Avaya Solution & Interoperability Test Lab

Application Notes for ClearOne Converge Pro VH20 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the ClearOne Converge Pro VH20 Conferencing Solution with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP interface. The Converge Pro VH20 links other Converge Pro products to create a complete audio conferencing system that is integrated with an Avaya SIP telephony network. In this compliance test, Converge Pro VH20 provided SIP connectivity for the Converge Pro 880T equipped with a speaker and microphone. The focus of these Application Notes is on the interoperability between Converge Pro VH20, Session Manager, and Communication Manager.

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 steps required to integrate the ClearOne Converge Pro VH20 Conferencing Solution with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP interface. The Converge Pro VH20 links other Converge Pro products to create a complete audio conferencing system that is integrated with an Avaya SIP telephony network. In this compliance test, Converge Pro VH20 provided SIP connectivity for the Converge Pro 880T equipped with a speaker and microphone. The focus of these Application Notes is on the interoperability between Converge Pro VH20, Session Manager, and Communication Manager.

2. General Test Approach and Test Results

To verify interoperability of ClearOne Converge Pro VH20, linking Converge Pro 880T, with Communication Manager and Session Manager, voice calls were made to Avaya IP telephones (SIP and H.323). All test cases were performed manually.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of Converge Pro VH20 with Session Manager.
- Voice calls between Converge Pro VH20 and Avaya IP telephones (SIP and H.323), including proper call disconnect.
- Long duration calls.
- G.711 and G.729 codec support and code negotiation.
- Audio mute on Converge Pro VH20 and Avaya endpoints.
- Converge Pro VH20 DTMF support.
- Proper handling of unsuccessful calls due to call abort, no answer, dialing invalid number, and called party busy.
- Proper system recovery after a restart of Converge Pro VH20 and loss of IP connectivity.

2.2 Test Results

All test cases passed. One observation is that the Communication Manager Preferred Minimum Session Refresh Interval setting in the SIP trunk group form and the Converge Pro VH20 Min-SE Timer have to match, or incoming calls to VH20 will fail with a SIP Status message indicating "Session Interval Too Small".

2.3 Support

For technical support and information on Converge Pro VH20, contact ClearOne at:

- Phone: 800-283-5936 (toll free)
- Email: techsupport@clearone.com
- Website: http://www.clearone.com/support

3. Reference Configuration

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

- Avaya Aura® Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway. Communication Manager was configured as an Evolution Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones and video endpoints.
- Avaya Aura® System Manager used to configure Session Manager.

In addition, ClearOne Converge Pro VH20 registered as a SIP endpoint to Session Manager. Converge Pro VH20 then connected to Converge Pro 880T, which provided connectivity to a speaker and tabletop microphone. The Converge Console was used to configure the Converge Pro VH20. All SIP devices registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

Note: The focus of these Application Notes is on the Converge Pro VH20.

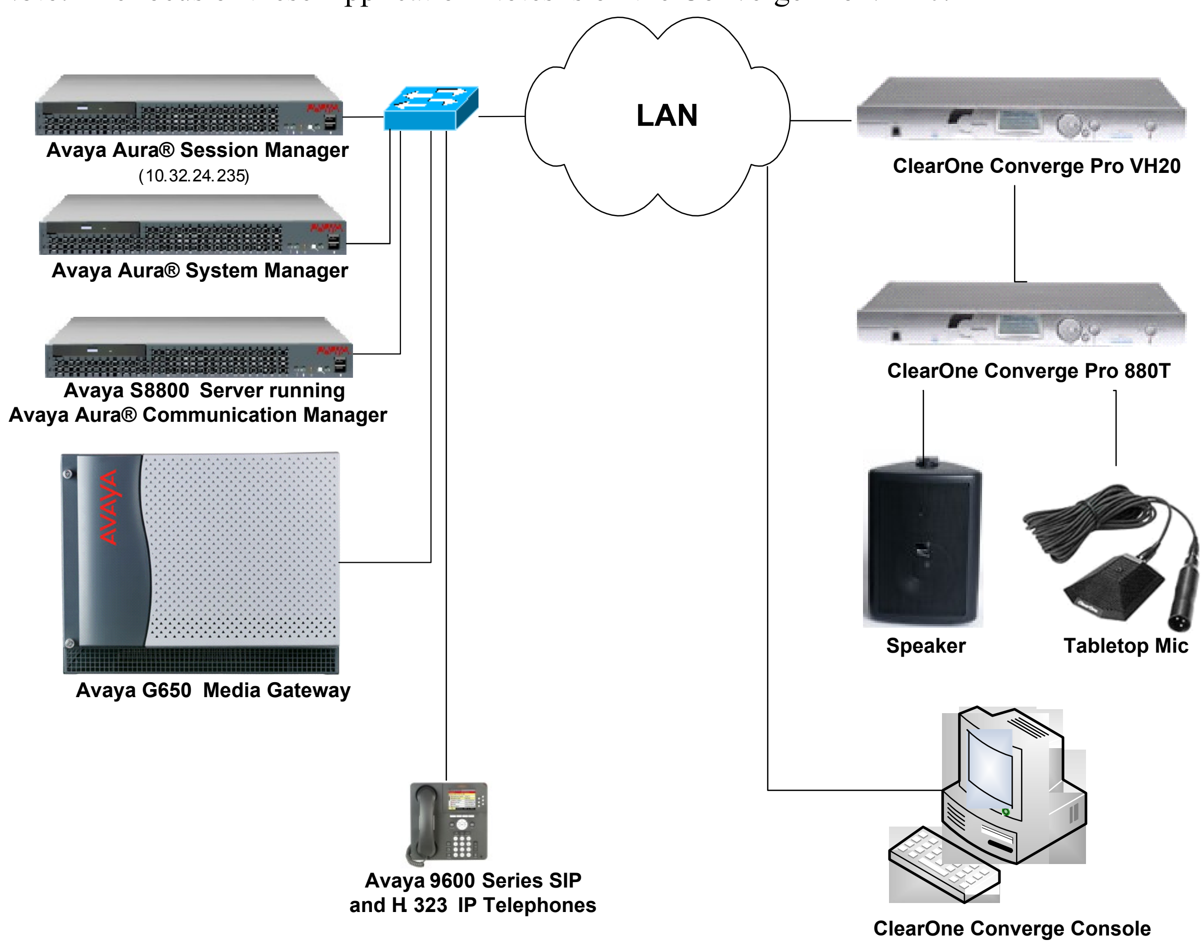


Figure 1: Avaya SIP Network with ClearOne Converge Pro VH20

4. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8800 Servers and G650 Media Gateway	Avaya Aura® Communication Manager 6.0.1 SP 3
Avaya Aura® Session Manager	6.1 (6.1.2.0-612004)
Avaya Aura® System Manager	6.1.0 (6.1.0.4.5072-6.1.4.113)
Avaya 9600 Series IP Telephones	3.011b (H.323) 2.6 (SIP)
ClearOne Converge Pro VH20	3.0.6.15
ClearOne Converge Pro 880T	3.0.6.15
ClearOne Converge Console	3.0.5

5. Configure Communication Manager

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

- Verify Communication Manager license
- Configure Converge Pro VH20 as an Off-PBX Station (OPS)
- Configure a SIP trunk between Communication Manager and Session Manager

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

5.1 Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS), video capable endpoints, and SIP Trunk options are 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  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                            USED
                                Platform Maximum Ports: 65000 161
                                      Maximum Stations: 41000 78
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 36000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 8
                    Maximum Off-PBX Telephones - PBFMC: 36000 0
                    Maximum Off-PBX Telephones - PVFMC: 36000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

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

On **Page 2** of the **system-parameters customer-options** form, verify that the number SIP trunks supported by the system is sufficient.

```
display system-parameters customer-options                       Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 30
           Maximum Concurrently Registered IP Stations: 18000 20
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2 Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the G650 Media Gateway and the Session Manager SIP interface. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.32.24.1
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
devcon-asm          10.32.24.235
medprocrm           10.32.24.21
procr               10.32.24.10
procr6              ::
 ( 8 of 8   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

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 G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
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? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        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 VH20. 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. Testing was also performed with the G.729B codec.

```
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:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- Set the **Transport Method** field to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The default values for the other fields may be used.

```
add signaling-group 50                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             Group Type: sip
  IMS Enabled? n       Transport Method: tcp
        Q-SIP? n                                              SIP Enabled LSP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM

   Near-end Node Name: clancrm               Far-end Node Name: devcon-asm
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
                                    Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP endpoints. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 50                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-asm              COR: 1        TN: 1        TAC: 1050
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 50
                                                     Number of Members: 10
```

On Page 2, verify that the **Preferred Minimum Session Refresh Interval (sec)** field matches the configuration on Converge Pro VH20 as described in **Section 7**. In this compliance test, Converge Pro VH20 was configured to match the setting on Communication Manager. That is, the Min SE Timer on Converge Pro VH20 was set to 180 sec. Alternatively, this timer can be changed in the trunk group to match the setting on Converge Pro VH20.

```
add trunk-group 50                                            Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 90

 Disconnect Supervision - In? y  Out? y


             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

5.3 Configure Station for ClearOne Converge Pro VH20

The **station** and **off-pbx-telephone station-mapping** configuration shown in this section was automatically performed after creating the **User** in Session Manager as described in **Section 6.7**. In this section, simply verify the settings. Note that the **User** has to be added in Session Manager first before it can be viewed on Communication Manager. Alternatively, this configuration could have also been performed manually.

Use the **display station** command to view the station created for Converge Pro VH20 and verify the settings in bold. Note that the **IP Video** field must be set to *y*.

```
add station 78305                                            Page   1 of   6
                                     STATION

Extension: 78401                         Lock Messages? n             BCC: 0
     Type: 9630SIP                       Security Code:                TN: 1
     Port: IP                          Coverage Path 1:               COR: 1
     Name: 78305, VH20                 Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 78305

        Display Language: english                  Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
```

Use the **display off-pbx-telephone station-mapping** command to view the mapping of the Communication Manager extensions (e.g., 78305) to the same extension configured in System Manager. Verify the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

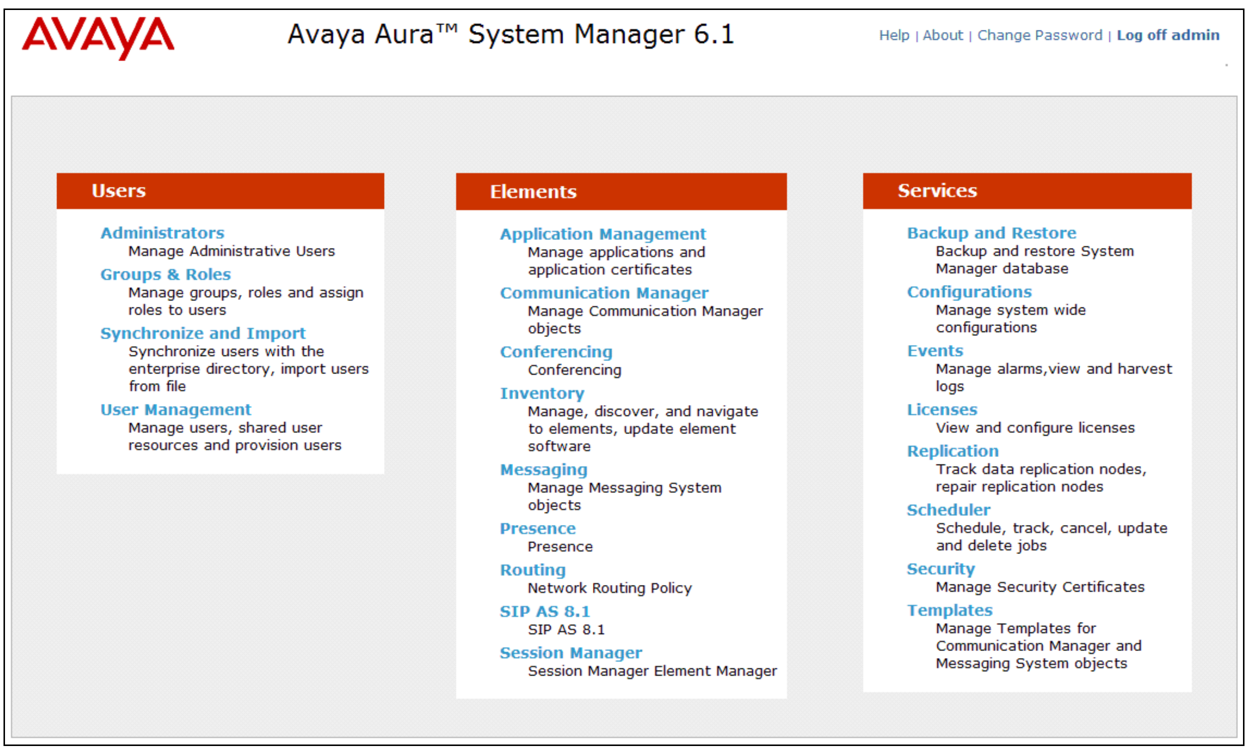
```
change off-pbx-telephone station-mapping 78305                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 78305            OPS           -     78305            aar        1
```

6. Configure Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Define Communication Manager as Administrable Entity (i.e., Managed Element)
- Application Sequence
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager
- Add SIP User

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.   The initial screen is displayed as shown below. The configuration in this section will be performed under **Routing** and **Session Manager** listed within the **Elements** box.



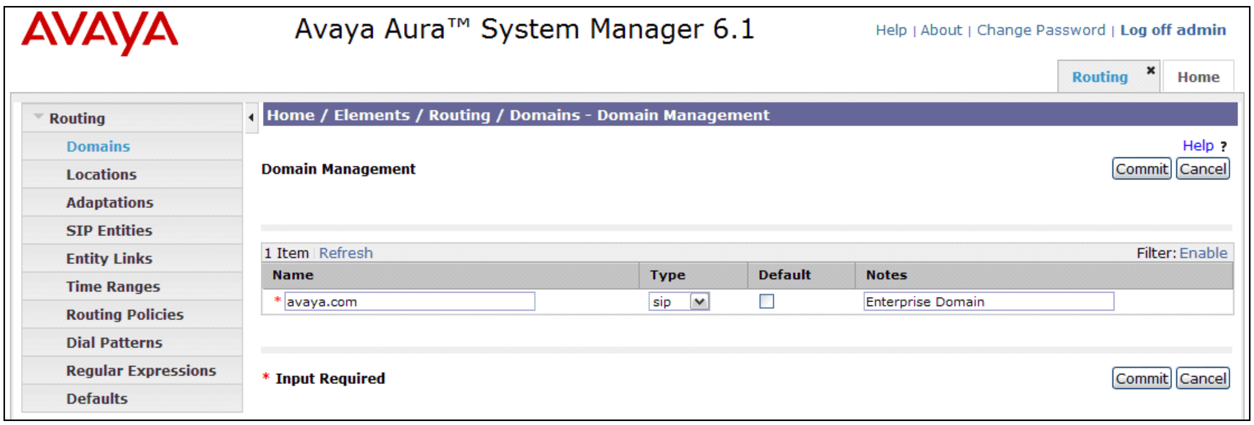
6.1 Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



6.2 Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

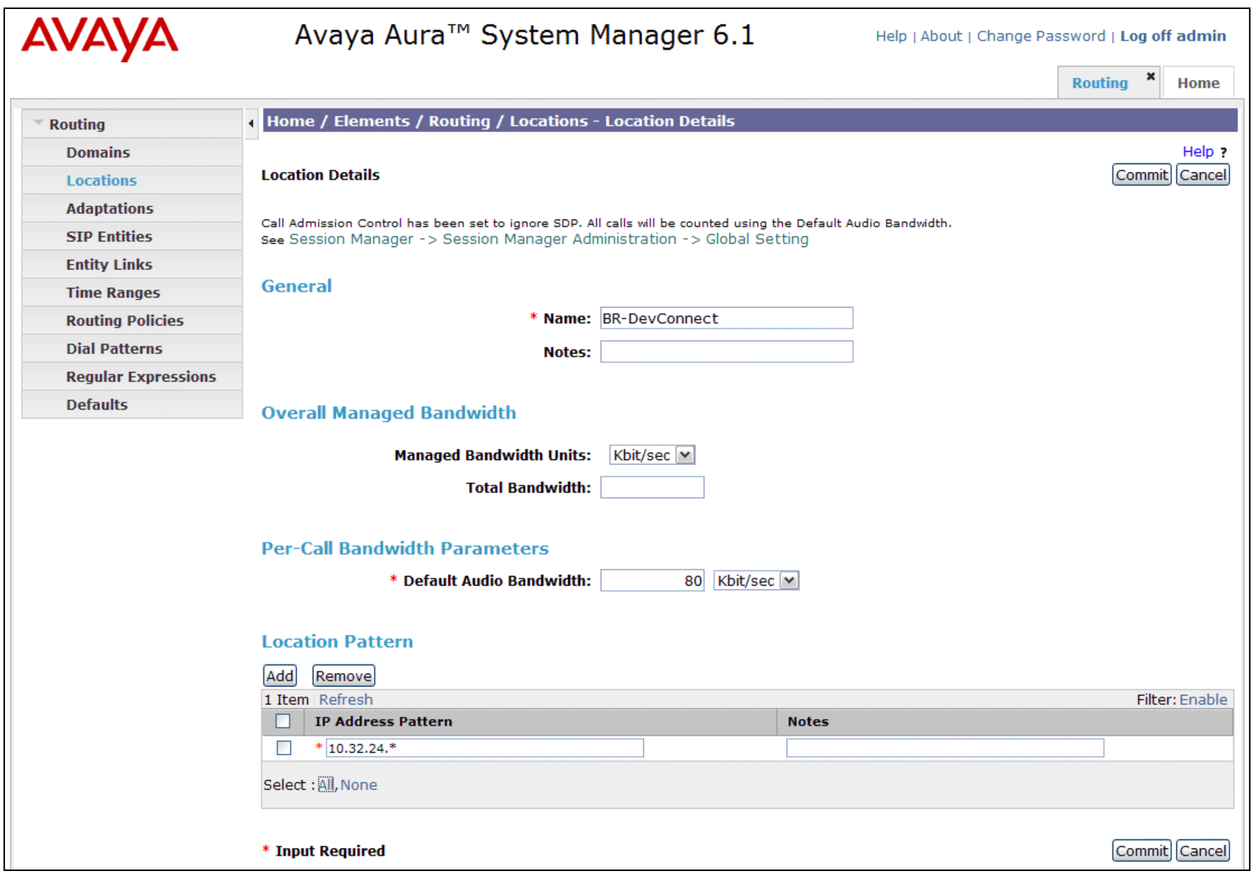
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *BR-DevConnect* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.



6.3 Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and the C-LAN in the G650 Media Gateway.

6.3.1 Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

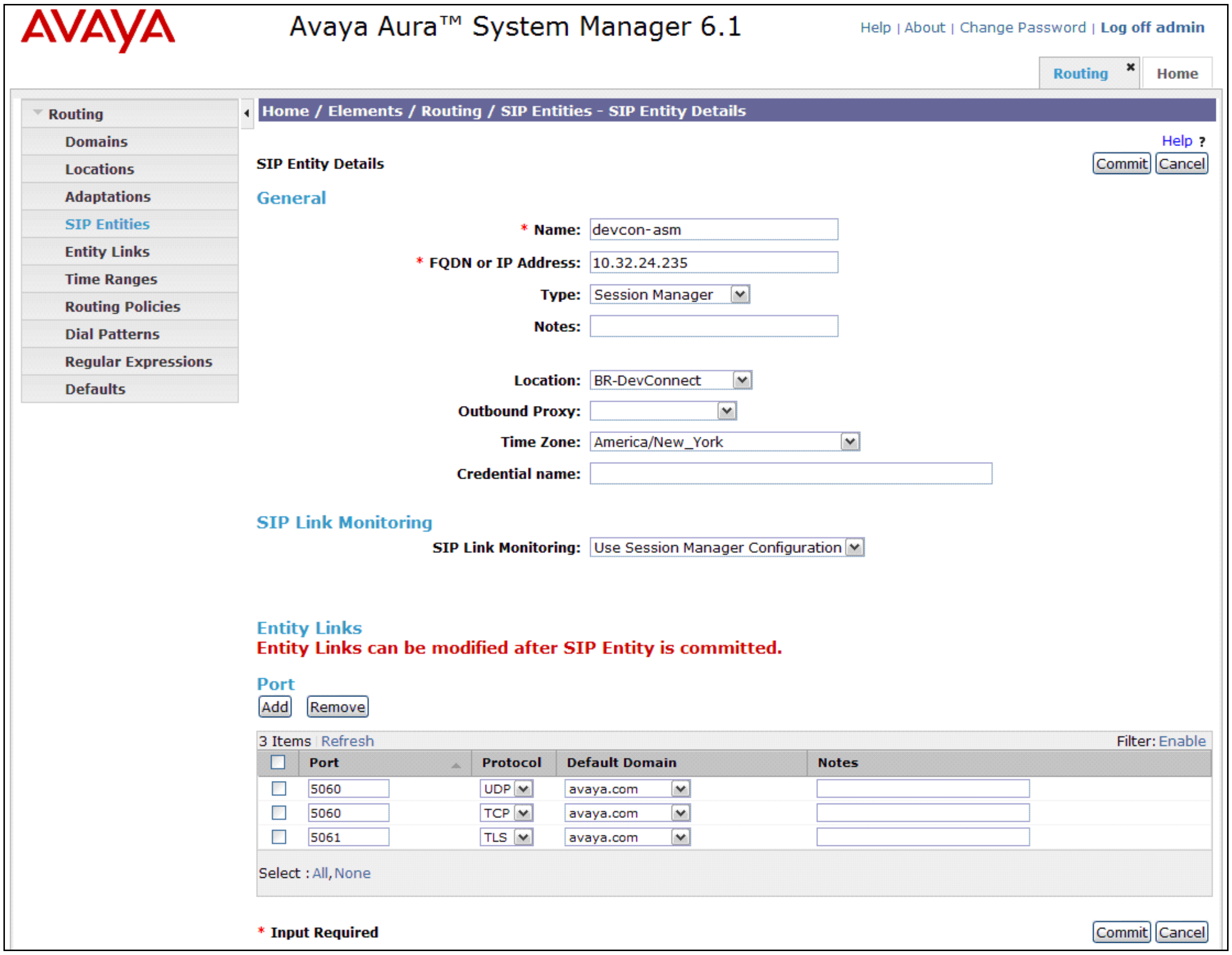
Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.


AVAYA
Avaya Aura™ System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / SIP Entities - SIP Entity Details
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: devcon-asm
* FQDN or IP Address: 10.32.24.235
Type: Session Manager
Notes:
Location: BR-DevConnect
Outbound Proxy:
Time Zone: America/New_York
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Entity Links can be modified after SIP Entity is committed.
Port
Add
Remove
3 Items Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 UDP avaya.com
5060 TCP avaya.com
5061 TLS avaya.com
Select : All, None
* Input Required
Commit
Cancel

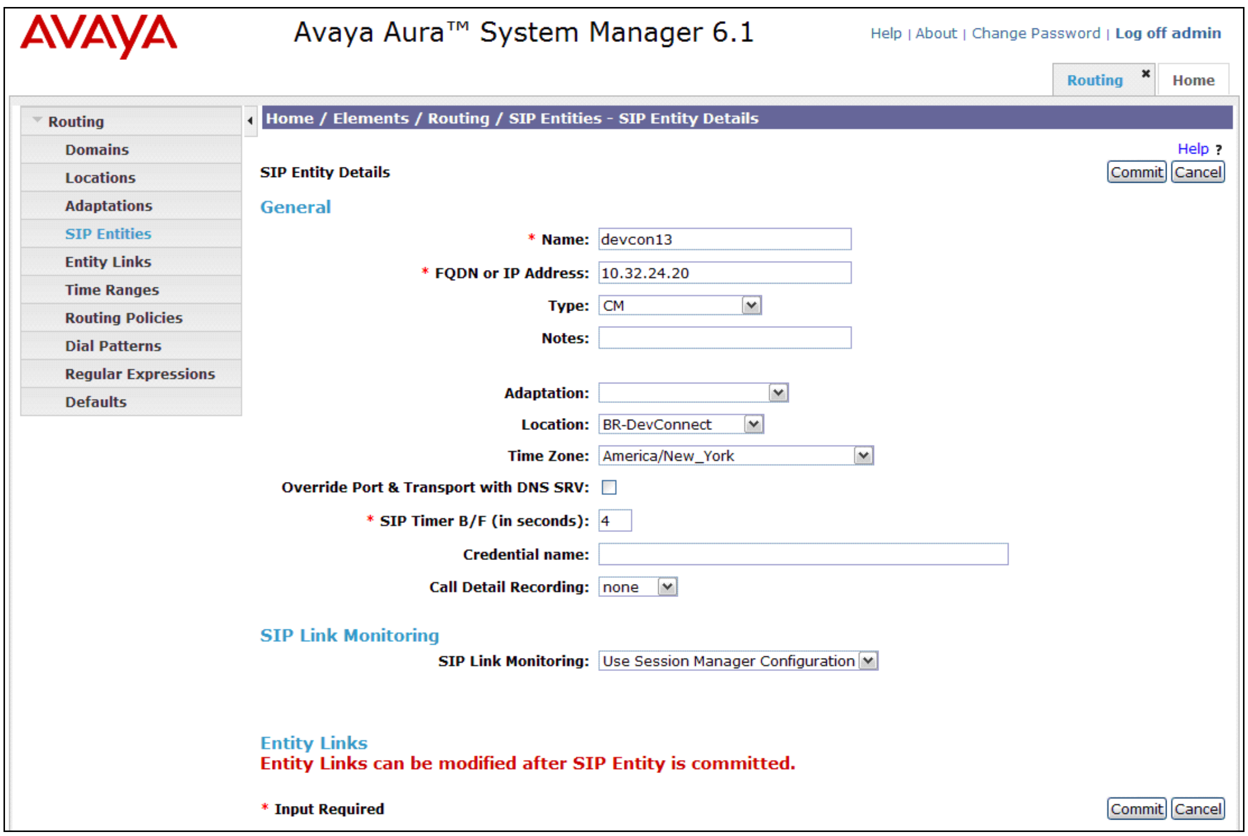
6.3.2 Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

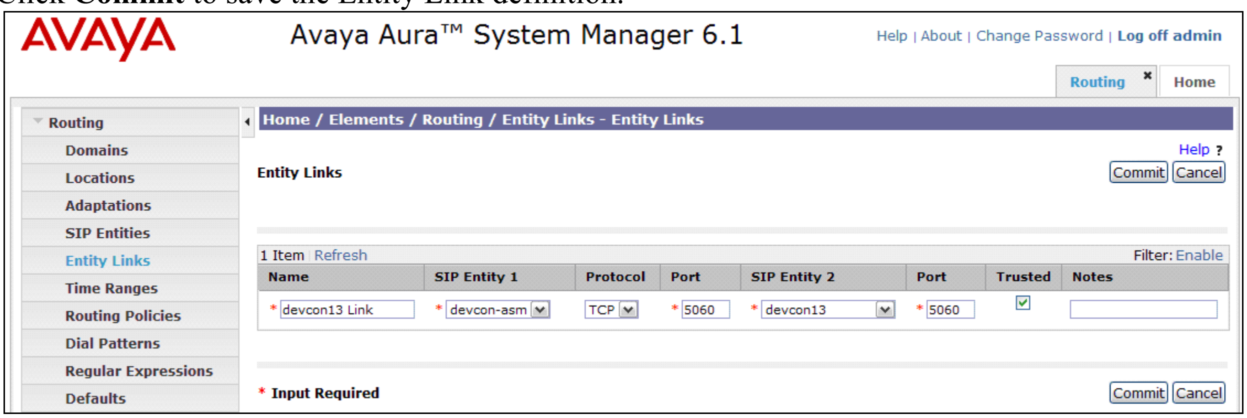


6.4 Add Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

Click **Commit** to save the Entity Link definition.



6.5 Define Communication Manager as Managed Element

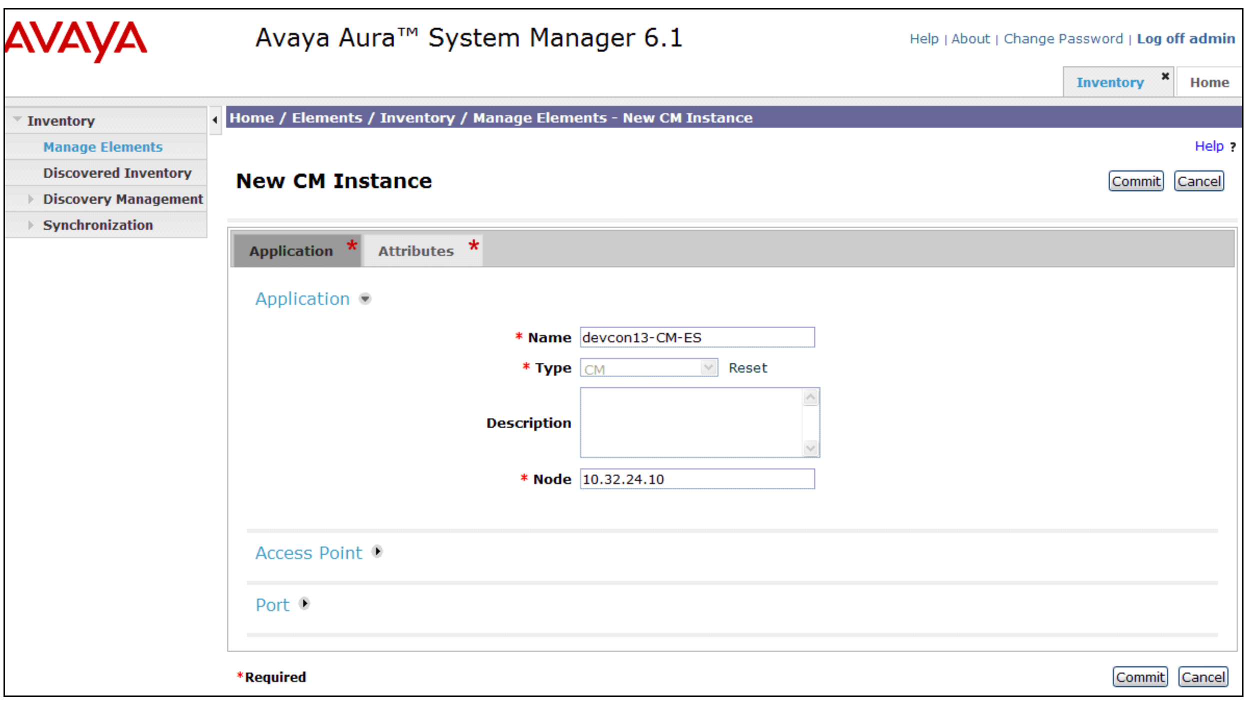
Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

To define Communication Manager as a managed element, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **New Entities Instance** screen (not shown), select *CM* in the **Type** field can click **Commit**.

In the **New CM Instance** screen, fill in the following fields as follows:

In the *Application* tab:

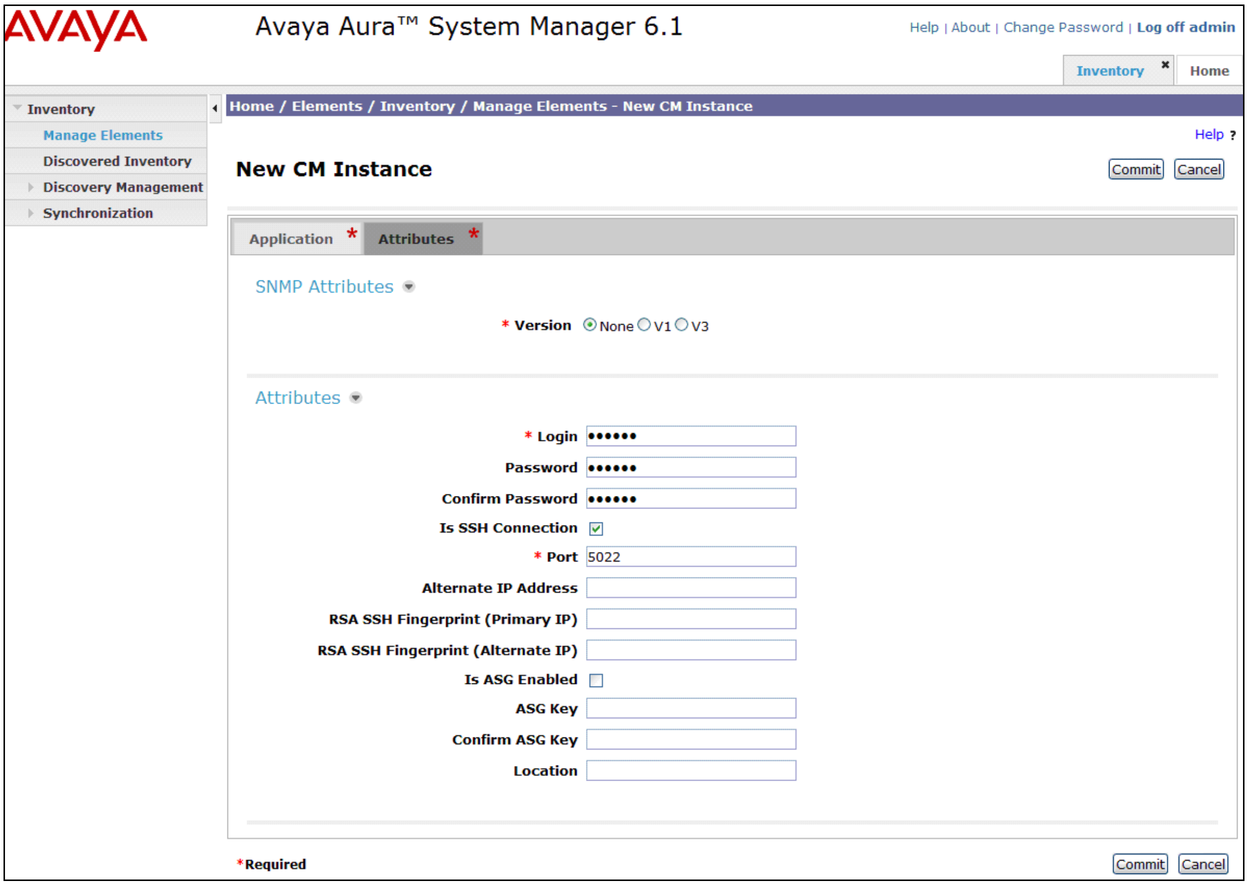
- **Name:** Enter an identifier for Communication Manager.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for Communication Manager.



In the *Attributes* tab:

- **Login / Password:** Enter the login and password used for administration access.
- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Defaults can be used for the remaining fields. Click **Commit** to save the settings.

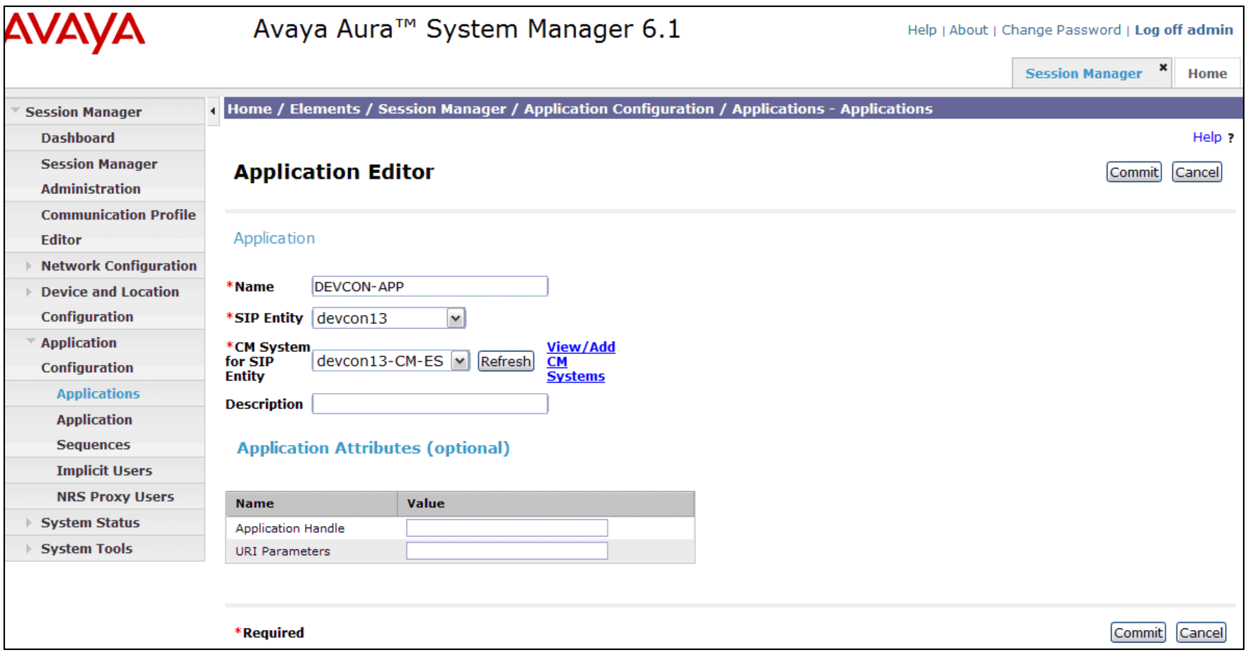


6.6 Add Application Sequence

To define an application for Communication Manager, navigate to **Elements → Session Manager → Application Configuration → Applications** on the left and select **New** button (not shown) on the right. Fill in the following fields:

- **Name:** Enter name for application.
- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity** Select the Communication Manager managed element.

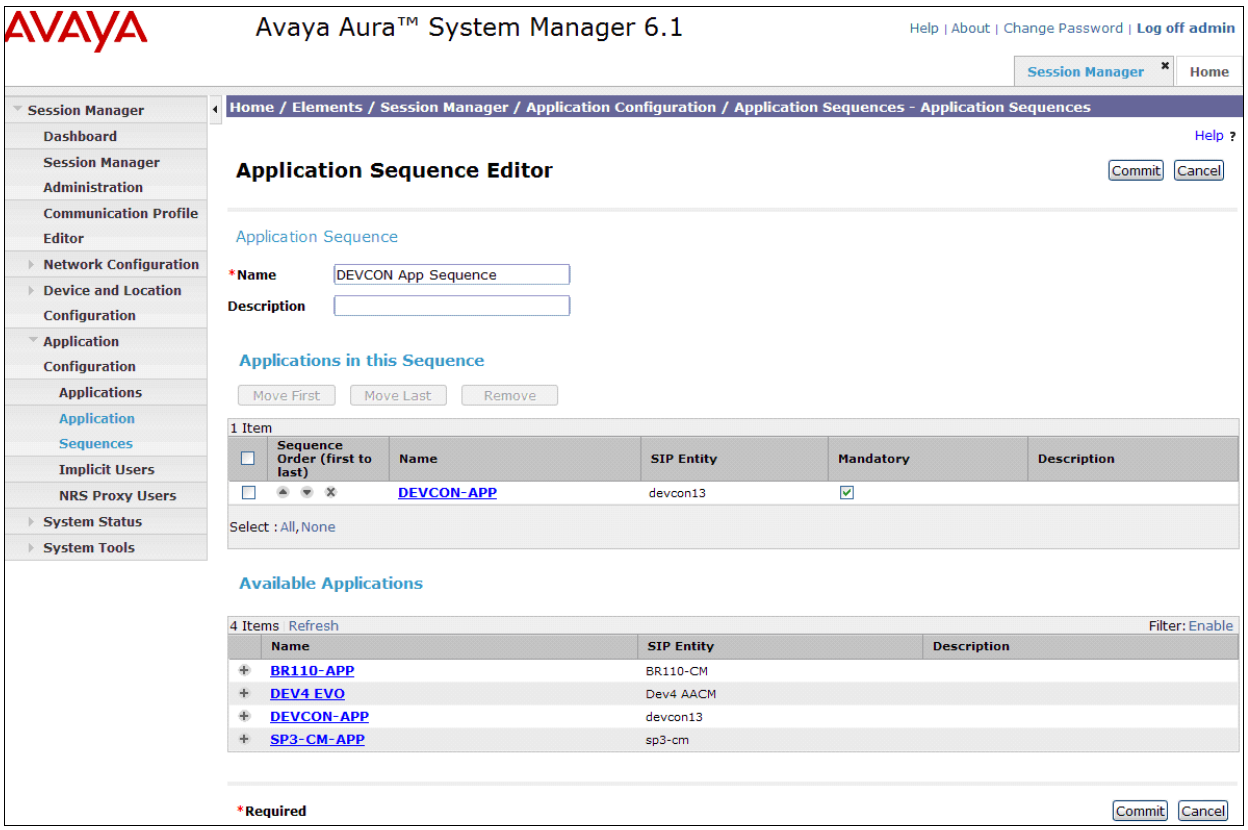
Click **Commit** to save the Application definition.



Next, navigate to **Elements → Session Manager → Application Configuration → Application Sequences** to define the Application Sequence for Communication Manager as shown below. Provide a **Name** (e.g., *DEVCON App Sequence*) for the Application Sequence and under **Available Applications**, click on the plus (+) sign by *DEVCON-APP* to add it under the **Application in this sequence** section.

Verify a new entry is added to the **Applications in this Sequence** table and the **Mandatory** column is ☑ as shown below.

Note: The Application Sequence defined for Communication Manager Evolution Server can only contain a single Application.



6.7 Add SIP User

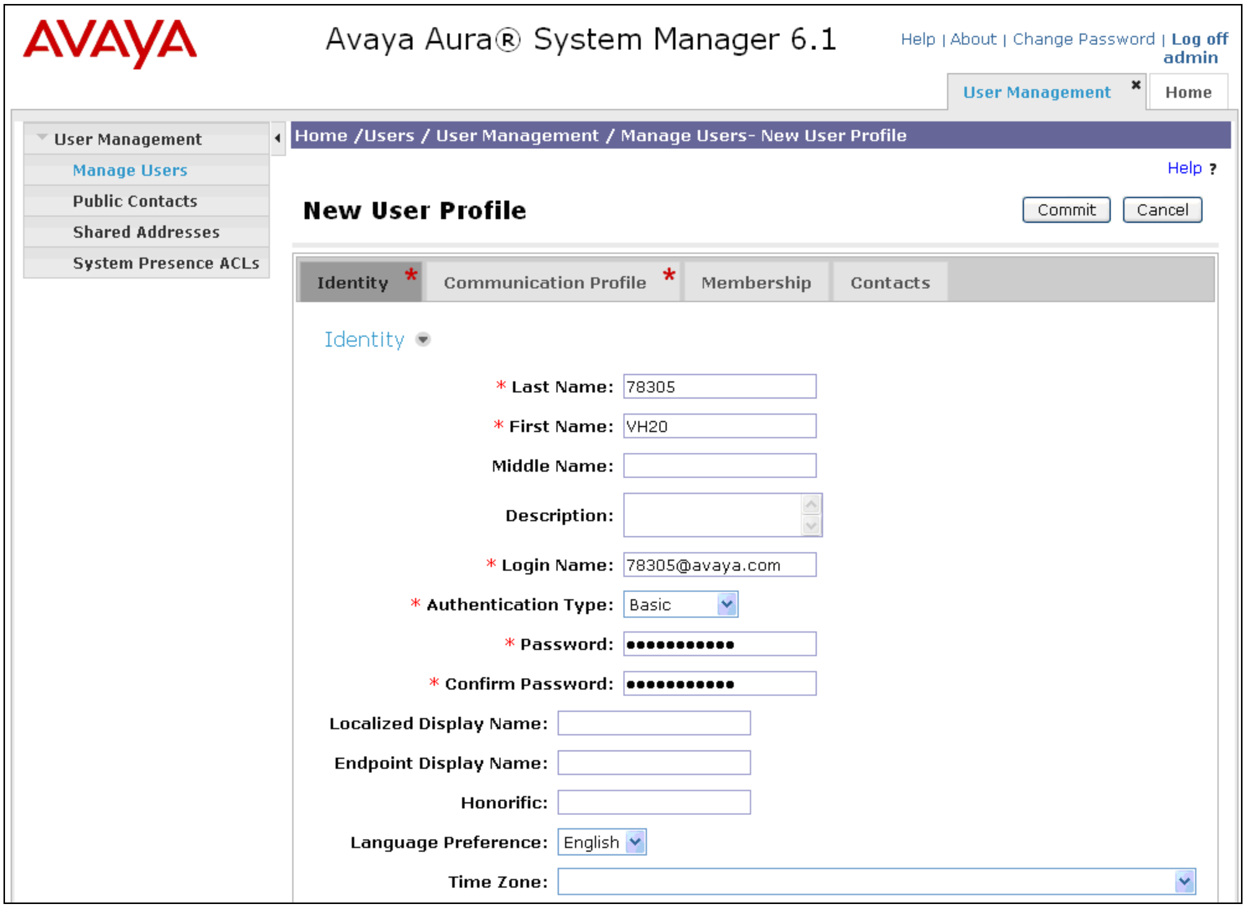
Add a SIP user for Converge Pro VH20**.** The following configuration will automatically create the SIP station on Communication Manager Evolution Server.

To add new SIP users, navigate to **Users → User Management → Manage Users** from the left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** tab of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>*@*<sip domain>* of the user (e.g., 78305@avaya.com).
- **Authentication Type:** Select *Basic***.**
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information when adding a new SIP user to the sample configuration.



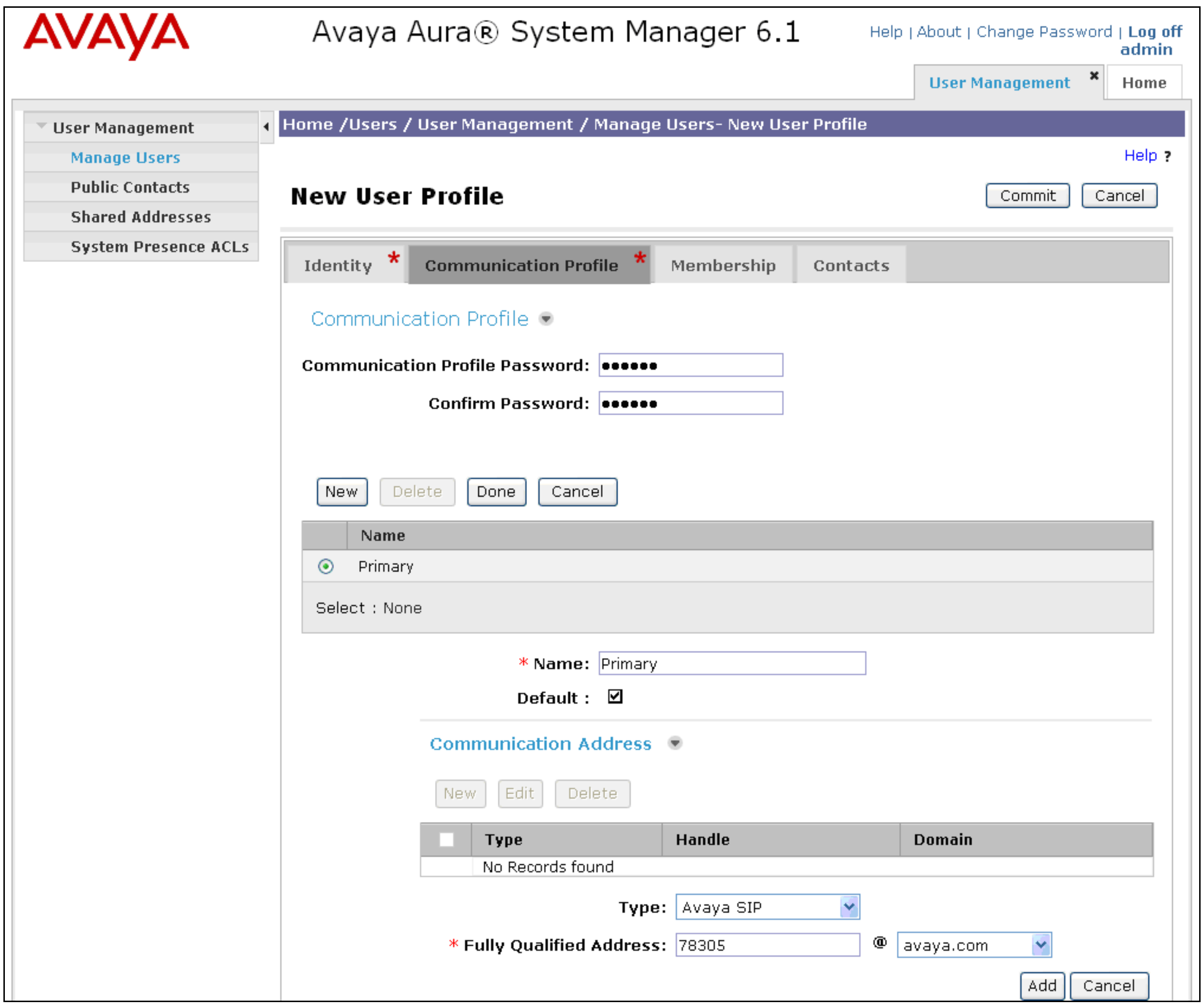
Enter values for the following required attributes for a new SIP user in the **Communication Profile tab** of the new user form.

- **Communication Profile Password:** Enter the password which will be used by Converge Pro VH20 to register with Session Manager.
- **Confirm Password:** Re-enter the password from above.

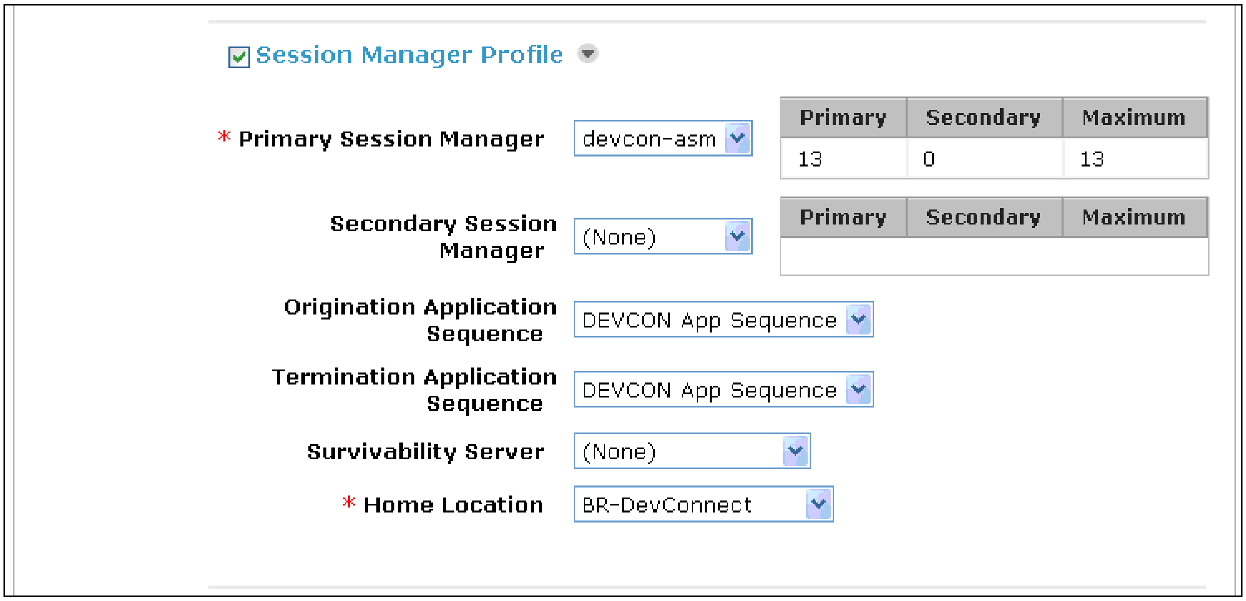
Scroll down to the **Communication Address** section and select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.

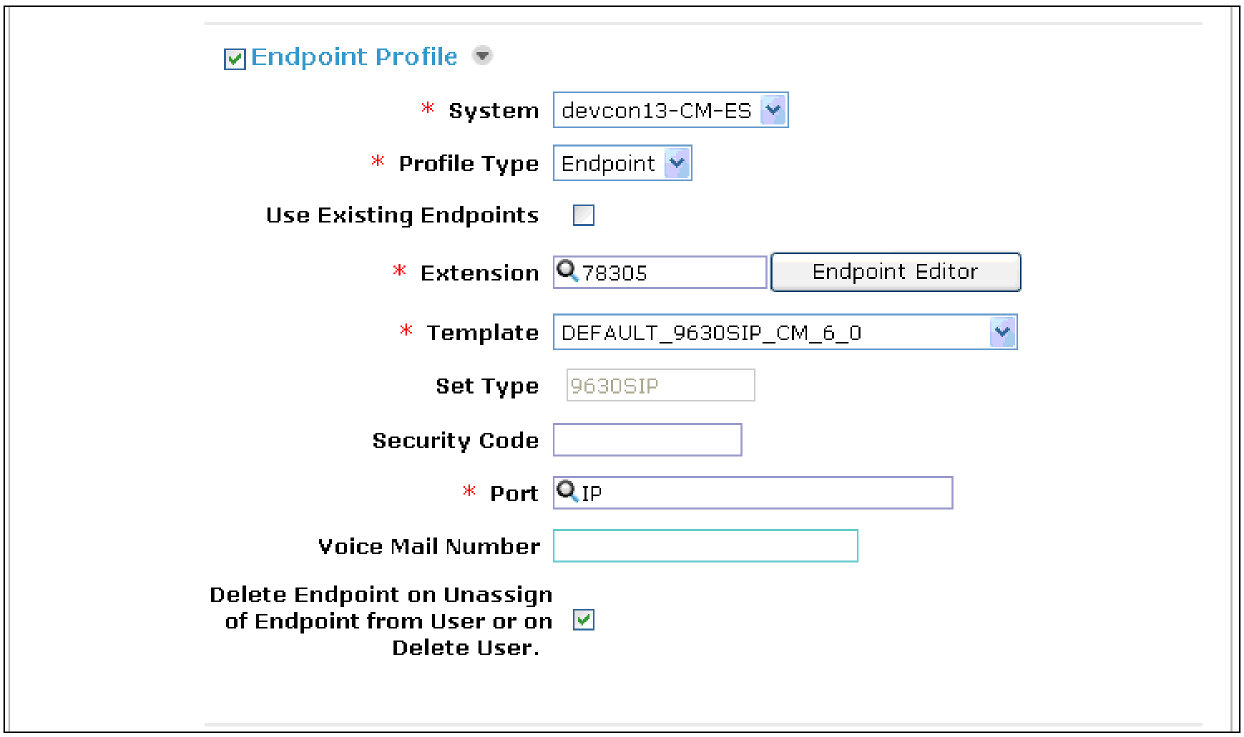


In the *Session Manager Profile* section, specify the Session Manager entity from **Section 6.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 6.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence. Set the **Home Location** field to the **Location** configured in **Section 6.2**.



In the **Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Endpoint on Unassign of Endpoint:** Enable field to automatically delete station when **Endpoint Profile** is un-assigned from user.



6.8 Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

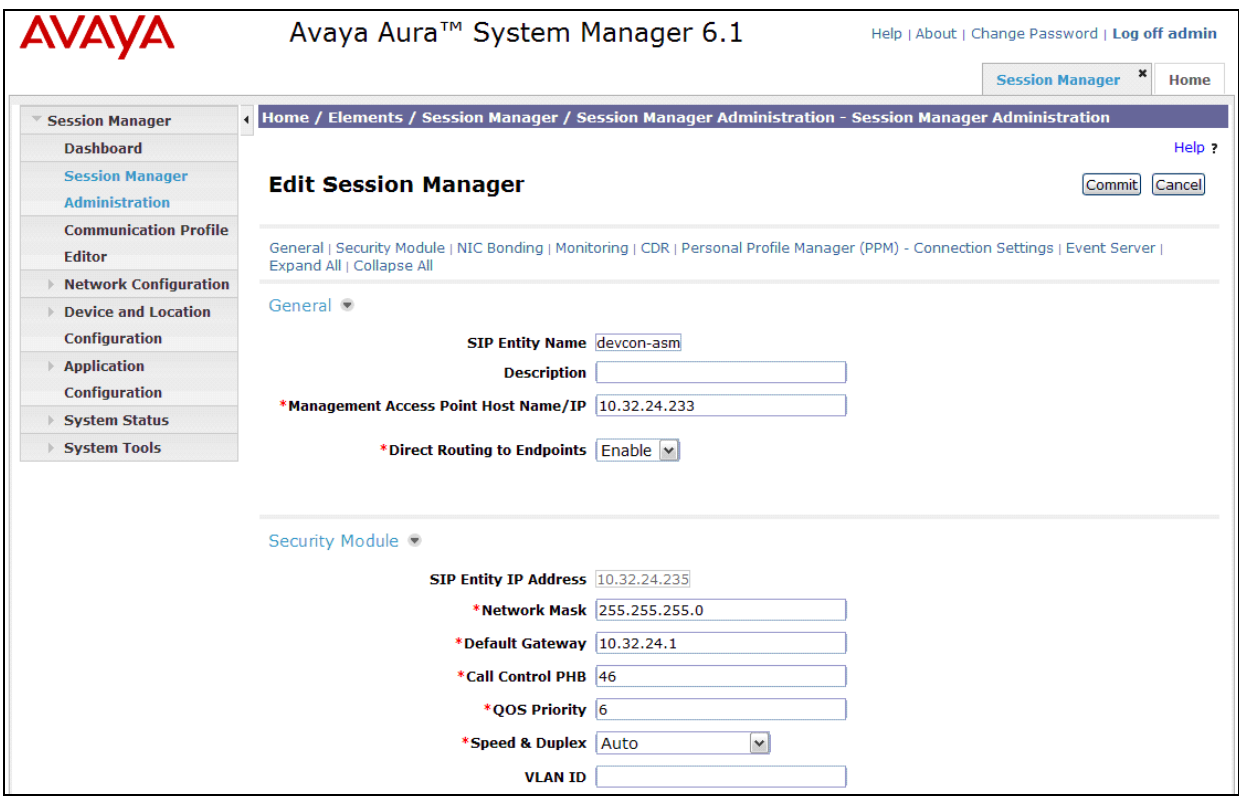
Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager

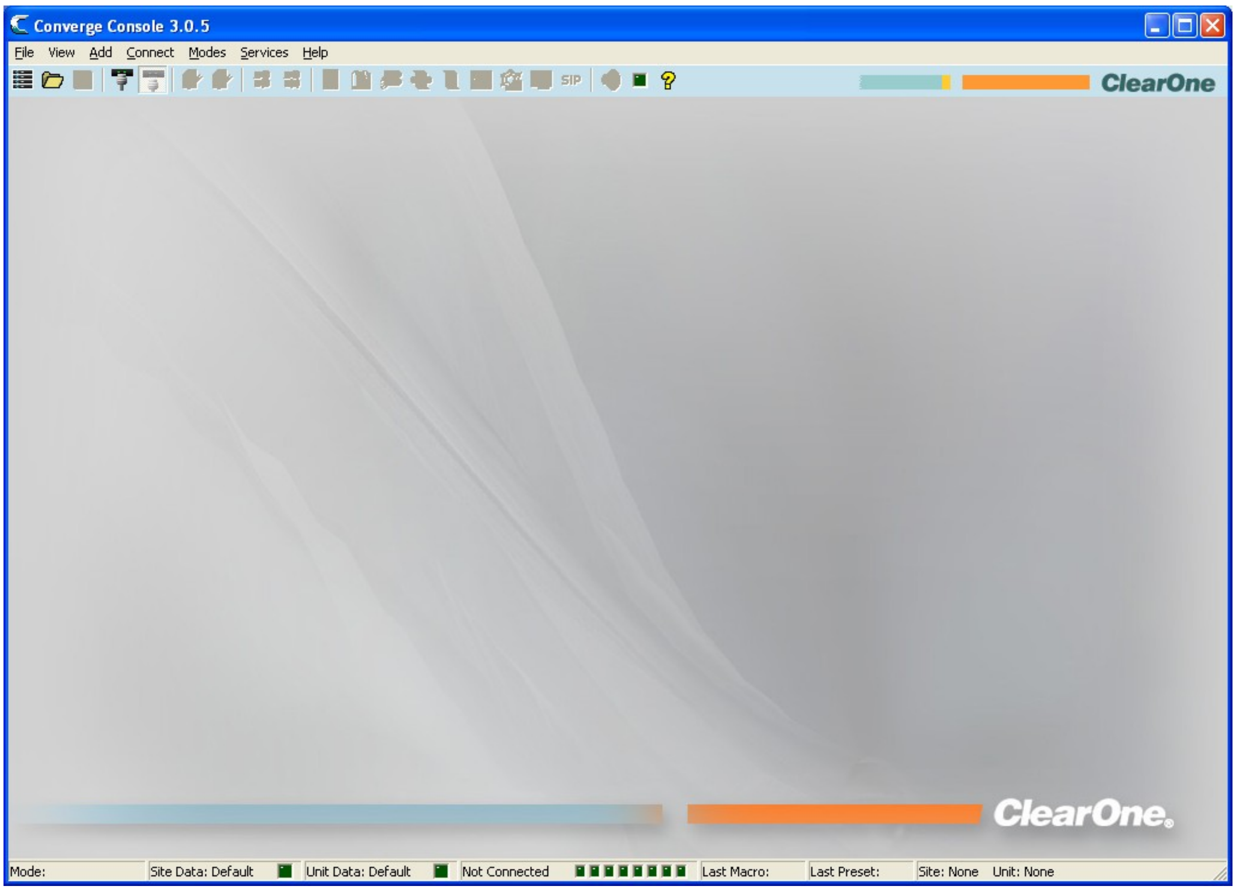
Use default values for the remaining fields. Click **Commit** to add this Session Manager.



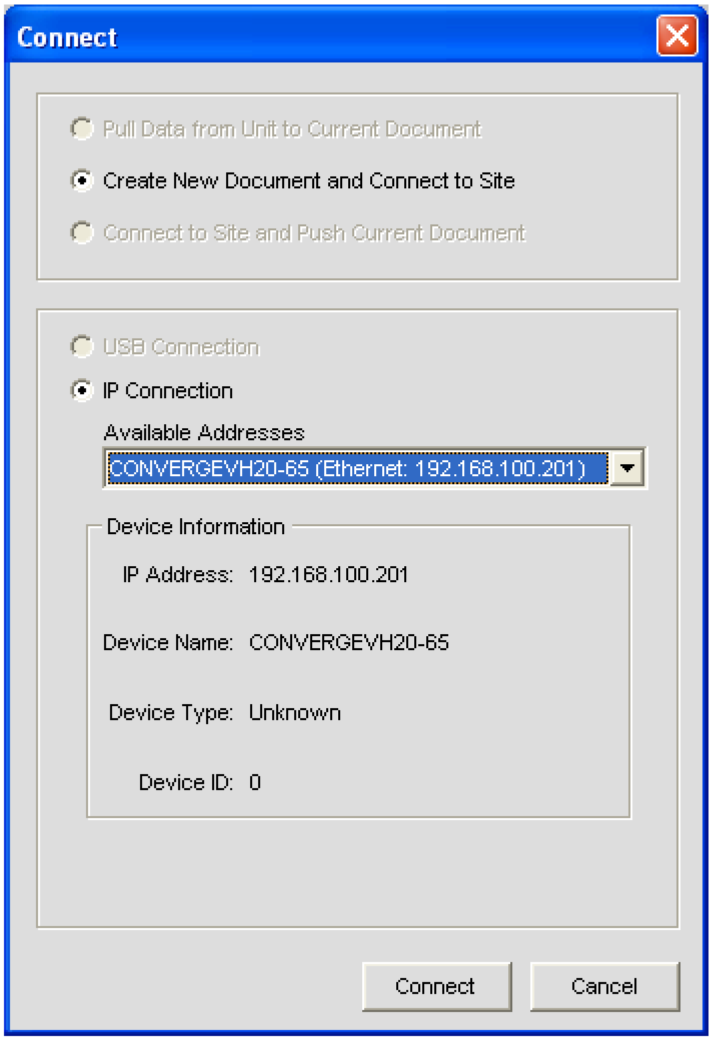
7. Configure ClearOne Converge Pro VH20

The configuration of Converge Pro VH20 was performed via the Converge Console. The Converge Pro VH20 used DHCP to obtain an IP address. Refer to [3] for additional information on configuring Converge Pro VH20.

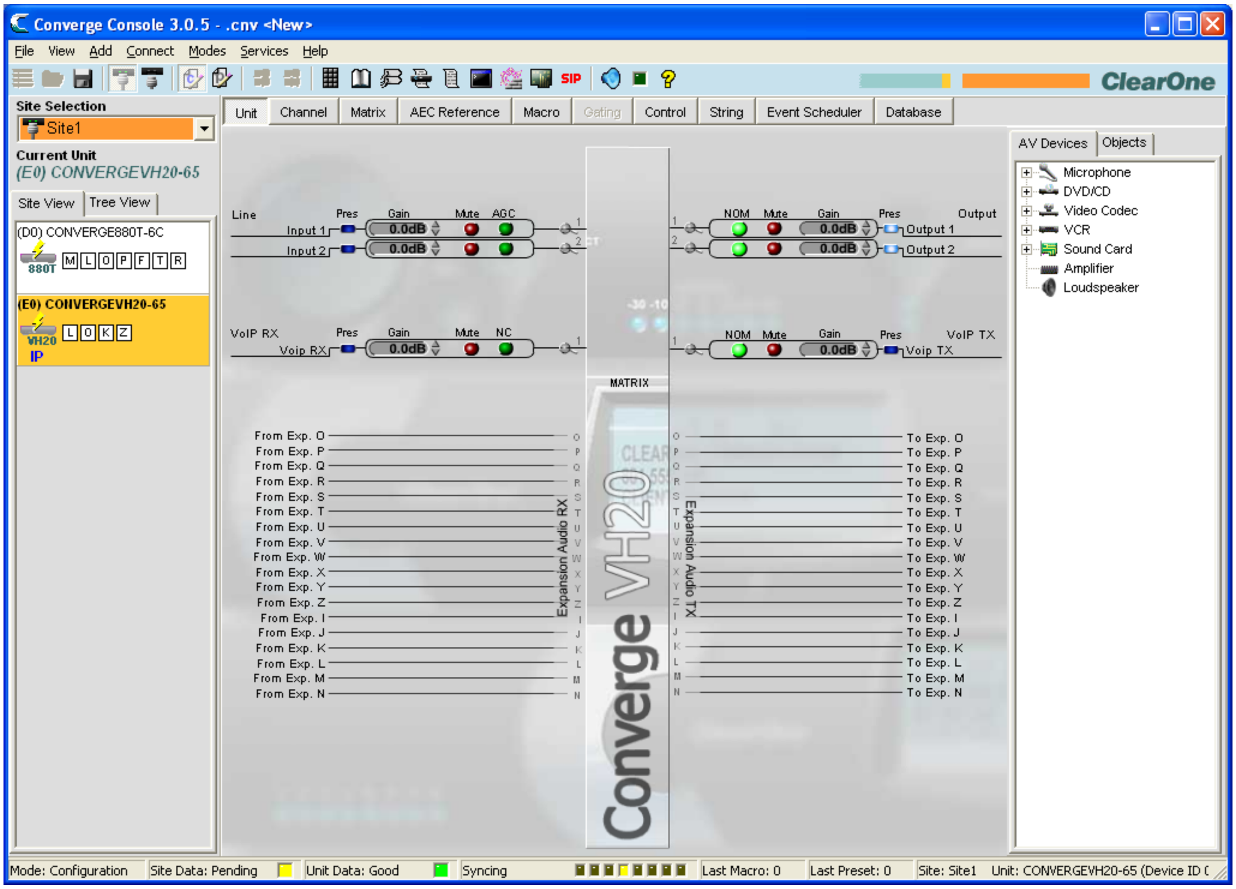
From a PC with the Converge Console installed, launch the application to display the window below.



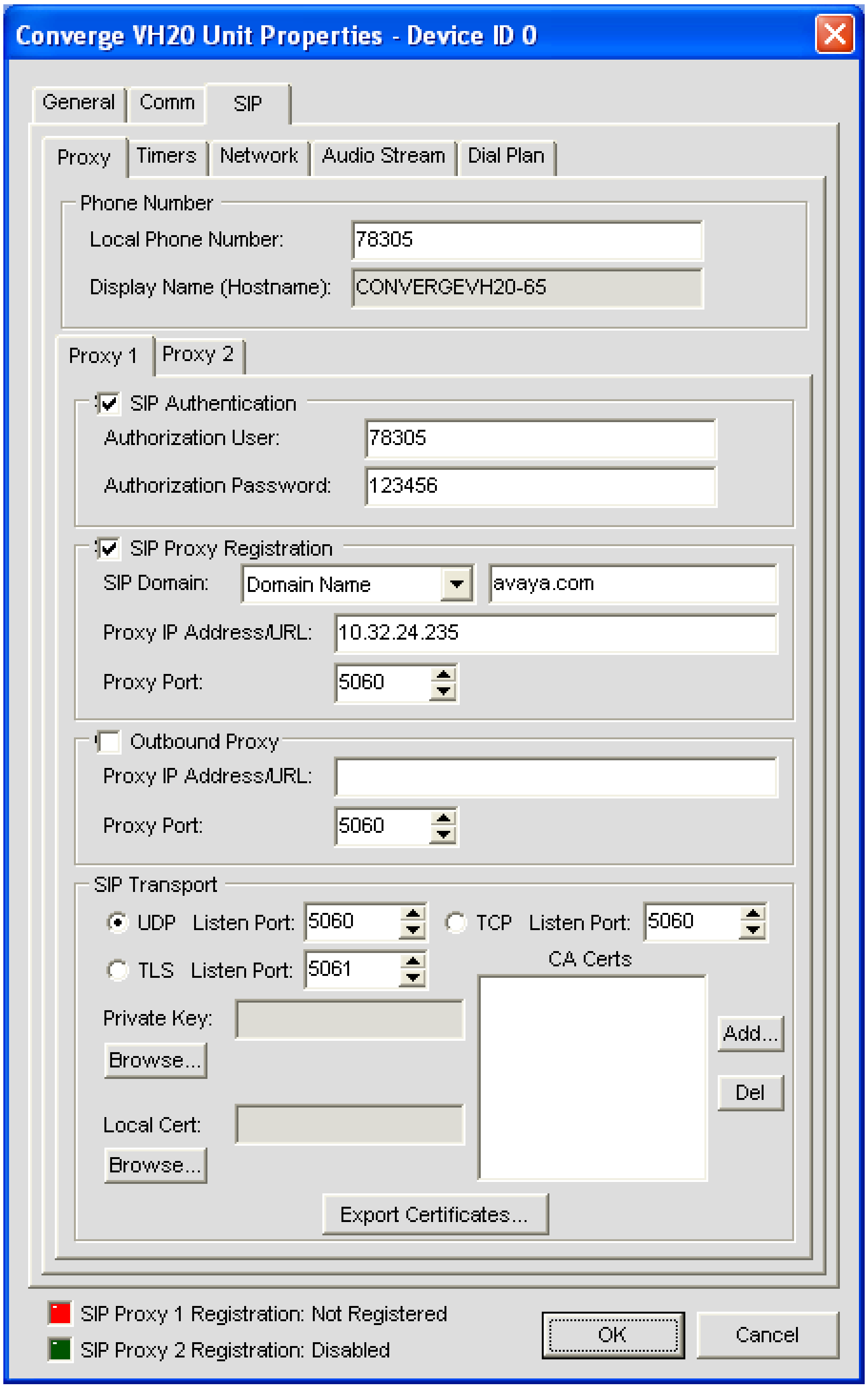
From the **Converge Console**, select the **Connect** option in the menu bar to display the **Connect** window shown below. Select the **IP Connection** radio button. In the **Available Addresses** field, select the entry for Converge VH20 with the IP address obtained from DHCP. Click the **Connect** button.



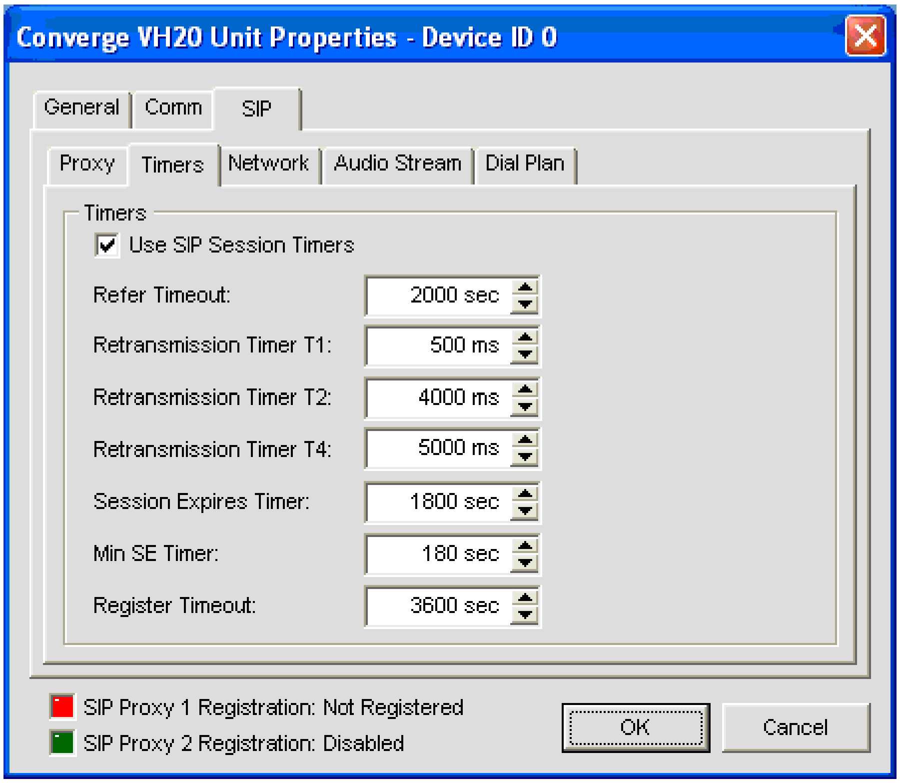
Once the Converge Console is connected to Converge Pro VH20, the following window is displayed. Select the Converge Pro VH20 in the left pane, and then click the **SIP** icon (in red) under the menu bar.



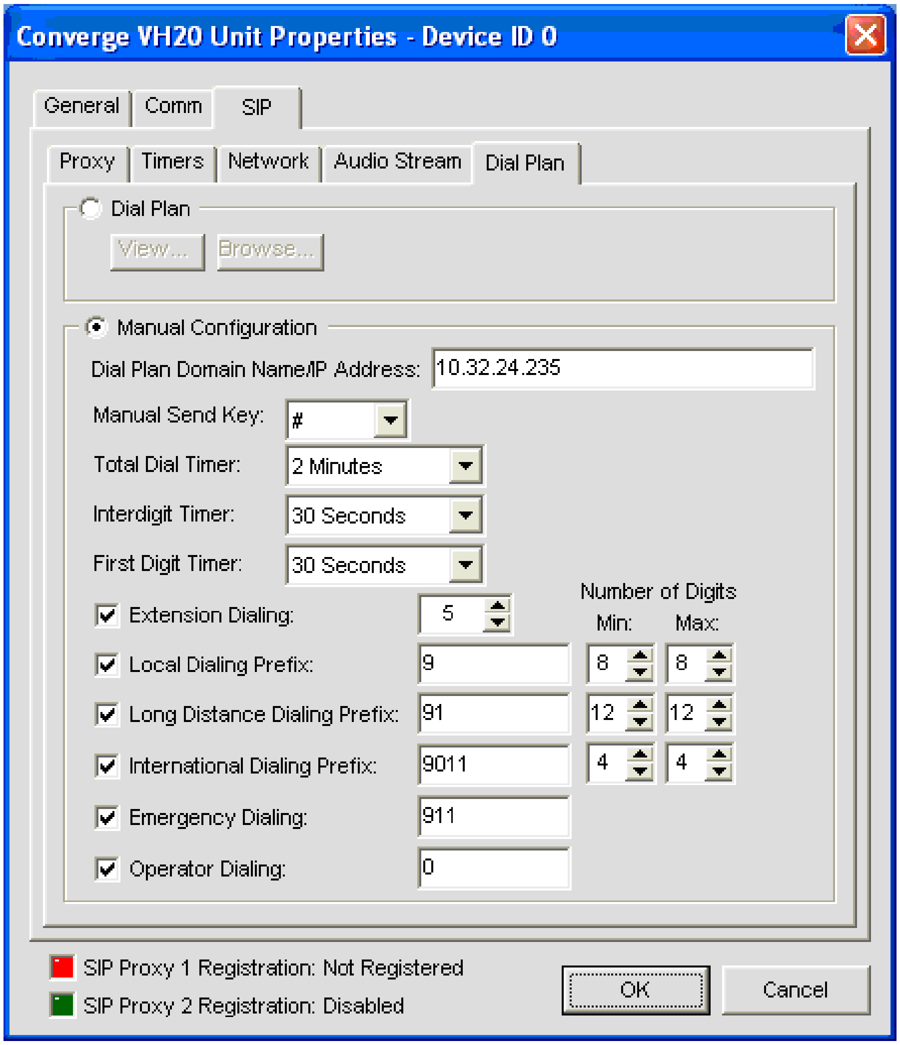
After selecting the SIP option, the **Converge VH20 Unit Properties** window is displayed as shown below. Select the **SIP** tab and configure the SIP parameters as shown below, including the VH20's extension number in the **Local Phone Number** and **Authorization User** fields and its **Authorization Password**. Next, configure the **SIP Domain**, the Proxy IP address/URL (i.e., Session Manager IP address), and the **Proxy Port**. Lastly, configure the **SIP Transport** method.



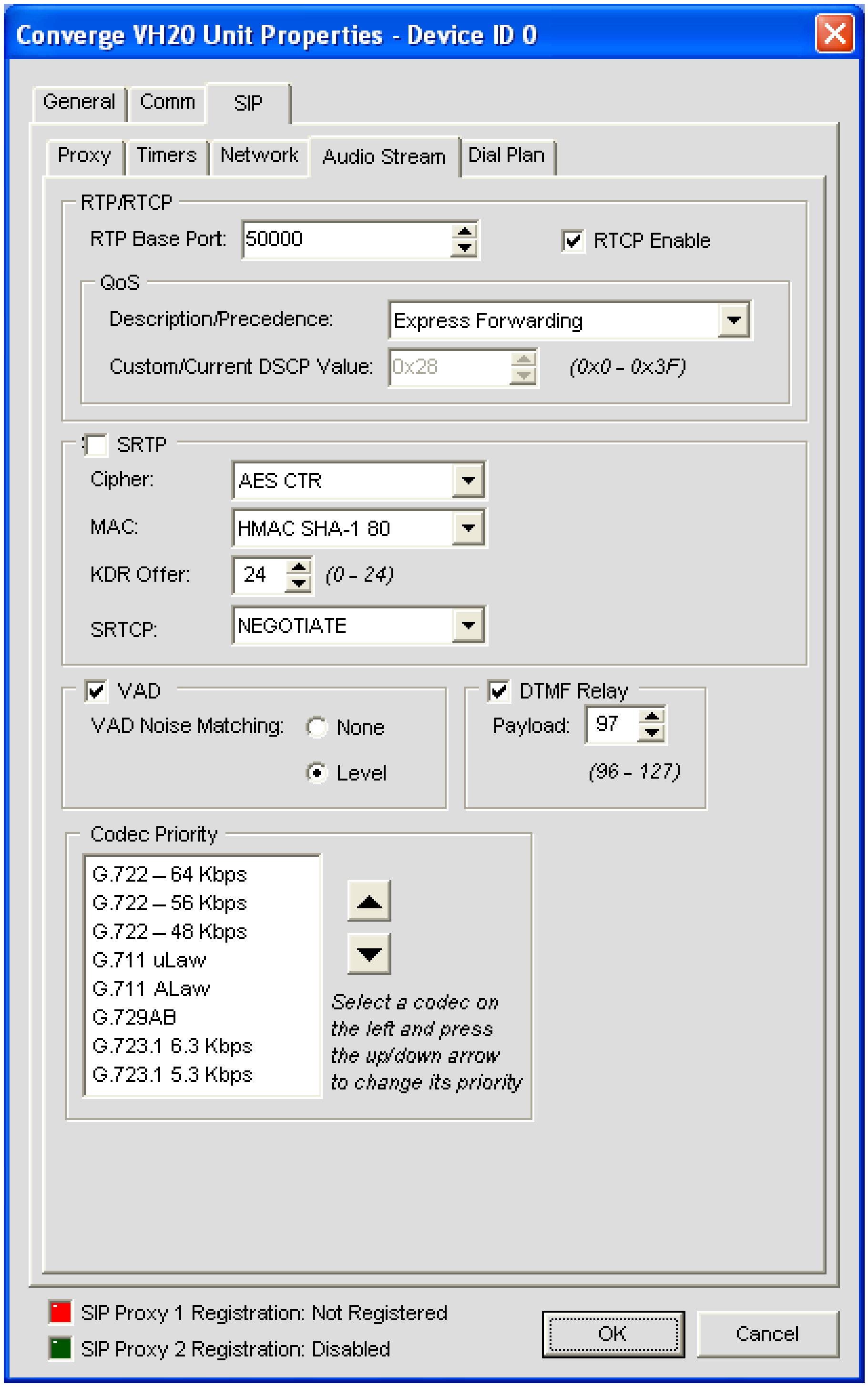
In the **Timers** tab, set the **Min SE Timer** to *180 sec*. This is required so that it matches the setting in the SIP trunk group in Communication Manager; otherwise, there will be a mismatch, and incoming calls to Converge Pro VH20 will with a status of Min-SE is too small. Alternatively, this timer can be changed in the Communication Manager SIP trunk group to match the setting on Converge Pro VH20.



In the Dial Plan tab, the Dial Plan Domain Name/IP Address was set to the Session Manager IP address and Extension Dialing was set to 5 since 5-digit extensions were being used. Any dialing prefixes can be entered here too.



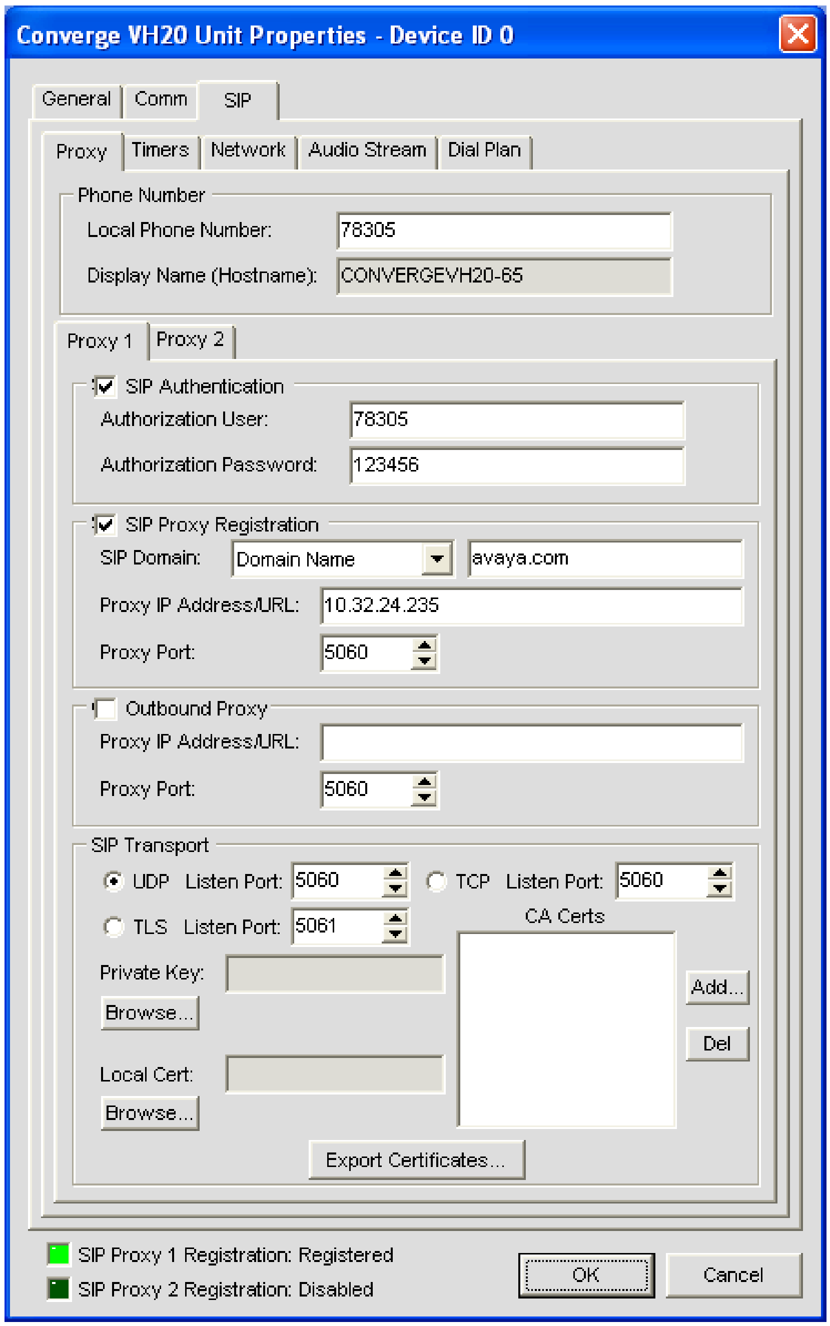
There is no configuration required in the **Audio Stream** tab; however, it is shown here to show the **Codec Priority** of Converge Pro VH20. Click **OK**.



8. Verification Steps

This section provides the steps that may be performed to verify proper configuration of the ClearOne ConvergePro VH20 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the ClearOne ConvergePro VH20 has successfully registered with Session Manager. Under **SIP Properties**, verify that the **SIP Proxy 1 Registration** status is *Registered* as shown at the bottom of the screen.



2. Place an incoming call to the ConvergePro VH20 and verify two-way audio.

3. Place an outgoing voice call from the ConvergePro VH20 to an Avaya IP telephone and verify that the call completes with two-way audio.

9. Conclusion

These Application Notes have described the administration steps required to integrate the ClearOne Converge Pro VH20 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. ClearOne Converge Pro VH20 successfully registered with Session Manager and voice calls were established successfully. All test cases passed with observations noted in **Section 2.2**.

10. References

This section references the Avaya and ClearOne documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, August 2010, Release 6.0, Issue 6.0, Document Number 03-300509.
[2] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.
[3] *ClearOne Converge Pro Installation & Operation Manual*, available at http://www.clearone.com.

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.