



Avaya Solution & Interoperability Test Lab

Application Notes for Aiphone IX Series 2 Video Door Stations (IX-DVM) with Avaya IP Office Server Edition - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Aiphone IX Series Video Door Stations (IX-DVM) Version 5.75 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. The Aiphone IX-DVM Video Door Station, which is part of the Aiphone IX Series 2 Video Door Stations, was used for the compliance test. The Aiphone IX-DVM Video Door Station is a surface mount, weather resistant video door station. It has one dry contact that can be used to release doors when activated by a phone and can provide one-way video to a video capable SIP phone. The Aiphone IX-DVM Video Door Station registers with Avaya IP Office as a SIP endpoint.

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 to integrate Aiphone IX Series Video Door Stations (IX-DVM) Version 5.75 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. The Aiphone IX-DVM Video Door Station, which is part of the Aiphone IX Series 2 Video Door Stations, was used for the compliance test. The Aiphone IX-DVM Video Door Station is a surface mount, weather resistant video door station. It has one dry contact that can be used to release doors when activated by a phone and can provide one-way video to a video capable SIP phone. The Aiphone IX-DVM Video Door Station (IX-DVM) registers with Avaya IP Office as a SIP endpoint.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing audio and video calls between Aiphone IX-DVM Video Door Station, Avaya SIP and H.323 telephones, Avaya Workplace Client for Windows, Avaya Vantage™ K155, and the PSTN, and exercising basic telephony features, such as hold/resume, mute/unmute, transfer, conference, call forwarding, and call coverage from an Avaya IP endpoint. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Aiphone IX-DVM Video Door Station come back into service after re-connecting the Ethernet cable.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Aiphone IX-DVM Video Door Station did not include use of any specific encryption features as requested by Aiphone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of IX-DVM with IP Office Server Edition and IP Office 500 V2 Expansion System.
- Audio calls between IX-DVM and Avaya SIP and H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Audio and video calls between IX-DVM, Workplace, and Vantage K155 with Direct IP Media (Shuffling) enabled and disabled. One-way video from IX-DVM to Workplace and Vantage K155 was verified.
- Audio calls between IX-DVM and the PSTN.
- G.711 codec support.
- UDP transport protocol.
- Door release by Avaya IP endpoint using DTMF.
- IX-DVM placing, answering, and terminating calls using contactless call sensor.
- Basic telephony features, including hold/resume, mute/unmute, redial, transfer, and 3-way conference, initiated from an Avaya IP endpoint.
- Proper system recovery after re-establishing IP connectivity to IX-DVM.

2.2. Test Results

All test cases passed.

2.3. Support

For technical support of Aiphone IX Series 2 Video Door Stations, contact Aiphone Technical Support via phone or website.

USA, Canada

- Phone: +1 (800) 692-0200
- Web: <https://www.aiphone.com/home/support>
- Email: tech@aiphone.com

Australia, New Zealand

- Phone: (02) 80364507
- Web: <https://www.aiphone.com.au/>

France

- Phone: 01 69 11 46 00
- Web: <https://www.aiphone.fr/>

Japan

- Phone: 052-228-9961
- Web: <https://www.aiphone.co.jp/>

Singapore

- Phone: 6534-1135
- Web: <http://www.aiphone.com.sg/>
- Email: admin@aiphone.com.sg

United Kingdom

- Phone: 020-7507-6250
- Web: <https://www.aiphone.co.uk/>

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion connected via a SCN trunk and configured via Avaya IP Office Manager (not shown).
- PSTN connectivity provided by a SIP trunk on Avaya IP Office Server Edition and an ISDN-PRI trunk on Avaya IP Office 500 V2 Expansion System.
- Avaya 96x1 Series H.323 Deskphones, Avaya J100 Series SIP Deskphones, Avaya Workplace Client and Avaya Vantage™ registered to Avaya IP Office Server Edition and Avaya IP Office 500 V2 Expansion.
- Aiphone IX-DVM Video Door Station registered to either IP Office Server Edition or IP Office 500 V2 Expansion System (not simultaneously).

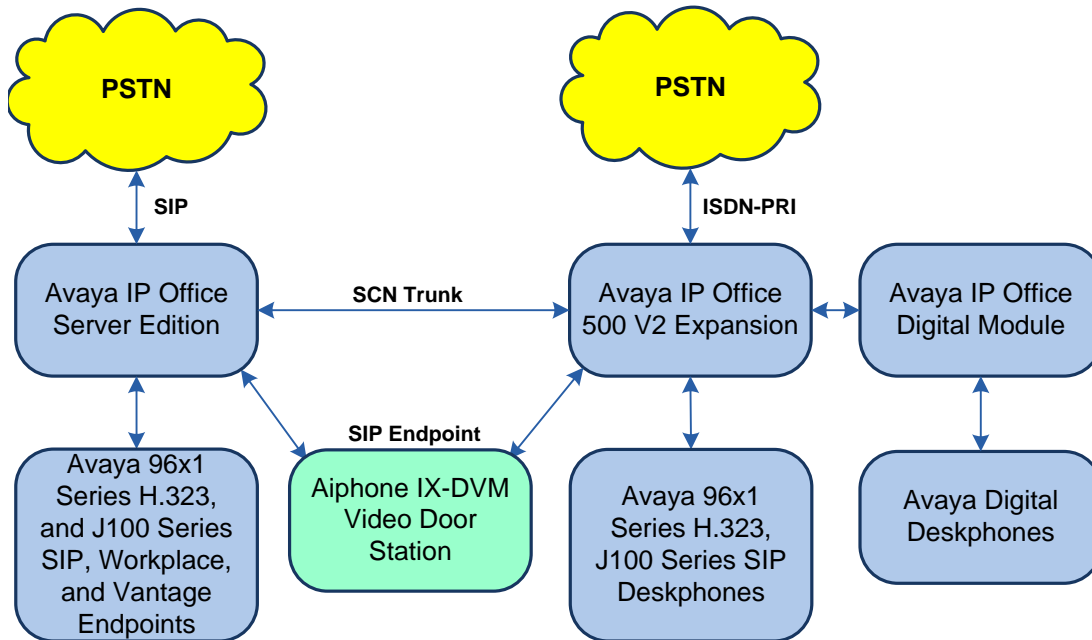


Figure 1: Aiphone IX-DVM Video Door Station with Avaya IP Office

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Server Edition	11.1.1.0.0 build 209
Avaya IP Office 500 V2 Expansion System	11.1.1.0.0 build 209
Avaya Workplace Client for Windows	3.19.0.72.19
Avaya Vantage™ K155	3.0.0.1.0006
Avaya 96x1 Series IP Deskphones	6.8304 (H.323)
Avaya J100 Series Deskphones	4.0.7.0.7 (SIP)
Aiphone IX-DVM Video Door Station	5.75

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office Server Edition

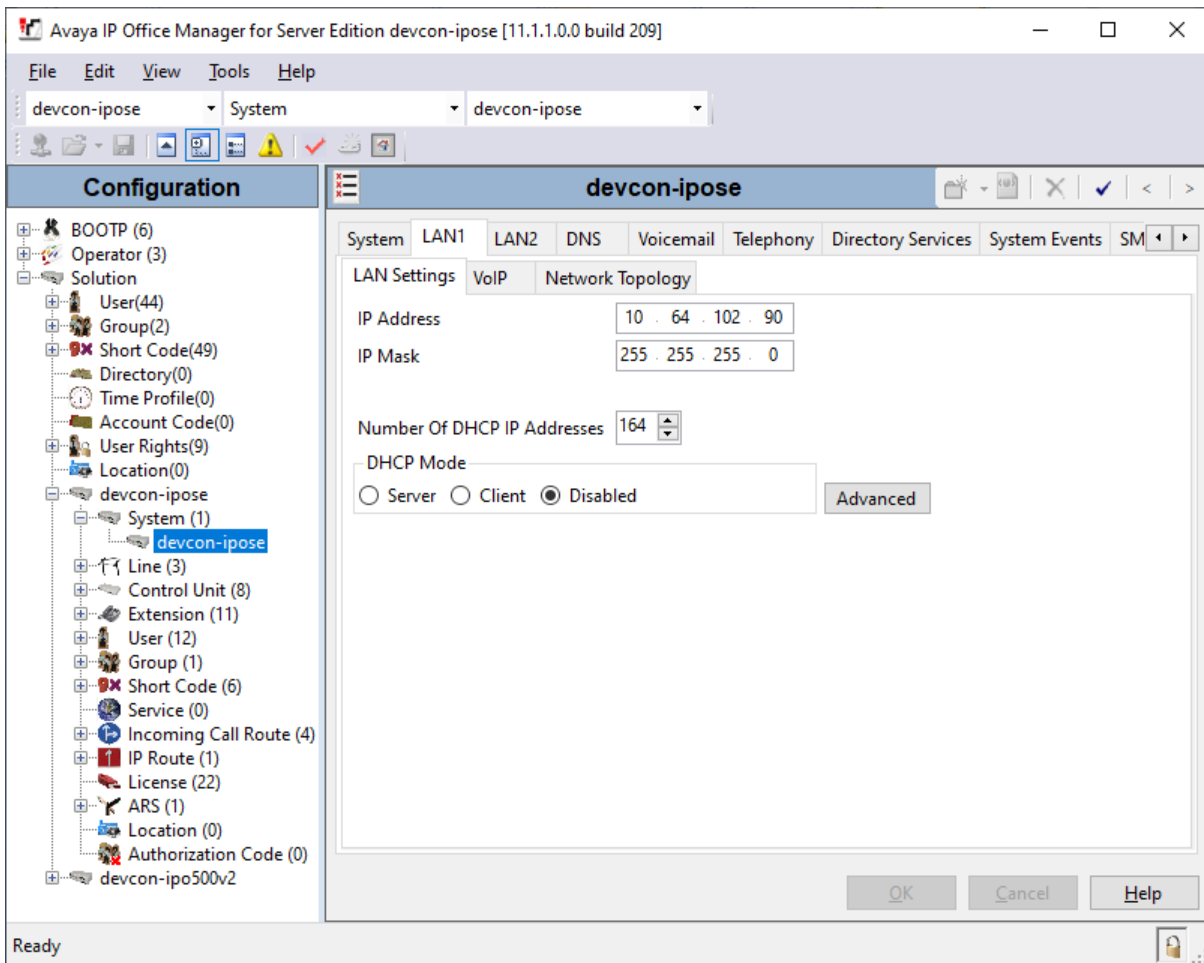
This section provides the procedures for configuring Avaya IP Office Server Edition. The procedures include the following areas:

- Obtain LAN IP Address
- Administer SIP Registrar
- Administer SIP Extension for IX-DVM
- Administer SIP User for IX-DVM

Note: This section covers the configuration of Avaya IP Office Server Edition, but the configuration is the same for Avaya IP Office 500 V2 Expansion System.

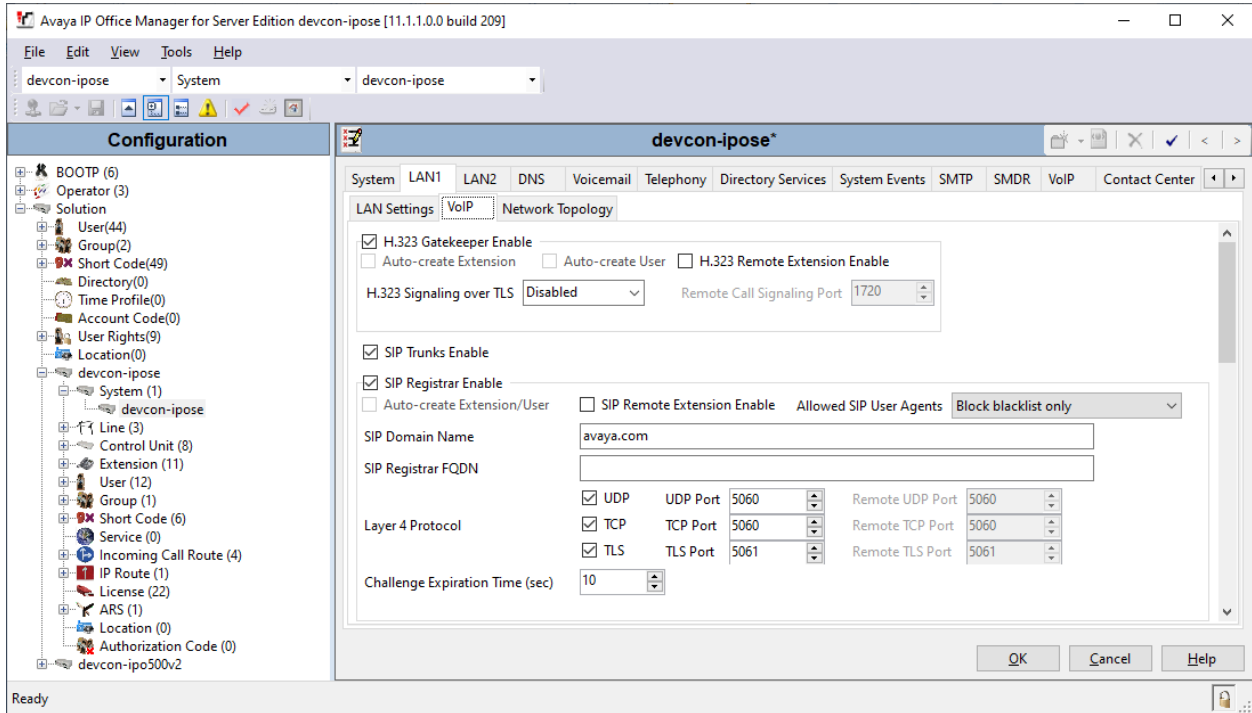
5.1. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **System** screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure IX-DVM.



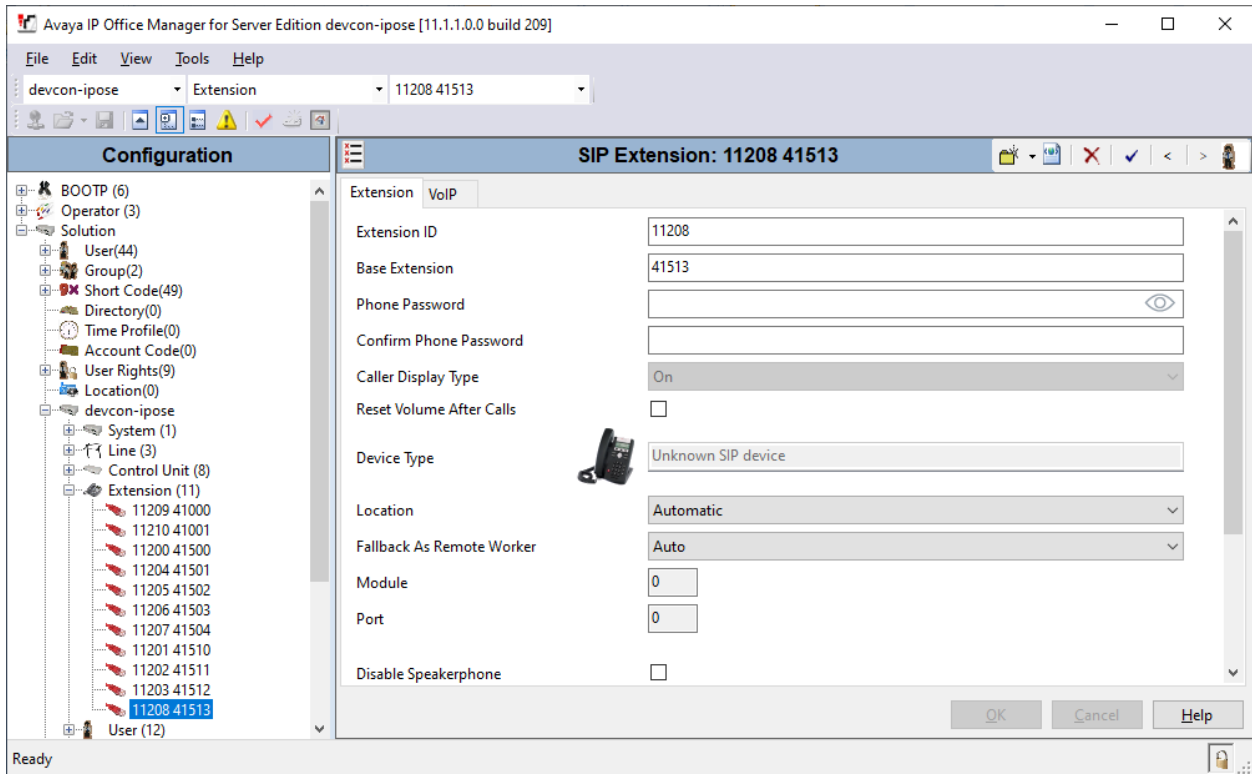
5.2. Administer SIP Registrar

Select the **VoIP** sub-tab. Ensure that **SIP Registrar Enable** is checked and enter a valid **SIP Domain Name**. In the compliance testing, the **SIP Domain Name** field was set to *avaya.com*. UDP transport protocol was enabled for the **Layer 4 Protocol**, which was used by IX-DVM.

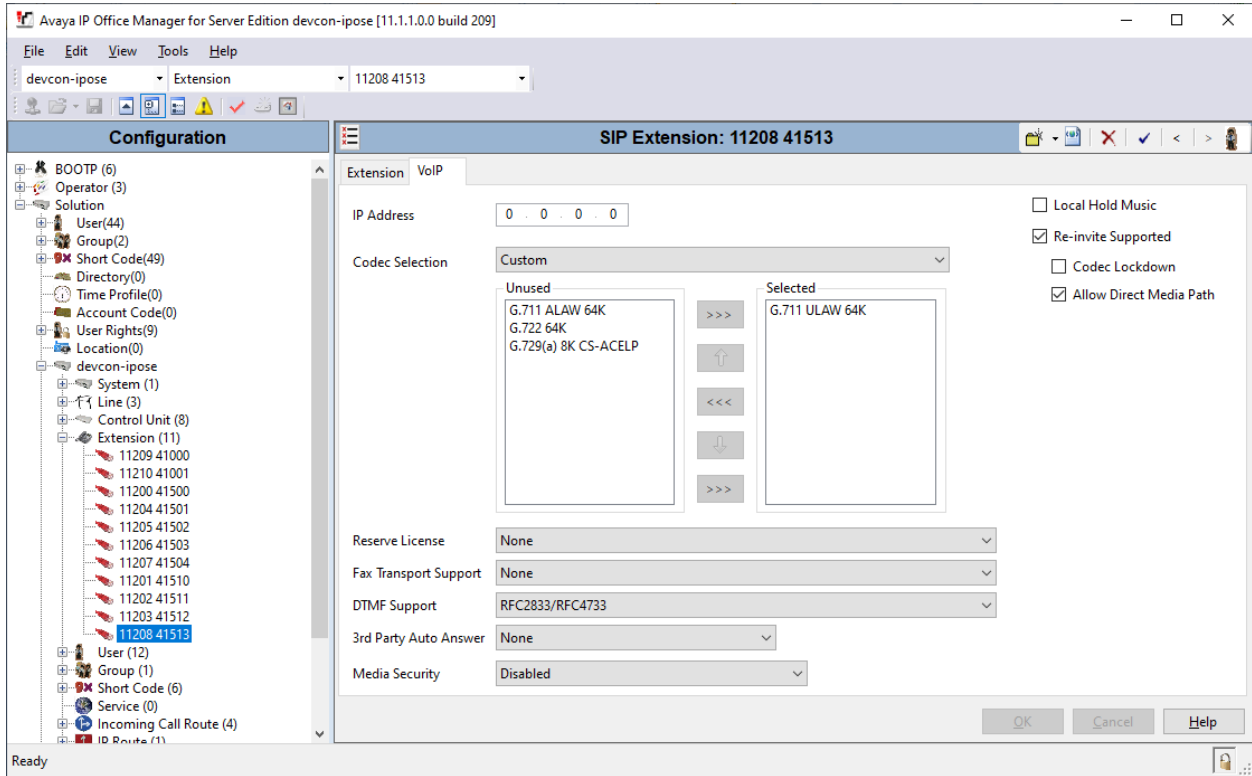


5.3. Administer SIP Extension for IX-DVM

From the configuration tree in the left pane, right-click on **Extension** and select **New → SIP** from the pop-up list to add a new SIP extension. Enter the desired extension for the **Base Extension** field as shown below. In this example, IX-DVM was assigned extension *41513*. This is the extension that IX-DVM will use to register with IP Office Server Edition.



Select the **VoIP** tab and retain the default values. During the compliance test, IX DVM was tested with *G.711 ULaw* codec. Enable **Allow Direct Media Path** so that audio/RTP flows directly between two SIP endpoints without using media resources in Avaya IP Office Server Edition. **Media Security** was *disabled* for IX-DVM.



5.4. Administer SIP User for IX-DVM

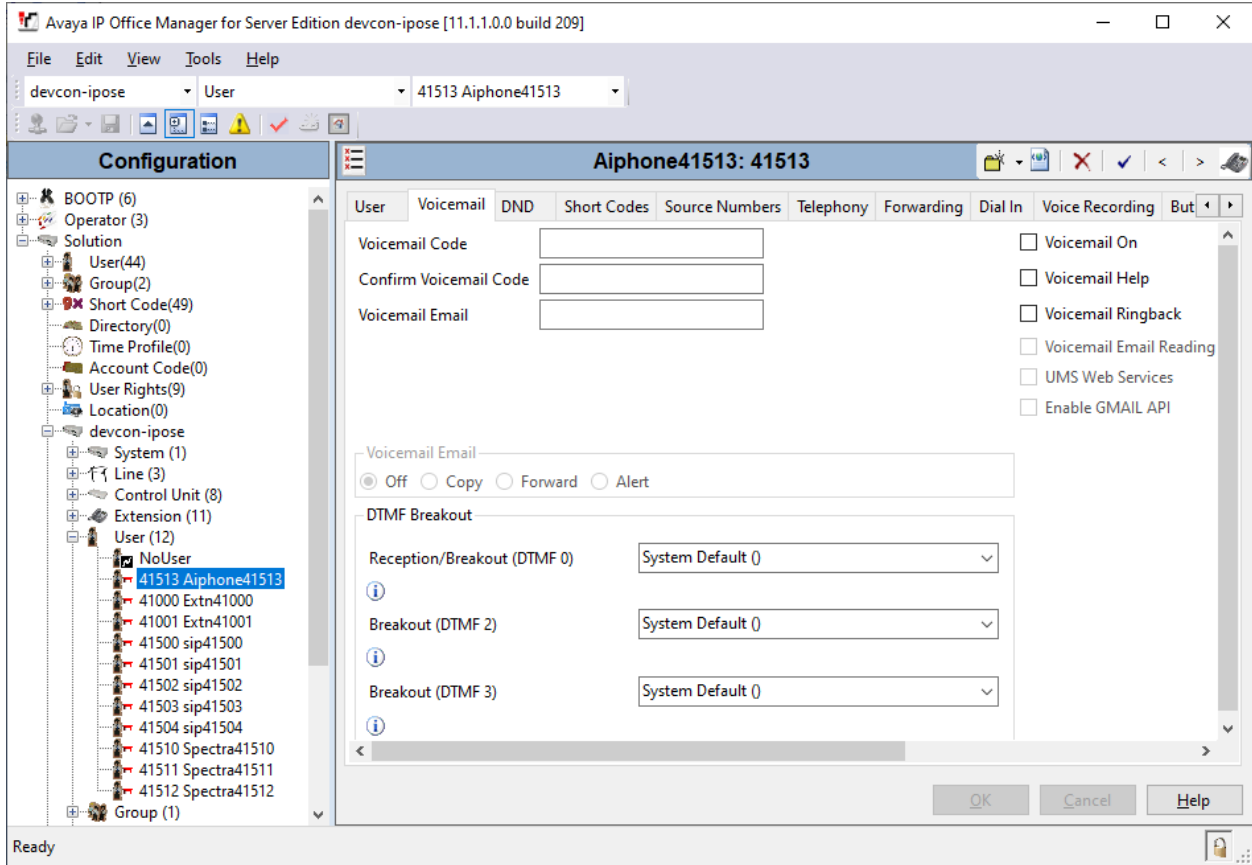
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list. Enter desired values for the **Name** and **Full Name** fields. For the **Extension** field, enter the SIP extension created above.

The screenshot displays the Avaya IP Office Manager for Server Edition interface. The left pane shows the configuration tree with 'User (12)' expanded, and '41513 Aiphone41513' selected. The right pane shows the configuration for this user, with the 'User' tab active. The configuration fields are as follows:

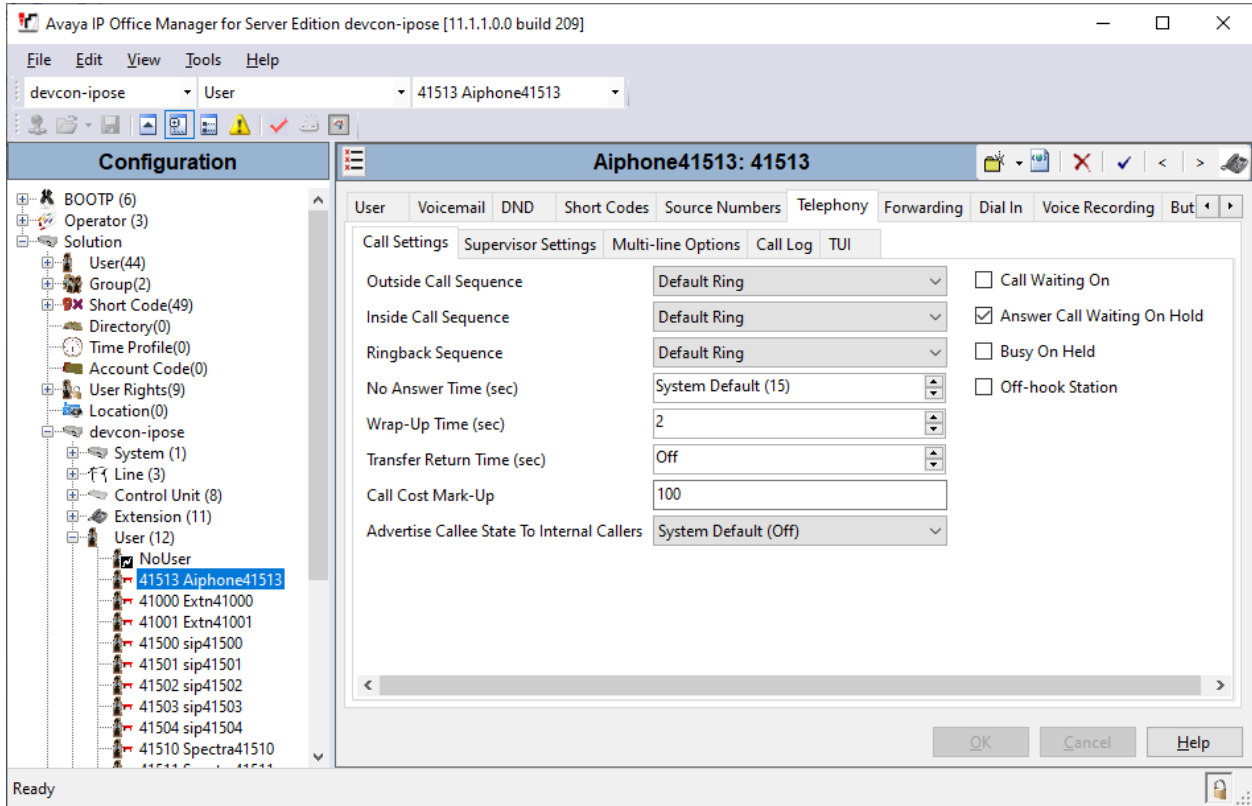
Field	Value
Name	Aiphone41513
Password	
Confirm Password	
Unique Identity	
Conference PIN	
Confirm Audio	
Conference PIN	
Account Status	Enabled
Full Name	Aiphone 41513
Extension	41513
Email Address	
Locale	
Priority	5
System Phone Rights	None
Profile	Basic User
Receptionist	<input type="checkbox"/>
Enable Softphone	<input type="checkbox"/>
Enable one-X Portal Services	<input type="checkbox"/>
Enable one-X TeleCommuter	<input type="checkbox"/>
Enable Remote Worker	<input type="checkbox"/>
Enable Desktop/Tablet VoIP client	<input type="checkbox"/>
Enable Mobile VoIP Client	<input type="checkbox"/>
Send Mobility Email	<input type="checkbox"/>
Web Collaboration	<input type="checkbox"/>
Exclude From Directory	<input type="checkbox"/>
Device Type	Unknown SIP device

At the bottom of the configuration pane, there are buttons for 'OK', 'Cancel', and 'Help'. The status bar at the bottom left shows 'Ready'.

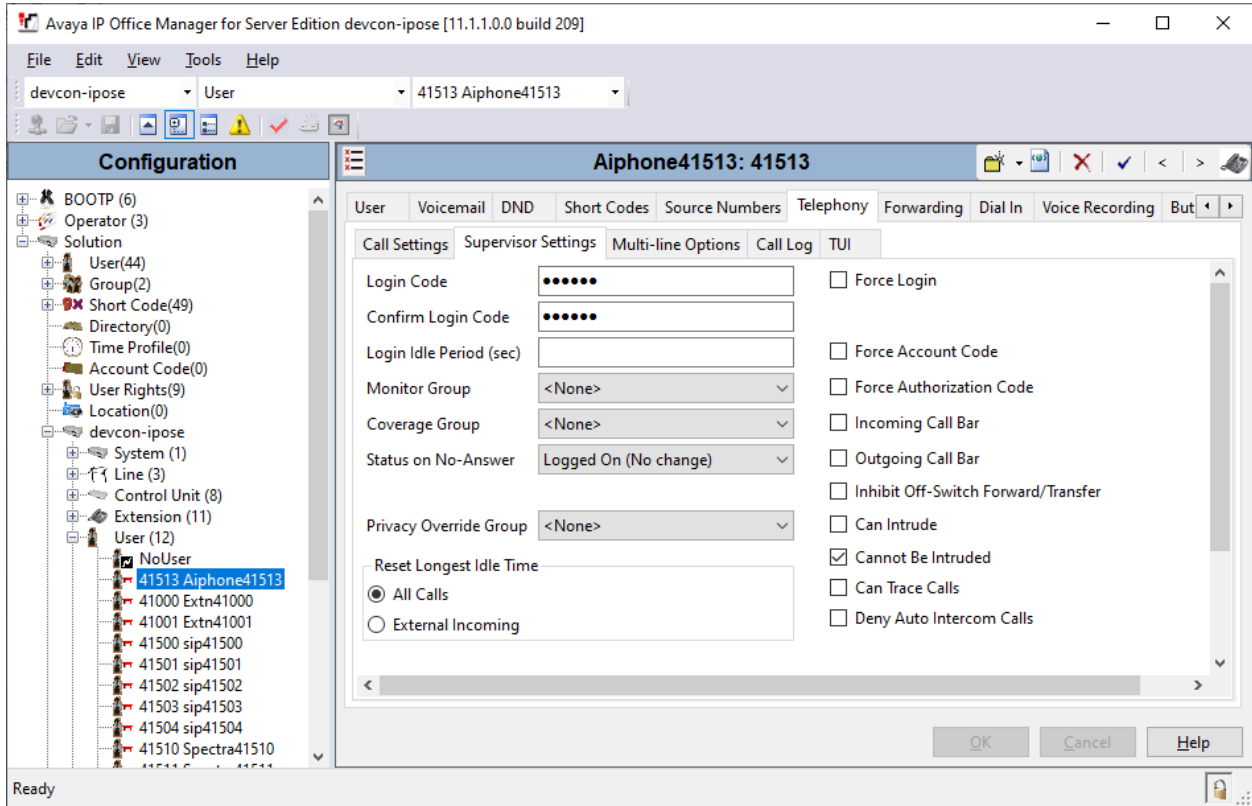
Select the **Voicemail** tab and disable voicemail for IX-DVM.



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings below for the user.



Select the **Supervisor Settings** sub-tab and enter a desired **Login Code**. The **Login Code** is the password that will be used by IX-DVM to register with IP Office Server Edition.



6. Configure Aiphone IX-DVM Video Door Station

This section provides the procedure for configuring IX-DVM to provide SIP connectivity to Session Manager. Configuration of IX-DVM is performed via Aiphone IX System web interface.

- Log into Aiphone IX System Web Interface
- Administer Station Information
- Administer SIP Parameters
- Administer Video SIP Channel
- Administer Audio Settings
- Administer Call Settings

6.1. Log into Aiphone IX System Web Interface

Access the Aiphone IX System Web Interface by using the URL <https://<ip-address>/webset.cgi?login> in an Internet browser, where <ip-address> is the IX-DVM IP address. Select language (not shown) and log in using the appropriate credentials.

AIPHONE IX System

Enter ID and password

ID:

Password:

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6.3. Administer SIP Parameters

Navigate to **Network Settings** → **SIP** from the left pane and configure the following parameters:

- **SIP Signaling Port:** Set to *5060*.
- **User Agent:** Enter desired value (e.g., *IX-DVM*).
- **ID:** Set to SIP extension (e.g., *41513*) from **Section 5.3**.
- **Password:** Enter SIP password from **Section 5.4**.
- **IPv4 Address:** Set to IP Office Server Edition IP address (e.g., *10.64.102.90*).
- **Port:** Set to *5060*.

Click **Update** to save changes.

The screenshot displays the 'AIPHONE IX System Setting' web interface. The top navigation bar includes 'Category: Video Stations' and 'Station Type: IX-DVM'. A blue header bar reads 'Network Settings'. On the left, a sidebar menu lists various settings categories: Station Information, Network Settings, System Information, and Call Settings. The main content area is titled 'SIP' and is divided into three sections: 'SIP Connections', 'SIP Server', and 'Primary Server'. Each section contains configuration fields with their current values and validation rules.

Section	Parameter	Value	Validation Rule
SIP Connections	SIP Signaling Port	5060	1-65535
	User Agent	IX-DVM	1-36 alphanumeric characters
SIP Server	SIP Compatibility Mode	Standard Mode	
	Primary Server		
Primary Server	ID	41513	1-24 alphanumeric characters
	Password	*****	1-24 alphanumeric characters
	IPv4 Address	10.64.102.90	1.0.0.1-223.255.255.254 or host
	IPv6 Address		::FF:0-FE:FF:FFFF:FFFF:FFFF
	Port	5060	1-65535

6.4. Administer Video SIP Channel

Navigate to **Network Settings** → **Video** in the left pane and configure the video settings as shown below.

The screenshot displays the 'AIPHONE IX System Setting' web interface. The left sidebar contains a navigation menu with categories: Station Information, Network Settings, System Information, Call Settings, Option Input / Relay, Output Settings, and Function Settings. The 'Network Settings' category is selected, and the 'Video' sub-option is active. The main content area is titled 'Network Settings' and is divided into several sections:

- Miscellaneous:**
 - Register Transmission Interval [sec]: 3600 (10-14400)
 - DTMF digit interval timeout [sec]: 5 (1-10)
 - Call health check timer: 80-3600 sec (dropdown) | 90 sec (input) | Do not transmit re-INVITE, 80-3600 sec
- Multicast Address:**
 - For Call IPv4: 224.0.0.0-239.255.255.255
 - IPv6: FF10::0-FF1F:FFFF:FFFF:FFFF:FFFF:FFFF:FFFF:FF
- Video:**
 - SIP Channel:**
 - Note: The "SIP Channel" RTP End Port should be greater than 90 digits from the RTP Start Port.
 - Resolution: 320x240(QVGA) 640x480(VGA)
 - Wide View: Enable Disable
 - Frame Rate [fps]: 30 (dropdown)
 - Select Profile: High (dropdown)
 - I-picture interval: 30 (input) | 1-100
 - Bit rate [kbps]: 1024 (dropdown)
 - RTP Start Port: 30000 (input) | 1-65534
 - RTP End Port: 31000 (input) | 1-65535

6.5. Administer Audio Settings

Navigate to **Network Settings** → **Audio** in the left pane and set **Audio Codec** to select *G.711 (u-law)*.

The screenshot displays the AIPHONE IX System Setting web interface. The top navigation bar includes the product name and an 'Update' button. The left sidebar contains a tree view with categories: Station Information, Network Settings, System Information, Call Settings, and Option Input / Relay. The main content area is titled 'Network Settings' and features a sub-section for 'Audio'. This section contains several configuration fields: 'Audio Codec' (radio buttons for G.711(μ-law), G.711(A-law), and G.722), 'Audio RTP Transmission Interval [msec]' (dropdown menu), 'RTP Idle Detection Time [sec]' (input field), 'SIP Channel' (RTP Start and End Port fields), 'ONVIF Transmit Channel' (RTP Start and End Port fields), and 'Audio Buffer' (Packets Buffered at Audio Start and Maximum Packets Buffered dropdown menus). Red text provides warnings about RTP port ranges and codec changes.

Station Information

- Identification
- ID and Password
- Language
- Time
- Expanded System

Network Settings

- IP Address
- DNS
- SIP
- Multicast Address
- Video
- Audio
- Packet Priority
- NTP

System Information

- Custom Sound Registry

Call Settings

- Station Settings
- Called Stations (for Door)
- Call Origination
- Incoming Call
- Contactless Call

Option Input / Relay

Output Settings

- Option Input
- Relay Output

Network Settings

Audio

The "SIP Channel" RTP End Port should be greater than 210 digits from the RTP Start Port.
The "ONVIF Transmit Channel" RTP End Port should be greater than 10 digits from the RTP Start Port.
Changing Audio Codec from G.711(μ-law) / G.711(A-law) to G.722, or from G.722 to G.711(μ-law) / G.711(A-law) will cause the station to restart after Update is clicked

Audio Codec: G.711(μ-law) G.711(A-law) G.722

Audio RTP Transmission Interval [msec]: 20

RTP Idle Detection Time [sec]: 10

This setting is ignored when transmitting to multiple stations (paging, etc.)
10-180 sec

SIP Channel

RTP Start Port: 20000 1-65534

RTP End Port: 21000 1-65535

ONVIF Transmit Channel

RTP Start Port: 22000 1-65534

RTP End Port: 23000 1-65535

Audio Buffer

Packets Buffered at Audio Start: 1

Maximum Packets Buffered: 3 Maximum Packet Buffer must be larger than Audio Start Buffer.

6.6. Administer Call Settings

Navigate to **Call Settings** in the left pane and set the **Call Button Function** to *Call, Answer Call, End Communication* in the **Station Information** section.

In the **Called Stations (for Door)** section, add an entry that specifies the number that should be dialed when the call button is pressed. Set the **Station Number** to the called number (e.g., 41503), set the **IPv4 Address** to IP Office Server Edition IP address (e.g., 10.64.102.90), and set **Station Type** to *VoIP Phone*. Only one VoIP phone may be specified.

AIPHONE IX System Setting
 Category: Video Stations Station Type: IX-DVM Update

Call Settings

Settings updated.

• **Station Information**

Call Button Function: + Required Settings

"Cancel Call, End Communication" disabled when using Option Input call.

• **Called Stations (for Door)**

Option Input #:

Station Number must be 3-5 digits. (3-32 digits for VoIP Phone)
 IPv4 must be 1.0.0.1-223.255.255.254 or hostname(1-64 alphanumeric characters).
 IPv6 must be ::FF:0:FEFF:FFFF:FFFF:FFFF:FFFF:FFFF or hostname(1-64 alphanumeric characters).
 Enter SIP Primary Server IP address for VoIP Phone, set only one VoIP Phone per call group.
 Station Type must be "VoIP Phone" when calling via SIP server.
 U = Unicast, M = Multicast
 If designating "M", multicast IP addresses must be configured for the station(s).

#	Station Number	IPv4 Address	IPv6 Address	Station Type	Protocol
1	41503	10.64.102.90		VoIP Phone	U
2					
3					

6.7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya IP Office Server Edition and Aiphone IX-DVM Video Door Station.

1. Verify that IX-DVM has successfully registered with IP Office Server Edition. Launch **IP Office System Status** and navigate to **Extensions** → **<SIP Extension>**, where **<SIP Extension>** is the IX-DVM extension. Verify that the **Current State** is *Idle* as shown below.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - devcon-ipose (10.64.102.90) - IP Office Linux PC 11.1.1.0.0 build 209". The AVAYA logo is in the top left, and "IP Office System Status" is in the top right. Below the logo is a menu bar with "Help", "Snapshot", "LogOff", "Exit", and "About".

The left sidebar contains a tree view with the following items: System, Alarms (7), Extensions (5), Trunks (3), Active Calls, Resources, Voicemail, IP Networking, and Locations. The "Extensions (5)" folder is expanded, and extension "41513" is selected.

The main content area is titled "Extension Status" and displays the following details for extension 41513:

Extension Number:	41513
IP address:	192.168.100.180
Standard Location:	None
Registrar:	Primary
Telephone Type:	Unknown SIP Device
User-Agent SIP header:	IX-DVM
Media Stream:	RTP
Layer 4 Protocol:	UDP
Current User Extension Number:	41513
Current User Name:	Aiphone41513
Forwarding:	Off
Twinning:	Off
Do Not Disturb:	Off
Message Waiting:	Off
Phone Manager Type:	None
SIP Device Features:	
License Reserved:	No
Last Date and Time License Allocated:	7/19/2021 11:56:09 AM
Packet Loss Fraction:	
Jitter:	
Round Trip Delay:	
Connection Type:	
Codec:	
Remote Media Address:	

Below the details is a table with the following columns: Call Ref, Current State, Time in State, Calling Number or Called Direction Number, and Other Party on Call. The table contains one row with the following data:

Call Ref	Current State	Time in State	Calling Number or Called Direction Number	Other Party on Call
	Idle	00:02:58		

At the bottom of the application, there are several buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time "11:59:07 AM" and the status "Online".

2. Establish inbound and outbound video calls to IX-DVM with Workplace and/or Vantage and verify two-way audio and one-way video.

7. Conclusion

These Application Notes describe the administration steps required to integrate Aiphone IX Series 2 Video Door Stations (IX-DVM) with Avaya IP Office Server Edition. The Aiphone IX-DVM Video Door Station successfully registered with Avaya IP Office Server Edition as a SIP endpoint and audio and video calls were verified. All test cases passed.

8. Additional References

This section references the Avaya and Aiphone documentation relevant to these Application Notes.

- [1] *Administering Avaya IP Office™ Platform Manager*, Release 11.1.1, Issue 29, February 2021.
- [2] *Aiphone IX-DVM Video Door Station Installation Manual*, Issue Date: Oct.2021, available from Aiphone.
- [3] *Aiphone IX Series Operation Manual*, Software version 5.75 or later, available from Aiphone.

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