



Avaya Solution & Interoperability Test Lab

Application Notes for ClearOne Converge Pro 2 48VT with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate ClearOne Converge Pro 2 48VT 8.5.15.0 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. ClearOne Converge Pro 2 48VT is a conferencing solution that can register to Avaya Aura® Session Manager as a SIP user. This solution also includes the ClearOne Converge Pro 2 Dialer for establishing calls and the ClearOne Converge Pro 2 Console AI for configuring the system.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 ClearOne Converge Pro 2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. ClearOne Converge Pro 2 provides a conferencing solution for any size venue and can register with Avaya Aura® Session Manager as a SIP user. This solution also includes the ClearOne Converge Pro 2 Dialer for establishing calls and the ClearOne Converge Pro 2 Console AI for configuring the system. For the compliance test, ClearOne Converge Pro 2 48VT was used, but these Application Notes also apply to other ClearOne Converge Pro models, which use the same SIP-based firmware. See Attachment 1 for additional details.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Converge Pro 2 48VT, Avaya SIP / H.323 desk phones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer and conference. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

The serviceability testing focused on verifying that Converge Pro 2 48VT comes back into service after re-connecting it to the telephony network and rebooting it.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and Converge Pro 2 48VT did not include use of any specific encryption features as requested by ClearOne.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP user registration of Converge Pro 2 48VT with Session Manager.
- Calls between Converge Pro 2 48VT and Avaya SIP / H.323 desk phones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between Converge Pro 2 48VT and the PSTN.
- G.711 and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, blind and attended call transfer and attended 3-party conference.
- Extended telephony features using Communication Manager FACs and FNEs for Call Forward, Call Park/Unpark, and Call Pickup.
- Use of programmable buttons on Converge Pro 2 48VT.
- Proper system recovery after a restart of Converge Pro 2 48VT and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations:

- Music on hold treatment is not applied to the H323 endpoint when it initiates a call to Converge Pro 2 48VT and then Converge Pro2 48VT initiates the hold. This occurs only when shuffling is administered. Audio call paths do function correctly. In this scenario, the long hold recall timer audio notification does not occur.
- Blind conference is currently not supported.
- When the Converge Pro 2 48VT user initiates an attended call transfer, that user cannot return to the held call while the call transfer is in progress. The Converge Pro 2 48VT user must either terminate the call to the transfer-to party or complete the call transfer.
- When the user goes off-hook on Converge Pro 2 48VT, and then dials the call, if the dialed digit format is not included in the Converge Pro 2 48VT dial plan, Converge Pro 2 48VT will attempt to establish the call by sending an INVITE message, but the call will fail. In this case, the user should terminate the call and retry the call with valid digits or modify the dial plan (see **Section 7.4**), as necessary.

2.3. Support

For technical support and information on Converge Pro 2, contact ClearOne General Technical Support at:

- Phone: +1-801-974-3760
- Website: <https://www.clearone.com/contact-support>
- Email: audiotechsupport@clearone.com

3. Reference Configuration

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

- Communication Manager running in a virtual environment.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Session Manager connected to the simulated PSTN via SIP trunk to Avaya Session Border Controller for Enterprise.
- System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 IP Deskphones and J100 Series SIP IP Phones.
- Converge Pro 2 48VT, Converge Pro 2 Dialer for establishing calls, and Converge Pro 2 Console AI for configuring the unit.

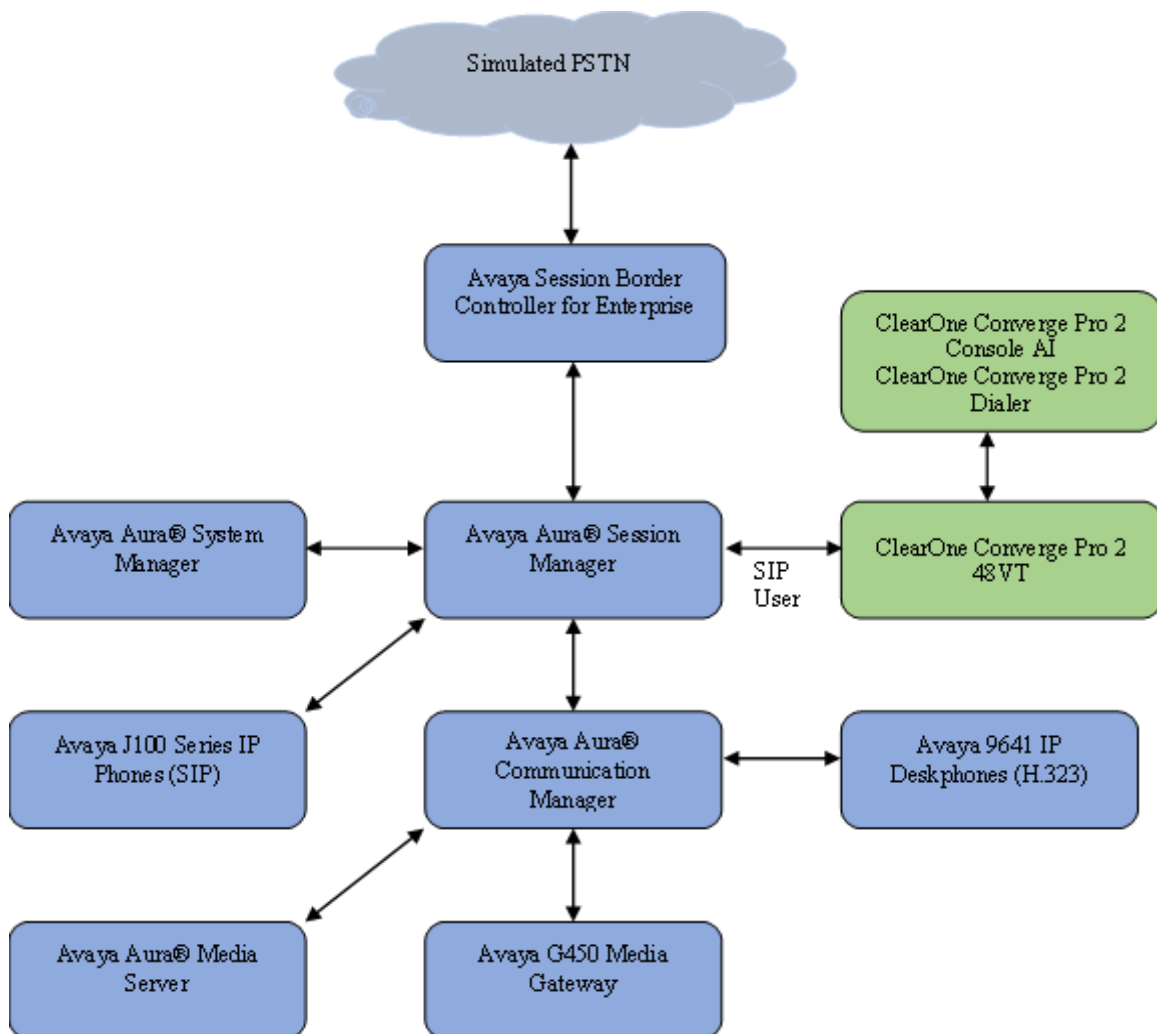


Figure 1: Avaya SIP Network with ClearOne Converge Pro 48VT

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 Virtual Machine	01.0.890.0-27168
Avaya Aura® Session Manager running on Virtual Machine	8.1.3.3_813310
Avaya Aura® System Manager running on Virtual Machine	8.1.3.3.1013529
Avaya Session Border Controller for Enterprise running on Virtual Machine	8.1.3.0-31-21052
Avaya G450 Media Gateway	41.34.3
Avaya Aura® Media Server	8.0.2.200
Avaya 9641G IP Deskphone	6.8511 (H.323)
Avaya J179 IP Phone	4.0.9.0.4 (SIP)
ClearOne Converge Pro 2 48VT (see note)	8.5.15.0
ClearOne Converge Pro 2 Dialer Windows client	2.0.5.4
ClearOne Converge Pro 2 Console AI Windows client	7.5.4.1

Note: These Application Notes also apply to the ClearOne Converge Pro 48VTD, 128VT, and 128VTD products, which use the same SIP-firmware and support the Converge Pro 2 Dialer for establishing calls and the Converge Pro 2 Console AI software for configuring the systems. The differences between the Converge Pro 2 products are as follows:

- Converge Pro 2 48VT (**tested model**). This model has 4 Mic/Line Inputs, 8 Mic/Line Outputs and supports VoIP (SIP)
- Converge Pro 2 48VTD. This model has 4 Mic/Line Inputs, 8 Mic/Line Outputs and supports VoIP (SIP)
- Converge Pro 2 128VT. This model has 12 Mic/Line Inputs, 8 Mic/Line Outputs and supports VoIP (SIP).
- Converge Pro 2 128VTD. This model has 12 Mic/Line Inputs, 8 Mic/Line Outputs and supports VoIP (SIP).

For additional details, contact ClearOne Technical Support.

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Network Region
- Administer IP Codec Set

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

Note: It is assumed that basic configuration of Communication Manager has already been completed, such as the SIP trunk to Session Manager. However, implementers should ensure sufficient Maximum Administered SIP Trunks licenses are available to accommodate the traffic between Communication Manager and Session Manager. The SIP station configuration for Converge Pro 2 48VT is configured through System Manager in **Section 6.1**.

5.1. Verify Communication Manager license

Using the SAT, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V18                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                          Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000      255
Maximum Stations: 150              120
Maximum XMOBILE Stations: 36000    0
Maximum Off-PBX Telephones - EC500: 150    0
Maximum Off-PBX Telephones - OPS: 150    65
Maximum Off-PBX Telephones - PBFMC: 150    0
Maximum Off-PBX Telephones - PVFMC: 150    0
Maximum Off-PBX Telephones - SCCAN: 0      0
Maximum Survivable Processors: 313      0

(NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Network Region

In the **ip-network-region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.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 G450 Media Gateway or Media Server. The **ip-network-region** form also specifies the **Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                     Page 1 of 20

                                IP NETWORK REGION
Region: 1              NR Group: 1
Location: 1           Authoritative Domain: avaya.com
Name: Main            Stub Network Region: n
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? n
                      UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
H.323 IP ENDPOINTS    AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y      RSVP Enabled? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

5.3. Administer IP Codec Set

In the **ip-codec-set** form, select the audio codec type supported for calls routed over the SIP trunk to Converge Pro 2 48VT. Note that IP codec set **1** was specified in IP Network Region **1** shown above. The settings of the **ip-codec-set** form are shown below. Ensure **none** is one of the **Media Encryption** options offered. Converge Pro 2 48VT was tested using G.711 and G.722 codecs.

```
change ip-codec-set 1                                         Page 1 of 2

                                IP MEDIA PARAMETERS
Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size (ms)
1: G.711MU      n          2          20
2: G.711A       n          2          20
3: G.722-64K    n          2          20
4:
5:
6:
7:

Media Encryption      Encrypted SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: none
```

6. Configure Avaya Aura® Session Manager

This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Converge Pro 2 48VT
- Administer SIP User

Note: It is assumed that Session Manager and Communication Manager SIP trunk connections are configured. This section will focus on the configuration of a SIP user for Converge Pro 2 48VT.

6.1. Launch System Manager

Access System Manager web interface by entering **http://<ip-address>/SMGR** in a web browser, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials.

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:

Password:

[Change Password](#)

Supported Browsers: Internet Explorer 11.x or Firefox (minimum version 65.0).

6.2. Set Network Transport Protocol for Converge Pro 2 48VT

From the System Manager **Home** screen, select **Elements** → **Routing** → **SIP Entities** and edit the SIP Entity for Session Manager shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Routing' tab is selected. The 'SIP Entity Details' form for 'sm81' is displayed. The 'General' section contains the following fields:

- Name:** sm81
- IP Address:** 10.64.110.212
- SIP FQDN:**
- Type:** Session Manager
- Notes:**
- Location:** DevConnect
- Outbound Proxy:**
- Time Zone:** America/Denver
- Minimum TLS Version:** Use Global Setting
- Credential name:**

Buttons for 'Commit' and 'Cancel' are visible at the top right of the form.

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Converge Pro 2 48VT is specified in the list below. For the compliance test, Converge Pro 2 48VT used *TCP* network transport.

Listen Ports

Add Remove					
4 Items Filter: Enable					
<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5062	TLS	avaya.com	<input type="checkbox"/>	
Select : All, None					

6.3. Administer SIP Users

A SIP user must be configured for Converge Pro 2 48VT by the following steps. This configuration is automatically synchronized with Communication Manager. In Session Manager, select **Users** → **User Management** → **Manage Users** to display the **User Management** screen (not shown). Click + **New** to add a user.

6.3.1. Identity

Enter values for the following required attributes for a new SIP user in the **User Profile** screen:

- **Last Name:** Enter the last name of the user, e.g., **ConvergePro2**.
- **First Name:** Enter the first name of the user, e.g., **User**.
- **Login Name:** Enter <extension>@<sip domain> of the user (e.g., **70130@avaya.com**).

The screenshot displays the Avaya Aura System Manager 8.1 interface for adding a new user profile. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, along with a search bar and a user profile icon labeled 'admin'. The main navigation pane on the left shows 'Home', 'Session Manager', and 'User Management' (selected). The 'User Profile | Add' form is shown with tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Identity' tab is active, displaying various input fields. The 'User Provisioning Rule' is set to a dropdown menu. The 'Last Name' field is filled with 'ConvergePro2', and the 'First Name' field is filled with 'User'. The 'Login Name' field is filled with '70130@avaya.com'. The 'Description' field is filled with 'Description Of User'. The 'Password' and 'Confirm Password' fields are empty. The 'Last Name (in Latin alphabet characters)' field is filled with 'ConvergePro2', and the 'First Name (in Latin alphabet characters)' field is filled with 'User'. The 'Middle Name' field is filled with 'Middle Name Of User'. The 'Email Address' field is filled with 'Email Address Of User'. The 'User Type' dropdown menu is set to 'Basic'. The 'Localized Display Name' field is filled with 'Localized Display Name Of'.

6.3.2. Communication Address

Select the **Communication Profile** tab. Select **Communication Address** in the left list and click + **New** (not shown).

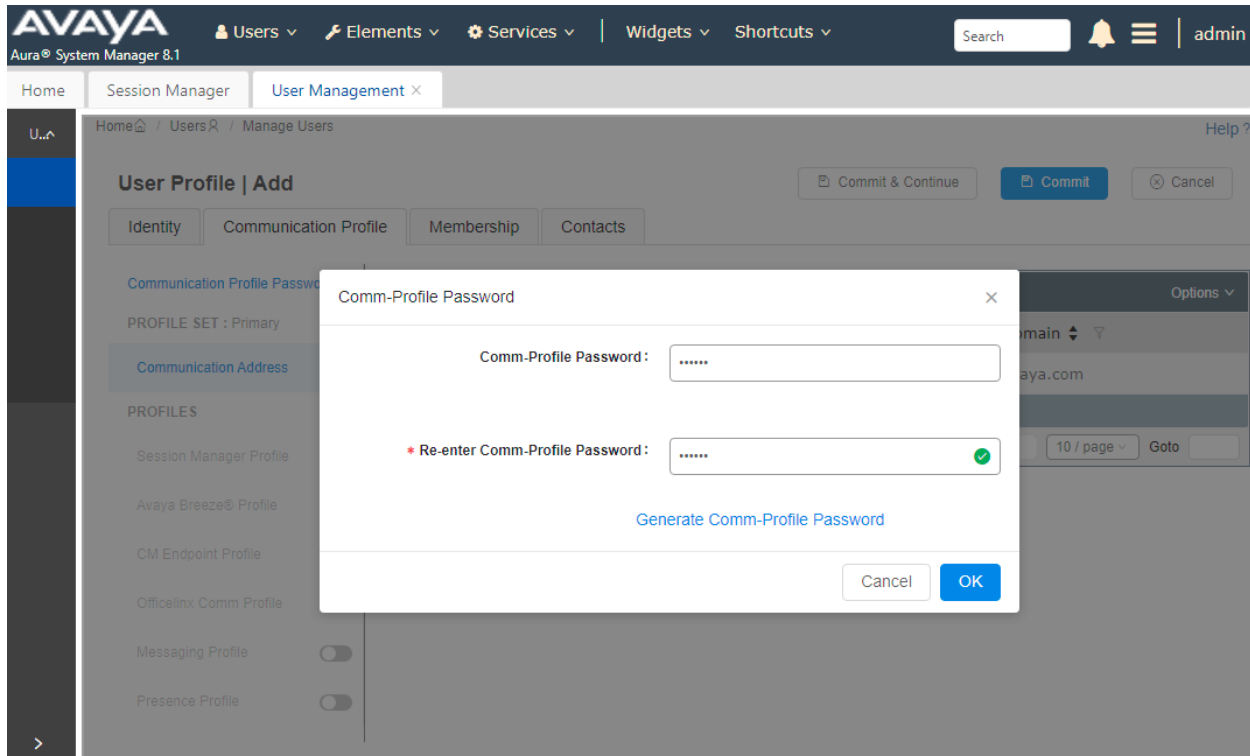
Enter the following attributes for the **Communication Address**:

- **Type:** Select **Avaya SIP** from the drop-down list.
- **Fully Qualified Address:** Enter the extension number (e.g., **70130**).
- **Domain:** Enter the domain (e.g., **avaya.com**).

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and tabs for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'User Profile | Add' and contains tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a list of profiles on the left and a 'Communication Address' section. A modal dialog box titled 'Communication Address Add/Edit' is open, featuring a 'Type' dropdown set to 'Avaya SIP' and a 'Fully Qualified Address' field with '70130' and 'avaya.com' entered. The dialog has 'Cancel' and 'OK' buttons at the bottom.

6.3.3. Communication Profile Password

Select **Communication Profile Password** on the left and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a password. This will be used to register the device. Click **OK**.



The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 8.1', and tabs for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section. A modal dialog box is open over this section, titled 'Comm-Profile Password'. It contains two password input fields: 'Comm-Profile Password' and 'Re-enter Comm-Profile Password'. The 'Re-enter' field has a green checkmark, indicating the passwords match. Below the fields is a link 'Generate Comm-Profile Password'. At the bottom of the dialog are 'Cancel' and 'OK' buttons. The background shows a list of profiles on the left and a table of users on the right.

6.3.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, "Aura® System Manager 8.1", and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile icon are also present. The main navigation bar shows "Home", "Session Manager", and "User Management". The left sidebar contains a "U..." button and a "Session Manager Profile" toggle switch, which is currently turned on. The main content area is titled "User Profile | Add" and includes tabs for Identity, Communication Profile, Membership, and Contacts. The "Communication Profile" tab is active, showing a "Communication Profile Password" field and a "PROFILE SET : Primary" dropdown. Below this, there are several profile toggles: Avaya Breeze® Profile, CM Endpoint Profile, OfficeInx Comm Profile, Messaging Profile, and Presence Profile. The "SIP Registration" section contains fields for "Primary Session Manager" (set to sm81), "Secondary Session Manager" (set to Start typing...), "Survivability Server" (set to Start typing...), "Max. Simultaneous Devices" (set to 1), and a checkbox for "Block New Registration When Maximum Registrations Active?". The "Application Sequences" section contains fields for "Origination Sequence" and "Termination Sequence", both set to cm81. At the top right of the main content area, there are buttons for "Commit & Continue", "Commit", and "Cancel".

6.3.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- **System:** Select the relevant Communication Manager SIP Entity (e.g., **cm81**).
- **Profile Type:** Select **Endpoint**.
- **Template:** Select **J179_DEFAULT_CM_8_1**.
- **Extension:** Enter the extension number (e.g., **70130**).

Click on **Endpoint Editor** in the **Extension** field to edit Communication Manager settings. Input the appropriate **Coverage Path** (not shown) configured to route unanswered calls to voicemail. Click **Done** to close the Endpoint Editor. Click **Commit** (not shown).

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, "Aura® System Manager 8.1", and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are on the right. The main content area is titled "User Profile | Add" and has tabs for Identity, Communication Profile, Membership, and Contacts. The "Communication Profile" tab is active. On the left, a sidebar shows profile sets: Primary, Session Manager Profile, Avaya Breeze® Profile, CM Endpoint Profile (selected), OfficeInx Comm Profile, Messaging Profile, and Presence Profile. The main form fields include: System (cm81), Profile Type (Endpoint), Extension (Start typing...), Template (J179_DEFAULT_CM_8_1), Set Type (J179), Security Code (Enter Security Code), Port (JP), Voice Mail Number, Preferred Handle (Select), Sip Trunk, SIP URI (Select), Delete on Unassign from User or on Delete User (checked), Override Endpoint Name and Localized Name (checked), and Allow H.323 and SIP Endpoint Dual Registration (unchecked). Buttons for "Commit & Continue", "Commit", and "Cancel" are at the top right.

7. Configure ClearOne Converge Pro 2 48VT

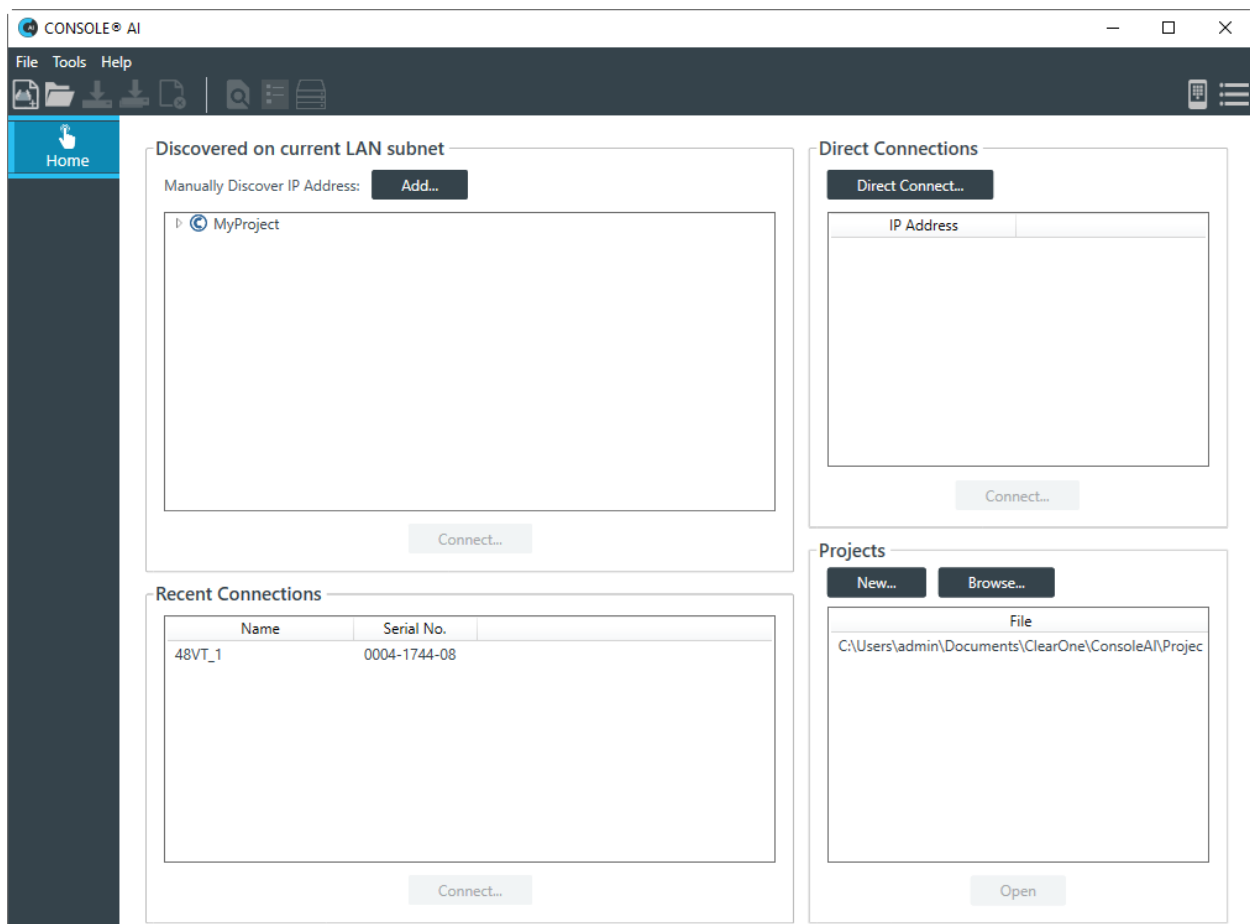
This section covers the Converge Pro 2 48VT configuration using the Converge Pro 2 Console AI. The procedure covers the following areas:

- Launch Converge Pro 2 Console AI
- Configure Channels and Routes
- Administer IP Settings
- Administer VoIP Properties
- Administer IP Phones
- Load Configuration to Converge Pro 2 48VT

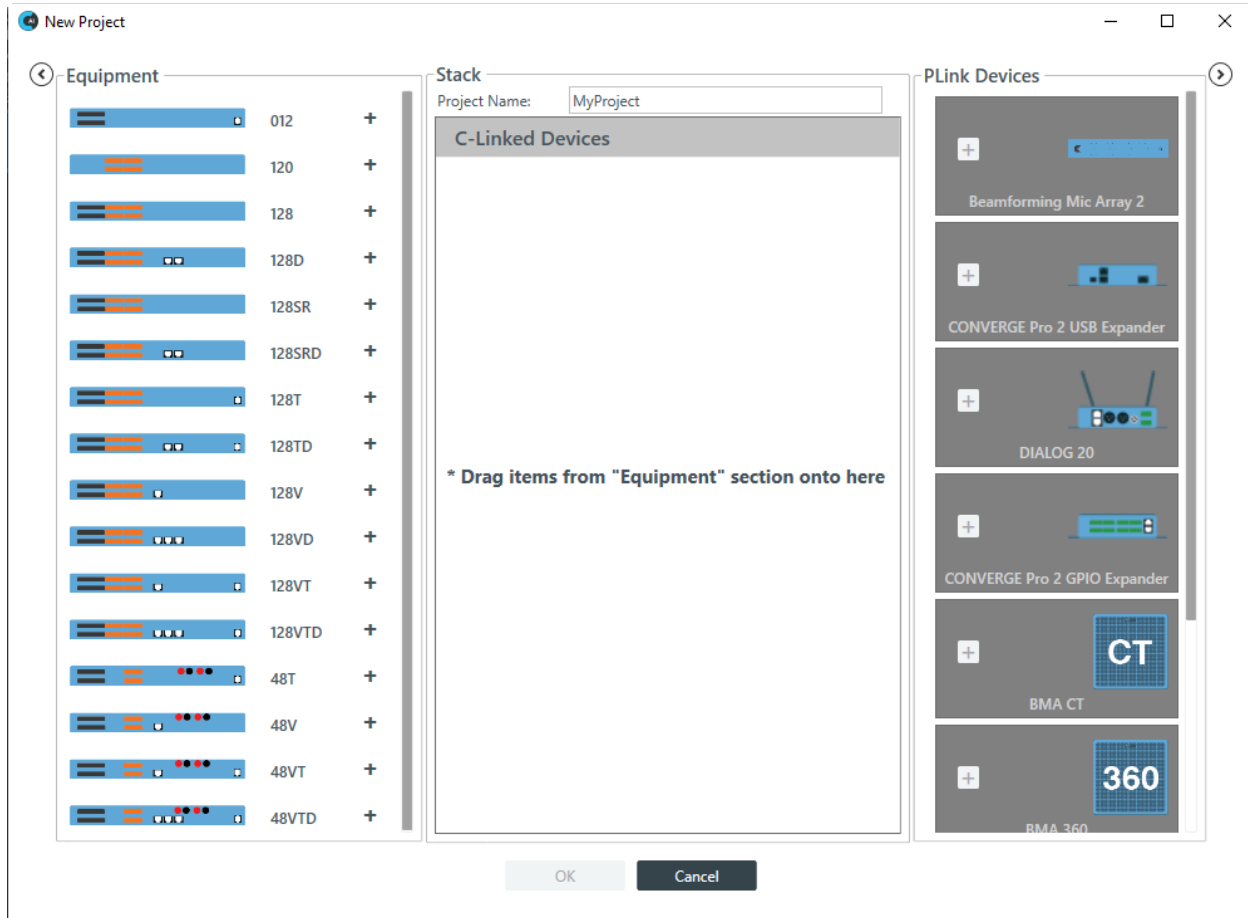
Refer to [3] and [4] for more information on configuring Converge Pro 2 48VT.

7.1. Launch Converge Pro 2 Console AI

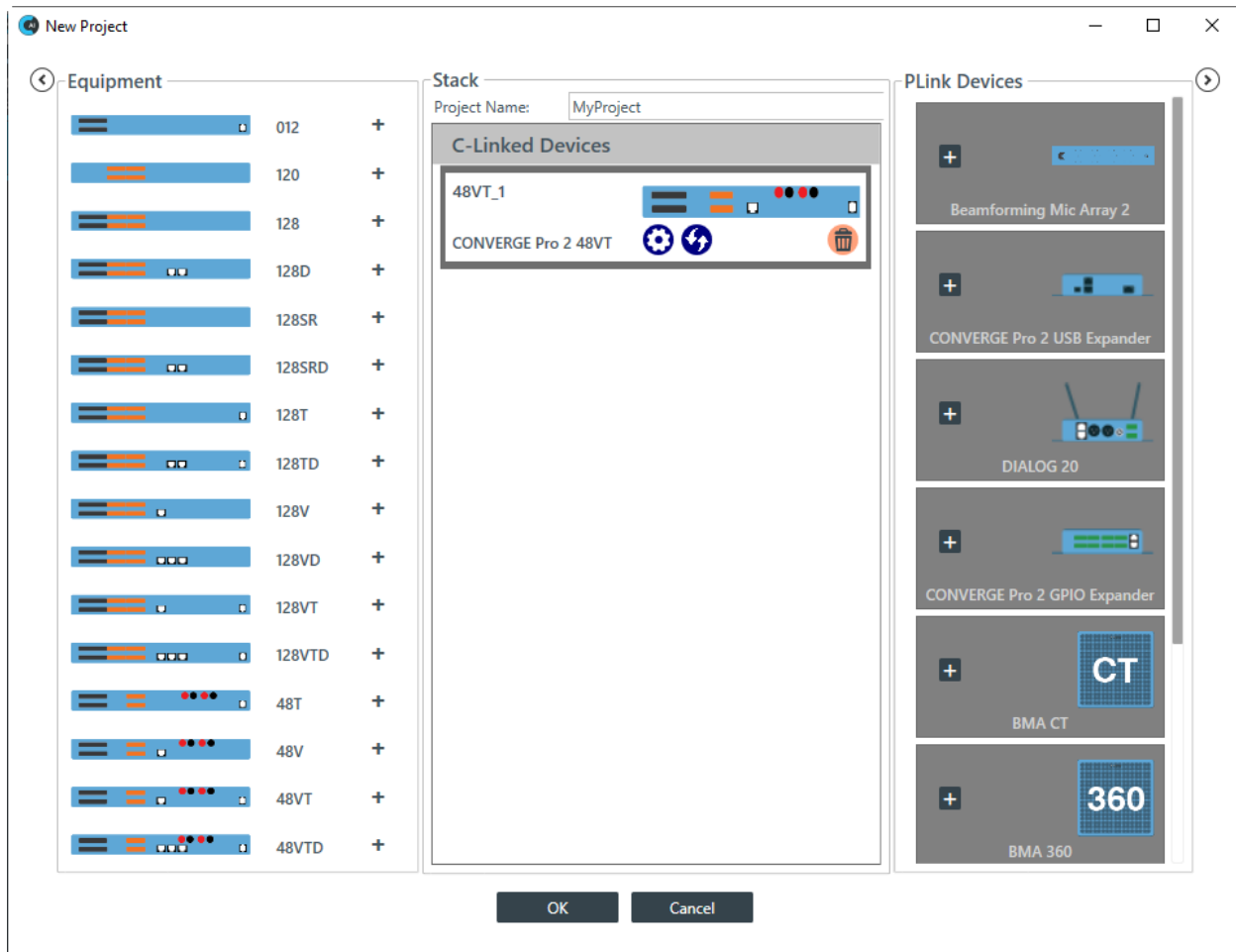
Converge Pro 2 48VT is configured using Converge Pro 2 Console AI which is installed on a Windows client. Launch Converge Pro 2 Console AI from Windows to display the window shown below.



To configure Converge Pro 2 48VT, either start a new project or open an existing project by selecting the appropriate project in the **Projects** section and then clicking the **Open** button. When a new project is opened, the Converge Pro 2 Console AI window will appear as shown below with **New Project** displayed at the top of the window.



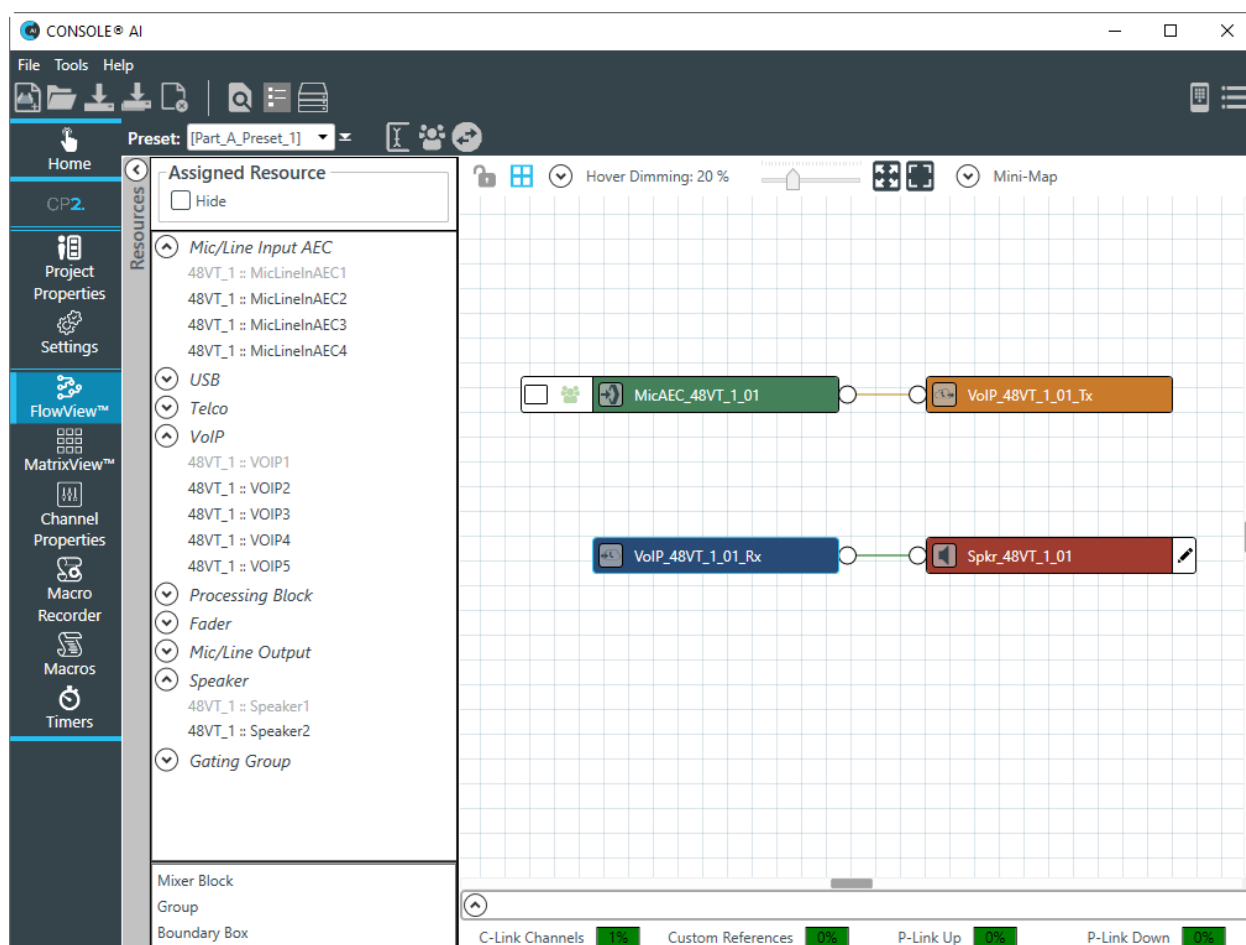
Select the appropriate **Equipment** (e.g., **48VT**) and drag it to the **Stack**. Click **OK**.



7.2. Configure Channels and Routes

The **FlowView** display should be selected which has available resources on the left side of the visualization. Resources can be dragged to the grid. Drag a **Mic/Line Input**, a **VoIP** (a transmit and receive object), and a **Speaker** object and arrange them on the grid.

Create routes between the resources by dragging the output connector (the circle outside of the resource) to the other resource. Connect the **Mic/Line Input** (from the physical microphone) to the **VoIP Tx** resources, and connect the **VoIP Rx** and **Speaker** (to the external speaker) resources in a similar manner.



7.3. Administer IP Settings

Converge Pro 2 48VT may acquire its IP network settings through DHCP or through manual configuration using a static IP address. For the compliance test, a static IP address was used. To configure the IP settings, click on the **Device** icon on the toolbar (shown on previous page). The Device window is displayed (not shown) with the 48VT equipment shown. Click on the **Settings** icon (gear). The **Device Settings** window is displayed as shown below with the **Network** tab and **Ethernet Port** sub tab active. Fill out the following fields.

- **IP Address:** Enter the Converge Pro 2 48VT IP address (e.g., **192.168.4.160**).
- **Subnet Mask:** Enter **255.255.255.0**.
- **Gateway:** Enter the relevant gateway IP (e.g., **192.168.4.1**).
- **DNS Address:** Enter the appropriate DNS server IP.

The screenshot shows the 'Device Settings' window with the 'Network' tab selected. Under the 'Ethernet Port' sub-tab, the 'Use Static IP' option is selected. The following fields are populated:

Field	Value
IP Address:	192.168.4.160
Subnet Mask:	255.255.255.0
Gateway:	192.168.4.1
DNS Address 1:	
DNS Address 2:	

* The DNS and Gateway Addresses from the Ethernet Port will be used on all network connections

Close

7.4. Administer VoIP Properties

In the **VoIP** tab, configure timers, audio codecs, dial plan, and SIP proxy. In the **VoIP→Timers** sub-tab, verify that the SIP timers are configured as desired. The default values shown below were used for the compliance test.

The screenshot shows the 'Device Settings' window with the 'VoIP' tab selected. The 'Timers' sub-tab is active, displaying a list of SIP-related settings. A note at the top states: 'If unit does not have a dedicated VoIP port voice traffic will be routed through the Ethernet port. The VoIP LAN Assignment will be ignored.' Below this, a dropdown menu 'Send VoIP Traffic To' is set to 'Ethernet Port', with a note: '* The DNS and Gateway Addresses from the Ethernet Port will be used on all network connections'. The settings are categorized under 'QoS', 'Timers', 'Audio', 'Proxy 1', and 'Proxy 2'. The 'Timers' section includes: 'Registration Expires (sec)' at 3600, 'Dial Tone Duration (ms)' at 60000, 'Refresh by Update' (unchecked), 'Min SE (sec)' at 90, 'Session Expires (sec)' at 180, 'Digit Map Short Timer (sec)' at 10, 'Digit Map Long Timer (sec)' at 50, and 'Digit Map Time Out (ms)' at 1500. At the bottom, 'SIP Log Level' is set to '0 - Disabled'. A 'Close' button is at the bottom right.

Setting	Value
Registration Expires (sec)	3600
Dial Tone Duration (ms)	60000
Refresh by Update	<input type="checkbox"/>
Min SE (sec)	90
Session Expires (sec)	180
Digit Map Short Timer (sec)	10
Digit Map Long Timer (sec)	50
Digit Map Time Out (ms)	1500

Navigate to the **VoIP** → **Audio** sub-tab to configure the **Dial Plan** and **Codec Priority** as shown below.

The **Dial Plan** included Communication Manager FACs that are 3-digits long starting with a '*' or '#', 5-digit local extensions starting with '7', 10-digit PSTN numbers prepended with the ARS access code '9' and prefix digit '1', 911, and '0' for the operator as shown below.

For the compliance test, G.711 and G.722 were prioritized in the **Codec Priority** field shown below.

The screenshot shows the 'Device Settings' window with the 'VoIP' tab selected. The 'Audio' sub-tab is active. The window title is 'Device Settings' with a close button. The 'VoIP' tab is highlighted, and the 'Audio' sub-tab is selected. The main content area shows the following settings:

- General** (selected), Network, VoIP, IP-Phone, Telco, TLS Certificates
- If unit does not have a dedicated VoIP port voice traffic will be routed through the Ethernet port. The VoIP LAN Assignment will be ignored.
- Send VoIP Traffic To: Ethernet Port (dropdown)
- * The DNS and Gateway Addresses from the Ethernet Port will be used on all network connections
- QoS**, Timers, **Audio** (selected), Proxy 1, Proxy 2
- The Settings below are GLOBAL to all devices in the stack
- DTMF Relay: Out of Band (dropdown)
- DTMF Payload: 101 (spin box)
- RTP Starting Port: 30000 (spin box)
- RTP Range: 200 (spin box)
- Dial Plan: *xx | #xx | 911 | 0 | 7xxxx | 91xxxxxxxxxx (text field)
- Codec Priority: G.711 U Law (selected), G.711 A Law, G.722 (dropdown menu)
- Enable SRTP/SRTCP: ☐
- SIP Log Level: 0 - Disabled (dropdown)
- Close button

Navigate to the **VoIP → Proxy 1** sub-tab to configure the SIP proxy settings. Configure the following fields:

- **User Domain:** Set to the **user domain** (e.g., **avaya.com**) as configured in **Section 5.2**.
- **Registrar Address:** Set to the IP address of the Session Manager SIP interface (e.g., **10.64.110.212**).
- **Registrar Port:** Set to the TCP port (e.g., **5060**).
- **Outbound Proxy Address:** Set to the IP address of the Session Manager SIP interface (e.g., **10.64.110.212**).
- **Outbound Proxy Port:** Set to the TCP port (e.g., **5060**).
- **Transport Type:** Set to the transport type (e.g., **TCP** which was used for the compliance test).
- **Enable Outbound Proxy:** Select the checkbox.

The screenshot shows the 'Device Settings' window with the 'VoIP' tab selected. Under the 'Proxy 1' sub-tab, various SIP settings are configured. A note at the top states: 'If unit does not have a dedicated VoIP port voice traffic will be routed through the Ethernet port. The VoIP LAN Assignment will be ignored.' Below this, 'Send VoIP Traffic To' is set to 'Ethernet Port' with a dropdown arrow. A note follows: '* The DNS and Gateway Addresses from the Ethernet Port will be used on all network connections'. The settings list includes: TCP Port (5060), UDP Port (5060), TLS Port (5061), User Domain (avaya.com), Registrar Address (10.64.110.212), Registrar Port (5060), Outbound Proxy Address (10.64.110.212), Outbound Proxy Port (5060), Transport Type (TCP), and Enable Outbound Proxy (checked). At the bottom, 'SIP Log Level' is set to '0 - Disabled' and a 'Close' button is present.

Setting	Value
TCP Port	5060
UDP Port	5060
TLS Port	5061
User Domain	avaya.com
Registrar Address	10.64.110.212
Registrar Port	5060
Outbound Proxy Address	10.64.110.212
Outbound Proxy Port	5060
Transport Type	TCP
Enable Outbound Proxy	<input checked="" type="checkbox"/>
SIP Log Level	0 - Disabled

7.5. Administer IP Phones

Navigate to **IP Phone** → **Phone 1** sub-tab to configure the SIP extension as follows:

- **Phone Number:** Set to the phone extension configured in **Section 6.1.3**.
- **Name / Label:** Set to an appropriate name e.g., **pro270130**.
- **User Name:** Set to the SIP user name configured in **Section 6.1.2**. (e.g., **70130**).
- **Password:** Set to the SIP user password from **Section 6.1.3**.

The screenshot shows the 'Device Settings' window with the 'IP-Phone' tab selected. The window has a title bar with a close button. Below the title bar are tabs for 'General', 'Network', 'VoIP', 'IP-Phone', 'Telco', and 'TLS Certificates'. The 'IP-Phone' tab is active, showing a message: 'The Settings below are for each VoIP phone found on the device.' and 'There is one VoIP Phones enabled which will support making five concurrent calls on each VoIP Phone'. Below this are tabs for 'Phone 1', 'Phone 2', 'Phone 3', 'Phone 4', and 'Phone 5'. The 'Phone 1' tab is selected, showing the 'Phone Properties' section with fields for 'Phone Number' (70130), 'Name / Label' (pro270130), 'Active Ringtone' (1), and checkboxes for 'Enable' (checked) and 'Auto Answer' (unchecked). Below this is the 'Proxy 1' section with fields for 'User Name' (70130), 'Password' (masked with dots), and 'Reenter Password' (masked with dots). The 'Proxy 2' section has empty fields for 'User Name', 'Password', and 'Reenter Password'. A 'Close' button is at the bottom.

Device Settings

General Network VoIP **IP-Phone** Telco TLS Certificates

The Settings below are for each VoIP phone found on the device.

There is one VoIP Phones enabled which will support making five concurrent calls on each VoIP Phone

Phone 1 Phone 2 Phone 3 Phone 4 Phone 5

Phone Properties

Phone Number: 70130

Name / Label: pro270130

Active Ringtone: 1

☒ Enable

☐ Auto Answer

Proxy 1

User Name: 70130

Password:

Reenter Password:

Proxy 2

User Name:

Password:

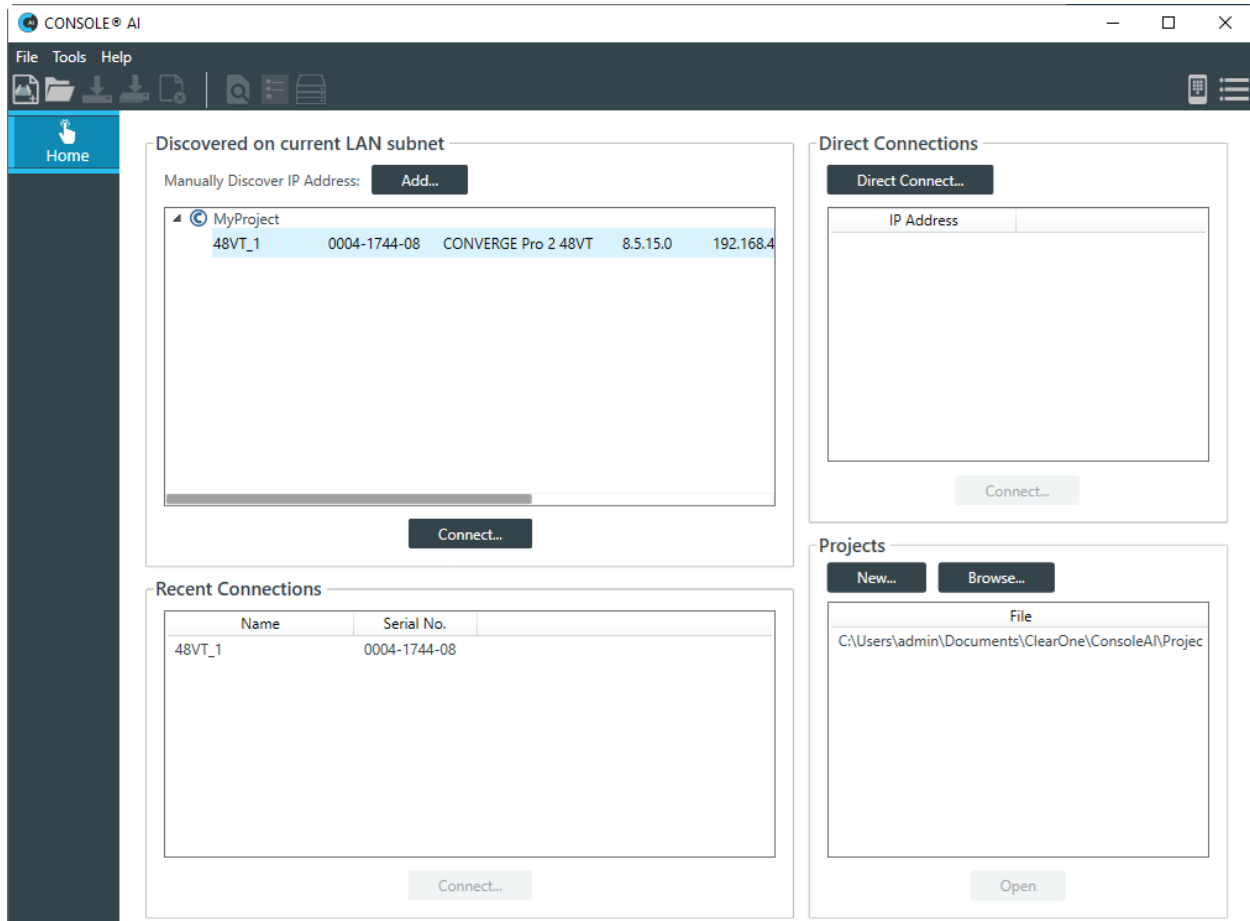
Reenter Password:

Close

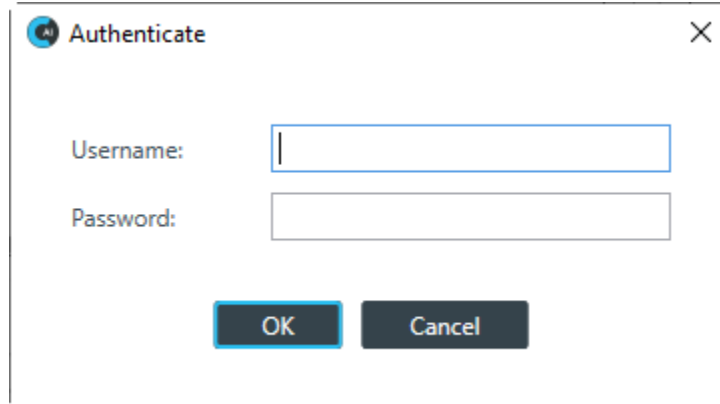
Once the Converge Pro 2 48VT configuration is completed, save the configuration by selecting the **File → Save** menu option (not shown) to save the working configuration to a project file. Select the **File → Close** menu option to close the current project.

7.6. Load Configuration to Converge Pro 2 48VT

To load the configuration to Converge Pro 2 48VT, expand **MyProject** in the **Discovered on current LAN subnet** box and select the device displayed. Click the **Connect** button. The **Connect** button is greyed out until the device is selected.



Next, log in with the appropriate credentials in the **Authenticate** window shown below.

A dialog box titled "Authenticate" with a close button (X) in the top right corner. It contains two input fields: "Username:" and "Password:". Below the input fields are two buttons: "OK" and "Cancel".

Authenticate

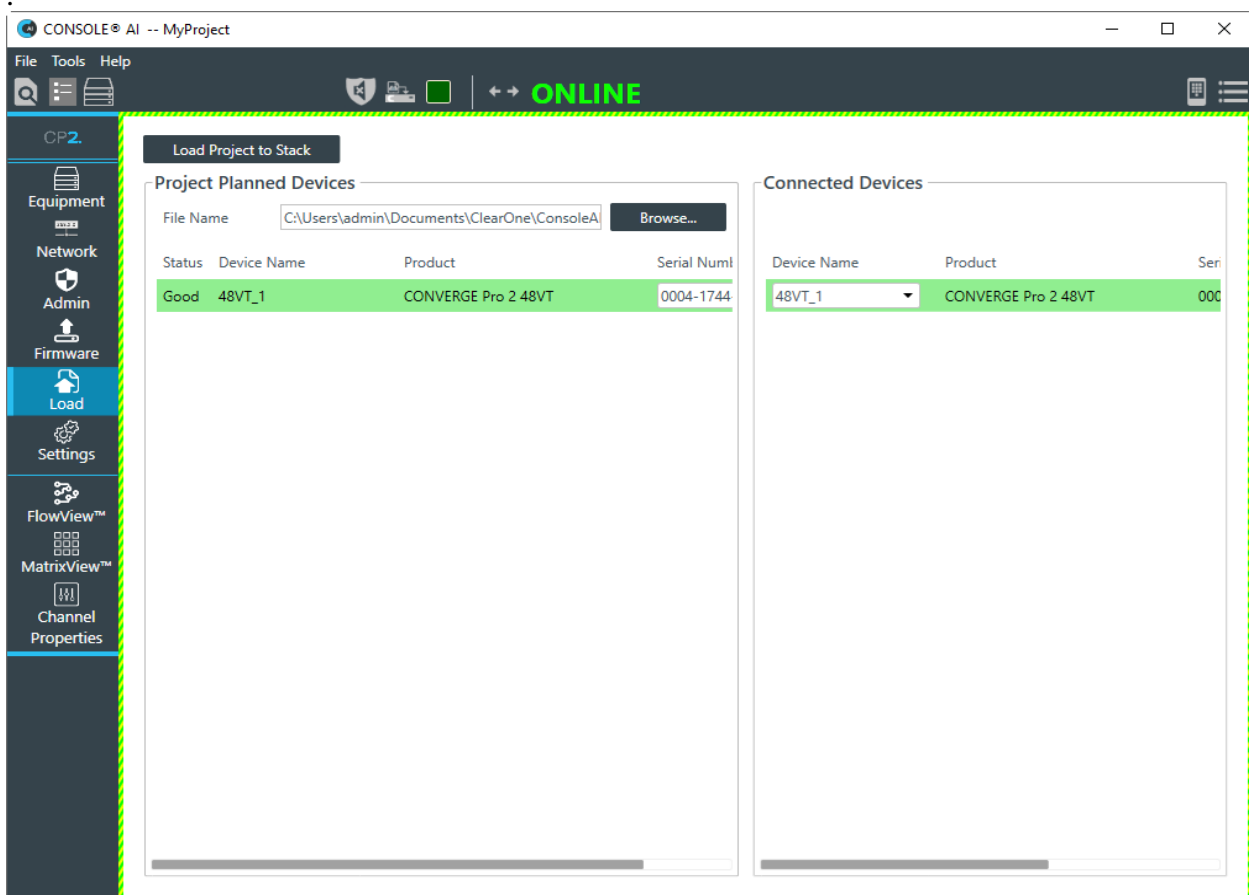
Username:

Password:

OK Cancel

Select **Load** from the left side selections. Under the **Project Planned Devices** section, select the **File Name** of the project file saved in **Section 7.5** using the **Browse** button if needed, and then click on the **Load Project to Stack** button.

Note: Changing the network setup to use a static IP address will cause a reset of Converge Pro 2 48VT.

The screenshot shows the CONSOLE AI software interface. The top bar includes a menu (File, Tools, Help), a status indicator (ONLINE), and window controls. The left sidebar contains navigation options: CP2, Equipment, Network, Admin, Firmware, Load (highlighted), Settings, FlowView™, MatrixView™, and Channel Properties. The main area is divided into two panels. The left panel, titled "Load Project to Stack", shows a "Project Planned Devices" table with columns for Status, Device Name, Product, and Serial Numl. It contains one row: "Good", "48VT_1", "CONVERGE Pro 2 48VT", and "0004-1744". Above this table is a "File Name" input field with a "Browse..." button. The right panel, titled "Connected Devices", shows a table with columns for Device Name, Product, and Serial. It contains one row: "48VT_1", "CONVERGE Pro 2 48VT", and "0004-1744".

CONSOLE AI -- MyProject

File Tools Help

ONLINE

CP2

Equipment

Network

Admin

Firmware

Load

Settings

FlowView™

MatrixView™

Channel Properties

Load Project to Stack

Project Planned Devices

File Name C:\Users\admin\Documents\ClearOne\ConsoleAI\ Browse...

Status	Device Name	Product	Serial Numl
Good	48VT_1	CONVERGE Pro 2 48VT	0004-1744

Connected Devices

Device Name	Product	Serial
48VT_1	CONVERGE Pro 2 48VT	0004-1744

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Converge Pro 2 48VT with Communication Manager and Session Manager.

8.1. Registration to Session Manager

Verify that Converge Pro 2 48VT has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status.

AVAYA Users Elements Services Widgets Shortcuts Search Admin

Aura System Manager 8.1

Home Session Manager x

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

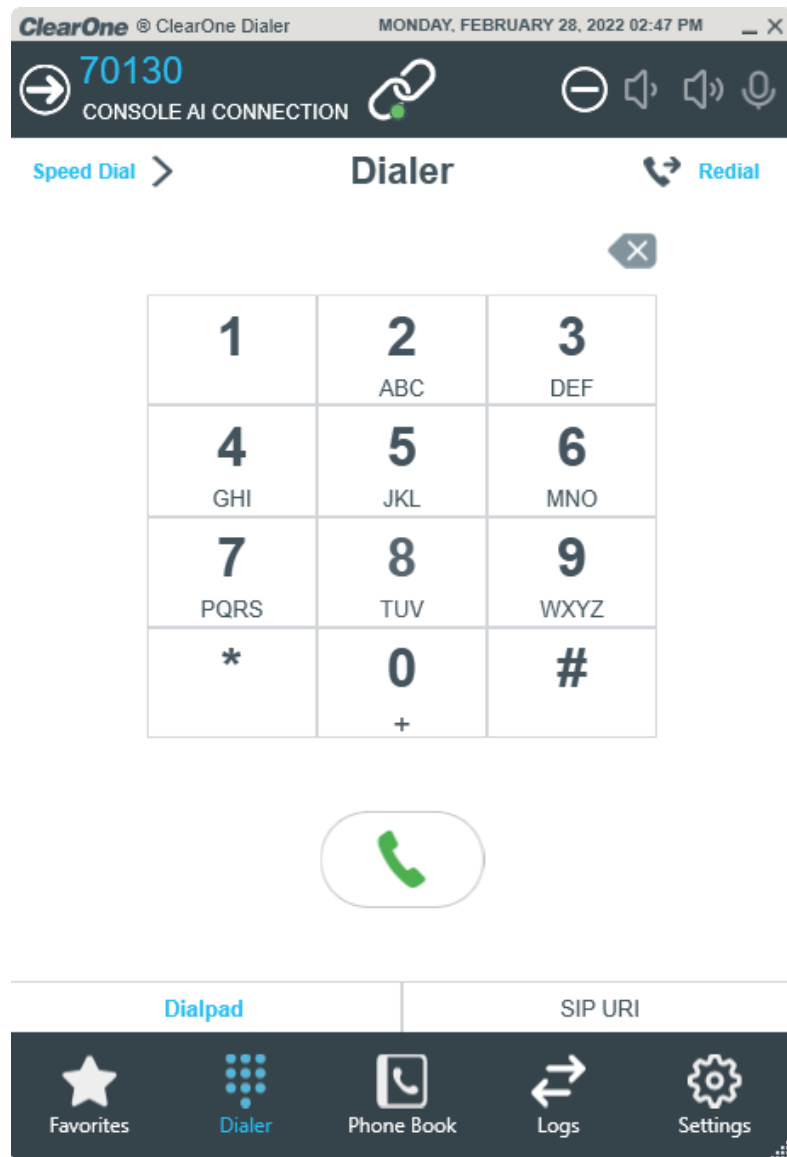
View Default Export Force Unregister AST Device Notifications: Reboot Reload Failback As of 2:38 PM Advanced Search Customize

56 Items Show 15 Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered			
											Prim	Sec	Surv	Visiting
<input type="checkbox"/>	Show	70130@avaya.com	User	ConvergePro2	DevConnect	192.168.4.160	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

8.2. Call Using Converge Pro 2 Dialer

Launch the Converge Pro 2 Dialer. It can be invoked from Converge Pro 2 Console AI or installed separately on a Windows client. Interoperability testing used Converge Pro 2 Dialer invoked from Converge Pro 2 Console AI. In Converge Pro 2 Console AI FlowView, double click on the VoIP RX object and then click the Dialer icon. Note that the green dot in the icon in the Dialer indicates that the unit is registered. Verify basic telephony features by initiating and answering calls between Converge Pro 2 48VT and another phone using the Converge Pro 2 Dialer application.



9. Conclusion

These Application Notes have described the administration steps required to integrate ClearOne Converge Pro 2 48VT 8.5.15.0 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. ClearOne Converge Pro 2 48VT successfully registered with Session Manager as a SIP user and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section** Error! Reference source not found..

10. Additional References

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

- [1] *Administering Avaya Aura® Communication Manager*, Issue 12, Release 8.1.x, July 2021
- [2] *Administering Avaya Aura® Session Manager*, Issue 10, Release 8.1.x, September 2021

The following ClearOne Converge Pro 2 documentation is available from ClearOne.

- [3] *ClearOne Converge® Pro 2 Console AI User Manual*, DOC-0040-001 (Revision 3.7) October 2021.
- [4] *ClearOne Converge® Pro 2 Dialer Installation & Getting Started*, DOC-0315-001 (Revision 1.0) December 2016.

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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.



March 3, 2022

To whom it may concern:

This letter serves as a declaration of model differences. The following ClearOne products share the same SIP firmware version (8.5.15.0):

- CONVERGE Pro 2 48VT. This model has 4 mic/line inputs, 8 mic/line outputs and supports VoIP (SIP) and POTS Telco.
- CONVERGE Pro 2 48VTD. This model has 4 mic/line inputs, 8 mic/line outputs and supports VoIP (SIP), POTS Telco, and Audinate Dante audio streaming.
- CONVERGE Pro 2 128VT. This model has 12 mic/line inputs, 8 mic/line outputs and supports VoIP (SIP) and POTS/Telco.
- CONVERGE Pro 2 128VTD. This model has 12 mic/line inputs, 8 mic/line outputs and supports VoIP (SIP), POTS/Telco, and Audinate Dante audio streaming.

Sincerely,

Derek L. Graham
Sr. V.P. of Research and Development
ClearOne, Inc.