...', 'Import...', 'Export...', and 'Delete' buttons. A table lists several nodes:

<input type="checkbox"/>	Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
<input type="checkbox"/>	510	1	LTPS, Gateway (SIPGw)	-	135.10.97.149	-	Synchronized
<input type="checkbox"/>	512	1	SIP Line, LTPS, Presence Publisher	-	135.10.97.187	-	Synchronized

At the bottom, there are checkboxes for 'Show: Nodes', 'Component servers and cards', and 'IPv6 address'.

On **Node Details** page of Node 512, take note of **Node IPv4 address** 10.10.97.187 in **Telephony LAN (TLAN)** section. This IP is needed for SSP-2110-S to register to SIP Line in **Section 6.3**. Click on **SIP Line** link to open SIP Line configuration detail of **512**

AVAYA CS1000 Element Manager

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

Node Details (ID: 512 - SIP Line, LTPS, Presence Publisher)

Node ID: * (0-9999)

Call server IP address: *

TLAN address type: IPv4 only
 IPv4 and IPv6

Embedded LAN (ELAN)

Gateway IP address: *

Subnet mask: *

Telephony LAN (TLAN)

Node IPv4 address: *

Subnet mask: *

Node IPv6 address:

IP Telephony Node Properties

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

Applications (click to edit configuration)

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

* Required Value.

On **SIP Line Configuration Details** page, take note of **SLG Local Sip port** 5060, this port number is needed for SSP-2110-S to register to SIP Line in **Section 6.3**.

AVAYA CS1000 Element Manager

Managing: 10.10.97.78 Username: admin
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: *

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:

Monitor addresses:

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.

5.2 Create a SIP Line User

To create a SIP Line user 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 at default values:

```
ld 20
REQ: prt
TYPE: UEXT - Universal extension type for SIP Line phone

TN 104 0 1 4
DES AEi
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL - Universal extension type is SIP Line
MCCL YES
SIPN 0 - For SIP phone third party, enter 0 in this field
SIP3 1 - For SIP phone third party, enter 1 in this field
FMCL 0
TLSV 0
SIPU 54504 - SIP phone user name
NDID 512 - Node ID of SIP Line
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
SCPW 1234 - Password of SIP Line user. Need to register SSP2110-S to SIP Line Node
VSIT NO
KEY 00 SCR 54356 0 MARP
    CPND
        CPND_LANG ROMAN
        NAME Room 504
        XPLN 13
        DISPLAY_FMT FIRST, LAST
    01 HOT U 2654504 MARP 0 - Key 1 hot U with prefix + DN
```

Repeat same step for other phone with extension **54332**

6 Configure G-Tek/AEi Communications SSP-2110-S

This section provides the procedures for configuring SSP-2110-S. The procedures include the following areas:

- Access Web Interface
- Configure LAN Port Settings
- Configure SIP Accounts
- Configure Codec Settings
- Configure Other Settings

6.1 Access Web Interface

Enter `http://<ip-addr>:8000/`, where `<ip-addr>` is the IP address of the SSP-2110-S phone, into the address bar of web browser and log in using a valid account. The **System Information** screen is displayed.

The screenshot shows a web browser window with the address bar containing `http://10.33.5.21:8000/mail`. The page title is "VOIP PHONE". The main content area is titled "Web Configuration" and features a "System Information" section. The "System Information" section contains a table with the following data:

Model Name:	VOIP
Serial number:	201212000541
Firmware Version:	190220.16.2.01D8

A sidebar on the left lists navigation options: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network.

6.2 Configure LAN Port Setting

On the left panel, select **Network** → **LAN Port Settings** and configure either as **DHCP Client** (default) or **Static IP** for the LAN connection. As DHCP Client, the LAN Port setting will be automatically populated. In the testing, DHCP Client was used.

LAN Port Settings

You could configure the LAN Port settings in this page.

LAN Port Setting	
IP Type:	<input type="radio"/> Static IP <input checked="" type="radio"/> DHCP Client
IP Address:	192.168.1.122
Netmask:	255.255.255.0
Gateway:	192.168.1.254
Primary DNS:	168.95.192.1
Secondary DNS:	168.95.1.1
Mac Address:	00:0e:43:d0:79:c0

VLAN Setting	
Activation:	<input type="radio"/> On <input checked="" type="radio"/> Off
Identifier:	1 (0~4094)
Priority:	0 (0~7)

6.3 Configure SIP Accounts

Select **SIP Accounts** from the left panel. Set the **Registration** radio button to **Enable**. Enter the IP address of SIP Line Node in **Section 5.1, in this case IP is 10.10.97.187**; since the SSP-2110-S register via SIP Line local SIP Port, enter port 5060 as shown below, for the **Registration Server** and **Proxy Server** fields. For the fields **Registration ID**, **Display Name**, **Authorization Name** and **Password**, enter the account details as shown below to match the user settings in CS1000 added in **Section 5.2**. Set **MWI** to **Disable**. Set the **Voice Mail** number to dial as 50000. This number is mapped to the message speed dial key on the base. Check the default **DTMF Type** settings is RFC2833. Click **Submit** to continue (not shown).

Phone Settings

System Settings

Global SIP Settings

SIP Accounts

Network

You could set information of service domains in this

SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	<input type="text" value="54504"/>
Display Name:	<input type="text" value="54504"/>
Authentication Name:	<input type="text" value="54504"/>
Password:	<input type="password" value="****"/>
Registration Server:	<input type="text" value="10.10.97.187:5060"/>
Proxy Server:	<input type="text" value="10.10.97.187:5060"/>
Realm Address:	<input type="text"/>
Voice Mail:	<input type="text" value="50000"/>
Expire Time:	<input type="text" value="600"/>
DTMF Type:	<input type="text" value="RFC2833"/>
MWI:	<input type="text" value="Disable"/>
DNSRV:	<input type="text" value="Disable"/>
Send KeepAlive:	<input type="text" value="Disable"/>
Send KeepAlive Type:	<input type="text" value="Dummy"/>
Send KeepAlive Interval:	<input type="text" value="240"/>
Status:	registered

6.4 Configure Codec Settings

Select **Global SIP Settings** → **Codec Settings** from the left panel. In the **Codec Priority** section, prioritize the audio codecs accordingly. Click **Submit** to continue.

Codec Settings
You could set the codec settings in this page.

Codec Priority

First Priority:

RTP Packet Length

G.711 Frame Size:

6.5 Configure Other Settings

Select **Global SIP Settings** → **Other Settings** from the left panel. Check that **RFC2833 Payload Type** is set to 101. Enter the appropriate name for the **Hotel**, this name will be displayed on the phone. Click **Submit** to continue.

Other Settings
You could set the other settings in this page.

Signaling Precedence(ToS):

Voice Precedence(ToS):

RFC2833 Payload Type: (96~127)

Hotel:

Room:

Log Server:

Tiger Server:

User Agent:

Call Waiting:

7 Verification Steps

This section provides the test that can be performed to verify proper configuration of CS1000 and SSP-2110-S.

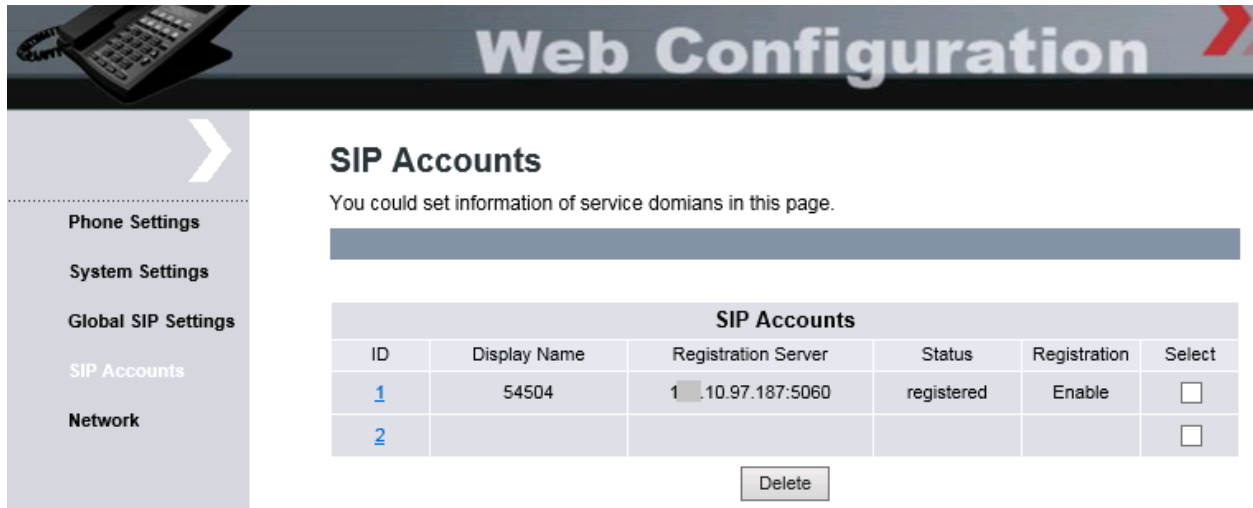
7.1 Verify Avaya Communication Server 1000E

To verify the registration status of SSP-2110-S on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 32, then issue the command “stat [TN]” where TN is terminal number of SSP-2110-S phone on CS1000, during compliance test TN for 2 SSP-2110-S phones are 104 0 04 and 104 0 1 4. Verify the status of each SSP-2110-S phone is registered as shown below:

```
>ld 32
.stat 104 0 0 4
IDLE REGISTERED 00
.stat 104 0 1 4
IDLE REGISTERED 00
```


7.2 Verify G-Tek/AEi Communications SSP-2110-S

Log on to the SSP-2110-S web interface by enter `http://<ip-addr>:8000/`, where `<ip-addr>` is the IP address of the SSP-2110-S phone, into the address bar of web browser and log in using a valid account. Click on **SIP Accounts** link on the left menu. Verify the status of extension **54504** is **registered** to CS1000.



The screenshot shows the 'Web Configuration' interface. On the left is a navigation menu with options: Phone Settings, System Settings, Global SIP Settings, SIP Accounts (highlighted), and Network. The main content area is titled 'SIP Accounts' and includes a sub-header 'You could set information of service domians in this page.' Below this is a table with the following data:

SIP Accounts					
ID	Display Name	Registration Server	Status	Registration	Select
1	54504	10.97.187:5060	registered	Enable	<input type="checkbox"/>
2					<input type="checkbox"/>

Below the table is a 'Delete' button.

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 SSP-2110-S firmware 190220.16.2.01D8 is considered to be in compliance with Avaya Communication Server 1000E SIP Line System Release 7.6.

9 Additional References

Product latest documentation for Avaya Communication Server 1000E products may be found at: <https://support.avaya.com/css/Products/>

- Avaya Communication Server 1000 Installation and Commissioning Release 7.6; NN43041-310.
- Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6; NN43001-508.
- Avaya Communication Server 1000 Element Manager System Reference – Administration; NN43001-632.
- Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals, Release 7.6; NN43001-509.
- Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals, Release 7.6; NN43001-116.

Product documentation for the SSP-2110-S can be obtain from G-Tek/AEi Communications:

- SSP-2210-S SSP-2110-S datasheet
- SSP-2210-S SSP-2110-S QIG Version 1.00

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