Avaya Solution & Interoperability Test Lab

Application Notes for Algo 8036 SIP Multimedia Intercom with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Algo 8036 SIP Multimedia Intercom with Avaya Aura® Session Manager and Avaya Aura® Communication Manager configured as an Evolution Server. The Algo 8036 SIP Multimedia Intercom provides hands-free intercom capability and entrance security with door unlock control. It is a SIP compliant device that registers with Avaya Aura® Session Manager.

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 steps required to integrate the Algo 8036 SIP Multimedia Intercom with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Algo 8036 SIP Multimedia Intercom provides hands-free video call capability and entrance security with door unlock control. It is a SIP compliant device that registers with Avaya Aura® Session Manager.

A visitor can interface with the Algo 8036 SIP Multimedia Intercom touch screen to call a preprogrammed extension(s). The called party can then answer the video call to communicate with the Algo 8036 SIP Multimedia Intercom. Using DTMF tones, the called party can press a digit on the phone keypad to activate the door control relay to open the door. Alternatively, a telephone can also originate a call to the Algo 8036 SIP Multimedia Intercom, which would be automatically answered. The Algo 8036 SIP Multimedia Intercom is configured via a web interface.

2. General Test Approach and Test Results

To verify interoperability of the Algo 8036 SIP Multimedia Intercom with Communication Manager and Session Manager, calls were made from the Algo 8036 SIP Multimedia Intercom to another specified telephone. The called telephone would ring and answer the call. Upon answering the call, a two-way video and audio path was established between the telephone and the Algo 8036 SIP Multimedia Intercom. The telephone can then press a digit on the keypad to open the door. In addition, incoming calls to the Algo 8036 SIP Multimedia Intercom were also verified.

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

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the Algo 8036 SIP Multimedia Intercom with Session Manager.
- Touch call button at 8036 SIP Multimedia Intercom to ring specified telephone, answer the call, and establish a two-way video and audio path. Caller ID on the telephone was also verified.
- Called telephone can press a DTMF digit to open the door.
- Incoming calls to the Algo 8036 SIP Multimedia Intercom.
- G.711 and H.264 video codec support.
- Proper system recovery after the Algo 8036 SIP Multimedia Intercom loses power.

2.2. Test Results

All test cases passed and the Algo 8036 SIP Multimedia Intercom successfully registered with Session Manager. Calls and delivery of DTMF tones to 8036 SIP Multimedia Intercom was successful.

2.3. Support

For technical support on the Algo 8036 SIP Multimedia Intercom, contact Algo Technical Support by phone or through their website.

Sales: (604)-454-3790
Technical Support: (604) 454-3792
Web: http://www.algosolutions.com/8036

3. Reference Configuration

Figure 1 illustrates a sample configuration with Avaya SIP-based network that includes Avaya Aura® Session Manager, Avaya Aura® Communication Manager running on Avaya S8300 Server with Avaya G450 Media Gateway, and the Algo 8036 SIP Multimedia Intercom. The Algo 8036 SIP Multimedia Intercom registered with Avaya Aura® Session Manager.

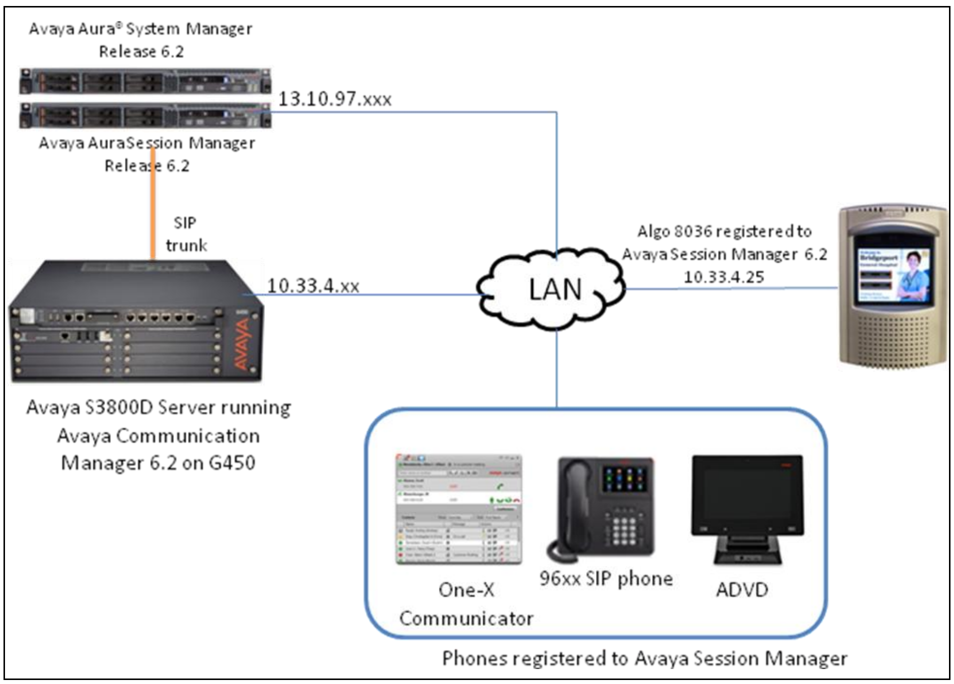


Figure 1: Avaya SIP Network with Algo 8036 SIP Multimedia Intercom

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on G450 with Avaya S8300 Server	R016x.02.0.823.0
Avaya Aura® Session Manager Avaya S8800 Server	6.2.3.0.623006
Avaya Aura® System Manager Avaya S8800 Server	R6.2 S12 1822
Avaya 9608 SIP Telephones	6.2.1.26
One-X Communicator	6.1 SP7
Avaya Desktop Video Device	Software version 1.1.0
Algo 8036 SIP Multimedia Intercom	Firmware 1.0

5. Configure Avaya Aura® Communication Manager

In this section it is assumed that Avaya Aura® Communication Manager has been installed and operational.

This section only describes the steps for configuring the Algo 8036 SIP Multimedia Intercom as an Off-PBX Station (OPS) and verifying shuffling configuration for a SIP trunk between the Communication Manager and Session Manager. Use the System Access Terminal (SAT) to configure Communication Manager. Log in the SAT using appropriate credentials.

Please note that the Algo 8036 SIP Multimedia Intercom is referred as the Algo 8036 SIP Multimedia Intercom in **Section 5** and **Section 6**.

5.1. Verifying shuffling setting for SIP Trunk

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. The form is accessed via the **change ip-network-region n** command where **n** is the ip network region that will be used when creating a signaling group. In this configuration, the domain name is *bvwdev.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the Algo 8036 SIP Multimedia Intercom. The form is accessed via the **change ip-codec-set n** command, where **n** is the codec set that was configured in ip-network-region form. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. The Algo 8036 SIP Multimedia Intercom supports G.711MU.

```
change ip-codec-set 1                                          Page   1 of   2
                         IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU          n          2        20
 2:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the signaling group using **add signaling-group n** command; where **n** is an available signaling group. Configure the signaling group as follows:

- Set **IP Video** to *y*.
- Set **Direct IP-IP Audio Connections** to *y*.
- Set **DTMF over IP** to *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. Use the default values for the other fields.

```
add signaling-group 5                                      Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                             SIP Enabled LSP? n
     IP Video? y                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: procr                Far-end Node Name: DevASM
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
                                    Far-end Secondary Node Name:
Far-end Domain: bvwdev.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the trunk group using **add trunk-group n** command; where **n** is an available trunk group. Configure the trunk group as follows:

- **Group Type** – Set the Group Type field to *sip*.
- **Group Name** – Enter a descriptive name.
- **TAC** (Trunk Access Code) – Set to any available trunk access code.
- **Service Type** – Set the Service Type field to *tie*.
- **Signaling Group** – Set to the Group Number field value for the signaling group configured in above section.
- **Number of Members –** Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 5                                               Page   1 of  21
                                TRUNK GROUP
Group Number: 92                   Group Type: sip           CDR Reports: y
Group Name: Non_Secure_SIP                COR: 1        TN: 1        TAC: 115
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 5
                                                     Number of Members: 20
```

5.2. Configure Station

Use the **add station n** command to add station for the Algo 8036 SIP Multimedia Intercom, where **n** is an available extension. Use *9621SIP* for the **Station Type**. The **IP Video** field is set to *y*. Use the default values for the other fields. Alternatively, the SIP station can also be configured in System Manager as described in **Section 0**.

```
add station 52173                                              Page   1 of   6
                                     STATION

Extension: 52173                        Lock Messages? n               BCC: 0
     Type: 9621SIP                      Security Code:                  TN: 1
     Port: IP                         Coverage Path 1:                 COR: 1
     Name: 52173, 8036 SIP Multimedia Intercom          Coverage Path 2:
COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
             Loss Group: 19
                                               Message Lamp Ext: 52173
        Display Language: english                 Button Modules: 0
          Survivable COR: internal
   Survivable Trunk Dest? y                          IP SoftPhone? n
                                                         IP Video? y
```

Note: In the station setting for Avaya one-X® Communicator (52172) and Avaya Desktop Video Device (ADVD) (52175) extension, the **IP SoftPhone, IP Video Softphone** options are set to *y*.

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extension to the same extension on Session Manager. Enter the field values shown below. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Config Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 52173                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                 Prefix                     Selection  Set     Mode
 52173            OPS           -     52173            aar       1
```

6. Configure Avaya Aura® Session Manager

It assumes that Avaya Aura® Session Manager has been installed and operational.
This section only provides the procedures for configuring SIP user for the Algo 8036 SIP Multimedia Intercom. It is assumed that the **Domain, SIP Entity, SIP Entity Link, Time Range** and **Routing Policy** for Communication Manager are pre-configured and are not shown in this document.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in using appropriate credentials.

6.1. Add SIP User

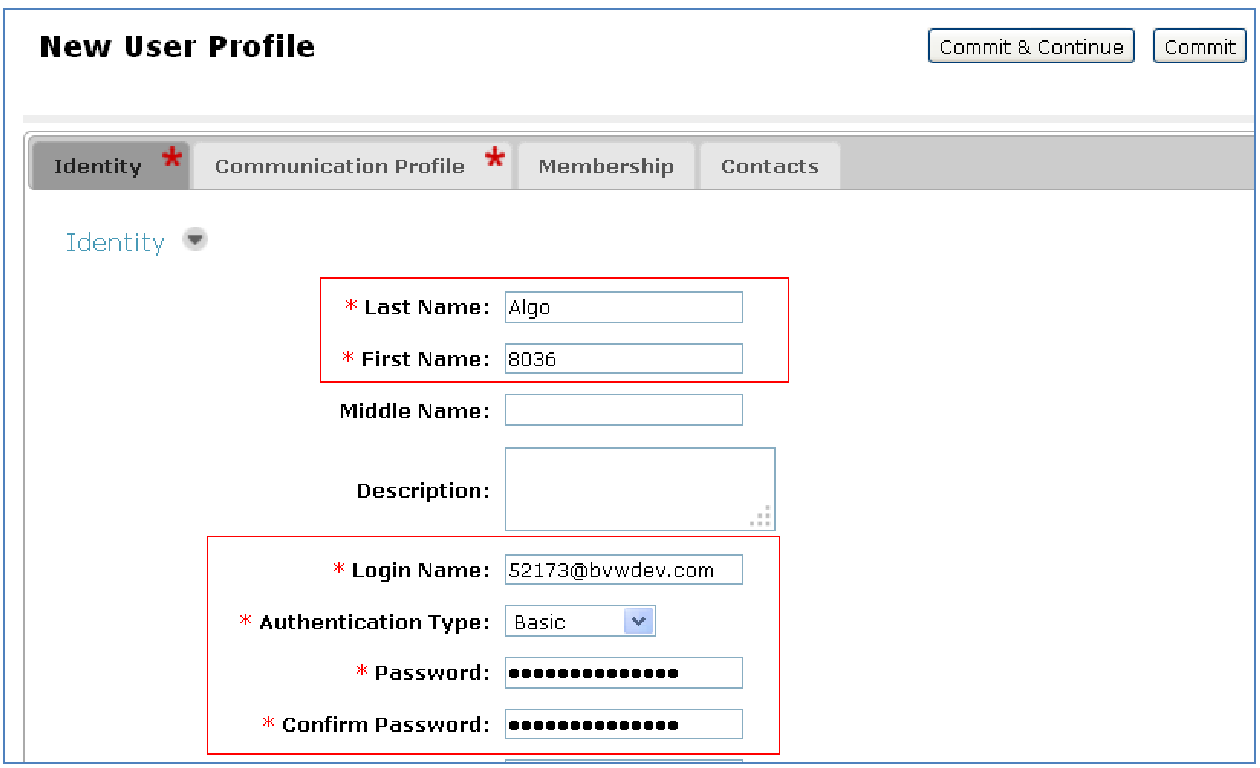
Add a SIP user for the Algo 8036 SIP Multimedia Intercom**.** The following configuration will automatically create the SIP station on Communication Manager.

To add a new SIP user, navigate to **Users → User Management → Manage Users** from the left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** tab of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., 52173@avaya.com).
- **Authentication Type:** Select *Basic***.**
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information when adding a new SIP user – **Identity** tab.



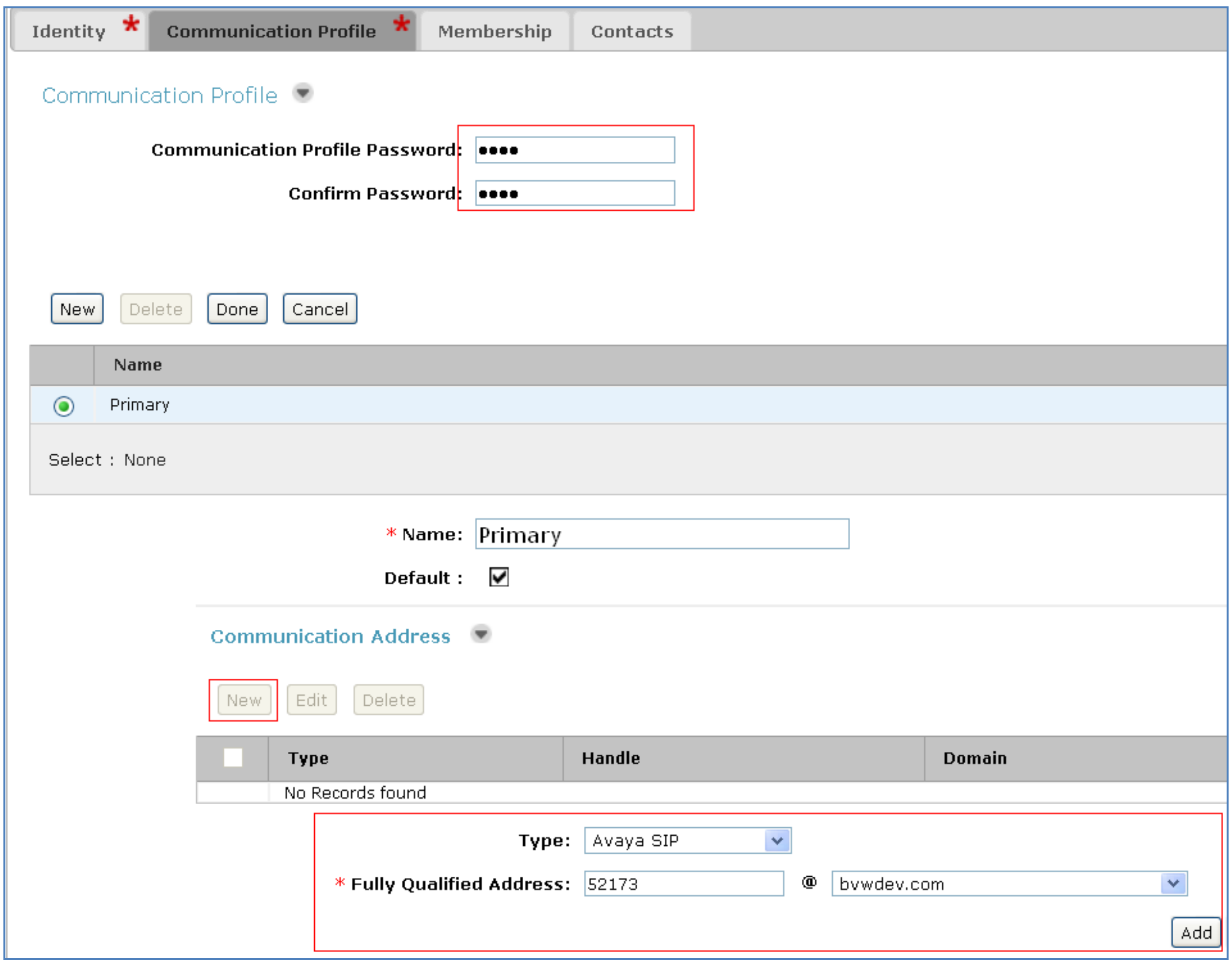
Enter values for the following required attributes for a new SIP user in the **Communication Profile** tab of the new user form.

- **Communication Profile Password:** Enter the password which will be used by the Algo 8036 SIP Multimedia Intercom to register with Session Manager in **Section 7.2**.
- **Confirm Password:** Re-enter the password from above.

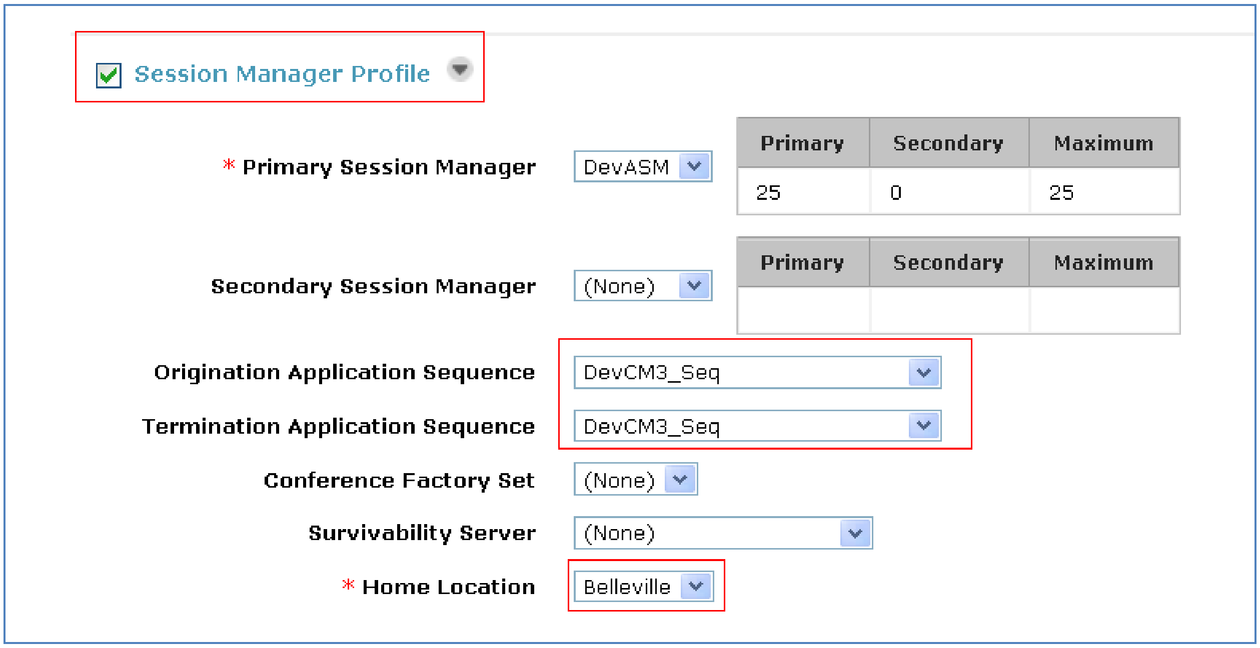
Scroll down to the **Communication Address** section and select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user in **Communication Profile** tab. Click **Add**.



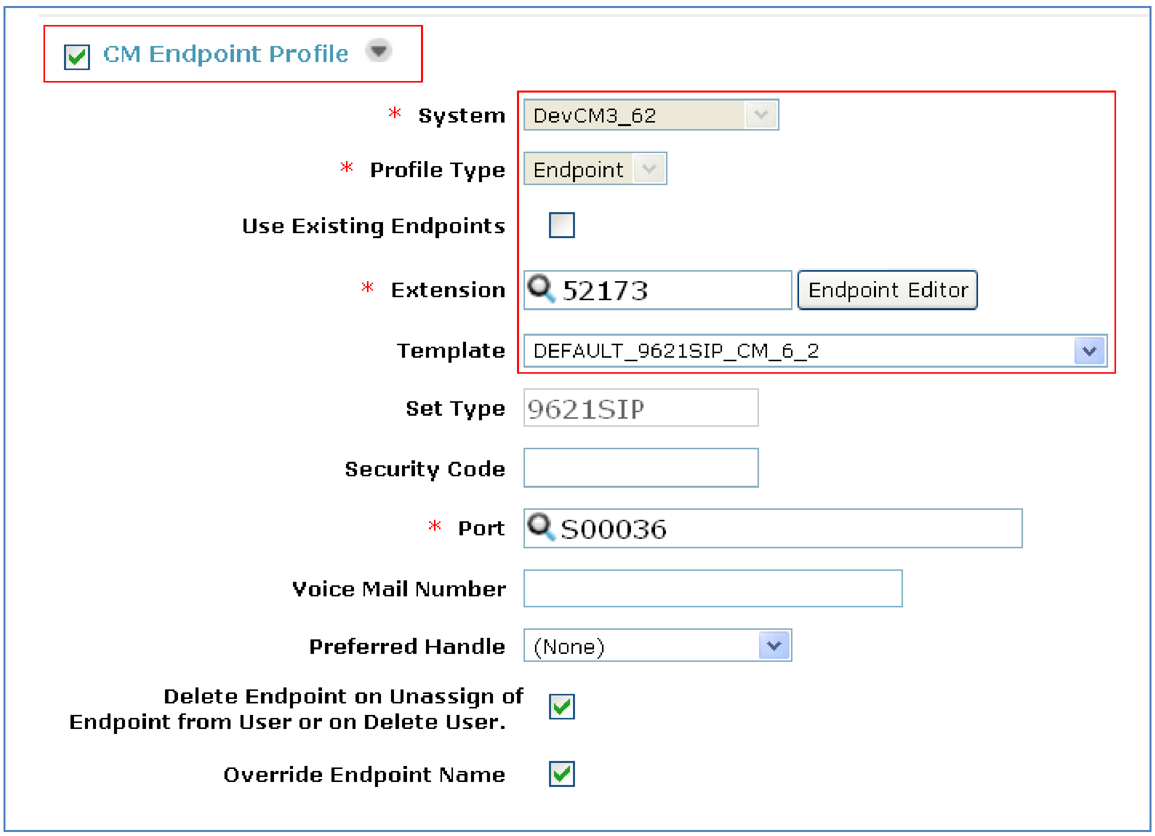
In the **Session Manager Profile** section, specify the Session Manager entity for **Primary Session Manager** and assign the **Application Sequence** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence. Set the **Home Location** field to the **Location**. This is assuming that Avaya Aura® System Manager, Avaya Aura® Session Manager and Avaya Aura® Communication Manager have been setup and operational.



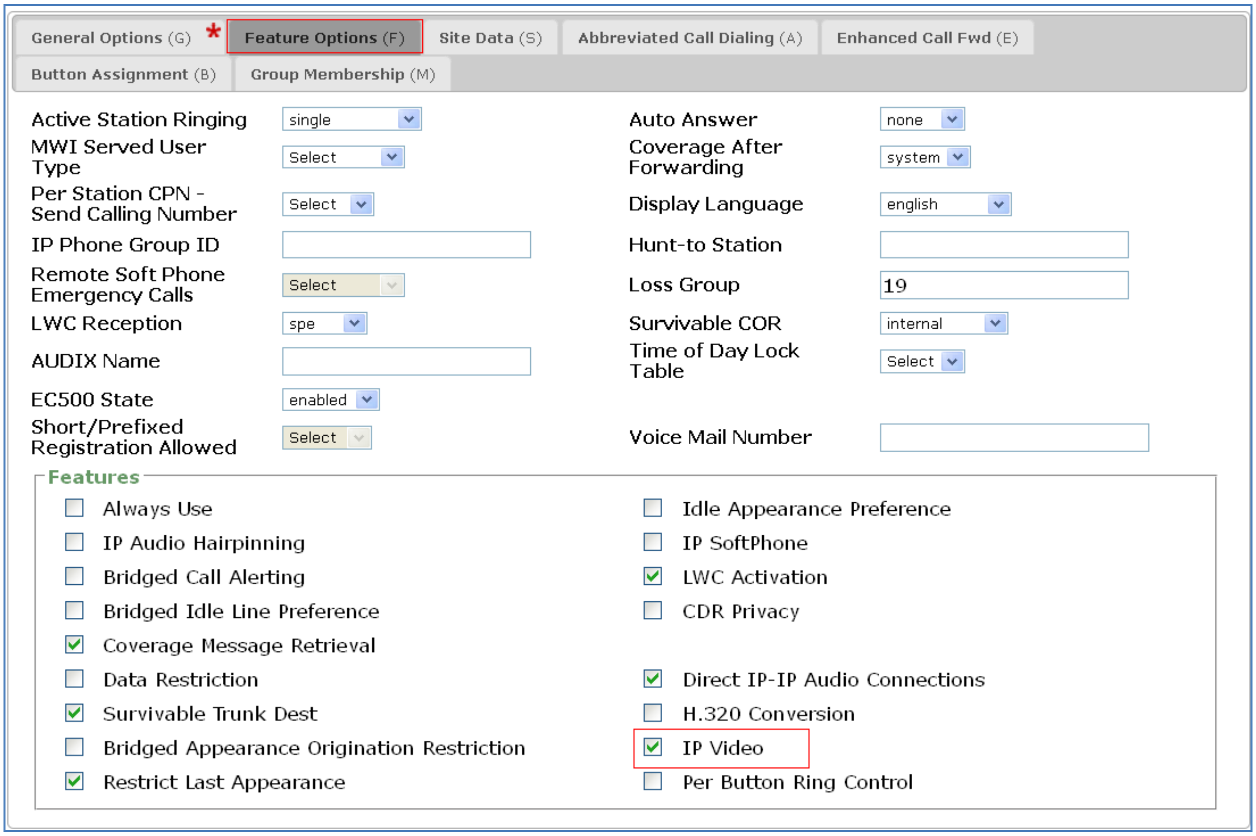
In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type** Select *Endpoint*.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Endpoint on Unassign of Endpoint:** Enable field to automatically delete station when **Endpoint Profile** is un-assigned from user.

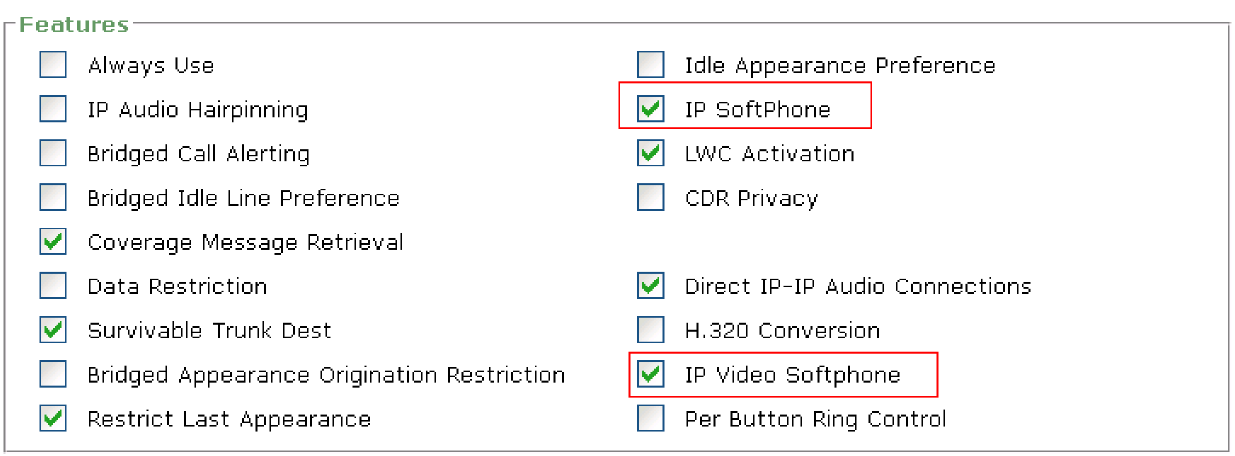
Note: To specify a coverage path for voicemail, click on the **Endpoint Editor** button.



In the **Enpoint Editor** form, select **Feature Options (F)** make sure **IP video** is enabled for Algo device.



Note: Repeat the same section **0** for the Avaya one-X® Communicator andADVD extension, make sure **IP SoftPhone** and **IP video Softphone** options are enabled as shown below.



7. Configure Algo 8036 SIP Multimedia Intercom

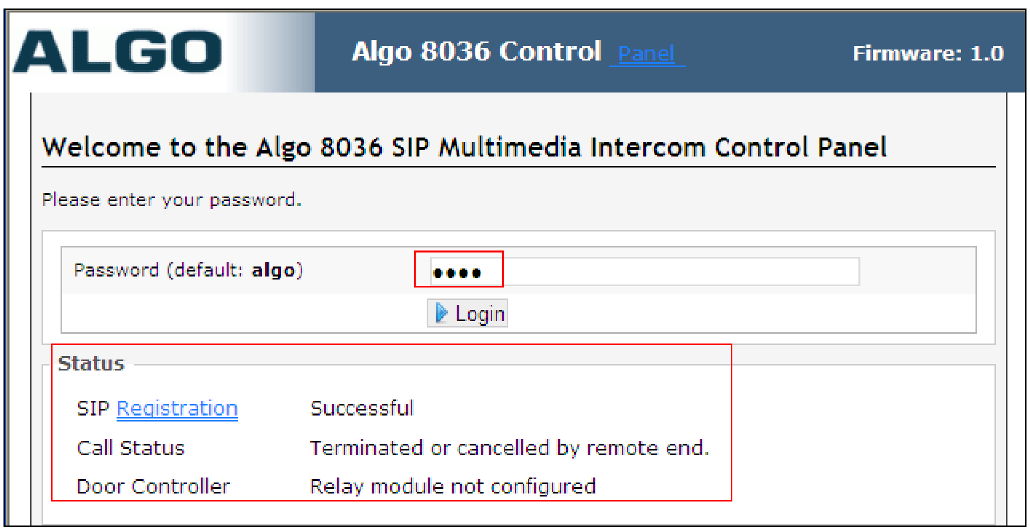
This section provides the procedures for configuring Algo 8036 SIP Multimedia Intercom. The procedures include the following areas:

- Launch web interface
- Administer Algo SIP Account
- Administer Dialing Extension
- Administer Algo Media

7.1. Launch Web Interface

Access the Algo 8036 SIP Multimedia Intercom web-based interface by using the URL "http://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Algo 8036 SIP Multimedia Intercom. This IP address can obtain from the touch screen during reboot process. The **Authorization Required** screen is displayed, as shown below. Log in using the appropriate credentials. Default password is "**algo**".

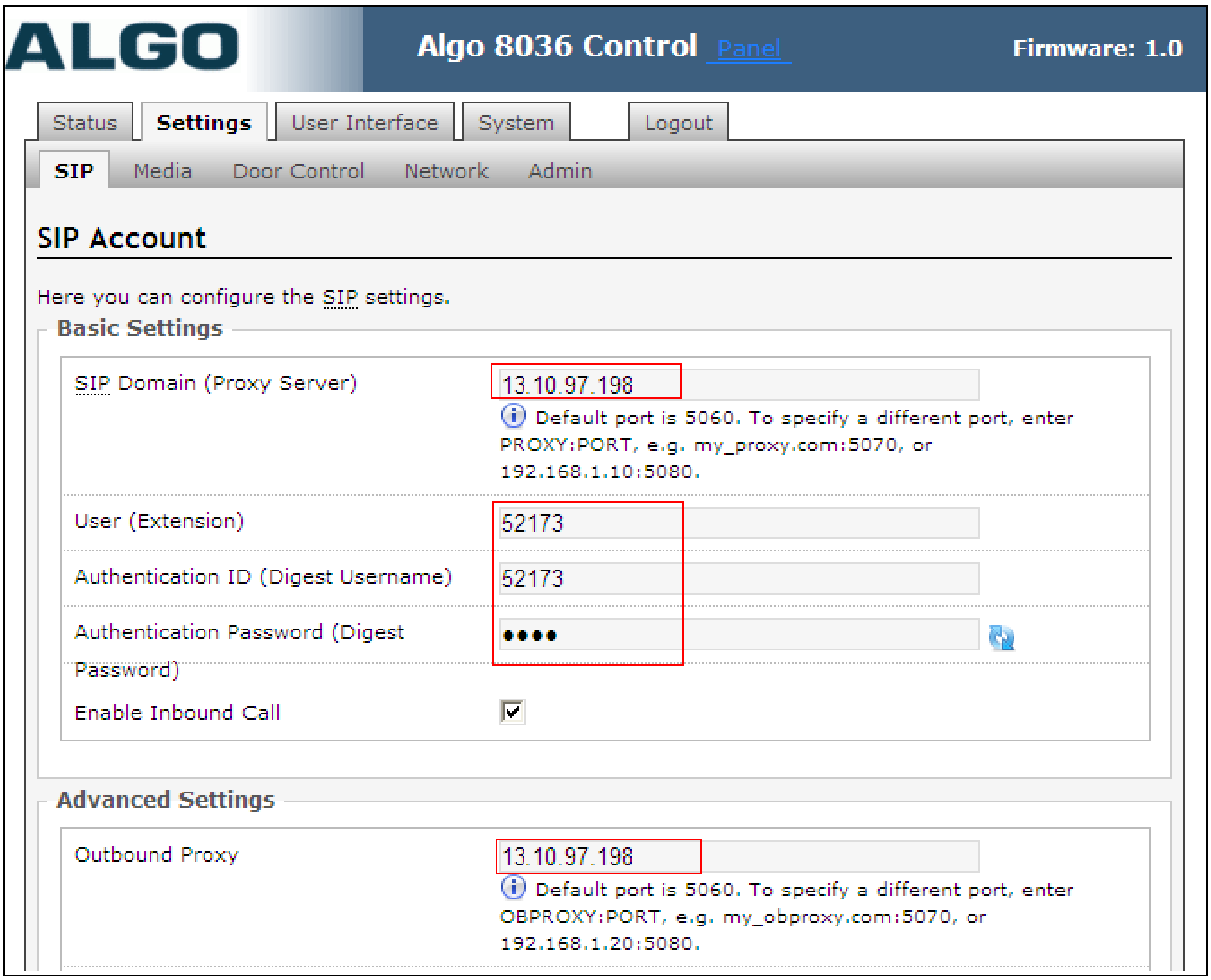
Note: The default IP address of the SIP Multimedia Intercom is 192.168.1.111.



7.2. Administer Algo SIP Account

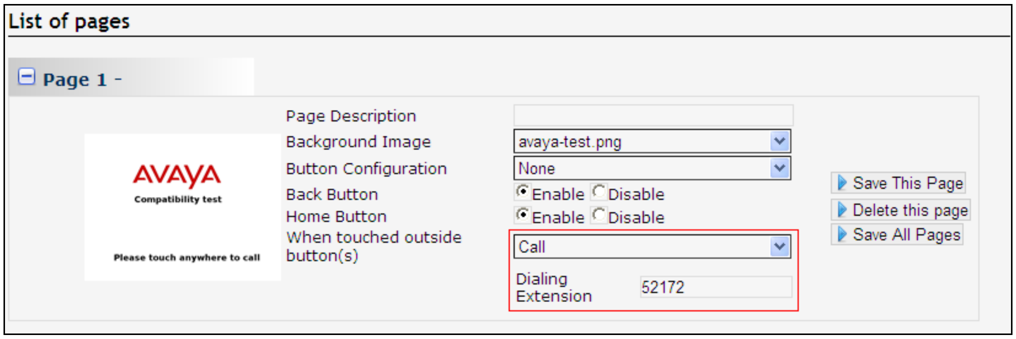
Select **Setting** from the top menu, to display the screen below. Configure the **SIP Account**, enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Sip Domain:** Enter the Session Manager signaling address.
- **User:** The extension value from **Section 5.2** or **0**.
- **Authentication ID:** The extension value from **Section 5.2** or **0**.
- **Authentication password:** The SIP user Security Code **Section 5.2** or the SIP user Communication Profile Password in **Section 0.**
- **Outbound Proxy** Enter the Session Manager's signaling address.



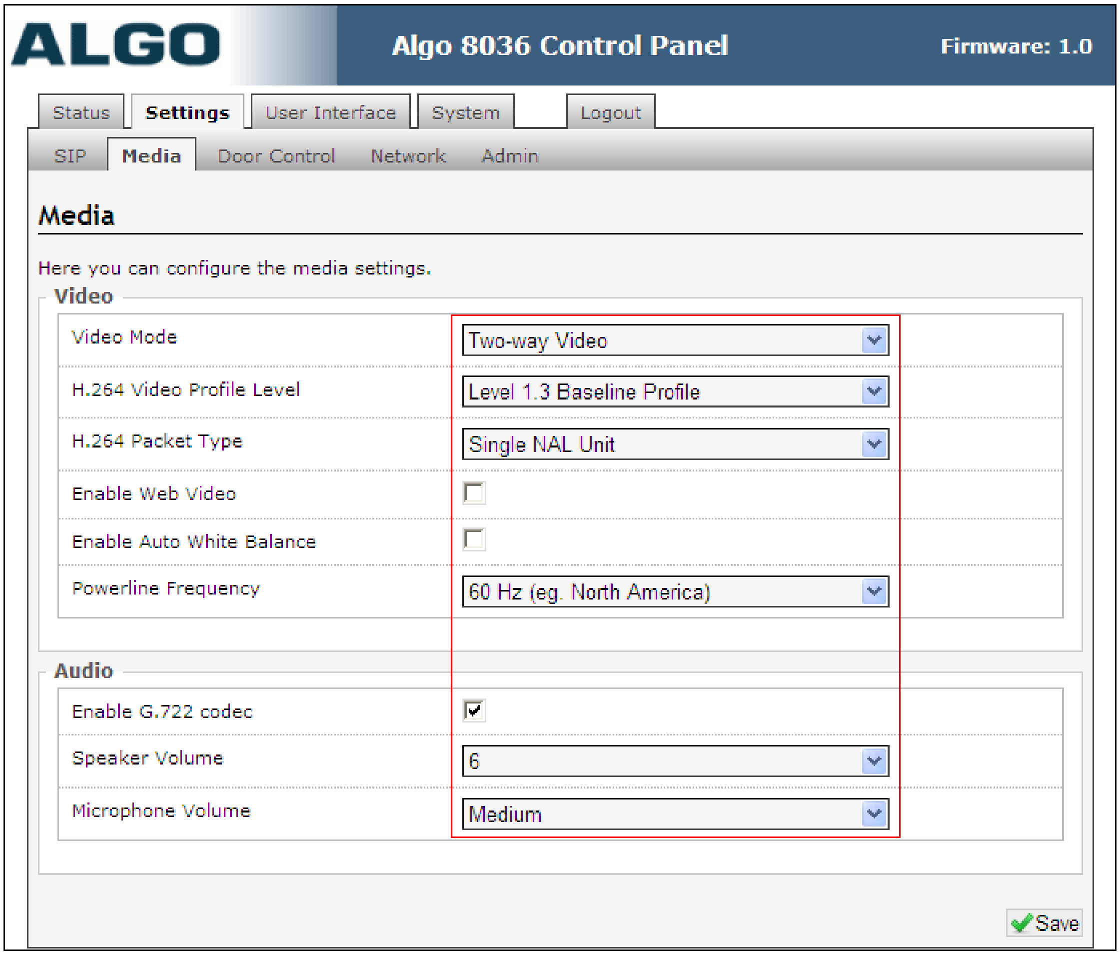
7.3. Administer Dialing Extension

Select **Screen** from the top menu, to display the screen below. Configure the **Dialing Extension** by scroll down toward the end, click on **+ Page 1**, enter designated extension that will ring when user touch the screen. During compliance testing, extension for Avaya one-X® Communicator was used to verify the voice/video call, it is the extension from **Section** Error! Reference source not found.. Click **Save Changes** to save changes.



7.4. Administer Algo Media

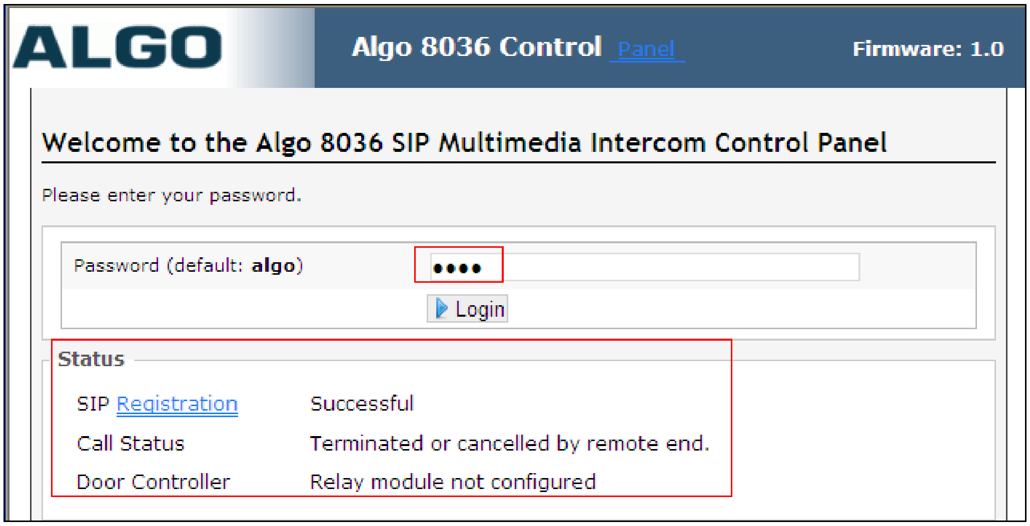
Select **Setting → Media** from the top menu, to display the screen below. Set **H264 Video Profile Level** to **Level 1.3 Baseline Profile** and retain the default values in the remaining fields.



8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the Algo 8036 SIP Multimedia Intercom has successfully registered with Session Manager. The **SIP Registration** field in the **Status** tab should indicate successful as shown below.



2. Verify that when the call button on the Algo 8036 SIP Multimedia Intercom is pressed, the specified telephone on Communication Manager rings, and upon answering the call, two-way video and audio path is established.
3. Verify that the Algo 8036 SIP Multimedia Intercom returns to the idle state when the call is terminated.
4. Verify that incoming video calls to the Algo 8036 SIP Multimedia Intercom are also successful.

9. Conclusion

These Application Notes describe the administration steps required to integrate the Algo 8036 SIP Multimedia Intercom with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Algo 8036 SIP Multimedia Intercom successfully registered with Avaya Aura® Session Manager and incoming and outgoing calls were successful. In addition, unlocking a door using DTMF tones was successful. All test cases passed.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya and Algo product documentation is available at http://support.avaya.com.

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 7.0, Release 6.2, February 2012, available at http://support.avaya.com
2. *Administering Avaya Aura®Session Manager* 03-603324 Release 6.2 July 2012 available at http://support.avaya.com
3. Algo 8036 documentation is available at http://www.algosolutions.com.

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.