



Avaya Solution & Interoperability Test Lab

Application Notes for ClearOne Converge Pro 2 128V 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 128V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. ClearOne Converge Pro 2 128V is a conferencing solution that registers with Avaya Aura® Session Manager via SIP. This solution also includes the ClearOne Converge Pro 2 Dialer for establishing calls and the ClearOne Converge Pro 2 Console for configuring the system. For the compliance test, ClearOne Converge Pro 2 128V was used, but these Application Notes also apply to the ClearOne Converge Pro 2 128VD and 48V products, which use the same SIP-based VoIP engine and support the Converge Pro 2 Dialer.

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 ClearOne Converge Pro 2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. ClearOne Converge Pro 2 provides a scalable conferencing solution for any size venue and registers with Avaya Aura® Session Manager via SIP. This solution also includes the ClearOne Converge Pro 2 Dialer for establishing calls and the ClearOne Converge Pro 2 Console for configuring the system. For the compliance test, ClearOne Converge Pro 2 128V was used, but these Application Notes also apply to the ClearOne Converge Pro 2 128VD and 48V products, which use the same SIP-based VoIP engine and support the Converge Pro 2 Dialer.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between ClearOne Converge Pro 2 128V, Avaya SIP / H.323 deskphones, 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 128V 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 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 ClearOne Converge Pro 2 128V 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 registration of Converge Pro 2 128V with Session Manager.
- Calls between Converge Pro 2 128V and Avaya SIP / H.323 deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between Converge Pro 2 128V 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 128V.
- Proper system recovery after a restart of Converge Pro 2 128V and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations:

1. When a call is terminated, it takes about 3 seconds for Converge Pro 2 128V to clear it from the ClearOne Converge Pro 2 Dialer. For example, when an active call is dropped, the user will have to wait about 3 seconds before placing another or resuming a held call.
2. Blind conference is currently not supported.
3. When the user initiates an attended call transfer, the user cannot return to the held call while the call transfer is in progress. The user must either terminate the call to the transfer-to party or complete the call transfer.
4. When the user goes off-hook on Converge Pro 2 128V, and then dials the call, if the dialed digit format is not included in the Converge Pro 2 dial plan, Converge Pro 2 128V 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 7.3**), as necessary.

2.3. Support

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

- Phone: 1-800-283-5936 or +1-801-974-3760
- Website: http://www.clearone.com/customer_support
- Email: tech.support@clearone.com

3. Reference Configuration

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

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway. Avaya G450 Media Gateway was connected to the PSTN via an ISDN-PRI trunk (not shown).
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server (not shown in figure).
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- ClearOne Converge Pro 2 128V, ClearOne Converge Pro 2 Dialer for establishing calls, and ClearOne Converge Pro 2 Console for configuring the unit.

ClearOne Converge Pro 2 128V registered with Session Manager and was configured as Off-PBX Stations (OPS) on Communication Manager.

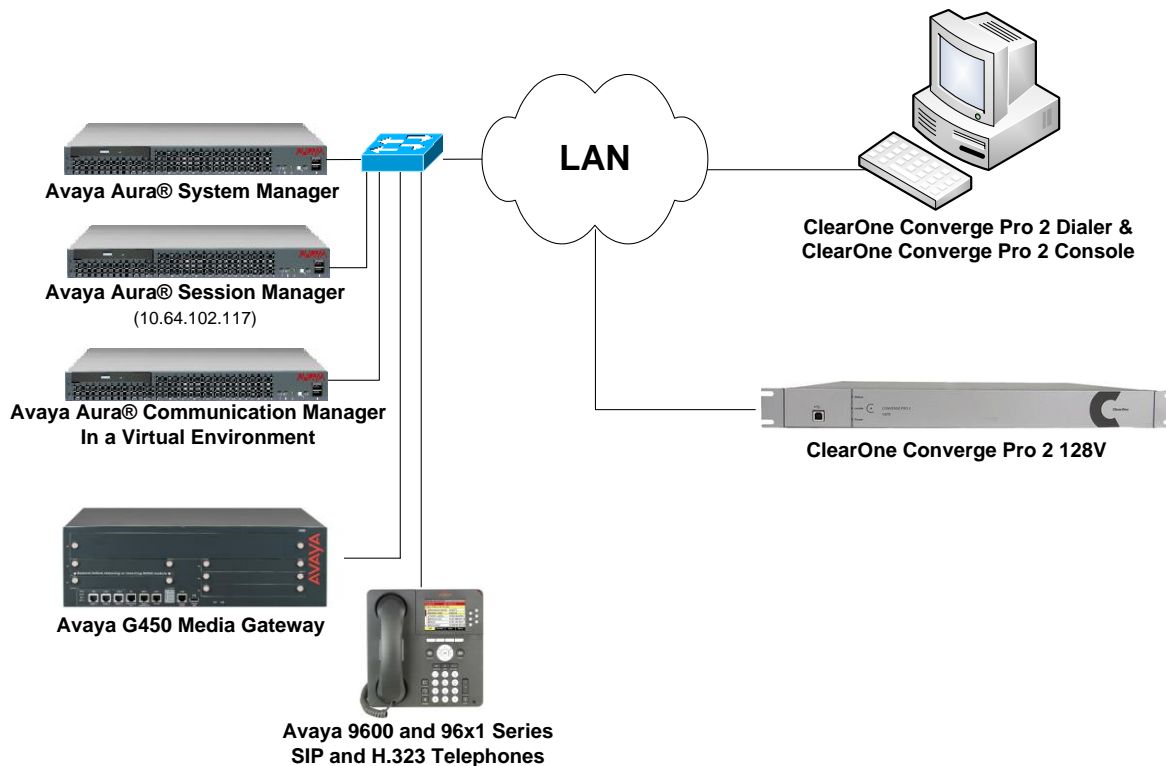


Figure 1: Avaya SIP Network with ClearOne Converge Pro 128V

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	7.0.1.2 SP2 (R017x.00.0.441.0 with Patch 23523)
Avaya G450 Media Gateway	36.16.0
Avaya Aura® Media Server	7.7.0.375
Avaya Aura® Session Manager	7.0.1.2 (7.0.1.2.701230)
Avaya Aura® System Manager	7.0.1.2 (Build No. 7.0.0.016266 Software Update Revision No: 7.0.1.2.086224 Service Pack 2)
Avaya 96x1 Series IP Deskphones	6.6401U (H.323) 7.0.1.4.6 (SIP)
ClearOne Converge Pro 2 128V (see note)	3.4.39.0
ClearOne Converge Pro 2 Dialer	2.0.5.2
ClearOne Converge Pro 2 Console	3.0.4.31

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

- 128V supports 12 Mic/Line Inputs and 8 Mic/Line Outputs.
- 128VD supports 12 Mic/Line Inputs and 8 Mic/Line Outputs and supports Dante™ audio networking.
- 48V supports 4 Mic/Line Inputs and 8 Mic/Line Outputs.

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 and 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 ClearOne Converge Pro 2 128V is configured through Avaya Aura® System Manager in **Section 6.2**.

5.1. Verify 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: V17                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 6400 60
                                Maximum Stations: 2400 22
                                Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 14
Maximum Off-PBX Telephones - PBFMC: 9600 0
Maximum Off-PBX Telephones - PVFMC: 9600 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 and IP Codec Set

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 Avaya G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP 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
Location: 1          Authoritative Domain: avaya.com
  Name:                               Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
  Codec Set: 1          Inter-region IP-IP Direct Audio: yes
                        IP Audio Hairpinning? n
  UDP Port Min: 2048
  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      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5

```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Converge Pro 2 128V. The form is accessed via the **change ip-codec-set 1** command. 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. Converge Pro 2 128V was tested using G.711 and G.722 codecs.

```

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.722-64K      n              2            20
2: G.711MU          n              2            20
3:
4:
5:
6:
7:

```

6. Configure Avaya Aura® Session Manager

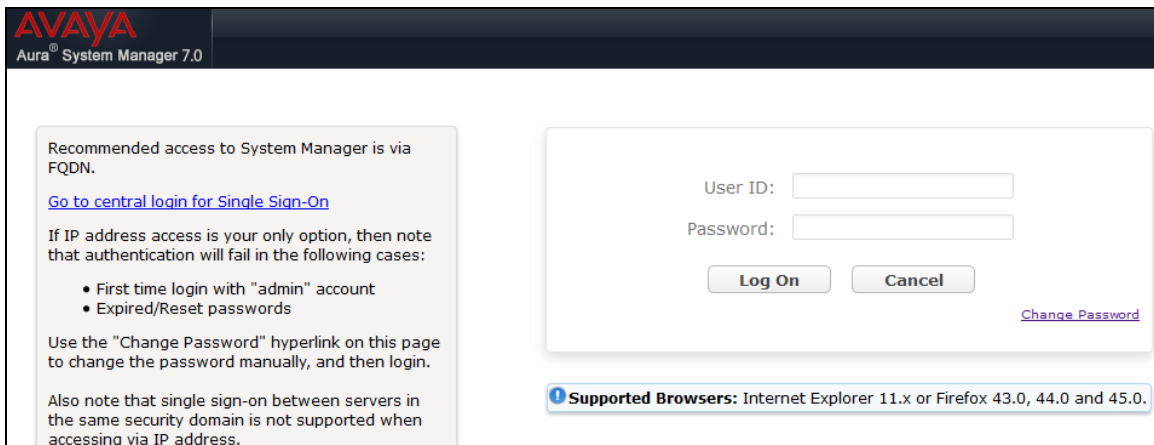
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Converge Pro 2 128V
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for ClearOne Converge Pro 2 128V.

6.1. Launch System Manager

Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.



AVAYA
Aura® System Manager 7.0

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.

User ID:

Password:

[Change Password](#)

Supported Browsers: Internet Explorer 11.x or Firefox 43.0, 44.0 and 45.0.

6.2. Set Network Transport Protocol for ClearOne Converge Pro 2

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 7.0 interface. The top navigation bar includes 'Home', 'Routing', and 'SIP Entities'. The main content area is titled 'SIP Entity Details' and contains a 'General' section with the following fields:

- Name: devcon-sm
- FQDN or IP Address: 10.64.102.117
- Type: Session Manager
- Notes: (empty)
- Location: Thornton
- Outbound Proxy: (empty)
- Time Zone: America/New_York
- Credential name: (empty)

At the bottom, there is a 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'.

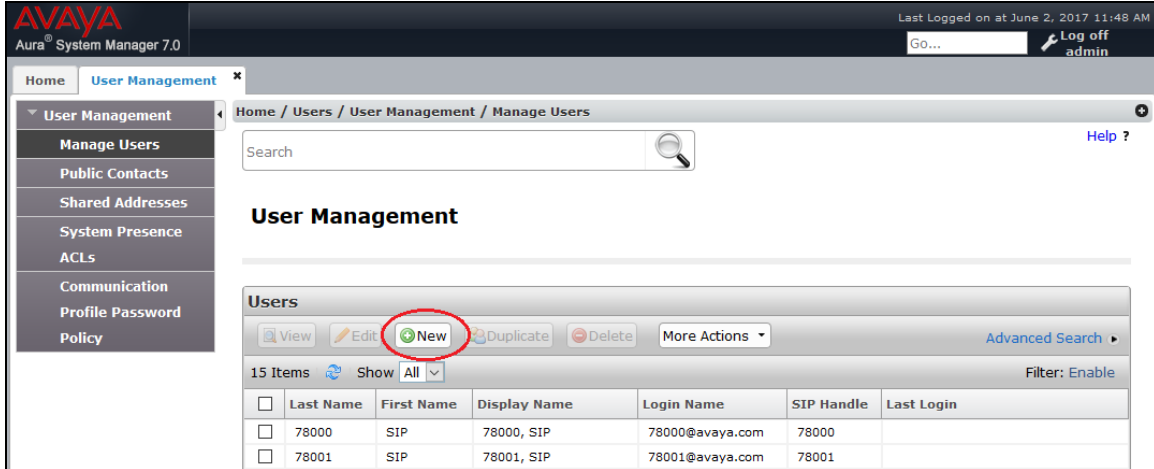
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Converge Pro 2 128V is specified in the list below. For the compliance test, Converge Pro 2 128V used *UDP* network transport.

The screenshot shows the 'Listen Ports' section of the interface. It includes input fields for 'TCP Failover port' and 'TLS Failover port', and a table with 3 items. The table has columns for 'Listen Ports', 'Protocol', 'Default Domain', 'Endpoint', and 'Notes'. The row for port 5060 with protocol UDP is highlighted with a red box.

Listen Ports	Protocol	Default Domain	Endpoint	Notes
5060	TCP	avaya.com	<input checked="" type="checkbox"/>	
5060	UDP	avaya.com	<input checked="" type="checkbox"/>	
5061	TLS	avaya.com	<input checked="" type="checkbox"/>	

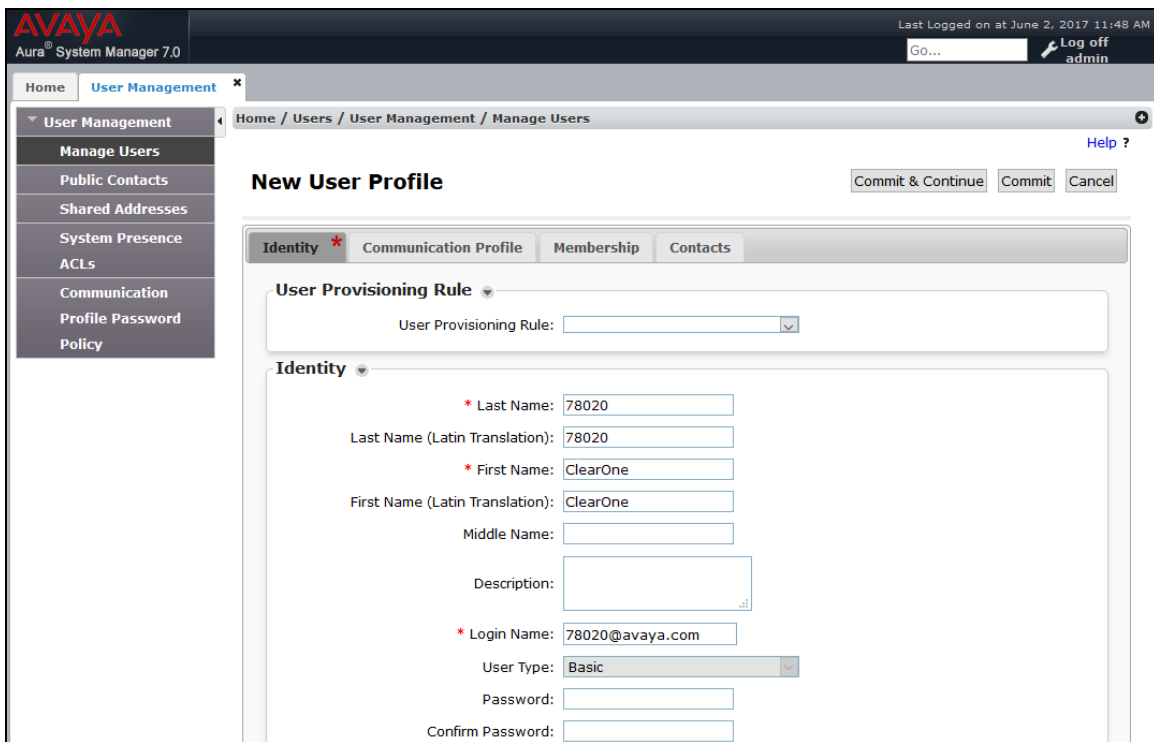
6.3. Administer SIP User

In the **Home** screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.



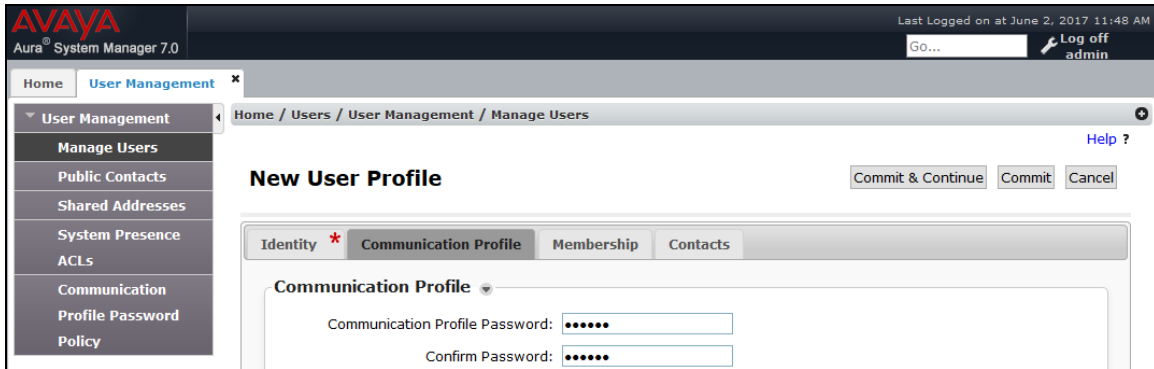
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter “<ext>@<domain>”, where “<ext>” is the desired Converge Pro 2 128V SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



6.3.2. Communication Profile

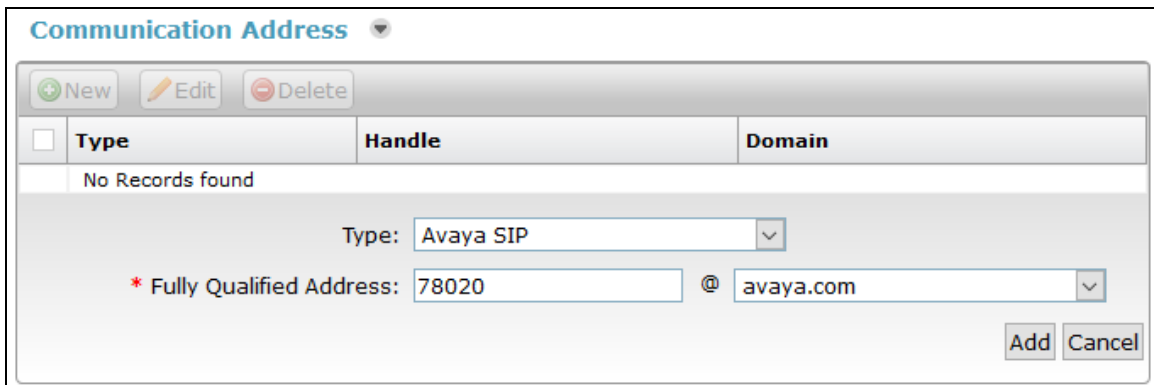
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



The screenshot shows the Avaya Aura System Manager 7.0 interface. The user is logged in as 'admin' and is viewing the 'New User Profile' form. The 'Communication Profile' tab is selected, and the 'Communication Profile Password' and 'Confirm Password' fields are visible. The form also includes tabs for 'Identity', 'Membership', and 'Contacts', and buttons for 'Commit & Continue', 'Commit', and 'Cancel'.

6.3.3. Communication Address

In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields as shown below. For **Type**, retain “Avaya SIP”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**. Click **Add**.



The screenshot shows the 'Communication Address' form. It includes a table with columns for 'Type', 'Handle', and 'Domain'. The table is currently empty, displaying 'No Records found'. Below the table, there are input fields for 'Type' (set to 'Avaya SIP') and 'Fully Qualified Address' (set to '78020@avaya.com'). There are also 'Add' and 'Cancel' buttons.

<input type="checkbox"/>	Type	Handle	Domain
No Records found			

Type: Avaya SIP

* Fully Qualified Address: 78020 @ avaya.com

6.3.4. Session Manager Profile

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

Session Manager Profile ▾

SIP Registration

* Primary Session Manager

Primary	Secondary	Maximum
15	0	15

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▾

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▾

Termination Sequence ▾

Call Routing Settings

* Home Location ▾

Conference Factory Set ▾

Call History Settings

Enable Centralized Call History?

6.3.5. CM Endpoint Profile

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9600SIP_DEFAULT_CM_7_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).

CM Endpoint Profile ▾

- * System ▾
- * Profile Type ▾
- Use Existing Endpoints
- * Extension [Display Extension Ranges](#)
- * Template ▾
- Set Type
- Security Code
- Port
- Voice Mail Number
- Preferred Handle ▾
- Calculate Route Pattern
- Sip Trunk
- Enhanced Callr-Info display for 1-line phones
- Delete Endpoint on Unassign of Endpoint from User or on Delete User
- Override Endpoint Name and Localized Name
- Allow H.323 and SIP Endpoint Dual Registration

In the **CM Endpoint Profile** sub-section, click the **Endpoint Editor** button to display the page below. In the **Button Assignments** tab, specify up to five call appearances as shown below. Converge Pro 2 can support up to five simultaneous calls per SIP line. Click **Done** to return to the previous page.

Note: To allow a maximum of five incoming calls, the **Restrict Last Appearance** option in the **Feature Options** tab should be disabled.

The screenshot displays the 'Edit Endpoint' configuration page. The breadcrumb path is 'Home / Users / User Management / Manage Users'. The page title is 'Edit Endpoint'. There are 'Done' and 'Cancel' buttons at the top right, and a '[Save As Template]' link below them.

Fields for configuration:

- System:** devcon-cm
- Template:** 9600SIP_DEFAULT_CM_7_0
- Port:** JP
- Name:** 78020,ClearOne
- Extension:** 78020
- Set Type:** 9600SIP
- Security Code:** (empty)

Below the fields are several tabs: 'General Options (G) *', 'Feature Options (F)', 'Site Data (S)', 'Abbreviated Call Dialing (A)', and 'Enhanced Call Fwd (E)'. The 'Button Assignment (B)' tab is selected, and the 'Group Membership (M)' sub-tab is also visible.

The 'Button Assignment' section contains a table with 8 rows, each representing a button assignment. The columns are 'Main Buttons', 'Feature Buttons', and 'Expansion Module'. The 'Main Buttons' column has dropdown menus with values 'call-appr' for rows 1-5 and 'None' for rows 6-8. The other columns have empty input fields.

At the bottom left, there is a '*Required' label. At the bottom right, there are 'Done' and 'Cancel' buttons.

7. Configure ClearOne Converge Pro 2 128V

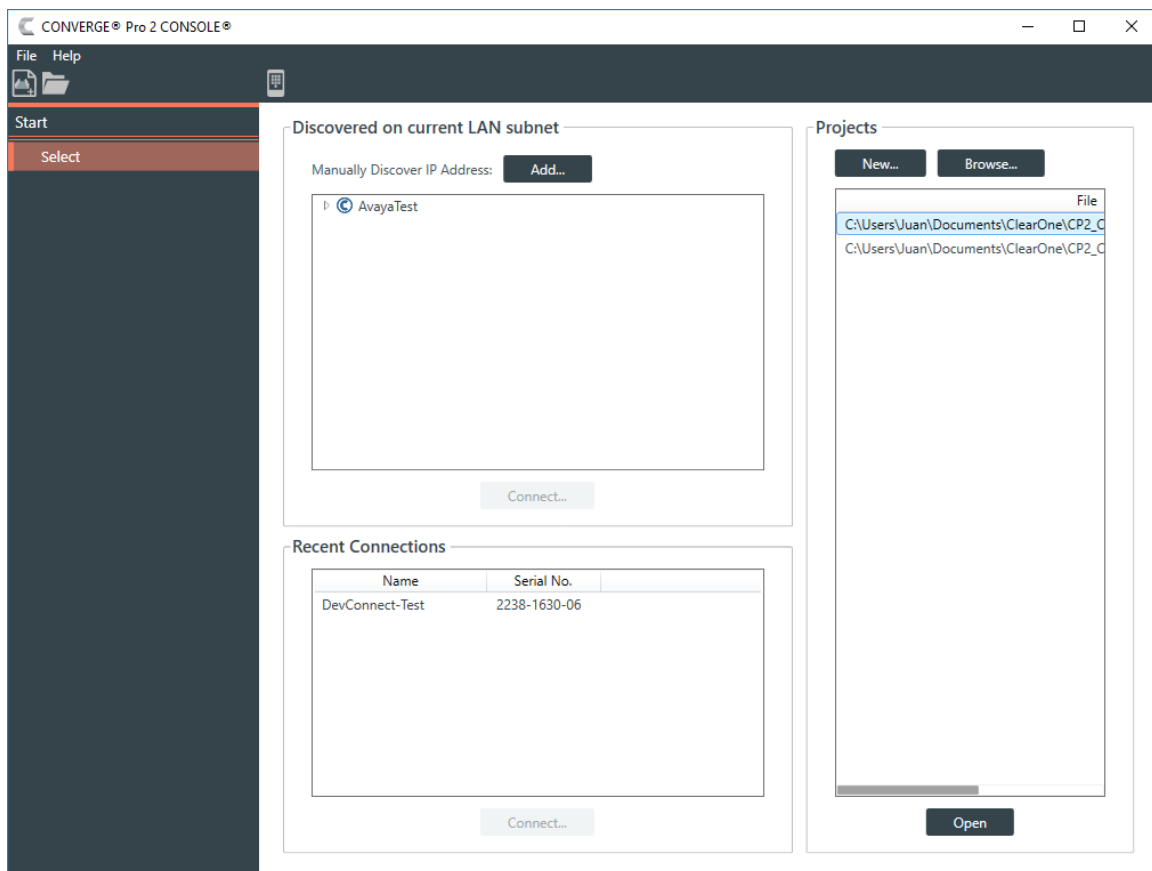
This section covers the Converge Pro 2 128V configuration using the Converge Pro 2 Console. The procedure covers the following areas:

- Launch Converge Pro 2 Console
- Administer IP Settings
- Administer VoIP Stack, including Timers, Audio, and SIP Proxy Settings
- Administer VoIP Phone Settings
- Load Configuration to Converge Pro 2 128V

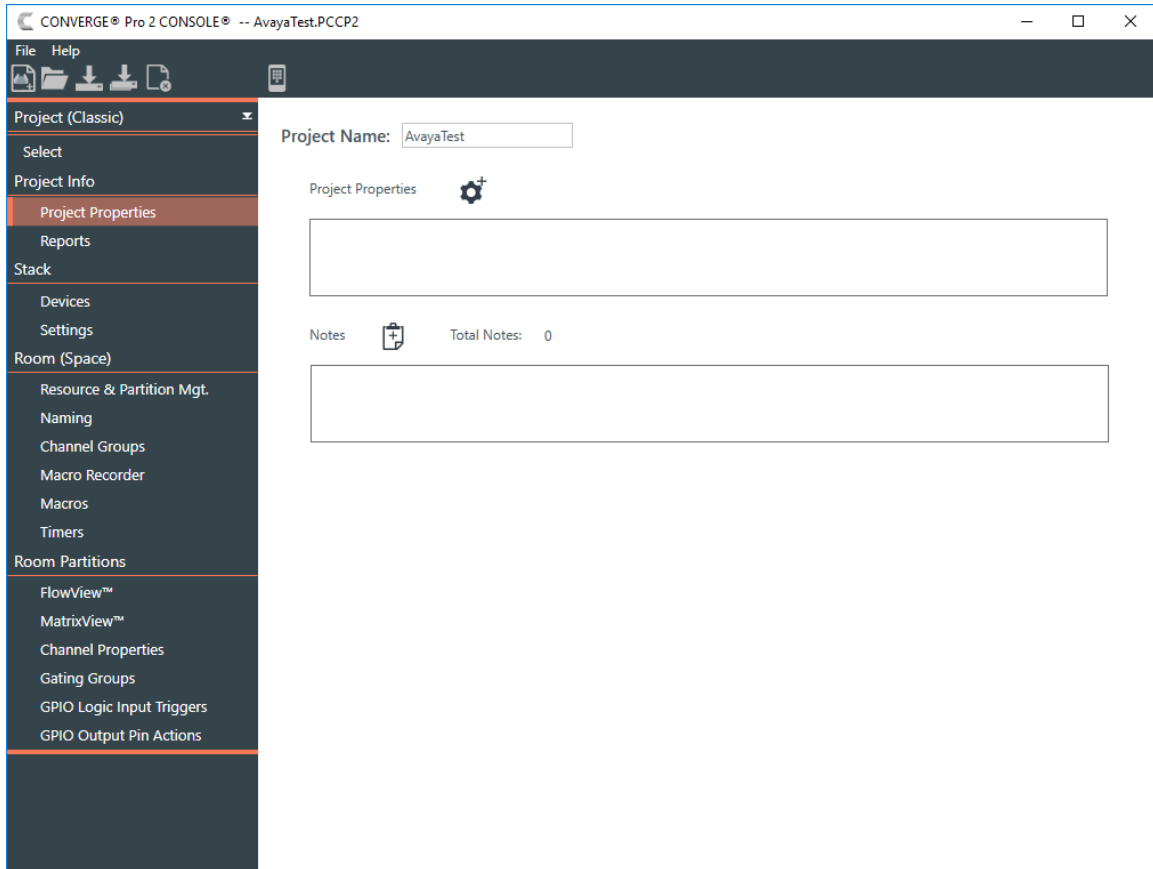
Refer to [3, 4] for more information on configuring ClearOne Converge Pro 2 128V.

7.1. Launch Converge Pro 2 Console

Converge Pro 2 128V is configured using the Converge Pro 2 Console. Launch the Converge Pro 2 Console to display the window shown below.

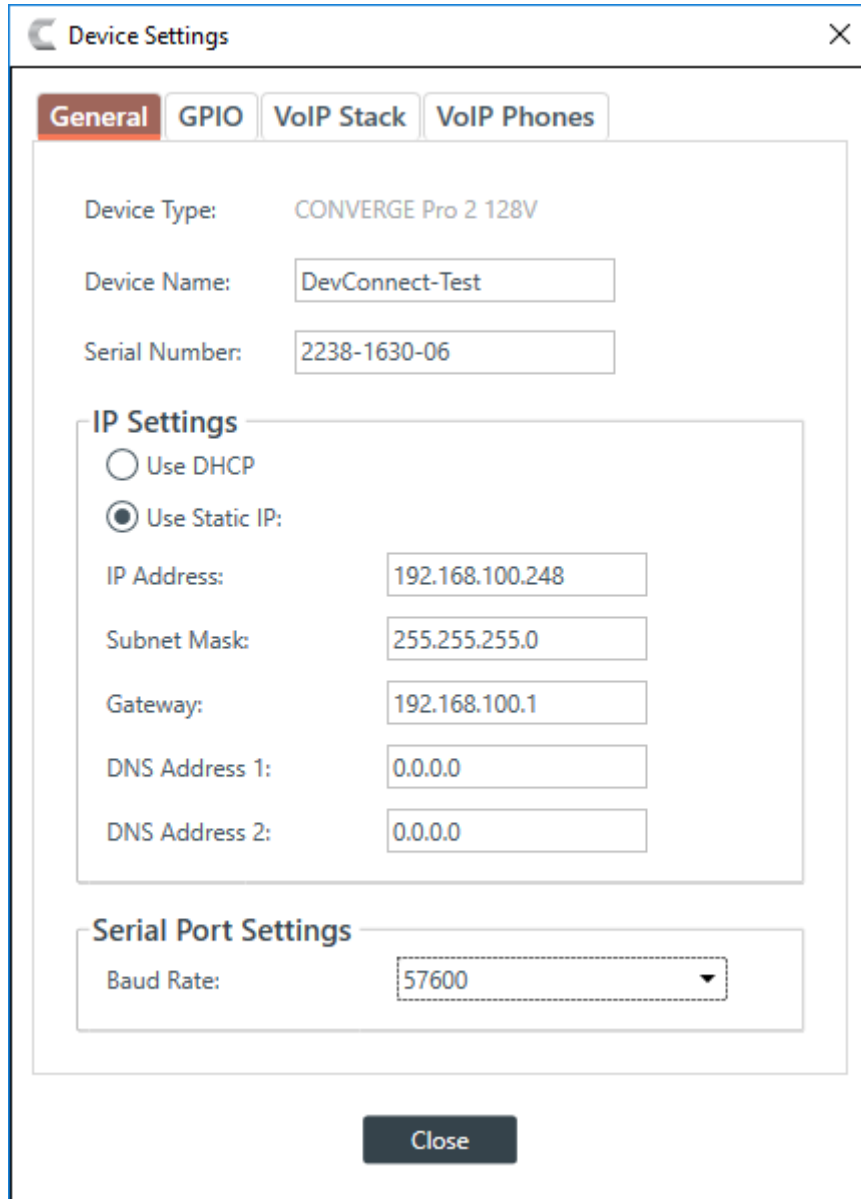


To configure Converge Pro 2 128V, either start a new project or open an existing project by selecting the appropriate project in the **Projects** section and then clicking the **Open** button. When the project is opened, the Converge Pro 2 Console window will appear as shown below with the **Project Name** displayed at the top of the window.



7.2. Administer IP Settings

Converge Pro 2 128V 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 **Devices** in the left pane (shown on previous page). The **Device Settings** window is displayed as shown below. In the **General** tab, configure the IP settings, including the **IP Address**, **Subnet Mask**, **Gateway**, and **DNS Address**, to correspond to the customer's network as shown below.



The screenshot shows a 'Device Settings' window with a title bar containing a back arrow, the text 'Device Settings', and a close button. Below the title bar are four tabs: 'General' (selected), 'GPIO', 'VoIP Stack', and 'VoIP Phones'. The 'General' tab contains the following fields:

- Device Type: CONVERGE Pro 2 128V
- Device Name: DevConnect-Test
- Serial Number: 2238-1630-06

The 'IP Settings' section is enclosed in a box and contains:

- Use DHCP:
- Use Static IP:
- IP Address: 192.168.100.248
- Subnet Mask: 255.255.255.0
- Gateway: 192.168.100.1
- DNS Address 1: 0.0.0.0
- DNS Address 2: 0.0.0.0

The 'Serial Port Settings' section is also enclosed in a box and contains:

- Baud Rate: 57600 (with a dropdown arrow)

A 'Close' button is located at the bottom center of the window.

7.3. Administer VoIP Stack

In the **VoIP Stack** tab, configure timers, audio codecs, dial plan, and SIP proxy. In the **VoIP Stack** → **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 a 'Device Settings' dialog box with the following configuration:

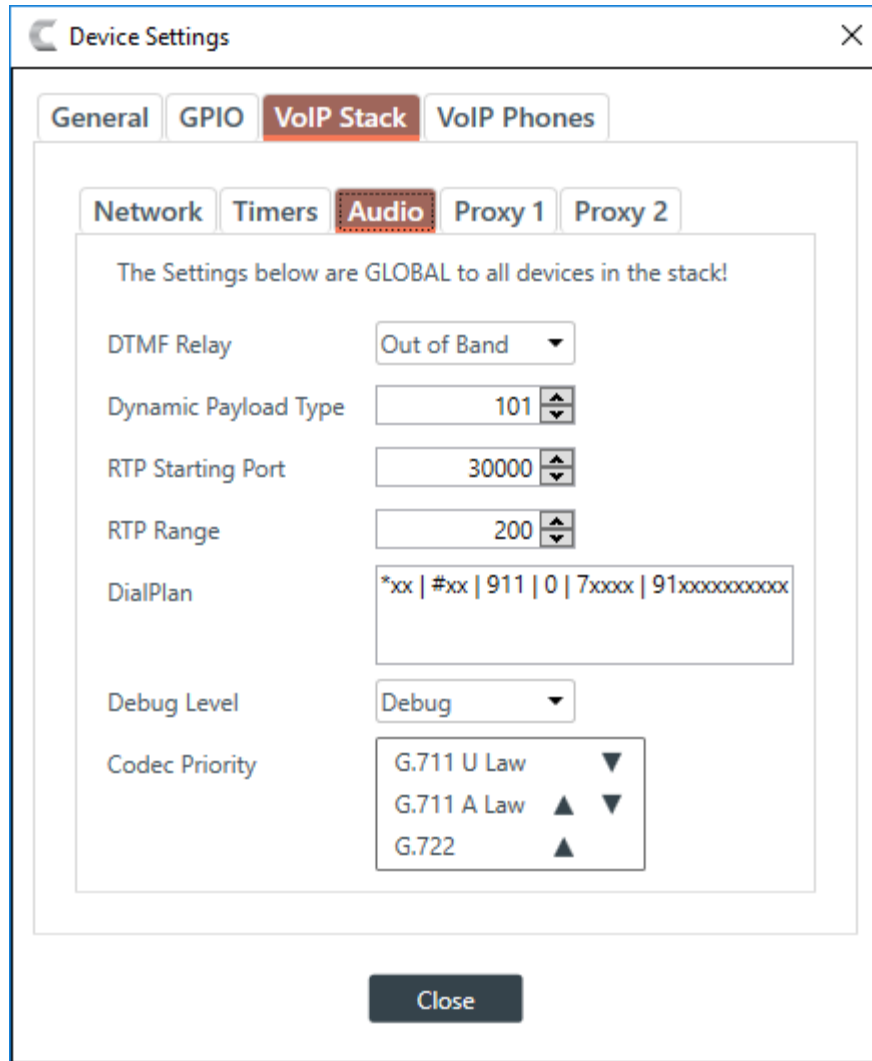
Setting	Value
Registration Expires (Sec)	3600
Dial Tone Duration (MSec)	60000
Refresh by Update	<input type="checkbox"/>
Min SE	90
Session Expires	180
Digit Map Short Timer	10
Digit Map Long Timer	50
Digit Map Time Out	1500

A 'Close' button is located at the bottom center of the dialog box.

Navigate to the **VoIP Stack** → **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.



Navigate to the **VoIP Stack** → **Proxy 1** sub-tab to configure the SIP proxy settings. In the following fields were configured:

- **UDP Port:** Set to the UDP port (e.g., 5060).
- **User Domain:** Set to the domain name (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., 10.64.102.117).
- **Registrar Port:** Set to the UDP port (e.g., 5060).
- **Outbound Proxy Address:** Set to the IP address of the Session Manager SIP interface (e.g., 10.64.102.117).
- **Outbound Proxy Port:** Set to the UDP port (e.g., 5060).
- **Transport Type:** Set to the transport type (e.g., *UDP* which was used for the compliance test).
- **OBP Enable:** Select the checkbox.

The screenshot shows a 'Device Settings' window with a 'VoIP Stack' tab selected. Underneath, the 'Proxy 1' sub-tab is active. A message states: 'The Settings below are GLOBAL to all devices in the stack!'. The configuration fields are as follows:

Field	Value
TCP Port	5060
UDP Port	5060
User Domain	avaya.com
Registrar Address	10.64.102.117
Registrar Port	5060
Outbound Proxy Address	10.64.102.117
Outbound Proxy Port	5060
Transport Type	UDP
OBP Enable	<input checked="" type="checkbox"/>

A 'Close' button is located at the bottom of the dialog.

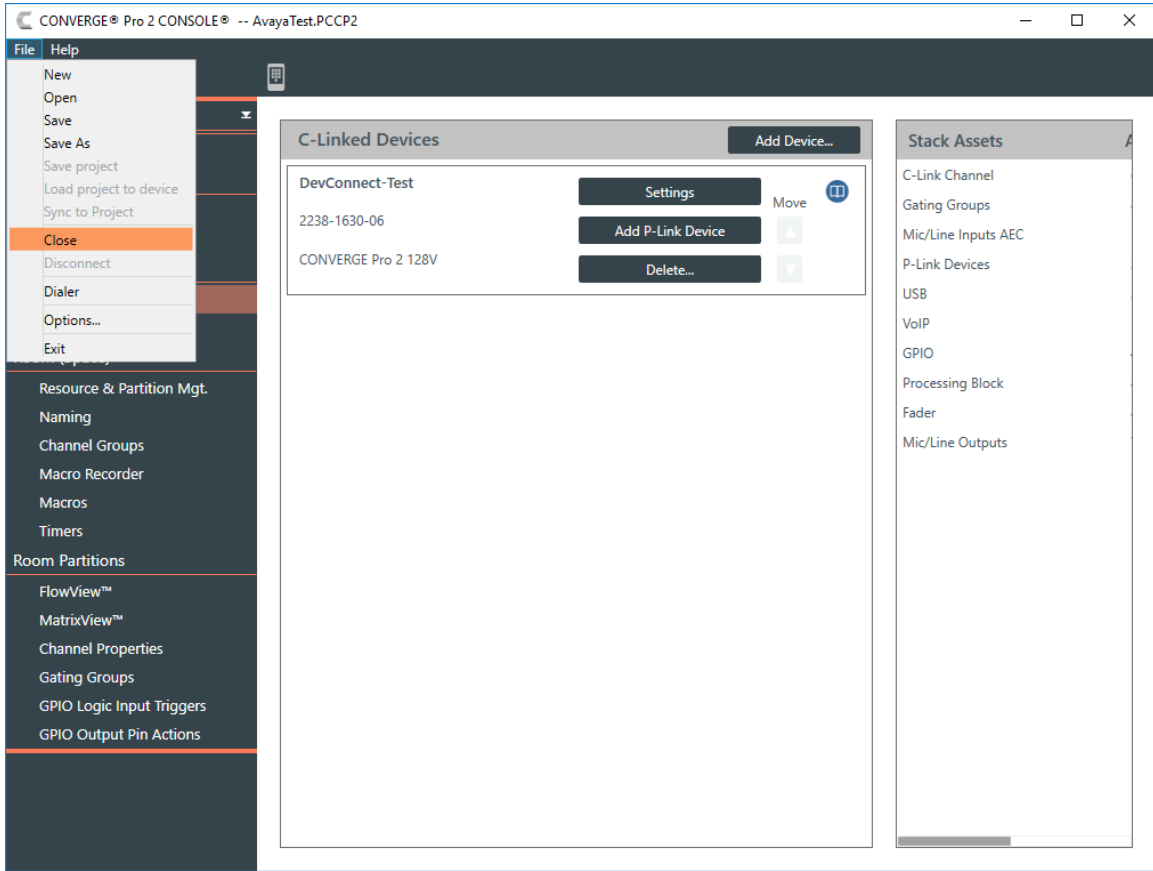
7.4. Administer VoIP Phones

Navigate to **VoIP Phones** → **Phone 1** sub-tab to configure the SIP extension, SIP registration credentials, and transport type as shown below. Click on **Close** button when the configuration is complete.

The screenshot shows a 'Device Settings' window with a 'VoIP Phones' tab selected. The window contains configuration fields for two phones, with 'Phone 1' currently active. The configuration is organized into three sections: 'Phone Properties', 'Proxy 1', and 'Proxy 2'. A 'Close' button is located at the bottom center of the window.

Section	Field	Value
Phone Properties	Phone Number	78020
	Name / Label	V78020
	UA Enable	<input checked="" type="checkbox"/>
Proxy 1	User Name	78020
	Password	••••••
	Reenter Password	••••••
	Transport Type	UDP
Proxy 2	User Name	
	Password	
	Reenter Password	
	Transport Type	UDP

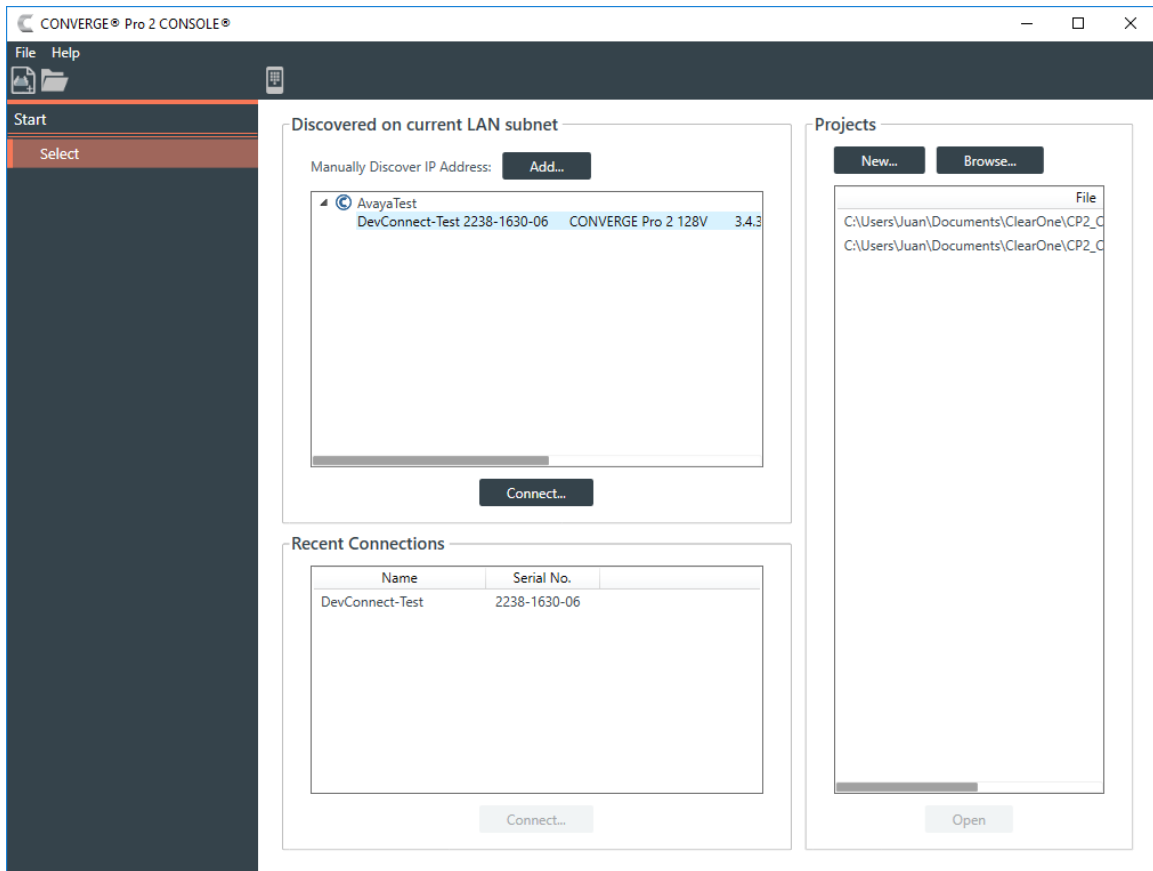
Once the Converge Pro 2 128V configuration is completed, close the configuration by selecting the **File** → **Close** menu option as shown below.



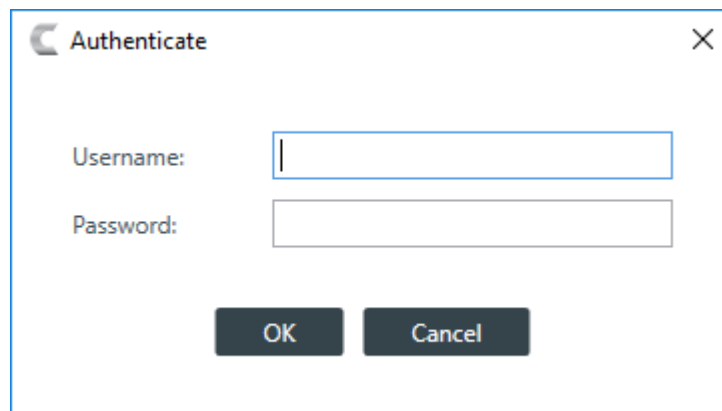
7.5. Load Configuration to Converge Pro 2 128V

To load the configuration to Converge Pro 2 128V, select the unit in the **Discovered on current LAN subnet** section, and then click the **Connect** button.

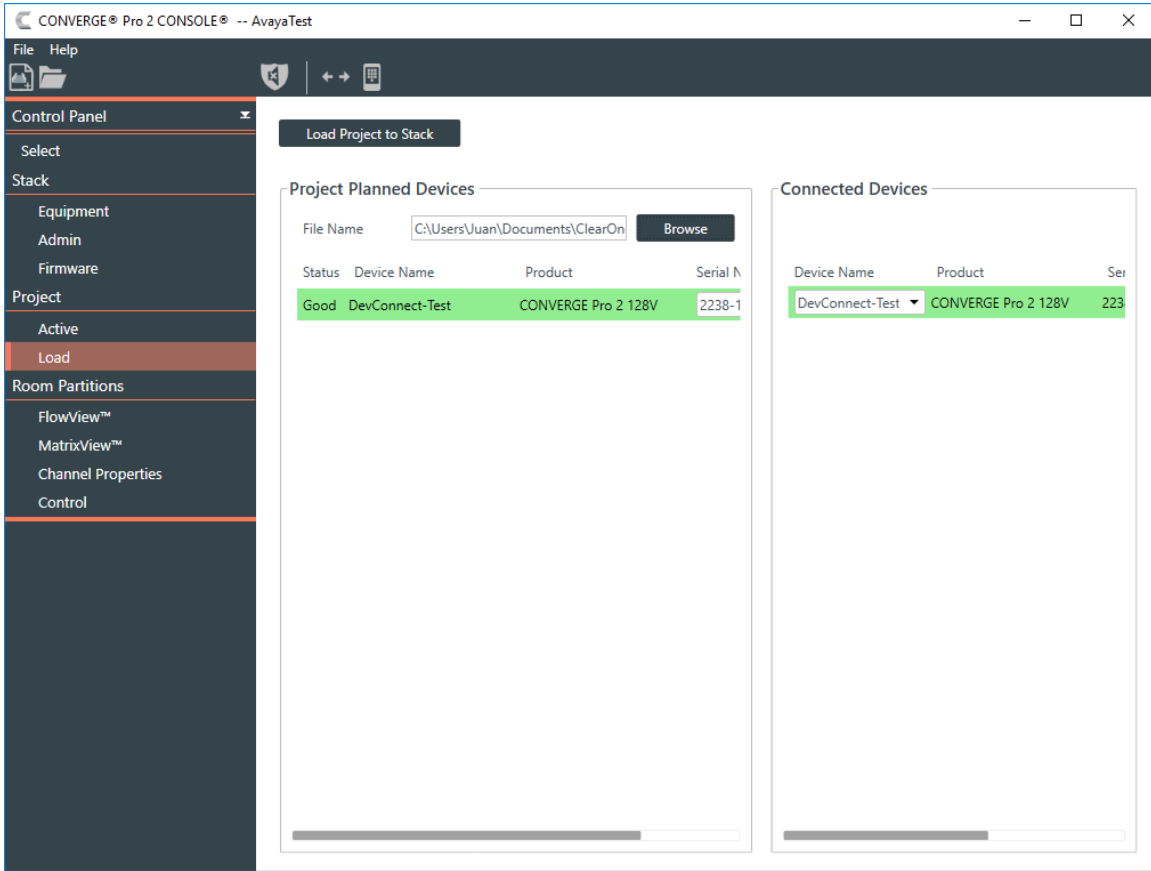
Note: The ClearOne Locator Service must be running in order for the Converge Pro 2 Console to discover any existing units.



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



Under the **Project Planned Devices** section, select the **File Name** using the **Browse** button, and then click on the **Load Project to Stack** button.



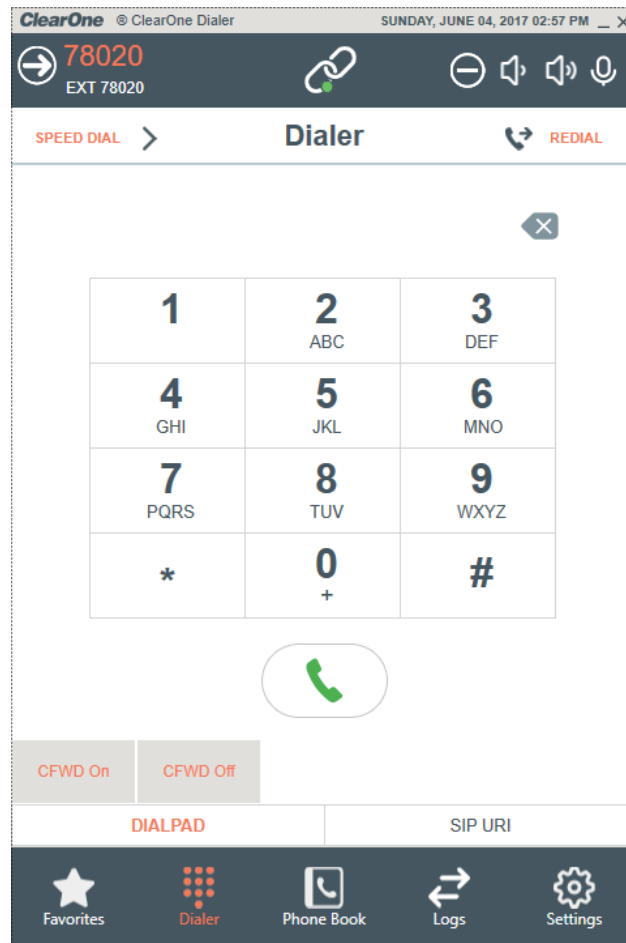
7.6. Verification Steps

This section provides the tests that can be performed to verify proper configuration of ClearOne Converge Pro 2 128V with Communication Manager and Session Manager.

1. Verify that Converge Pro 2 128V has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status.

2. Launch the ClearOne Converge Pro 2 Dialer and verify that it has registered with Session

Manager. Note that the green dot in the  icon in the Dialer indicates that the unit is registered.



3. Verify basic telephony features by establishing calls between Converge Pro 2 128V and another phone using the ClearOne Converge Pro 2 Dialer application.

8. Conclusion

These Application Notes have described the administration steps required to integrate ClearOne Converge Pro 2 128V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. ClearOne Converge Pro 2 128V successfully registered with Session Manager and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

9. 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*, Release 7.0.1, Issue 2, May 2016, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016.

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

- [3] *ClearOne Converge® Pro 2 Console® User Manual*, DOC-0309-001 (Revision 1.2) January 2017.
- [4] *ClearOne Converge® Pro 2 Dialer User Manual*, DOC-0316-001 (Revision 1.0) January 2017.

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