Avaya Solution & Interoperability Test Lab

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@aeicommunications.com
- **Website:** www.aeicommunications.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 UNIStim 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)

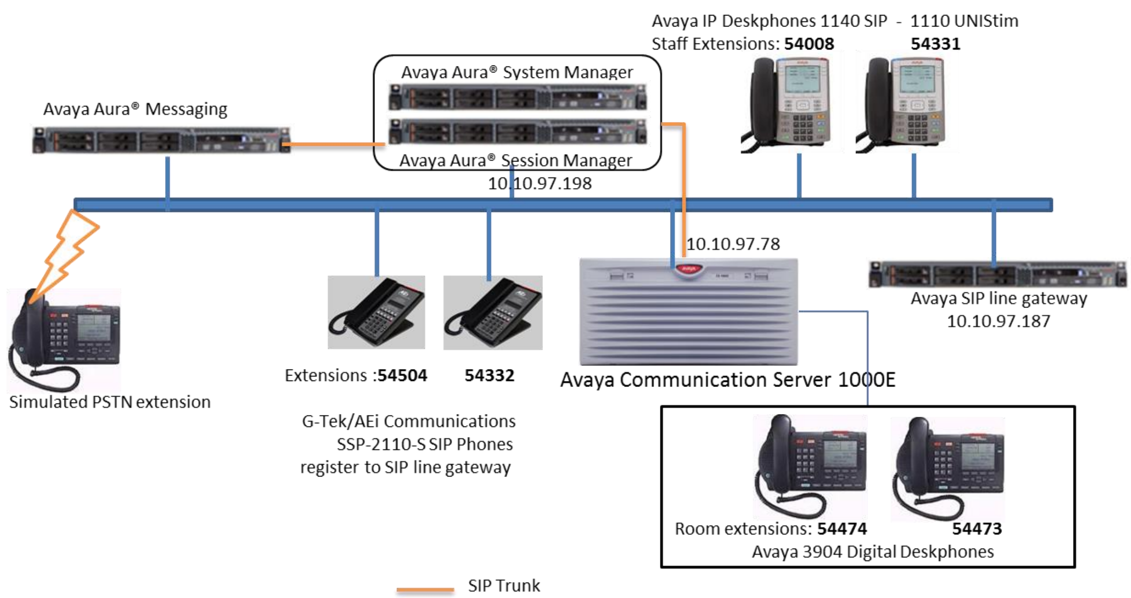


Figure 1: Network Configuration Diagram

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

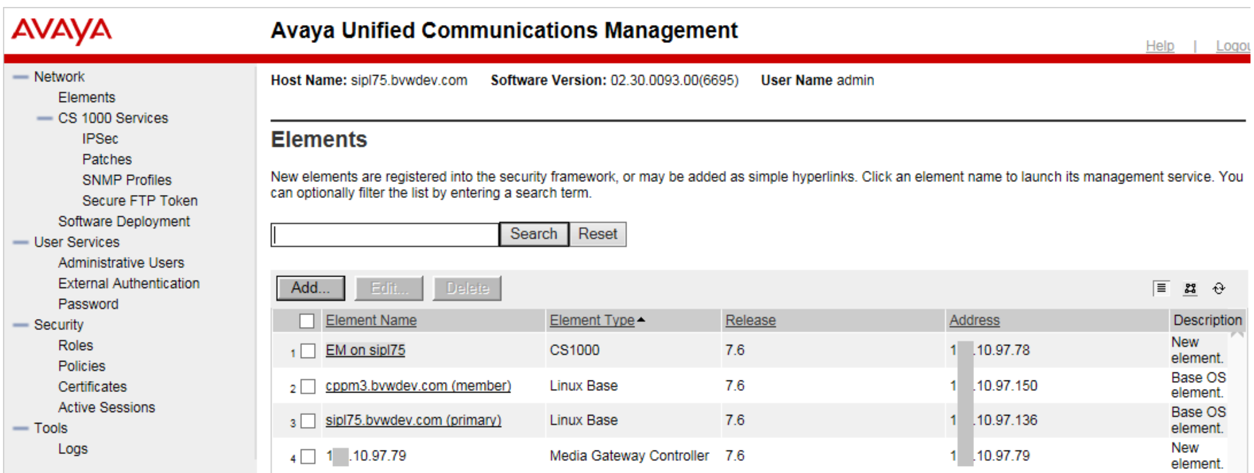
This section shown information of SIP Line Node used during compliance test on CS1000 Element Manager and describes the steps to configure an Avaya CS1000 SIP Line user using the CS1000 command line interface (CLI). For detailed information on how to configure and administer the CS1000 SIP Line, please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

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

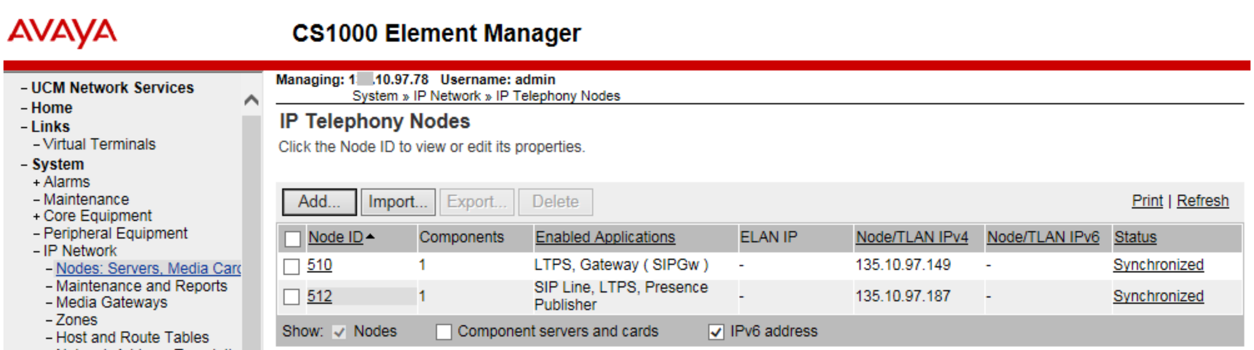
5.1 Administer SIP Line Node

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.

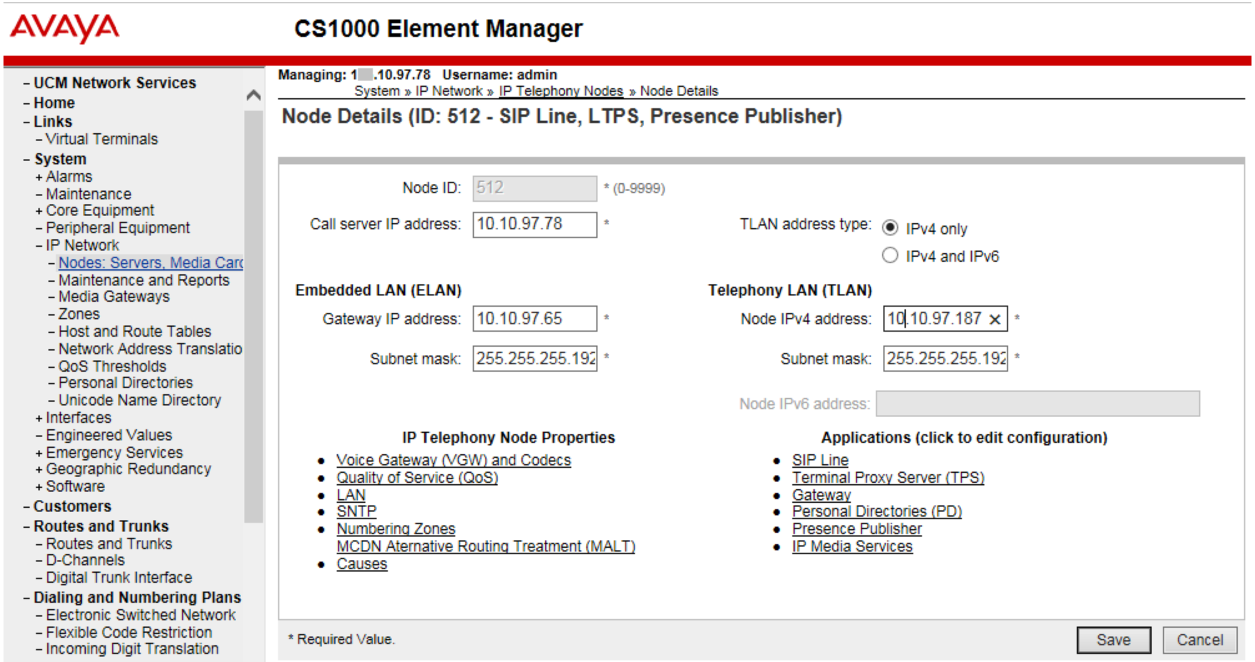
Use the Microsoft Internet Explorer browser to launch CS 1000 UCM web portal at http://<IP Address or FQDN> where <IP address or FQDN> is the UCM Framework IP address or FQDN for UCM server. Log in with appropriate username/password (not shown). On the UCM home page, under the **Element Name** column, click on the Element Manager name of CS 1000 system in this case **EM on sipl75**.



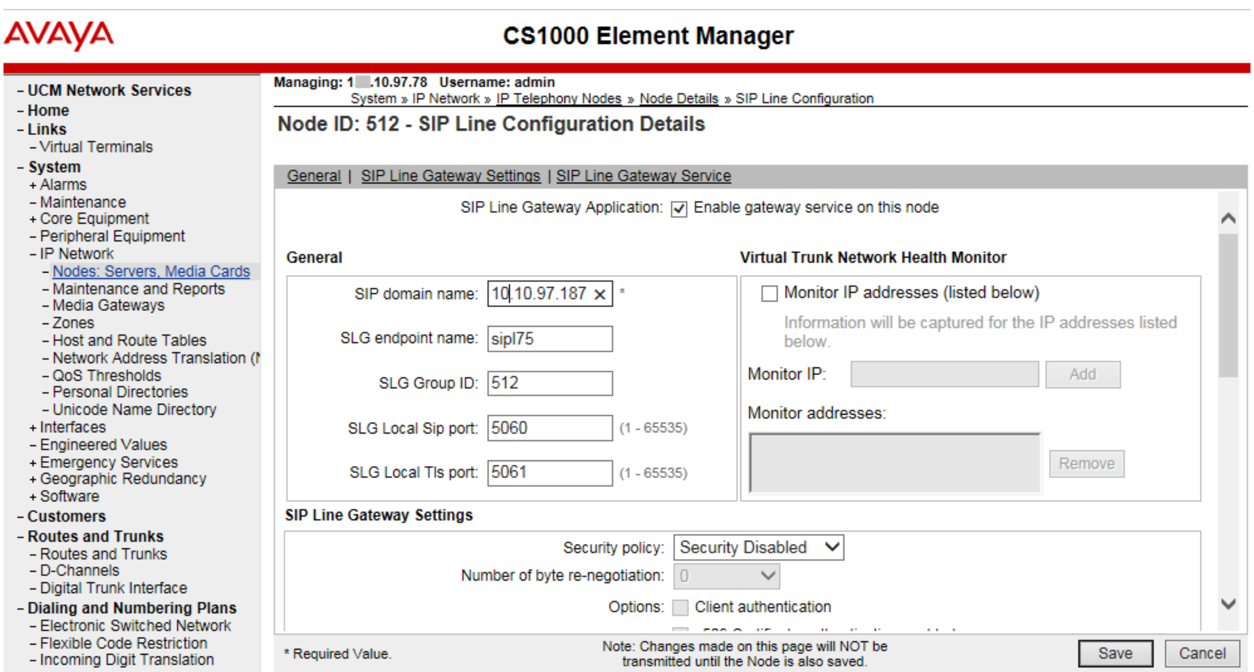
On **CS1000 Element Manager** page, navigate to **System → IP Network → Nodes: Server, Media Cards**. Click on selected Node, **512**.



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**



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**.



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**

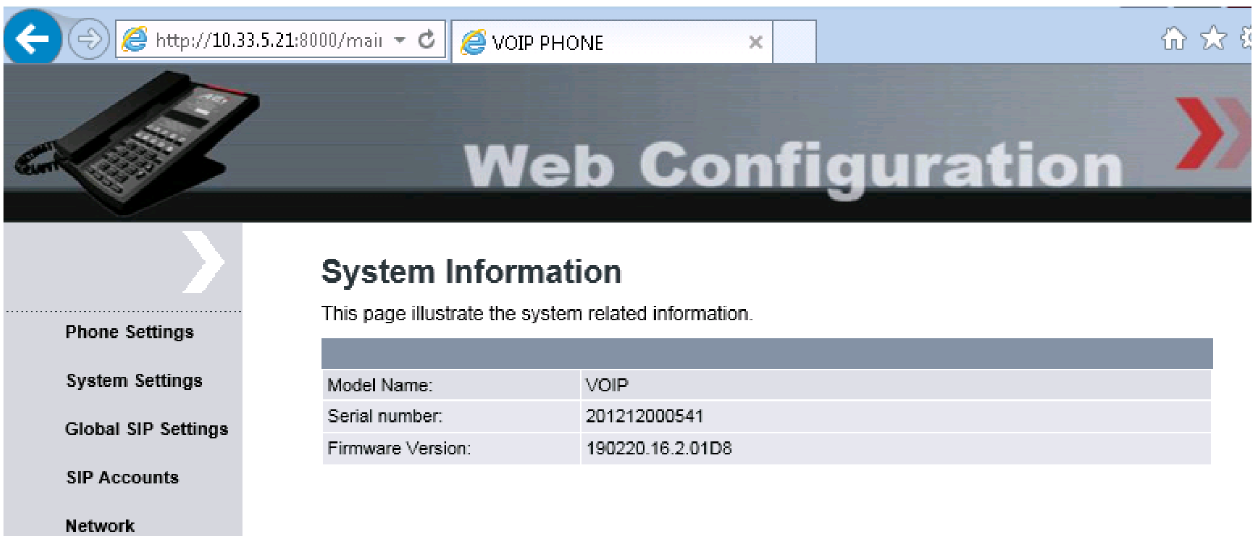
6 Configure G-Tek/AEi Communications SSP-2110-S

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

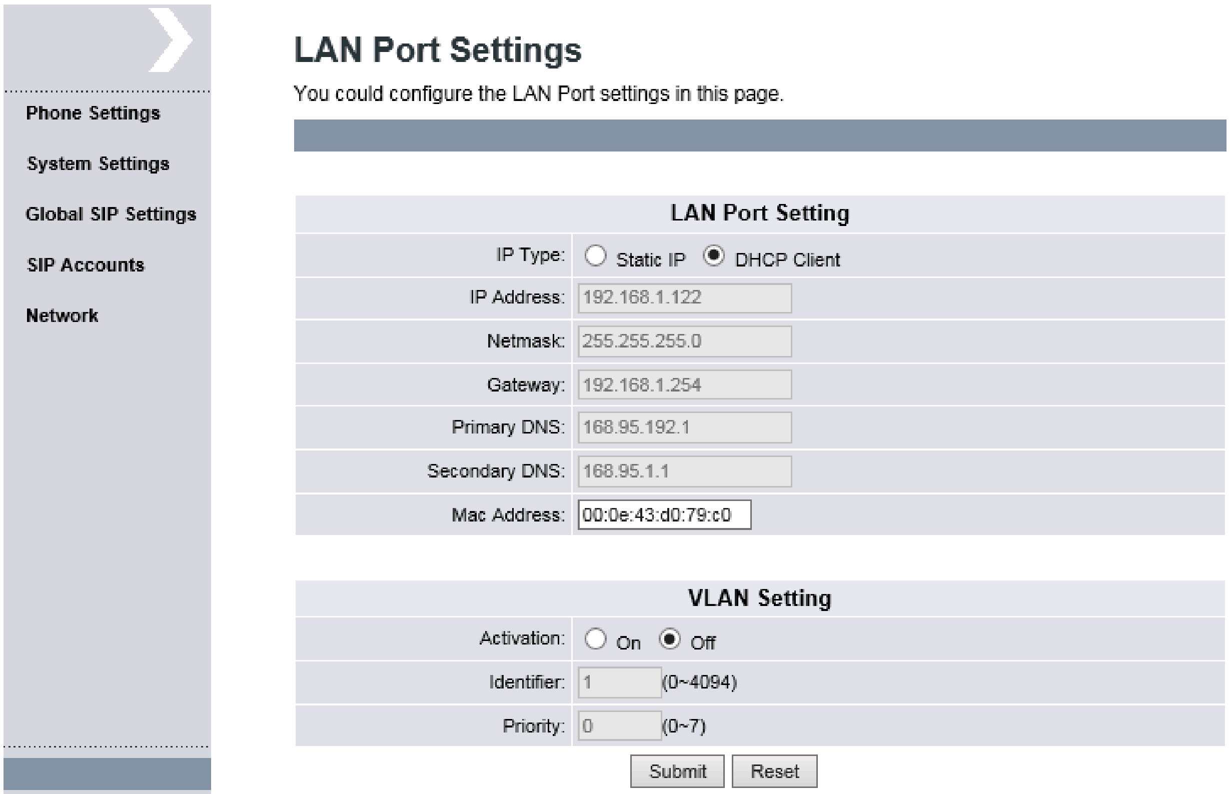
6.1 Access Web Interface

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.



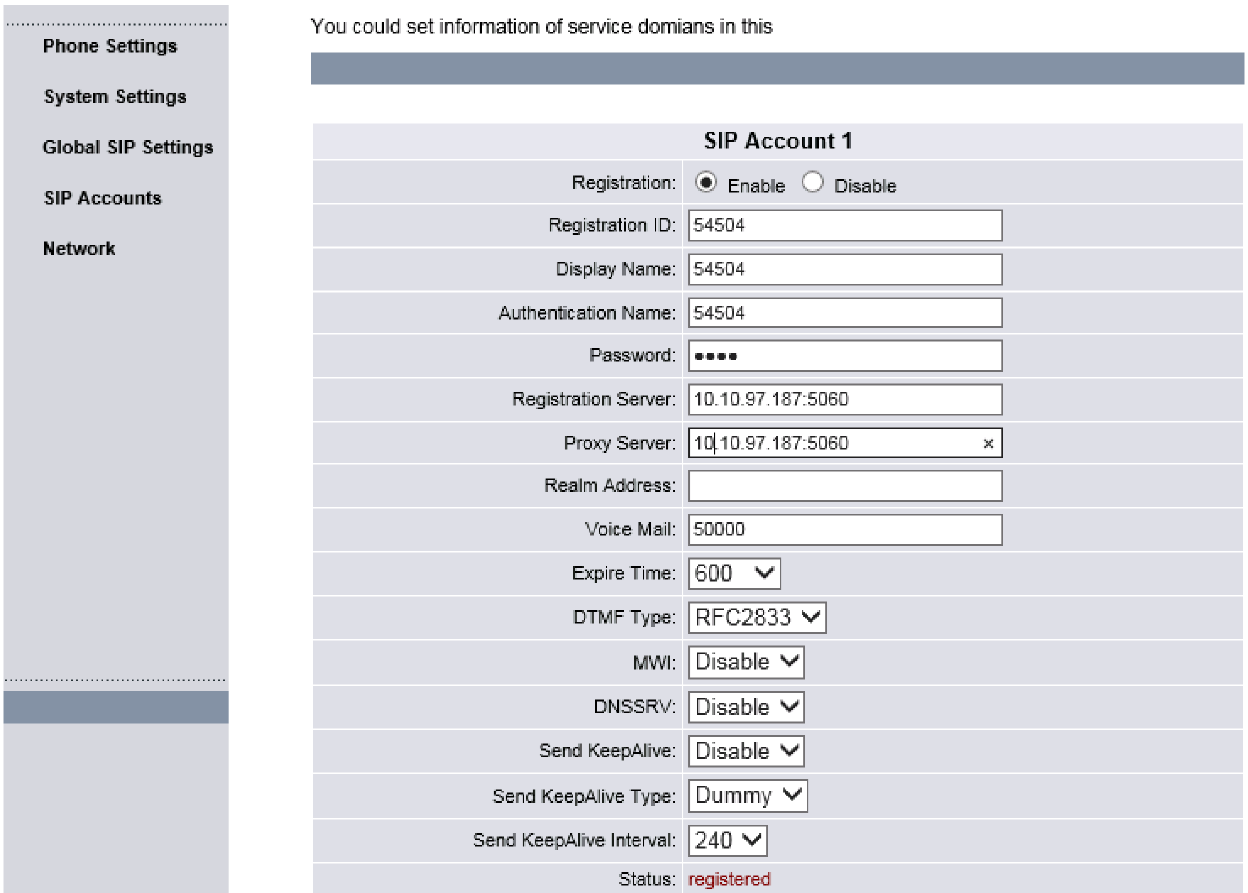
6.2 Configure LAN Port Setting

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.



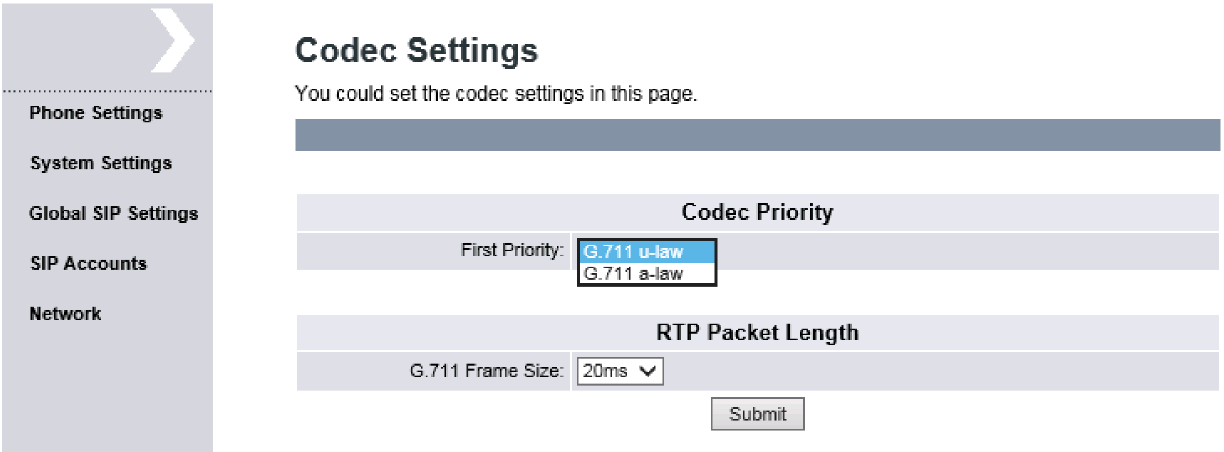
6.3 Configure SIP Accounts

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 **Section5.2**. 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).



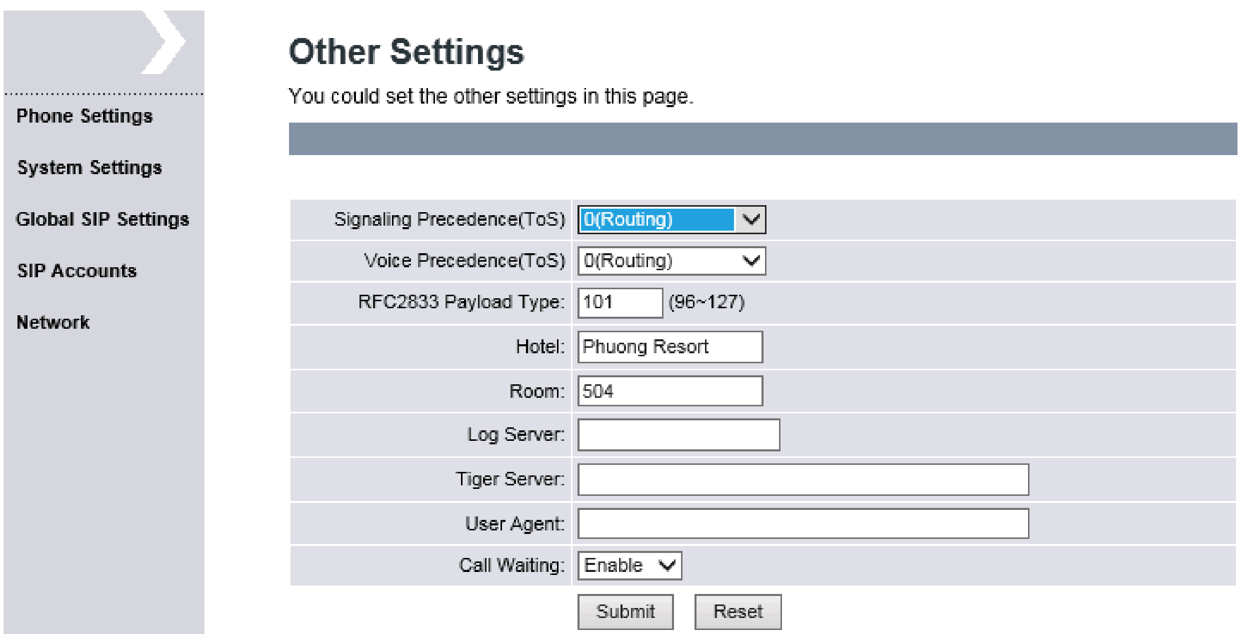
6.4 Configure Codec Settings

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



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.



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
```

To verify the status of SIP Line phone on SIP Line Signaling server, log in to the SIP Line Signaling server as administrator and issue **slgSetShowAll** command:

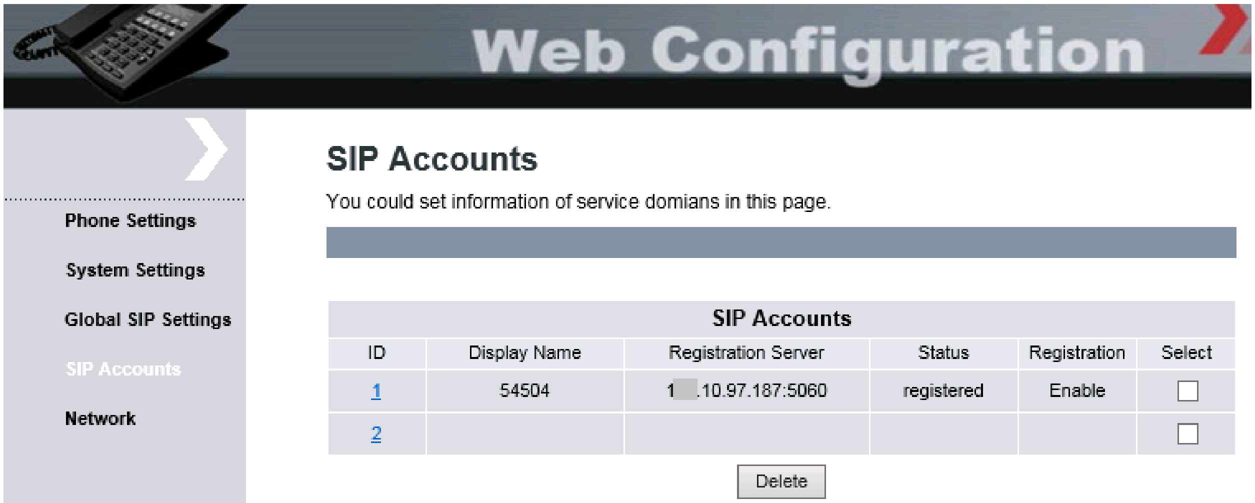
```
[admin@sip175 ~]$ slgSetShowAll

=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------   ---------
--------------- IPV4 Endpoints   --------------------------
          54353       54353     104-00-00-13        1      0  0xb2715aa0         SIP Lines
          54332       54332     104-00-00-04        1      0  0xb2724ec0         SIP Lines
          54504       54504     104-00-01-04        1      0  0x9cdd360         SIP Lines


Total User Registered = 3   V4 Registered = 3  V6 Registered = 0
```

To verify the detail information of SIP Line phone on SIP Line Signaling server, log in SIP Line Signaling server as administrator and issue **slgSetShowByUID** command for extension 54504 created in **Section 5.2**.

```
[admin@sip175 ~]$ slgSetShowByUID 54504

=== VTRK ===
UserID           AuthId      TN               Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  ---------------  -------  -----  ---------  -------   ---------
          54504       54504     104-00-01-04        1      0  0x9cdd360         SIP Lines
        StatusFlags = Registered Controlled KeyMapDwld SSD
        FeatureMask =
        CallProcStatus = 0

        Current Client = 0, Total Clients = 1

         == Client 0 ==
         IPv4:Port:Trans = 10.33.5.21:5060:udp
         Type            = Unknown
         UserAgent       = Gtek GE20X sipagent-MAC-000e43d079c0 V-190220.16.2.01D8-SIP
         x-nt-guid       = 53b1ab8a9a0b4407111ff46a7da2002b
         RegDescrip      =
         RegStatus       = 1
         PbxReason       = OK
         SipCode         = 200
         hTransc         = (nil)
         Expire          = 600
         Nonce           = 84ed00a8edbad913d86c84ce6b3be0dc
         NonceCount      = 2
         hTimer          = 0x9c6d3b0
         TimeRemain      = 171
         Stale           = 0
         Outbound        = 0
         ClientGUID      = 0
         MSec CLS        = MSNV (MSEC-Never)
         Contact         = sip:54504@10.33.5.21
         KeyNum          = 255
         AutoAnswer      = NO
```

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.



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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.