

# APPLICATION NOTES

**PRODUCTS SUPPORTED:**  
ClearOne CONVERGE® Pro 2 and  
Avaya Aura® Session Manager

CLEARONE DOCUMENT NTS-0050-001  
(REVISION 1.0) July 2017

CONFIGURING THE CONVERGE Pro 2  
WITH AVAYA AURA COMMUNICATION MANAGER  
AND AVAYA AURA SESSION MANAGER

## Overview

This application note describes the configuration steps required to integrate ClearOne Converge Pro 2 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Converge Pro 2 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.

## 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 the 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 the Session Manager. The SIP station configuration for ClearOne Converge Pro 2 128V is configured through Avaya Aura® System Manager in Section 6.2.

## Verify License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled 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.

On Page 1, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```

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.)

```

**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
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      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.711MU      n            2           20
2:
3:
4:
5:
6:
7:
```

## 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

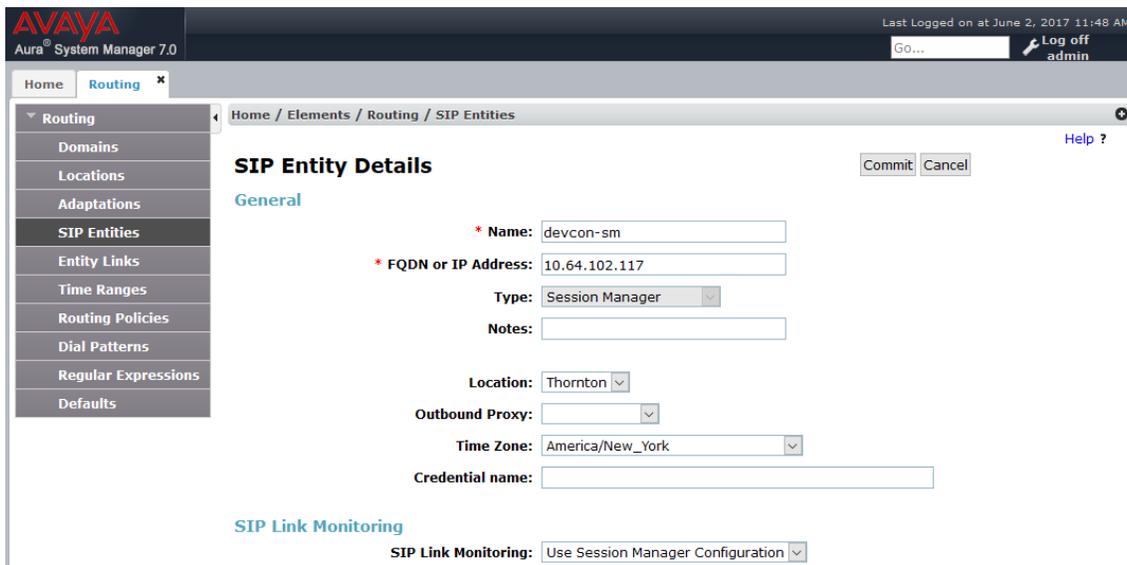
**NOTE:** It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for ClearOne Converge Pro 2 128V.

### 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.

### 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.



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.

#### Listen Ports

TCP Failover port:

TLS Failover port:

Add Remove

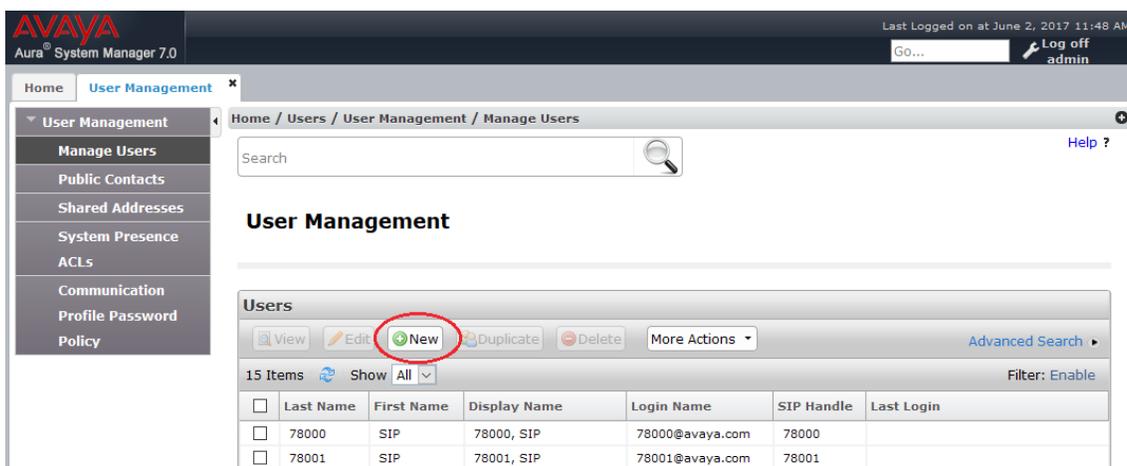
3 Items Filter: Enable

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

Select : All, None

### 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.



### 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.

AVAYA  
Aura System Manager 7.0

Last Logged on at June 2, 2017 11:48 AM  
Go... Log off admin

Home User Management x

Home / Users / User Management / Manage Users

Manage Users  
Public Contacts  
Shared Addresses  
System Presence  
ACLs  
Communication  
Profile Password  
Policy

**New User Profile** Commit & Continue Commit Cancel

Identity \* Communication Profile Membership Contacts

User Provisioning Rule  
User Provisioning Rule: [v]

Identity  
\* Last Name: 78020  
Last Name (Latin Translation): 78020  
\* First Name: ClearOne  
First Name (Latin Translation): ClearOne  
Middle Name: [ ]  
Description: [ ]  
\* Login Name: 78020@avaya.com  
User Type: Basic [v]  
Password: [ ]  
Confirm Password: [ ]

## 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.

AVAYA  
Aura System Manager 7.0

Last Logged on at June 2, 2017 11:48 AM  
Go... Log off admin

Home User Management x

Home / Users / User Management / Manage Users

Manage Users  
Public Contacts  
Shared Addresses  
System Presence  
ACLs  
Communication  
Profile Password  
Policy

**New User Profile** Commit & Continue Commit Cancel

Identity \* Communication Profile Membership Contacts

Communication Profile  
Communication Profile Password: [ ]  
Confirm Password: [ ]

## 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.

## Communication Address ▼

+ New✎ Edit✖ Delete

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

Type:

\* Fully Qualified Address:  @

## 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?

## 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

[Display Extension Ranges](#)

\* Extension

\* 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

## Configure ClearOne Converge Pro 2

This section covers the Converge Pro 2 128V configuration using the Converge Pro 2 CONSOLE.

These instructions apply to the following CP2 models: 128V, 128VD, 48V.

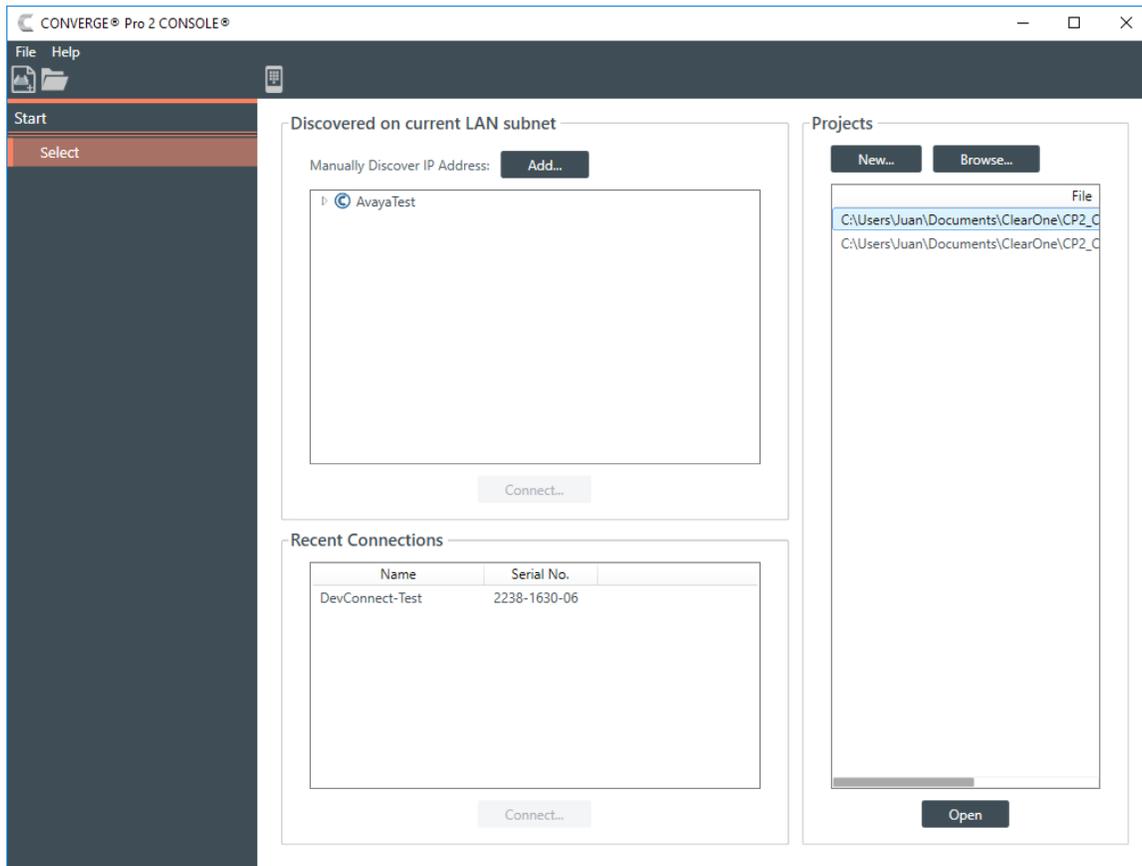
The procedure covers the following areas:

- 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

Refer to the CONVERGE Pro 2 CONSOLE User Manual for more information on configuring ClearOne Converge Pro 2.

## 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.

## 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. 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.

Device Settings ×

General GPIO VoIP Stack VoIP Phones

Device Type: CONVERGE Pro 2 128V

Device Name:

Serial Number:

### IP Settings

Use DHCP

Use Static IP:

IP Address:

Subnet Mask:

Gateway:

DNS Address 1:

DNS Address 2:

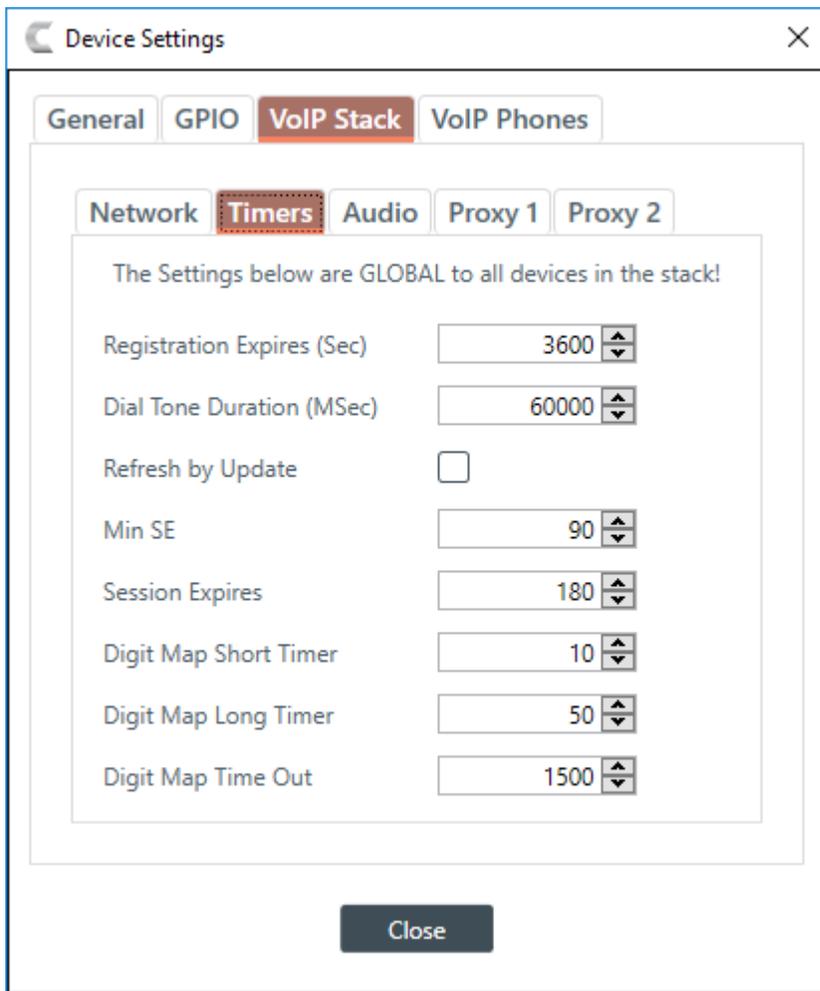
### Serial Port Settings

Baud Rate:

Close

### **Administer VoIP Stack**

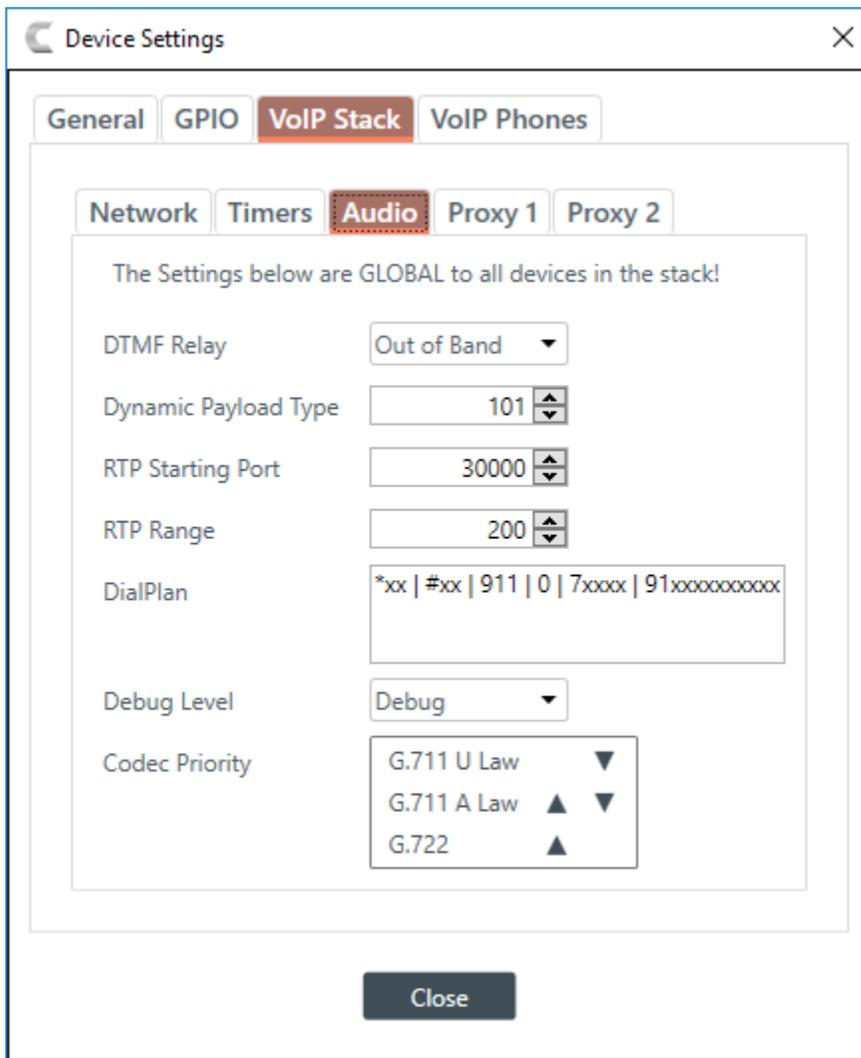
In the VoIP Stack tab, configure timers, audio codecs, dial plan, and SIP proxy. In the VoIP Stack  Timers sub-tab, verify that the SIP timers are configured as desired. The default values shown below were used for the compliance test.



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 6.3.
- **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.

Device Settings

General GPIO **VoIP Stack** VoIP Phones

Network Timers Audio **Proxy 1** Proxy 2

The Settings below are GLOBAL to all devices in the stack!

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"/>

Close

### **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.

Device Settings

General GPIO VoIP Stack **VoIP Phones**

The Settings below are for each VoIP phone found on the device. Additional VoIP licences may be required.

**Phone 1** Phone 2

**Phone Properties**

Phone Number	78020
Name / Label	V78020
	<input checked="" type="checkbox"/> UA Enable

**Proxy 1**

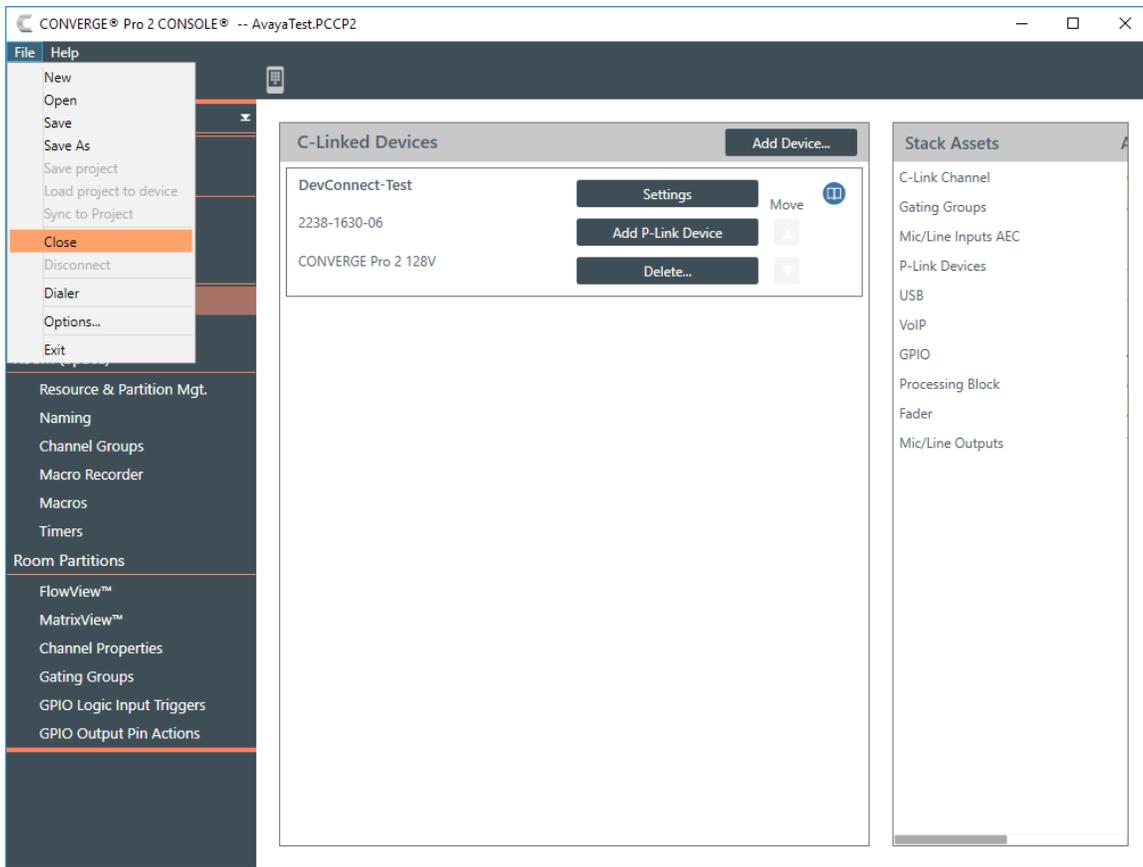
User Name	78020
Password	••••••
Reenter Password	••••••
Transport Type	UDP

**Proxy 2**

User Name	
Password	
Reenter Password	
Transport Type	UDP

Close

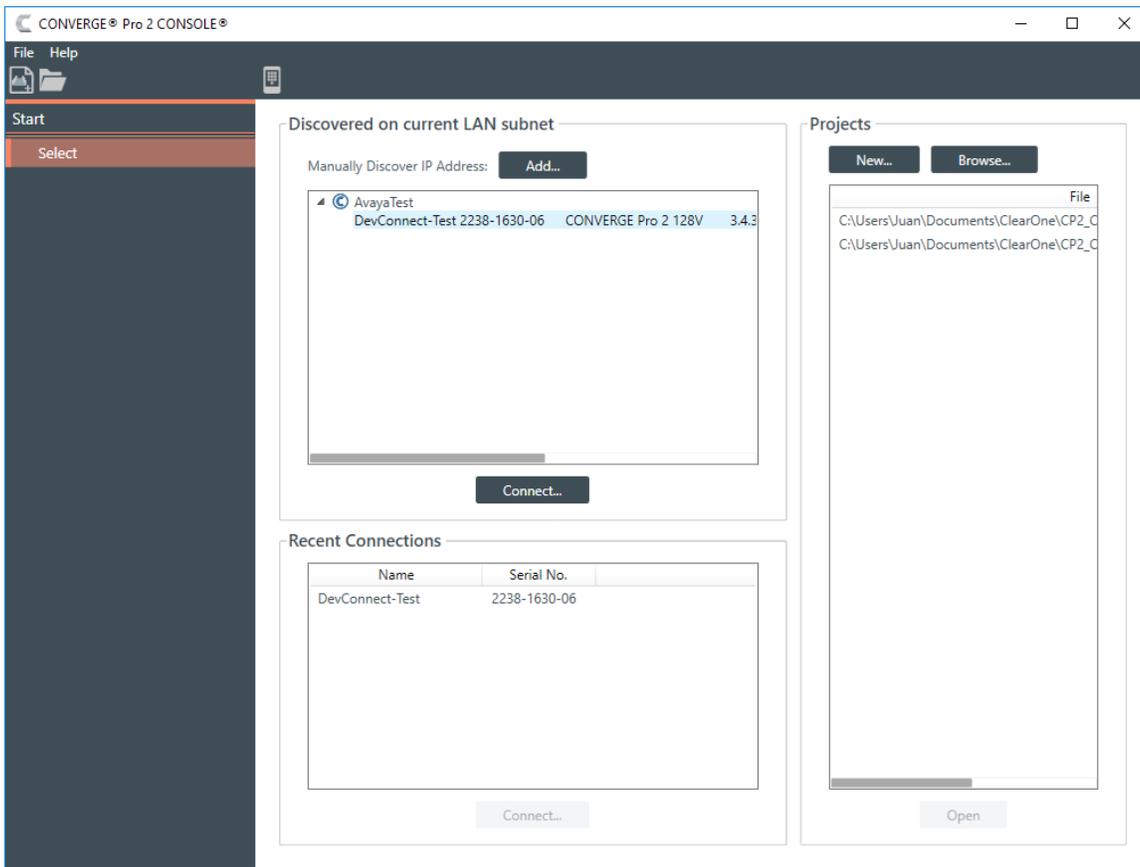
Once the Converge Pro 2 128V configuration is completed, close the configuration by selecting the File > Close menu option as shown below.



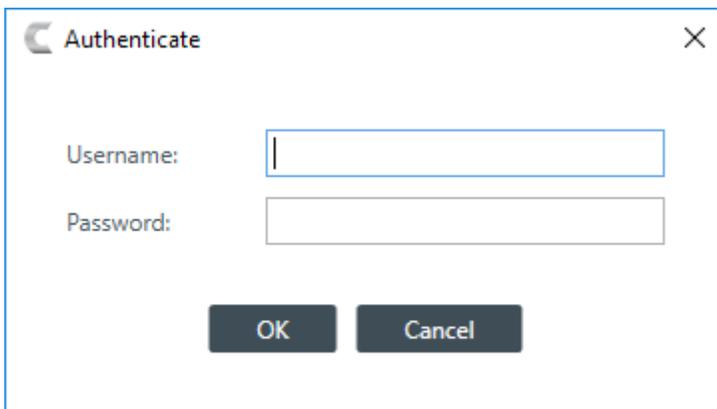
### **Load Configuration to CONVERGE Pro 2**

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.

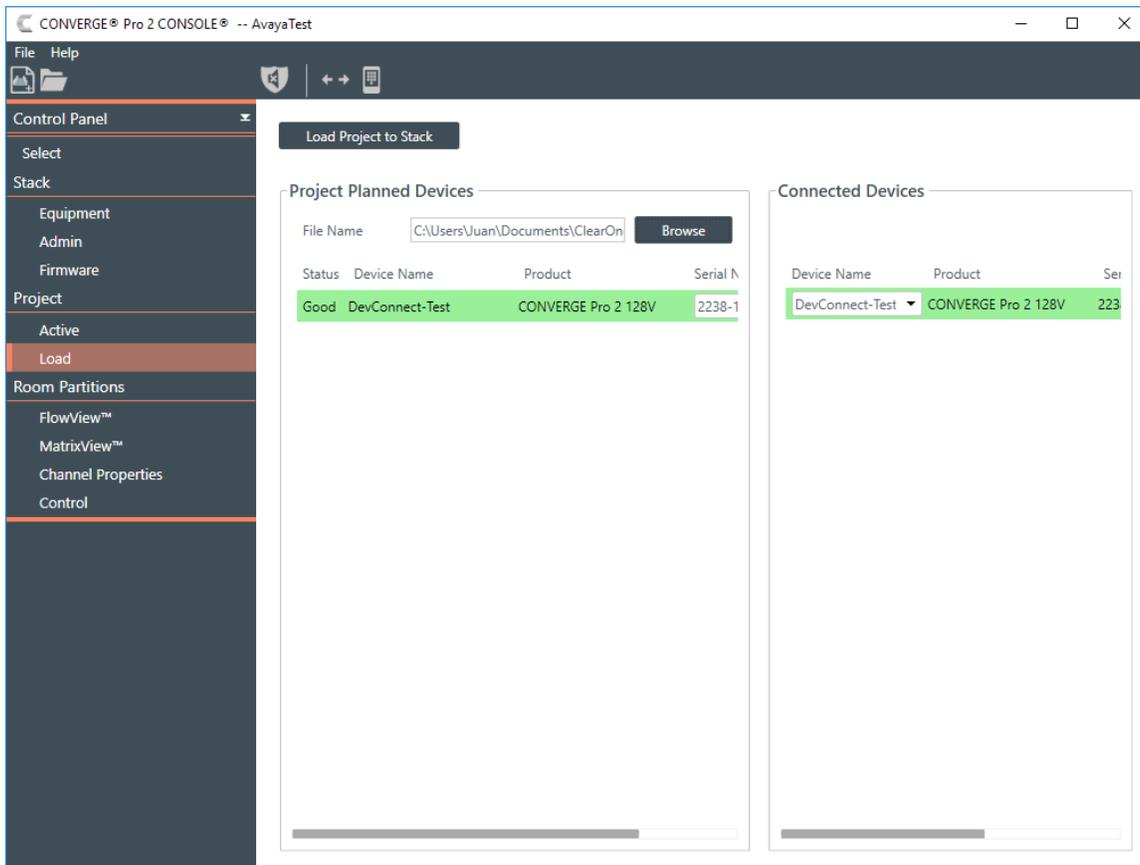
**NOTE:** The ClearOne Locator Service must be running in order for the Converge Pro 2 Console to discover any existing units.



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.

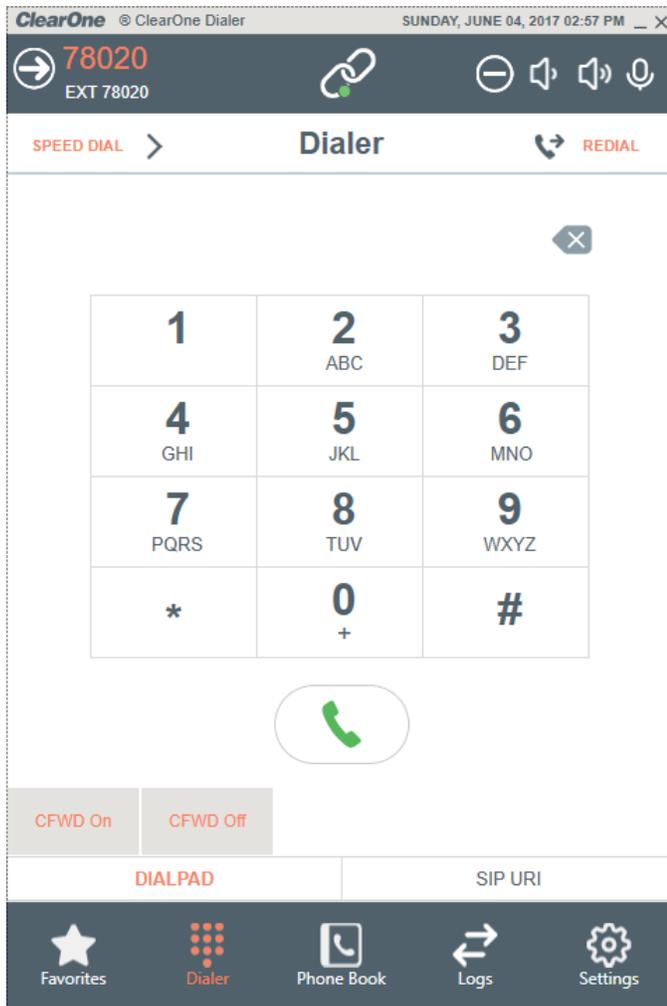


### Verification Steps

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

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