Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Algo 8188 SIP Ceiling Speaker Version 1.1.1 with Avaya Communication Server 1000 Release 7.6 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Algo 8188 SIP Ceiling Speaker to interoperate with Avaya Communication Server 1000. Algo 8188 SIP Ceiling Speaker is a SIP-based device that can register with Avaya Communication Server 1000 as two separate SIP endpoints, one for loud ringing and one for voice paging.

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 describe the configuration steps required for Algo 8188 SIP Ceiling Speaker to interoperate with Avaya Communication Server 1000. Algo 8188 SIP Ceiling Speaker is a SIP-based device that can register with Avaya Communication Server 1000 (CS 1000) SIP Line server as two separate SIP endpoints, one for loud ringing and one for voice paging.

For loud ringing, Algo 8188 SIP Ceiling Speaker can be configured to ring whenever the associated desk phone receives an incoming call. The loud ringing is useful for users that require louder ringing than what is available from the desk phone. The simultaneous ringing at the desk phone and Algo 8188 SIP Ceiling Speaker is accomplished via the Personal Call Assistant (PCA) feature.

For voice paging, Algo 8188 SIP Ceiling Speaker can auto-answer an incoming call and allow the caller to broadcast audio over the Algo 8188 SIP Ceiling Speaker.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually placed to the loud ringing and voice paging extensions, with call controls such as hold/resume, unattended, attended transfer and conference performed from the caller.

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 interoperability compliance test included feature and serviceability testing. The loud ringing feature testing included registration, internal and external caller, interactions with the voice paging extension, and interactions with desk phone features such as coverage, call forwarding, and do not disturb. The voice paging feature testing included registration, media shuffling, G.722, internal and external caller, interactions with the loud ringing extension, and interactions with caller actions such as drop, hold/reconnect, blind/attended transfer, and blind/attended conference.

The serviceability testing focused on verifying the ability of Algo 8188 SIP Audio Alerter to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the device.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All test cases are execute and passed.

2.3. Support

Technical support on Algo 8188 SIP Ceiling Speaker can be obtained through the following:

- Phone: + 1 604 454 3792
- Web: http://www.algosolutions.com/support
- Email: support@algosolutions.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between Algo 8188 SIP Ceiling Speaker and Avaya Communication Server 1000 and Avaya Aura® Session Manager. The Algo 8188 communicated with Avaya CS1000 through Avaya switch with Power over Ethernet (PoE) and registered with Avaya CS1000 SIP Line server as two separate SIP endpoints, and the extensions used for the testing: one for Voice Paging and one for Loud Ringer. The IP Office Solution that consists of a primary Server Edition in Virtual Environment and an expansion 500V2. The testing used Avaya Aura® Session Manager to route calls between Avaya CS1000 and Avaya Communication Manager via SIP trunk for test cases require external call from/to CS1000.

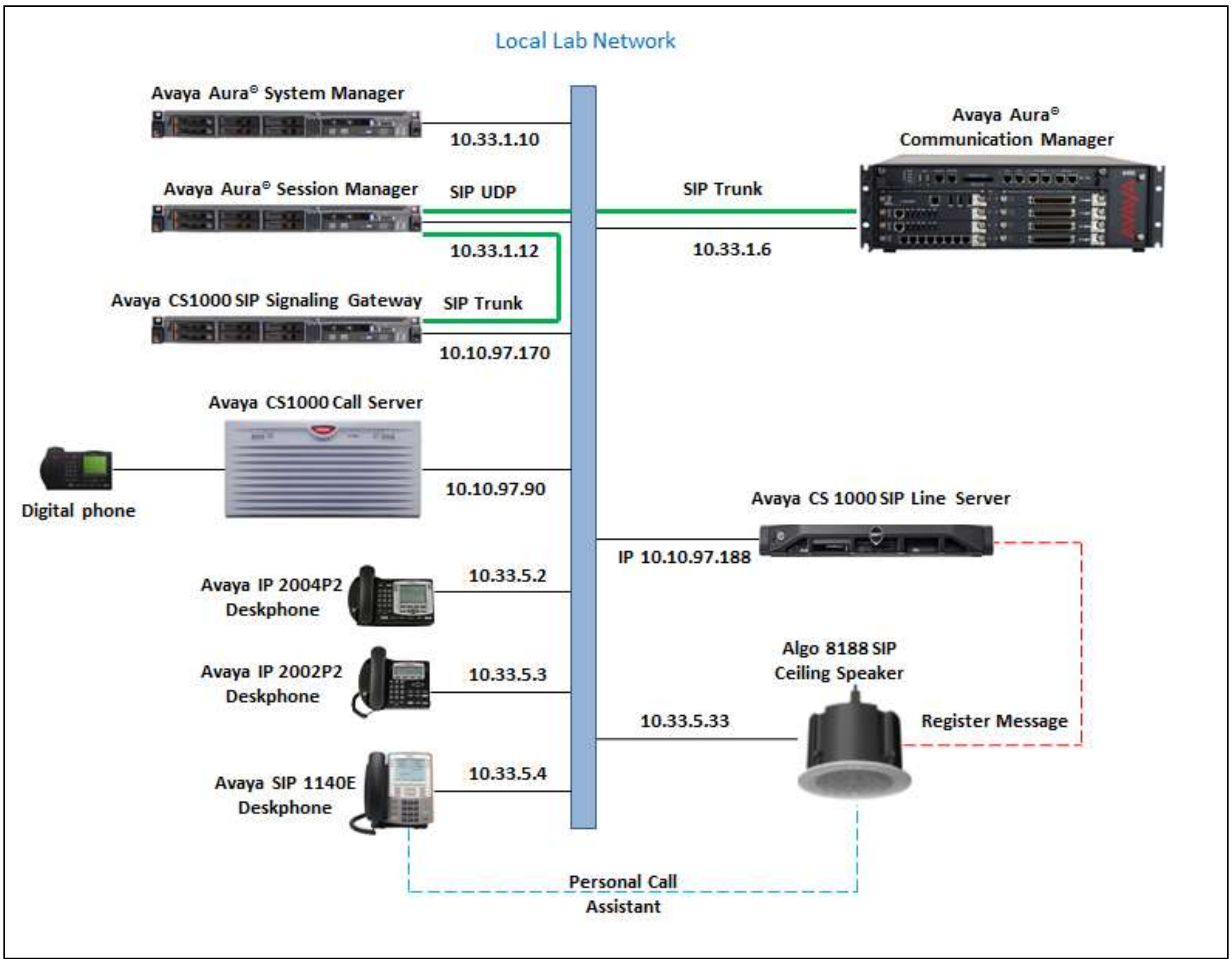


Figure 1: Test Configuration Diagram

4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® System Manager in Virtual Environment	7.0
Avaya Aura® Session Manager in Virtual Environment	7.0
Avaya Communication Server 1000 Call server running on CPPM	7.6 SP7
Avaya Communication Server 1000 SIP Signaling Server running on CPPM	7.6 SP7
Avaya Communication Server 1000 SIP Line Server running on IBM 3550 (COST)	7.6 SP7
Avaya Aura® Communication Manager in Virtual Environment	7.0
Avaya 2002P2 IP Deskphone	0604DCO
Avaya 1140E IP Deskphone	0625D8Q
Avaya 1140E SIP Phone	4.4.23
Avaya 3904 Digital Phone	Core 2.4, Flash 9.3
Avaya Analog Phone	-
Algo 8188 SIP Ceiling Speaker Firmware Base Version System Version	 1.1.1 r1.2 r1.2

5. Configure Avaya Communication Server 1000

This section provides the configuration for the CS 1000 SIP Line server and SIP user that Algo 8188 SIP phone used during the testing. The document assumes that the CS 1000 installation and configuration are already in place, this section only show the relevant and important configuration that used for Algo 8188. The procedures include the following areas:

- Verify CS1000 Prerequisite.
- Configure SIP User.
- Configure Personal Call Assistant.
- Administer SIP Line Server.

5.1. Verify CS1000 Prerequisite

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. Administer SIP User

Access to the overlay command in the CS 1000 call server, use the overlay command LD 20/11 to create a new terminal number for a SIP user. The screen below shows the previously configured SIP user that used by Algo 8188 ring, in the detail of the terminal number configuration the **SIP3** set to **1** as this is 3rd party SIP endpoint, SIP user (**SIPU**) set to **4694**, Node ID (**NDID**) set to **2005** the node ID will be mentioned in the next section, station control password (**SCPW**) set to **1234** this is the password of SIP user, and the extension is configured for this SIP user in the Key 0 which is **4694**.

```
TN   108 0 01 09  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 1
UXTY SIPL
MCCL YES
SIPN 0
SIP3 1
FMCL 0
TLSV 0
SIPU 4694
NDID 2005
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
MRT
ERL  0
ECL  0
VSIT NO
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
    MWD 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 RCBD
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXD ARHD CLTD ASCD
    CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3
MCBN
    FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
    MSNV FRA  PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
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 4694 0     MARP
      CPND
        CPND_LANG ROMAN
          NAME Algo 8188 Ring
          XPLN 14
          DISPLAY_FMT FIRST,LAST
    01 HOT U 1114694 MARP 0
    02
    03
    04
    05
    06
```

The second SIP user was created for Algo 8188 page, the screen blow shows the terminal number (TN) configuration for this SIP user.

```
TN   108 0 01 10  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 1
UXTY SIPL
MCCL YES
SIPN 0
```

```
SIP3 1
FMCL 0
TLSV 0
SIPU 4670
NDID 2005
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
MRT
ERL  0
ECL  0
VSIT NO
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
     MWD 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 RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXD ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3
MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
     MSNV FRA  PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
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 4670 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Algo 8188 Page
            XPLN 14
            DISPLAY_FMT FIRST,LAST
     01 HOT U 1114670 MARP 0
     02
     03
     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
     28
     29
     30
     31
DATE 16 DEC 2015
```

5.3. Administer Personal Call Assistant

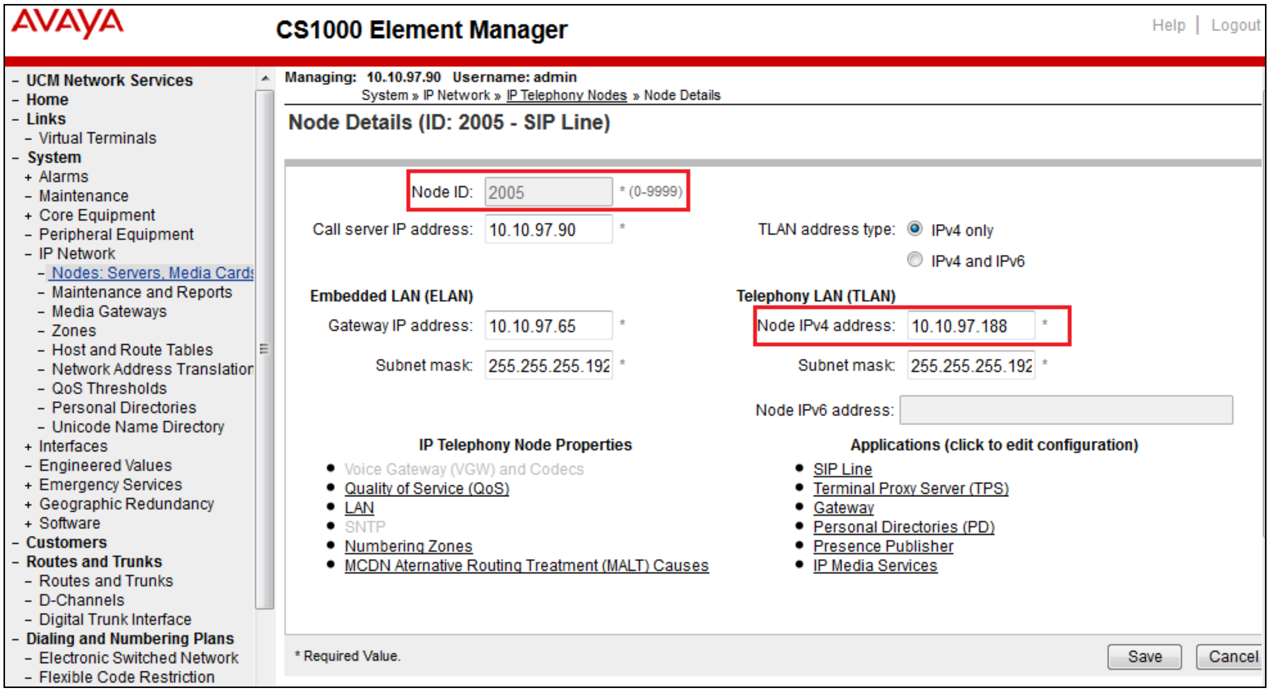
Using the overlay command LD 20 to create a personal call assistant (PCA) terminal number to associate a CS 1000 extension with the SIP user that is created in the **Section 5.2** above. The SIP user will ring when this extension is called. In this configuration, the extension is **4605** which is configured in the Key 0 of PCA and the Key 1 is pointed to the extension of the SIP user.

```
DES  PCA
TN   108 0 01 11  VIRTUAL
TYPE PCA
CDEN 8D
CTYP XDLC
CUST 1
NUID
NHTN
MRT
ERL  0
ECL  0
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
     MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDD
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXD ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3
MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
     MSNV FRA  PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
```

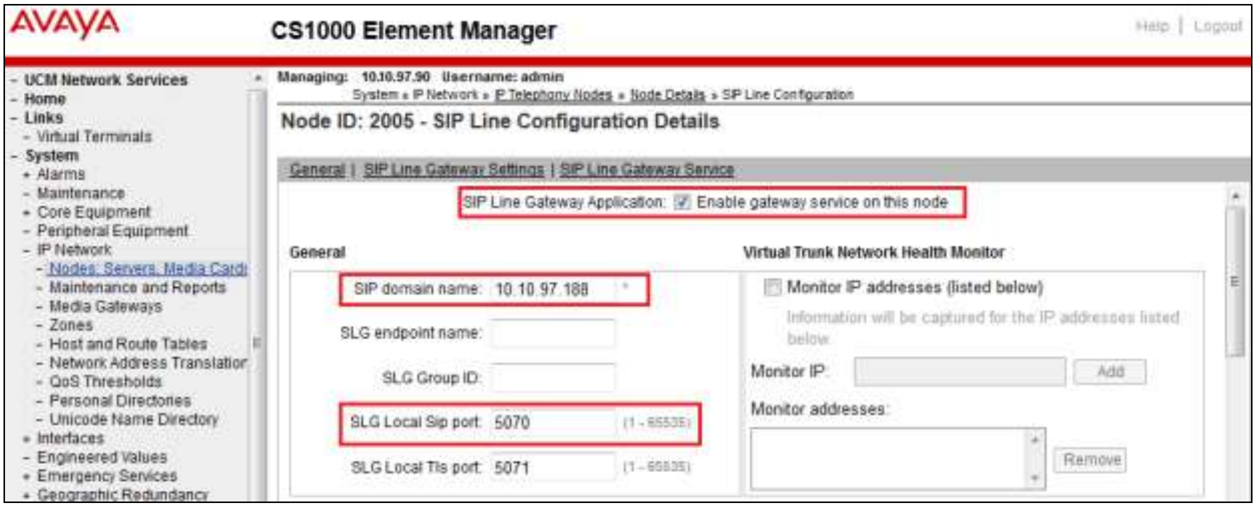
```
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 4605 0
        CPND
          CPND_LANG ROMAN
            NAME SCR 4605
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 HOT P 4 4694
     02
     03
     04
     05
     06
     07
```

5.4. Administer CS 1000 SIP Line Server

The CS 1000 SIP Line server can be accessed and configured via Element Manager, the screen below shows the **Node Details** configuration of the SIP Line server that was used for the testing. The Node ID is **2005** this node ID is matched with the node ID configured in the SIP user **4694** and **4670** in **Section 5.2** and the node IP address is **10.10.97.188**.



Click on the **SIP Line** link under the **Applications** as shown in the screen above to edit or display the detail configuration of the SIP Line. The screen below shows the **SIP Line Configuration Details** page; ensure the "**SIP Line Gateway Application**" check box is checked to enable gateway service on this node. In the **General** section, enter a domain in the **SIP domain name** field in the testing the SIP domain name used as "**10.10.97.188**" and the SLG Local Sip port set to **5070**.



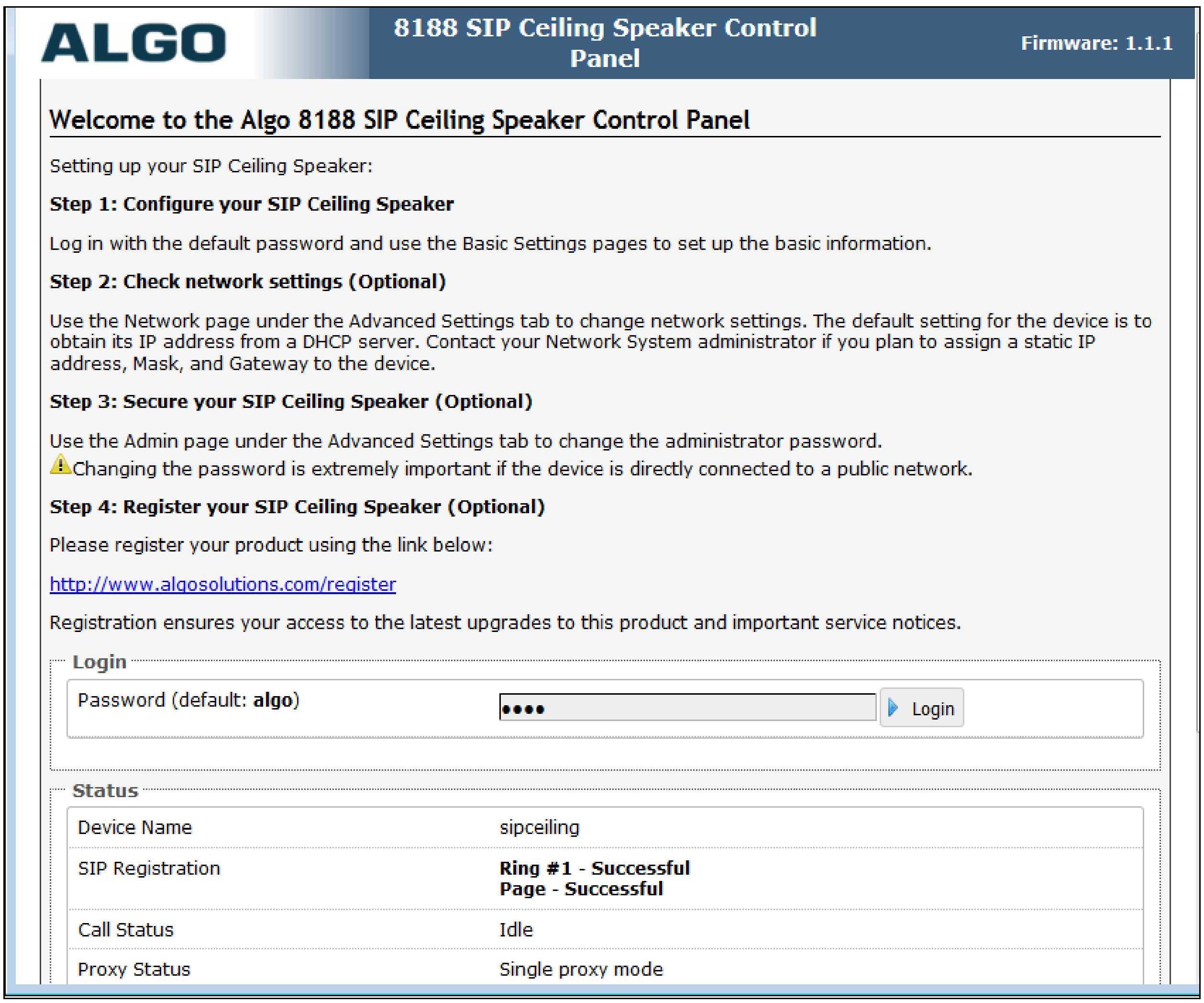
6. Configure 8188 SIP Ceiling Speaker

This section provides the procedures for configuring Algo 8188 SIP Ceiling Speaker. The procedures include the following areas:

- Launch web interface.
- Administer configuration.

6.1. Launch Web Interface

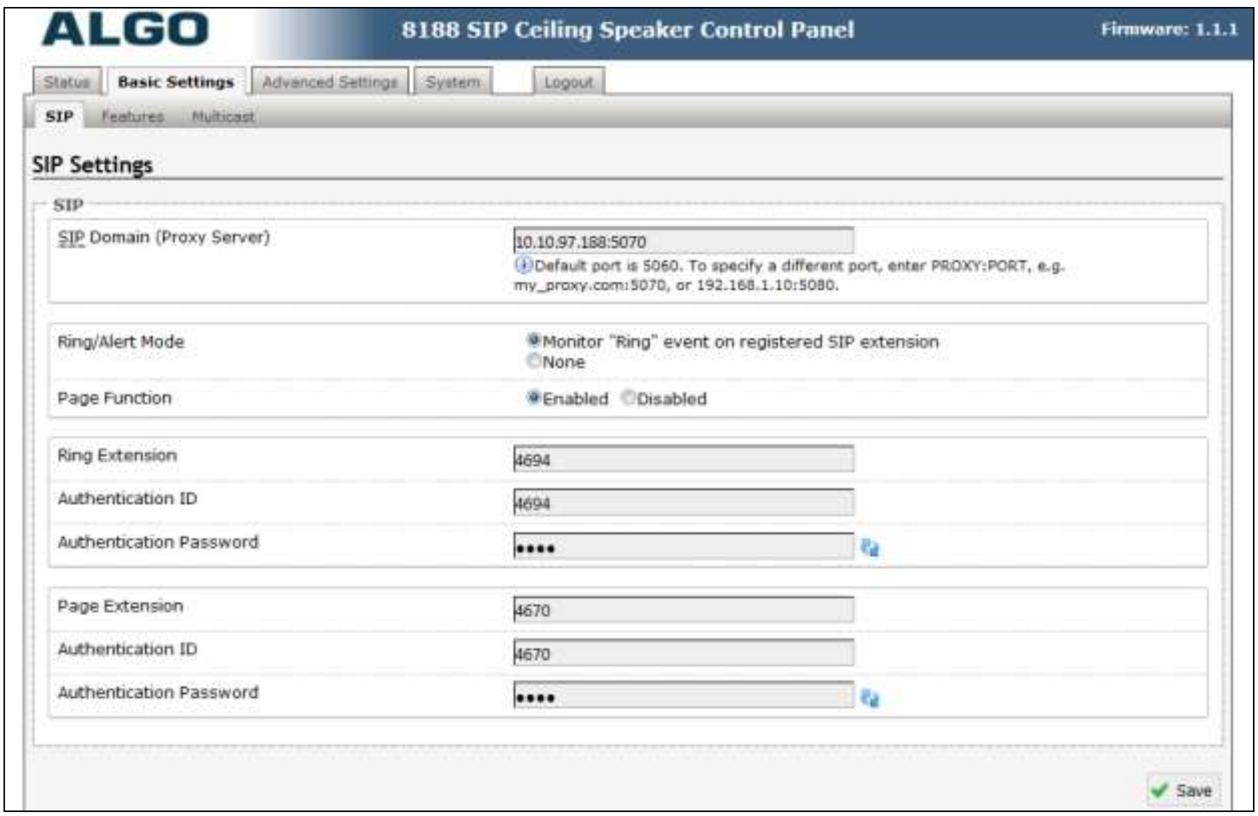
Access the 8188 SIP Ceiling Speaker web-based interface by using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the 8188 SIP Ceiling Speaker. The IP address of the 8188 can be spoken by pressing the reset button on the front of the 8188. The **Welcome to the Algo 8188 SIP Ceiling Speaker Control Panel** screen is displayed, as shown below. Log in using the appropriate credentials.



6.2. Administer Algo 8188

Select **Basic Settings → SIP** from the top menu, to display the screen below. Configure the **SIP Settings** section toward the bottom of the screen as desired to match the configuration. Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Sip Domain (Proxy Server):** enter the node IP address and the port **10.10.97.188:5070** as configured in **Section 5.2**.
- **Ring/Alert Mode:** Select **Monitor "Ring" event on the registered SIP extension**.
- **Page Function:** Select **Enabled**.
- **Ring Extension:** Enter the loud ringing SIP user from **Section 5.2**.
- **Authentication ID:** Enter the loud ringing SIP user from **Section 5.2**.
- **Ring Password:** Enter the loud ringing SIP user password from **Section 5.2**.
- **Page Extension:** Enter the voice paging SIP user from **Section 5.2**.
- **Page Auth ID:** Enter the voice paging SIP user from **Section 5.2**.
- **Page Password:** Enter the voice paging SIP user password from **Section 5.2**.



7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya CS 1000 and Algo 8188 SIP Ceiling Speaker.

7.1. Verify Avaya CS 1000

Use the overlay command LD 32 to show the terminal number of the 8188 ring and page SIP user it should show as IDLE REGISTERED if the 8188 ring and page SIP users register successfully to the SIP Line server.

```
>ld 32
NPR000
.stat 108 0 1 9
IDLE REGISTERED 00
.stat 108 0 1 10
IDLE REGISTERED 00
```

7.2. Verify Algo 8188

From the Algo 8188 SIP Ceiling Speaker web-based interface, select **Status** from the top menu. Verify that **SIP Registration** displays "Ring – Successful" and "Page – Successful", as shown below.

The following tests were conducted to verify the solution between the Algo 8188 and Avaya CS 1000.

- Verify that the incoming call to the desk-1 extension on the CS 1000 rings that associated with the 8188 ring will ring the 8188 loud and the 8188 ring stops ringing if the desk-1 extension answers the call.
- Verify that the incoming call to the 8188 Voice page is automatically answered with clear audio path.
- Verify that the telephone that places the incoming call to the 8188 Page can do conference, transfer, mute, un-mute and provide busy tone if it is on another call.
- Verify that the solution works with different Avaya clients (e.g. digital, analog, IP etc).
- Verify that 8188 goes into an idle state when the call is completed.
- Verify that the 8188 re-registers without issues if the Ethernet cable is unplugged and plugged back in.

8. Conclusion

These Application Notes describe the configuration steps required to integrate the Algo 8188 Ceiling Speaker with Avaya Communication Server 1000. 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**.

9. Additional References

Product documentation for the Avaya IP Office may be found at:
https://support.avaya.com/css/Products/

Avaya Communication Server 1000 Documents:

[1] Avaya Communication Server 1000E Installation and Commissioning
[2] Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6
[3] Avaya Communication Server 1000 Element Manager System Reference – Administration
[4] Avaya Communication Sever 1000 Co-resident Call Server and Signaling Server Fundamentals

Product documentation for the Algo 8188 SIP Audio Alerter products may be found at:
http://www.algosolutions.com/8188

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.