



**Application Notes for Algo 8036 SIP Multimedia Intercom
with Avaya Communication Server 1000 SIP Line Release
7.5 – Issue 1.0**

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and Algo 8036 SIP Multimedia Intercom. During the compliance testing, the Algo 8036 controller was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Algo 8036 SIP Multimedia Intercom was able to place and receive calls from the Communication Server 1000 Release 7.5 non-SIP and SIP Line telephones.

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 detailed information on how to create SIP Endpoints for Algo 8036 and One-X Communicator used during the compliance testing. All the applicable telephony feature test cases of release 7.5 SIP line were executed on the Algo 8036, where applicable, to ensure interoperability with the CS1000.

2 General Test Approach and Test Results

The general test approach was to have the Algo 8036 register to the CS1000 SIP line gateway successfully. From the CS1000 telephone clients/users video calls were placed to and from the other telephony features such as busy, hold, DTMF, MWI and codec negotiation were tested.

2.1 Interoperability Compliance Testing

The focus of this testing was to verify that the Algo 8036 SIP Multimedia Intercom was able to interoperate with the CS1000 SIP line system. The following areas were covered:

- Registration of the Algo 8036 to the CS1000 SIP Line Gateway.
- Video call establishment of Algo 8036 with CS1000 SIP One-X Communicator, Avaya Desktop Video Device and non-SIP telephones, and ensure two way speech paths.
- CS1000 telephones can send DTMF tones to Algo 8036 controller to exercise the remote door release function.
- Telephony features: Video call, basic calls, conference, blind and consultative transfer (from Avaya phone), DTMF (dual tone multi frequency), busy, mute, call forward no answer, busy and all calls.

2.2 Test Results

The objectives outlined in the **Section 2.1** were verified. The following observations were made during the compliance testing:

- In order to have the normal quality video call to Avaya Flare Experience on ADVD, make sure the **H264 Video Profile Level** is set to “Level 1.3 Baseline Profile” see configuration screen in **Section 6.4**.

2.3 Support

For technical support on Algo 8036, please contact Algo technical support team:

- **Telephone:** 1-877-884-2546
- **Email:** support@algosolutions.com
- **Website:** <http://www.algosolutions.com>

3 Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Algo 8036.

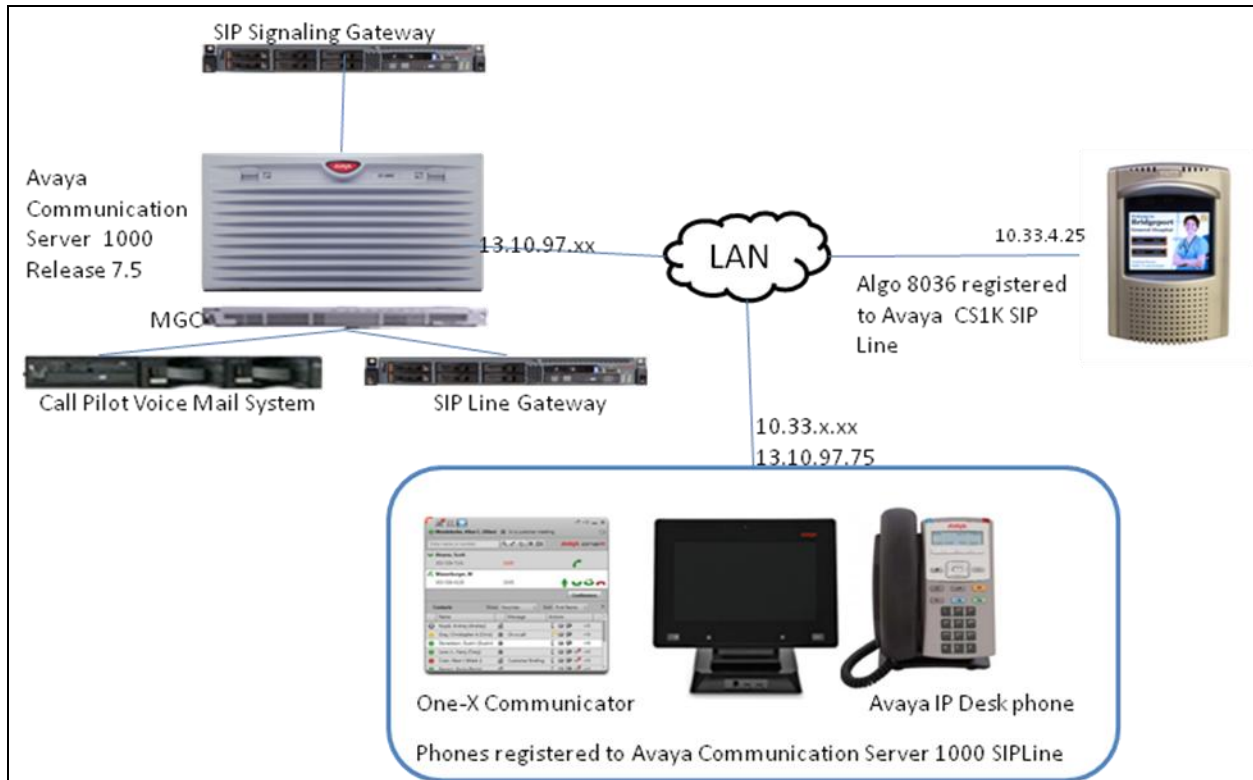


Figure 1: Network Configuration Diagram

4 Equipment and Software Validated

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

Equipment	Software Version
Avaya CS1000E	Call Server (CPPM): 7.50Q Signaling Server (CPPM): 7.50.17
Avaya CallPilot™ Messaging System	5.0.1
Avaya One-X Communicator	CS1K version
Avaya Desktop Video Device	Model version A175D01A-003 Kernel version 2.6.31.6-17.1.moblin2-mojo
Avaya SIP 1140	02.02.21.00
Algo 8036 SIP Multimedia Intercom	Firmware version 00.11.10

5 Configure Avaya CS 1000 – SIP extension for 8036

This section describes the steps to configure the Avaya CS1000 SIP Line using the CS1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS1000 system. For detailed information on how to configure and administer the CS1000 SIP Line, please refer to **Section 9[1]**.

This document assume that CS100 SIP Line configuration had been setup and operational. For more detail on how to configure please refer document Application Notes for Algo 8028 SIP Doorphone with Avaya Communication Server 1000 SIP Line Release 7.5 listed in **Section 9**. The following is a summary of tasks that need to be done to configure the CS1000 SIP User for Algo, One-X Communicator: Create SIP Line phones.

5.1 Prerequisite

This document assumes that the CS1000 SIP Line server has:

- Been installed with CS1000 Release 7.5 Linux Base.
- Joined CS1000 Release 7.5 Security Domain.
- Been deployed with SIP Line Application.
- SIP Line Node has been setup and in operational state. The IP address of SIP Line used in this test is **13.10.97.187** and its domain is **sip175.com**.

5.2 Create a SIP Line Phone for Algo 8036

To create a SIP Line phone 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 it at default values.

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

TN 104 0 0 13
DES ALGO
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 54354u - SIP phone user name
NDID 512 - Node ID of SIP Line
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
```

```

ECL 0
VSIT NO
FDN 54443 - Forward No Answer to this DN, need to enable class of service FNA
TGAR 1
LDN NO
NCOS 7 - Network Class of Service, 7 is highest level
SCPW 1234 - Password of SIP Line user. Need to register Algo to SIP Line Node
SFLT NO
CAC_MFC 0
CLS FNA FBA HTA MWA DNDA CNDA CFXA
CPND_LANG ENG
RCO 0
HUNT 54443 - Forward busy to this DN, need to enable class of service FBA, HTA
LHK 0
PLEV 02
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 54354 0 MARP
      CPND
        CPND_LANG ROMAN
        NAME Algo 1
        XPLN 13
        DISPLAY_FMT FIRST, LAST
      01 HOT U 2654354 MARP 0 - Key 1 hot U with prefix + DN

```

5.3 Create a SIP Line Phone for One-X Communicator

To create a SIP Line phone 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 it at default values.

```

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

TN 104 0 0 14
DES ALGO
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL - Universal extension type is SIP Line
MCCL YES
SIPN 1 - For SIP phone third party, enter 0 in this field
SIP3 0 - For SIP phone third party, enter 1 in this field
FMCL 0
TSLV 0
SIPU 54353u - SIP phone user name
NDID 512 - Node ID of SIP Line
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001

```

```

ECL 0
VSIT NO
FDN 54443 - Forward No Answer to this Call Pilot DN, need to enable class of
service FNA
TGAR 1
LDN NO
NCOS 7 - Network Class of Service, 7 is highest level
SCPW 1234 - Password of SIP Line user. Need to register Algo to SIP Line Node
SFLT NO
CAC_MFC 0
CLS FNA FBA HTA MWA DNDA CNDA CFXA
CPND_LANG ENG
RCO 0
HUNT 54443 - Forward busy to this Call Pilot DN, need to enable class of service
FBA, HTA
LHK 0
PLEV 02
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 54353 0 MARP
      CPND
        CPND_LANG ROMAN
        NAME Algo 2
        XPLN 13
        DISPLAY_FMT FIRST, LAST
01 HOT U 2654353 MARP 0 - Key 1 hot U with prefix + DN

```

Repeat the same configuration to create another SIP user for other SIP Phone such as Avaya ADVD, Avaya 1140E.

6 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 configuration

6.1 Launch Web Interface

Access the 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 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.

Welcome to the Algo 8036 SIP Multimedia Intercom Control Panel

Please enter your password.

Password (default: algo)

••••

Login

6.2 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:** The SIP Line Node domain **Section 5.1**.
- **User:** The extension value from **Section 5.2**.
- **Authentication ID:** The SIPU user name from **Section 5.2**.
- **Authentication password:** The SIP user login code from **Section 5.2**.

SIP Account

Here you can configure the basic SIP settings.

SIP domain

sipl75.com

User

54354

Authentication ID

54354u

Authentication password

 ••••

Enable inbound call



6.3 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 testing, extension that login One-X Communicator is used to verify the voice/video call, it is the extension from **Section 5.3**. Click **Save Changes** to save changes.

List of pages

You can configure individual pages.

Page 1 -

Page Description:

Background Image:

Button Configuration:

Back Button: Enable Disable

Home Button: Enable Disable

When touched outside button(s):

Dialing Extension:

Button 1

Enable Disable

Button Text:

Action Type:

Extension:

Button 2

Enable Disable

Button Text:

Action Type:

Target Page:

Button 3

Enable Disable

Button Text:

Action Type:

Target Page:

6.4 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.

ALGO Algo 8036 Control Panel Firmware: 0.11.10

Status Settings User Interface System Logout

SIP **Media** Door Control Network Admin

Media

Here you can configure the media settings.

Video

Video Mode:

H.264 Video Profile Level:

H.264 Packet Type:

Audio

Enable G.722 codec:

Speaker Volume:

Microphone Volume:

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7 Verification Steps

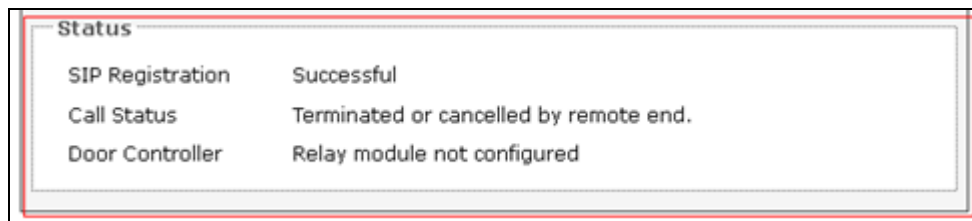
7.1 Verify SIP Line phones are registered

To verify the status of a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 32, verify the status of SIP Phone is registered as shown below:

```
>ld 32
.stat 104 0 0 13
IDLE REGISTERED 00
.stat 104 0 0 14
IDLE REGISTERED 00
```

7.2 Verify Algo 8036 is registered

Access the 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 SIP Multimedia Intercom. The Status section below the Login section display the status of Algo 8036, example below shown the Algo 8036 is successfully registered to CS1K.



Status	
SIP Registration	Successful
Call Status	Terminated or cancelled by remote end.
Door Controller	Relay module not configured

Status section is also available in **Status** tab in the Algo web interface.

7.3 Login One-X Communicator CS1K

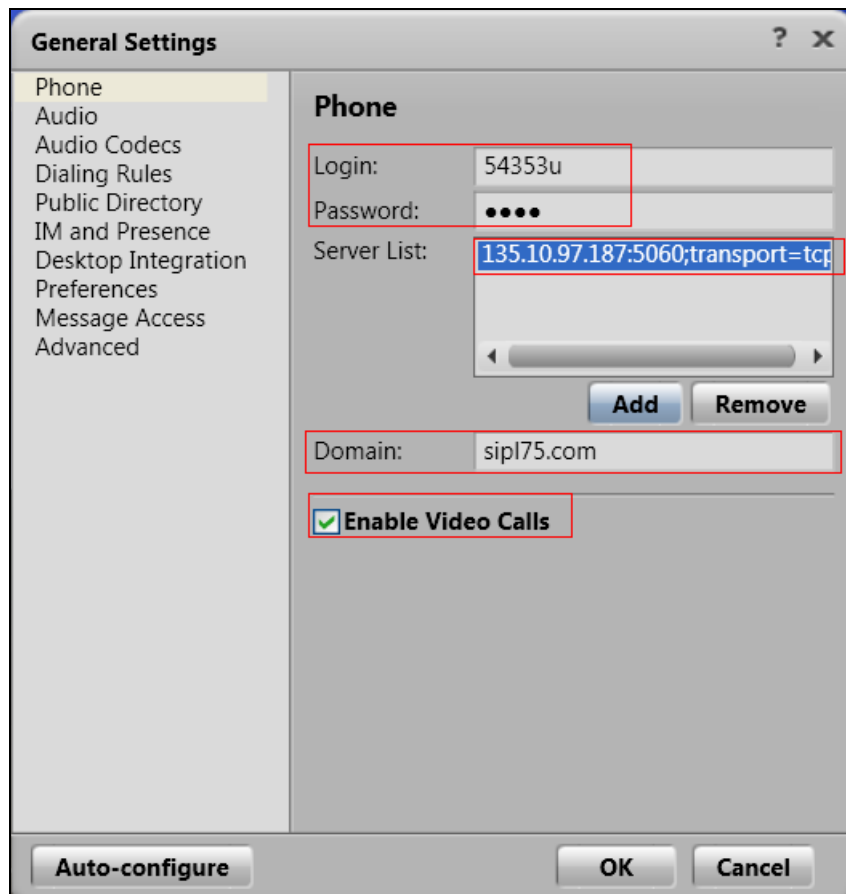
This section describe the steps need to be setup in order to login One-X Communicator such as server configuration, login credential.

Select **Start** → **Programs** → **Avaya one-X Communicator** → **Avaya One-X Communicator** click on **Menu** → **Setting** as show below:

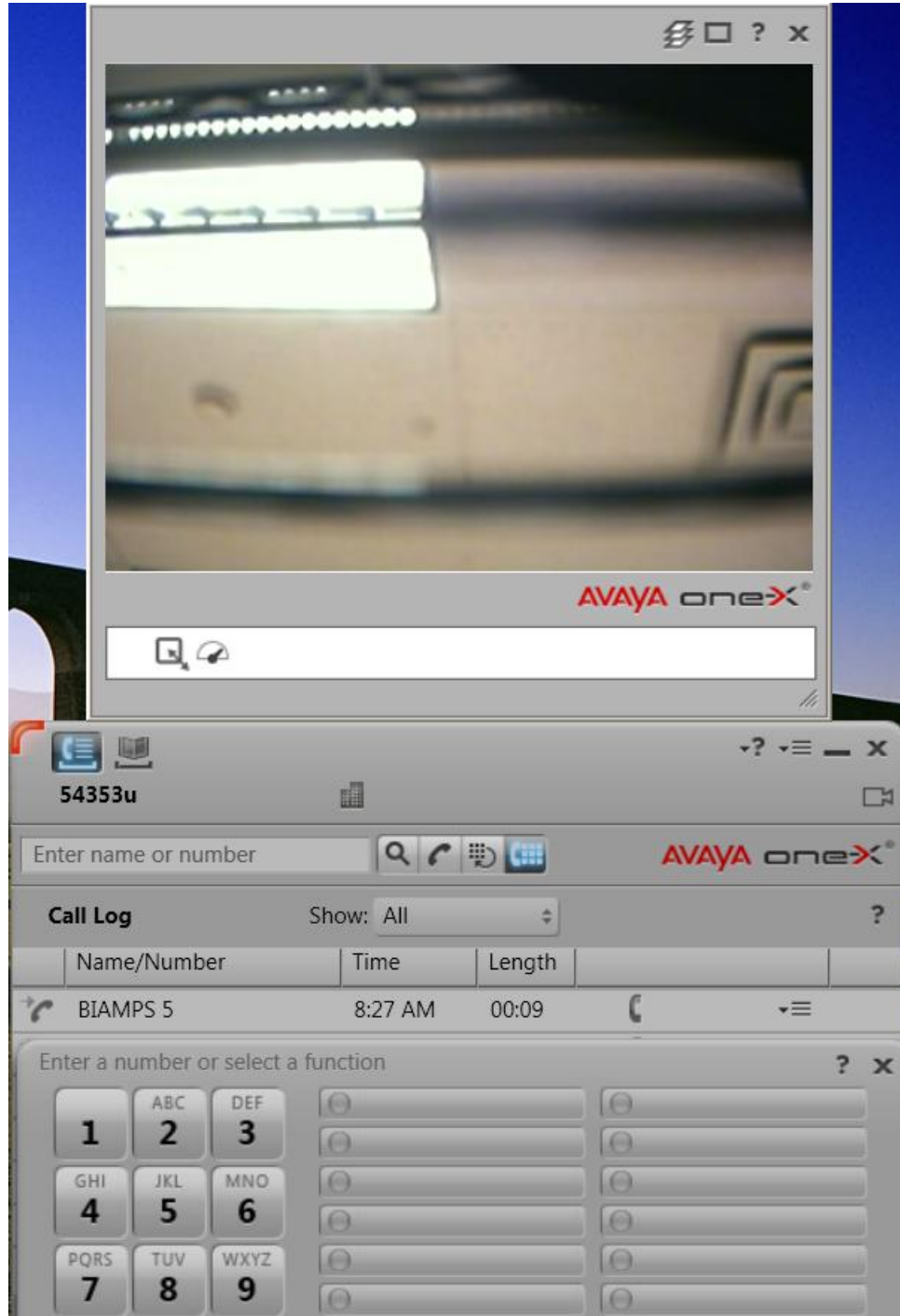


In the **General Setting** enters the following values for the specified fields, and retains the default values in the remaining fields.

- **Login:** The SIP user name from **Section 5.3**.
- **Password:** The password of **User** tab from **Section 5.3**.
- **Server List:** The IP address of SIP Line Node from **Section 5.1**;transport=tcp;port=5060.
- **Domain:** The domain of SIP Line Node from **Section 5.1**



Click **OK** then click **Log On** in the login screen. Below shown the screen when user login successfully.



7.4 Make a video call from Algo 8036 to one-X Communicator

Touch the screen to initiate video call to one-X Communicator, the one-X Communicator is ringing then clicks on the answer button to establish the call. Verify that the video call is established with a good quality of voice and video. Below is a screenshot of video call capture from one-X Communicator.



8 Conclusion

All of the executed test cases have passed and met the objectives outlined in the **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Algo 8036 firmware version 0.11.10 is considered to be in compliance with Avaya CS1000 SIP Line System Release 7.5.

9 Additional References

[1] Product documentation for the Avaya CS1000 products may be found at:

<https://support.avaya.com/css/Products/>

Avaya Communication Server 1000E Installation and Commissioning
Avaya Communication Server 1000 SIP Line Fundamental, Release 7.5
Avaya Communication Server 1000 Element Manager System Reference – Administration
Avaya Communication Server 1000 Co-resident Call Server and Signaling Server
Fundamentals
Avaya Communication Server 1000 Unified Communications Management Common Services
Fundamentals
Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and
Commissioning
Application Notes for Algo 8028 SIP Doorphone with Avaya Communication Server 1000 SIP
Line Release 7.5 – Issue 1.0

[2] Product documentation for the Algo 8036 products may be found at:
www.algosolutions.com/8036

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