Avaya Solution & Interoperability Test Lab

Application Notes for Intuition Advanced Console to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Intuition Advanced Console 7.0 to interoperate with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1.

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 outline the steps necessary to configure Intuition Advanced Console from Enghouse Interactive to interoperate with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager). Intuition Advanced Console connects to the Communication Manager using a SIP trunk via the Session Manager.

Intuition Advanced Console (IAC) is a client/server-based application running on Windows Server operating systems. Intuition Advanced Console provides users with an attendant answering position for Communication Manager. The Intuition Advanced Console Attendant client provides a view of contacts, schedules, and communication tasks and was installed on the same server as the IAC Server but can be installed on a separate platform if required.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using Communication Manager. The Intuition Advanced Console communicates with the Communication Manager using a SIP trunk through the Session Manager. See **Figure 1** for a network diagram. A Dial plan was configured on the Communication Manager to route calls to Intuition Advanced Console. Calls placed to the Intuition Advanced Console automatically place a call to the telephone the attendant is using for answering purposes. When the attendant answers the call, the Intuition Advanced Console server bridges the two calls. When the attendant extends the call to another telephone, Intuition Advanced Console server performs a SIP Refer method, and the caller and the called user are now directly connected.

It is possible to have multiple attendant positions on a Communication Manager system. A variety of Avaya telephones were installed and configured on the Communication Manager.

Note: During compliance testing Avaya SIP and H.323 endpoints were used as the attendant's telephone.

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 Intuition Advanced Console did not include use of any specific encryption features as requested by Enghouse Interactive.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if Intuition Advanced Console could resume after a link failure with Communication Manager. The testing included:

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls from busy extensions and extensions that do not answer
- Call queuing and retrieval

2.2. Test Results

Tests were performed to ensure full interoperability between Intuition Advanced Console and Avaya Communication Manager. The tests were all functional in nature and performance testing was not included. All test cases passed successfully. The following observations were made during compliance testing;

- All testing was done using G.711MU or G.711A codecs
- The audio for the attendant console used Avaya SIP and H.323 Deskphones.
- The attendant console does not support conferencing. As such no conferencing interoperability tests were done

2.3. Support

For technical support for Enghouse Interactive products, please use the following web link.
https://mysupport.enghouse.com

Enghouse Interactive can also be contacted as follows.
Phone: +44 870 220 2205
E-mail: UKSupport@Enghouse.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, which has a SIP connection to the Intuition Advanced Console via the Session Manager. H.323 and SIP stations were used as the Intuition Advanced Console Attendant telephone during compliance testing. SIP and H.323 stations were configured on the Communication Manager to generate outbound/inbound calls to/from the PSTN. A PRI/T1 trunk on Media Gateway G450 was configured to connect to the simulated PSTN.

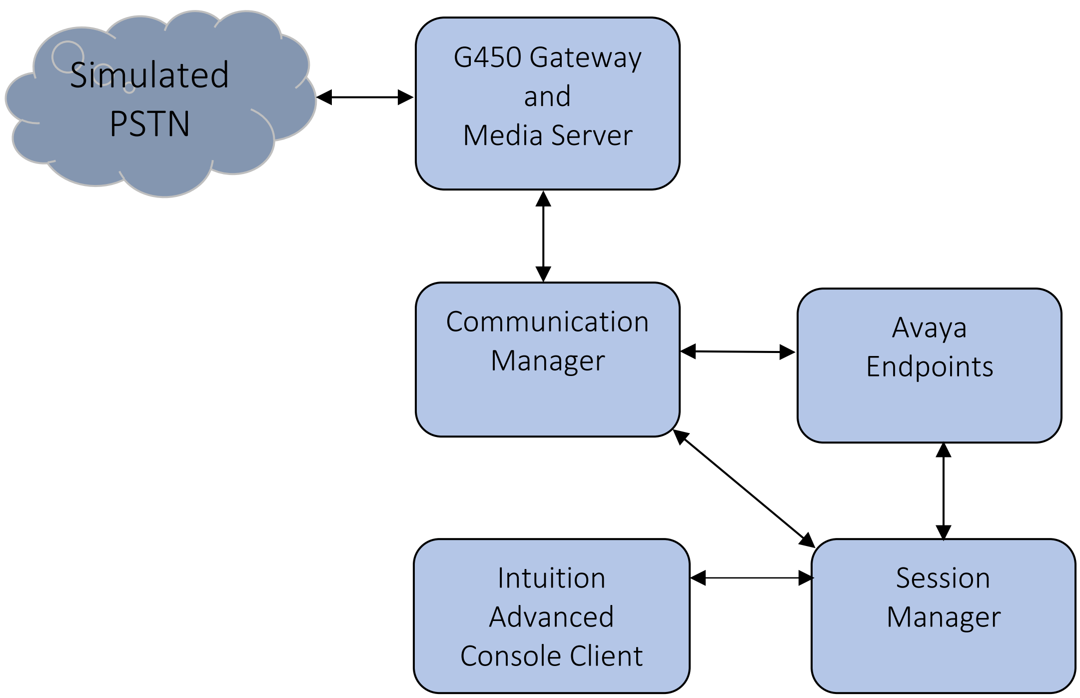


Figure 1: Avaya and Intuition Advanced Console Reference Configuration

4. Equipment and Software Validated

The following equipment and virtualized software versions were used in the reference configuration described above:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.0.1.1.890.25763
Avaya Aura® Session Manager	8.1.0.0.811021
Avaya Aura® System Manager	8.1.1.0.0310782
Avaya Aura® Media Server	8.0.0.21
Avaya G450 Media Gateway	41.24.0/2
Avaya IP Deskphones - J100 Series (SIP) - 96xx Series (H.323)	 4.0.6.0 6.8.3
Intuition Advanced Console Server and Client running on Microsoft Windows 2016 Server	7.0

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on the Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of the Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**.

It is implied a working system is already in place. The configuration operations described in this section can be summarized as follows: (Note: During Compliance Testing all inputs not highlighted in Bold were left as Default)

- Verify License
- Administer System Parameters Features
- Administer IP Node Names
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP Trunk Group Members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer AAR analysis

5.1. Verify License

Log in to the System Access Terminal to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2** and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 20
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 3
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 12000 58
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer System Parameter Features

It was suggested during compliance testing of Intuition Advanced Console to set the Station Call Transfer Recall Timer to 20 seconds. Use the “change system-parameters features” command to change the **Station Call Transfer Recall Timer** on **page 6**.

```
change system-parameters features                               Page   6 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
         Public Network Trunks on Conference Call: 5             Auto Start? n
    Conference Parties with Public Network Trunks: 6              Auto Hold? n
 Conference Parties without Public Network Trunks: 6         Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180        Bridging Tone? n
                 Short Interdigit Timer (seconds): 3        Conference Tone? n
              Unanswered DID Call Timer (seconds):           Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30    Mode Code Interface? n
                 Long Hold Recall Timer (seconds): 0
                      Reset Shift Timer (seconds): 0
     Station Call Transfer Recall Timer (seconds): 20       Recall from VDN? n
           Trunk Alerting Tone Interval (seconds): 15
                              DID Busy Treatment: tone
               Allow AAR/ARS Access from DID/DIOD? n
                 Allow ANI Restriction on AAR/ARS? n
Use Trunk COR for Outgoing Trunk Disconnect/Alert? n
                  7405ND Numeric Terminal Display? n                 7434ND? y



    DTMF Tone Feedback Signal to VRU - Connection:          Disconnection:
```

Enable **Create Universal Call ID (UCID)**, which is located on **Page 5**. For **UCID Network Node ID**, enter an available node ID.

```
change system-parameters features                               Page   5 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint:              Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                     Switch Name:
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT: station
                EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
      Delay Sending RELease (seconds): 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n
              Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? y    UCID Network Node ID: 1
```

5.3. Administer IP Node Names

Use the "change node-names ip" command (not shown) and add an entry for Session Manager. In this case, **sm81** and **10.64.110.212** are entered as **Name** and **IP Address**. Note the **procr** and **10.64.110.213** entry, which is the node **Name** and **IP Address** for the processor board. These values will be used later to configure the SIP trunk to Session Manager in **Section 5.5**.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
aes81               10.64.110.215
ams81               10.64.110.214
procr               10.64.110.213
sm81                10.64.110.212
```

5.4. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** "sip".
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie".

```
add trunk-group 1                                           Page   1 of   5
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: SM Trunk                  COR: 1        TN: 1        TAC: 101
   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: 1
                                                   Number of Members: 10
```

Navigate to **Page 3** and enter "private" for **Numbering Format**.

```
add trunk-group 1                                           Page   3 of   5
TRUNK FEATURES
          ACA Assignment? n           Measured: both
                                                         Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                              UUI Treatment: shared
                                            Maximum Size of UUI Contents: 128
                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n

                                                Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? y



 Show ANSWERED BY on Display? y
```

5.5. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** "sip".
- **Transport Method:** "tls".
- **Near-end Node Name:** An existing C-LAN node name or "procr" from **Section 5.3**.
- **Far-end Node Name:** The existing node name for Session Manager from **Section 5.3**.
- **Near-end Listen Port:** An available port for integration with Session Manager.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Session Manager.
- **Direct IP-IP Audio Connections?:** "y".

```
add signaling-group 1                                           Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM                        Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
 Near-end Node Name: procr                    Far-end Node Name: sm81
Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain:
                                              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? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.4**. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.5**.
- **Number of Members:** The desired number of members, in this case “10”.

```
change trunk-group 1                                            Page   1 of   5
                                TRUNK GROUP

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: SM Trunk                    COR: 1        TN: 1        TAC: 101
   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: 1
                                                     Number of Members: 10
```

5.7. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.5**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “yes” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Intuition Advanced Console.

```
change ip-network-region 1                                      Page   1 of  20
                              IP NETWORK REGION
  Region: 1
Location:           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
```

5.8. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.7**. Update the audio codec types in the **Audio Codec** fields as necessary. Configure the codec as shown below.

```
display 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:
Media Encryption                   Encrypted SRTCP: enforce-unenc-srtcp
 1: none
 2:
 3:
 4:
 5:
```

5.9. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Intuition Advanced Console, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.4**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name: Main
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No. Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 1    0                    0                                         n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  none
```

5.10. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Intuition Advanced Console. Add an entry for the trunk group defined in **Section 5.4**. In the example shown below, all calls originating from a 5-digit extension beginning with "7" will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext                 Trk        Private          Total
Len Code                Grp(s)     Prefix           Len
 5  5                                               5  Total Administered: 2
 5  7                                               5     Maximum Entries: 540
```

5.11. Administer AAR Analysis

Use the “change aar analysis 0” command and add an entry to specify how to route calls to 51xxx. In the example shown below, calls with digits 51 will be routed as an AAR call using route pattern “1” from **Section 5.9**.

```
change aar analysis 51                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 0

          Dialed            Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    51                    5    5     1         aar          n
```

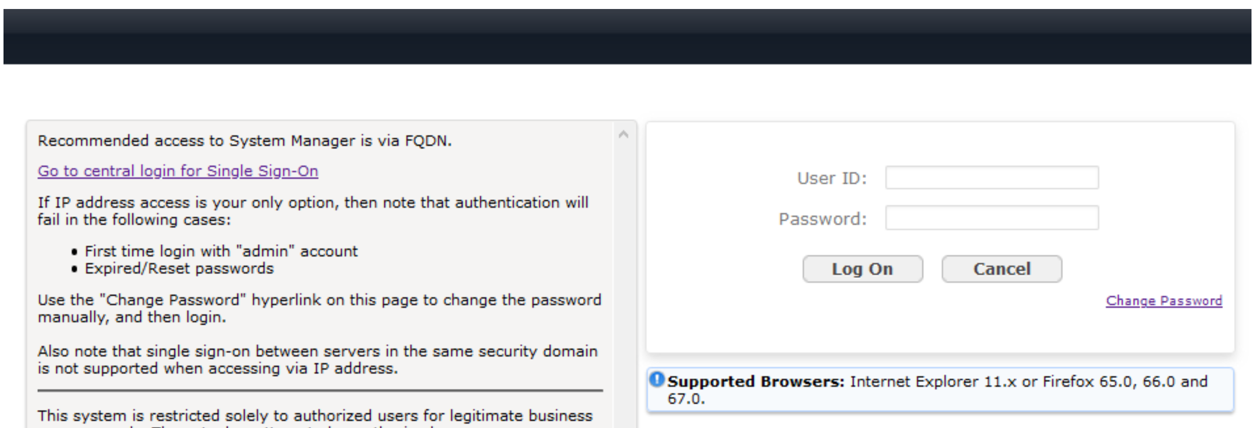
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer Domain
- Administer locations
- Administer Adaptation
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

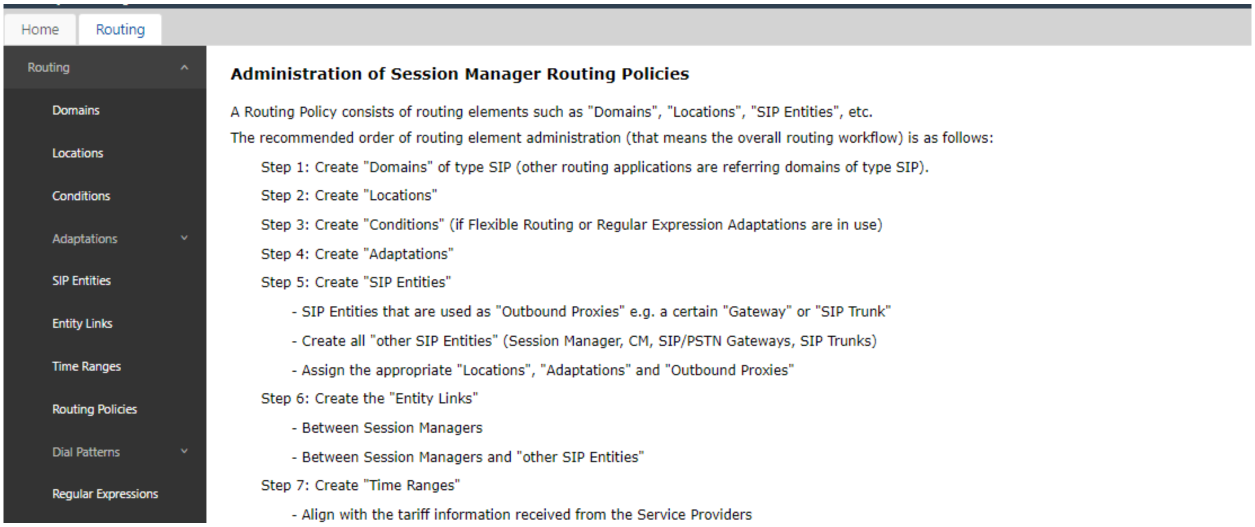
6.1. Launch System Manager

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



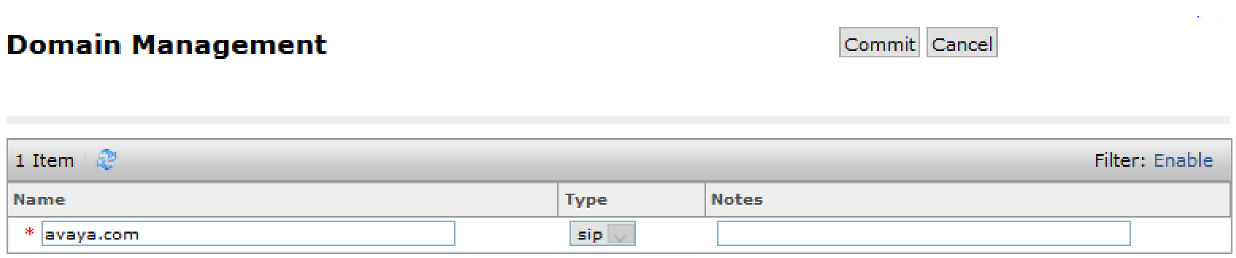
6.2. Administer Domain

In the subsequent screen (not shown), select **Elements → Routing** to display the **Administration of Session Manager Routing Policies** screen below.



Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen to add a new domain. The **Domain Management** screen is displayed.

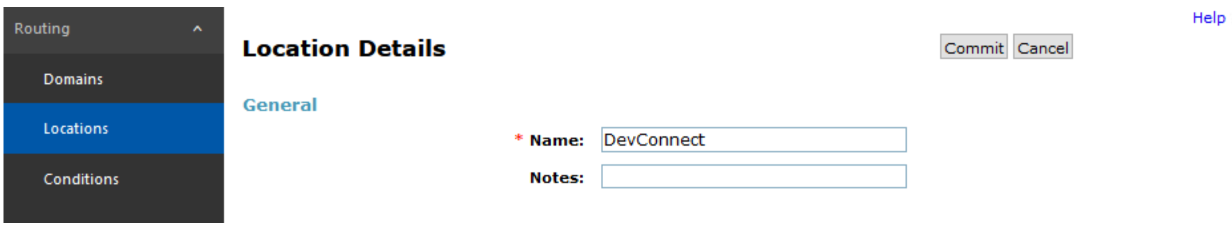
In the **Name** field, enter the domain name. Select "sip" from the **Type** drop down menu and provide any optional **Notes**.



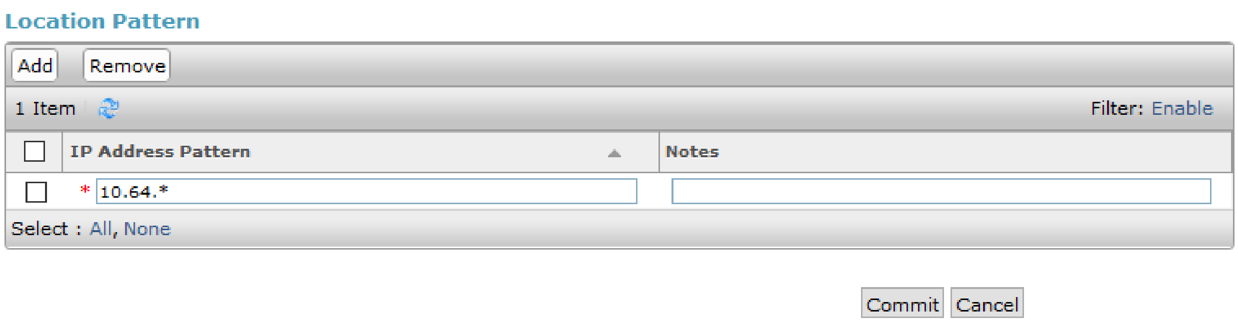
6.3. Administer Locations

Select **Routing → Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a new location for Intuition Advanced Console.

The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



6.4. Administer Adaptation

During compliance test, to make the call from and to Communication Manager via Session Manager, administer an adaptation to translate IP address into domain name for Intuition Advanced Console SIP entity. Below are the steps that were used during compliance testing to create the needed Adaptation. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown).

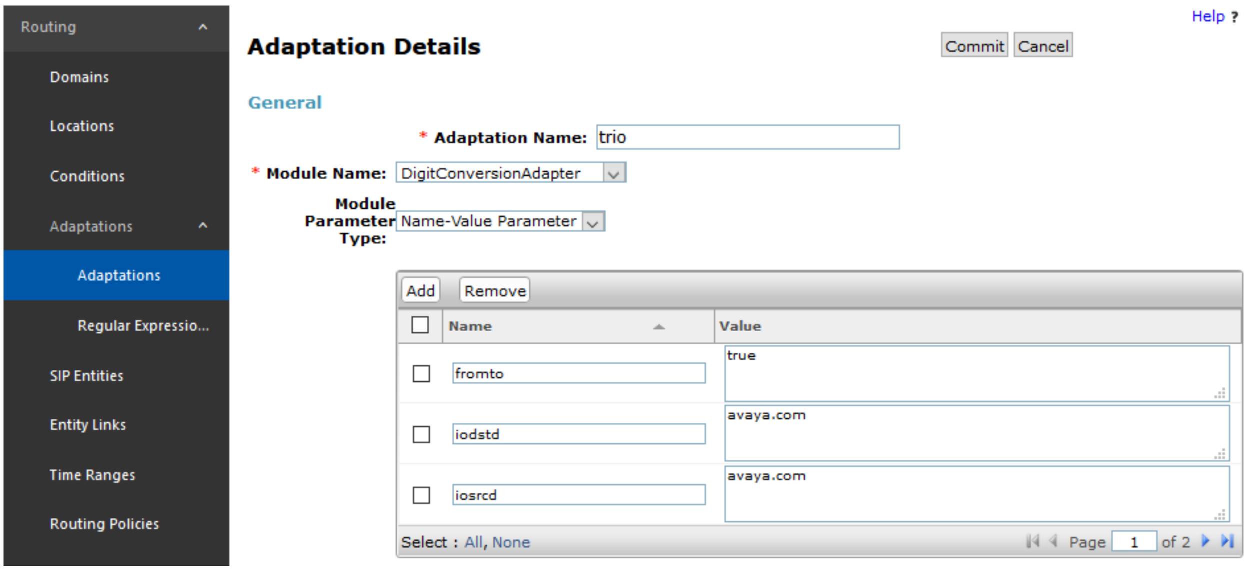
Enter the following for the Intuition Advanced Console Adaptation.

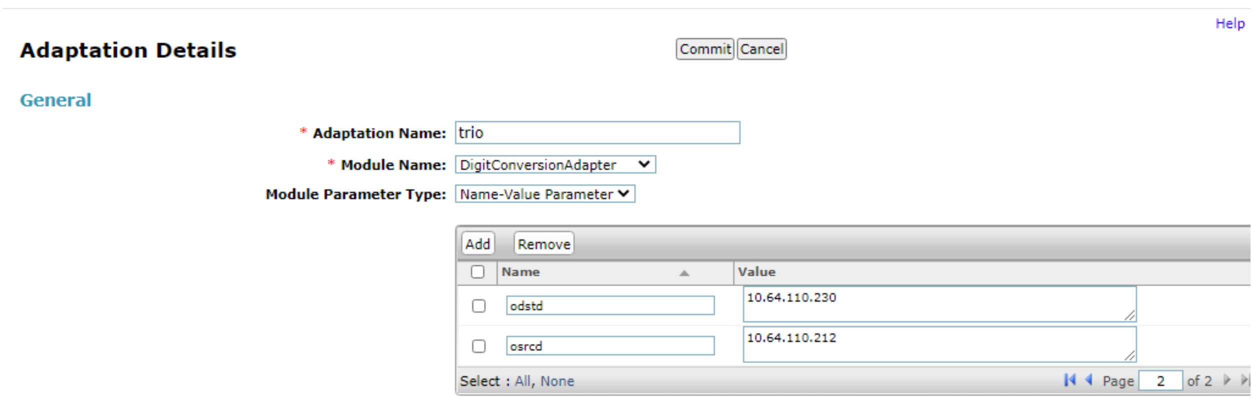
- **Adaptation Name:** An informative name (e.g., change IP to Domain of Intuition Advanced Console).
- **Module Name:** Select "DigitConversionAdapter".
- **Module Parameter Type:** Select "Name-Value Parameter".

Click **Add** to add a new row for the following values as shown below table:

Name	**Value**
fromto	true
iodstd	Enter the domain name of system, e.g.: **avaya.com**
iosrcd	Enter the domain name of system, e.g.: **avaya.com**
odstd	Enter IP address of Intuition Advanced Console SIP Server, e.g.: **10.64.110.230**
osrcd	Enter IP address of Session Manager Server, e.g.: **10.64.110.212**

Once the correct information is entered, click the **Commit** button. Below are screenshots showing the Adaptation created for Intuition Advanced Console. Select next to see the 2nd page to view the rest of the adaptation.





6.5. Administer SIP Entities

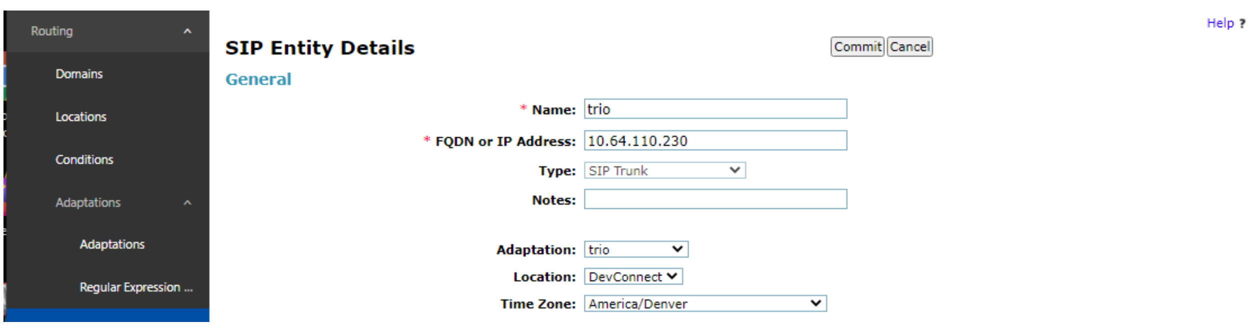
Add two new SIP entities, one for Intuition Advanced Console and one for the new SIP trunks with Communication Manager.

6.5.1. SIP Entity for Intuition Advanced Console

Select **Routing → SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Intuition Advanced Console.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

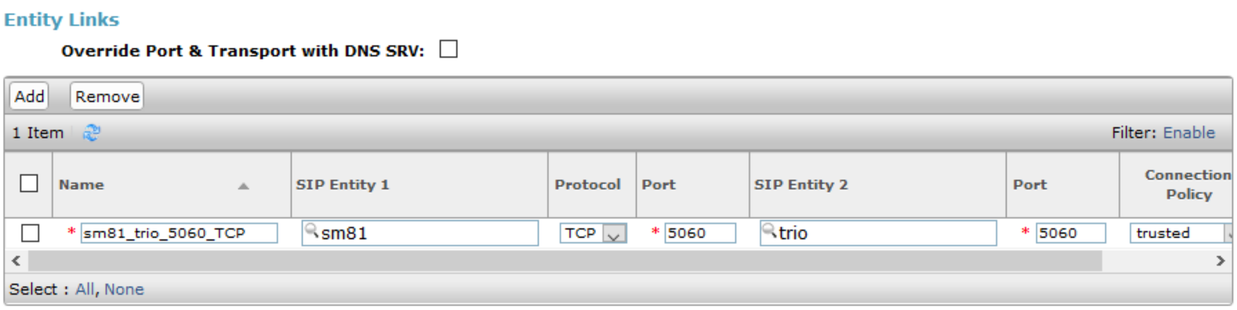
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of Intuition Advanced Console SIP Server.
- **Type:** "SIP Trunk"
- **Adaptation:** Select the adaptation configured in **Section 6.4**
- **Location:** Select the Intuition Advanced Console location name from **Section 6.3**
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name.
- **Protocol:** "TCP".
- **Port:** "5060".
- **SIP Entity 2:** The Intuition Advanced Console entity name from this section.
- **Port:** "5060".
- **Connection Policy:** "trusted".

Note that only TCP protocol was tested.

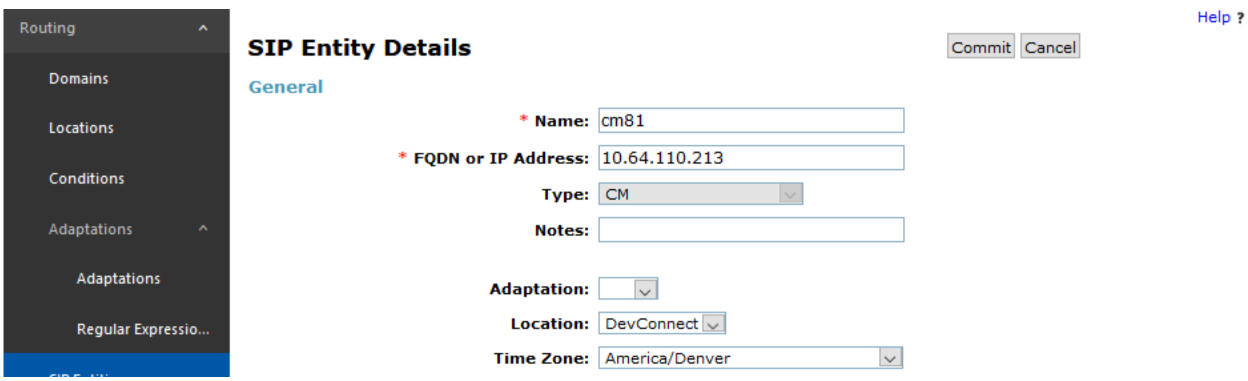


6.5.2. SIP Entity for Communication Manager

Select **Routing → SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with Intuition Advanced Console.

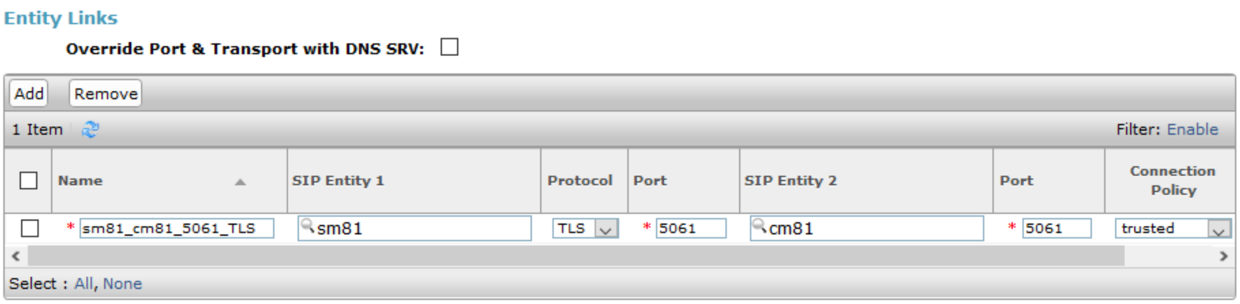
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name.
- **Protocol:** The signaling group transport (TLS) method from **Section 5.5**.
- **Port:** The signaling group listen port (5061) number from **Section 5.5**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port (5061) number from **Section 5.5**.
- **Connection Policy:** “trusted”



6.6. Administer Routing Policies

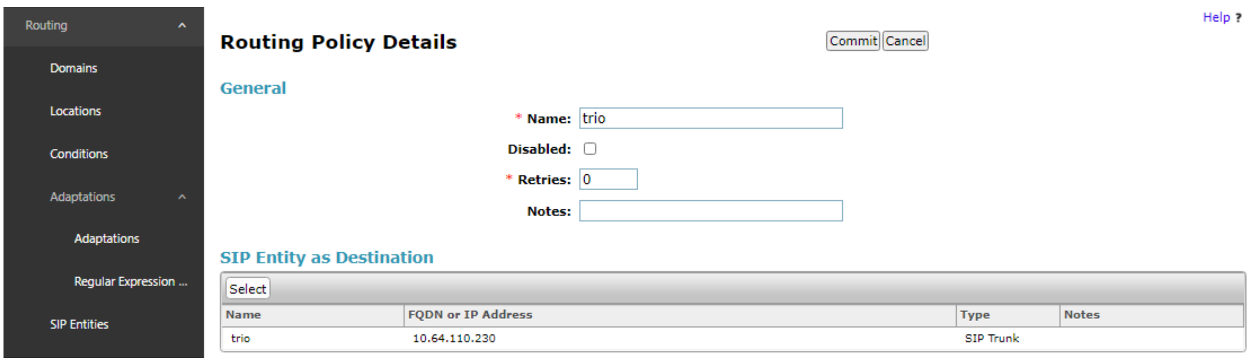
Add two new routing policies, one for Intuition Advanced Console and one for the new SIP trunks with Communication Manager.

6.6.1. Routing Policy for Intuition Advanced Console

Select **Routing → Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Intuition Advanced Console.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Intuition Advanced Console entity name from **Section 6.5.1**. The screen below shows the result of the selection.

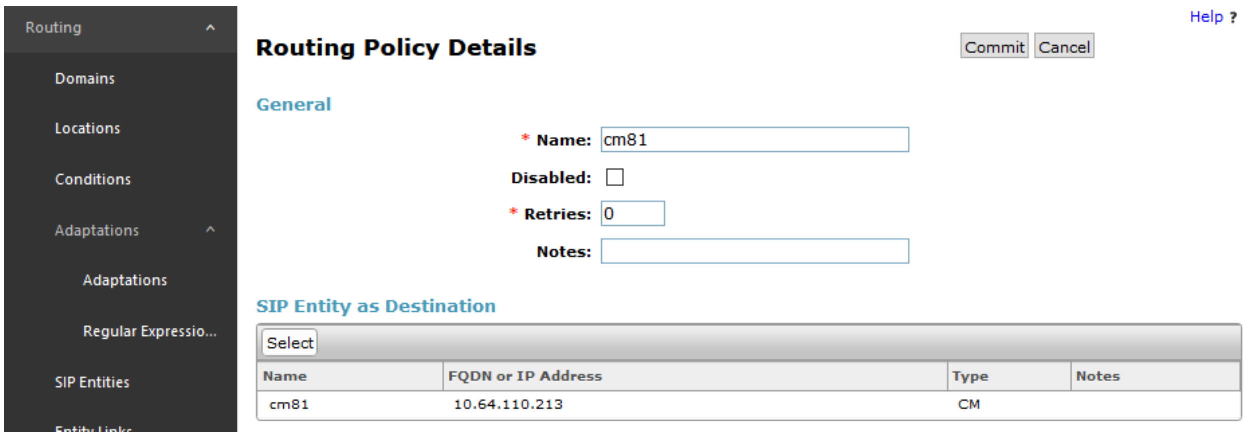


6.6.2. Routing Policy for Communication Manager

Select **Routing → Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.5.2**. The screen below shows the result of the selection.



6.7. Administer Dial Patterns

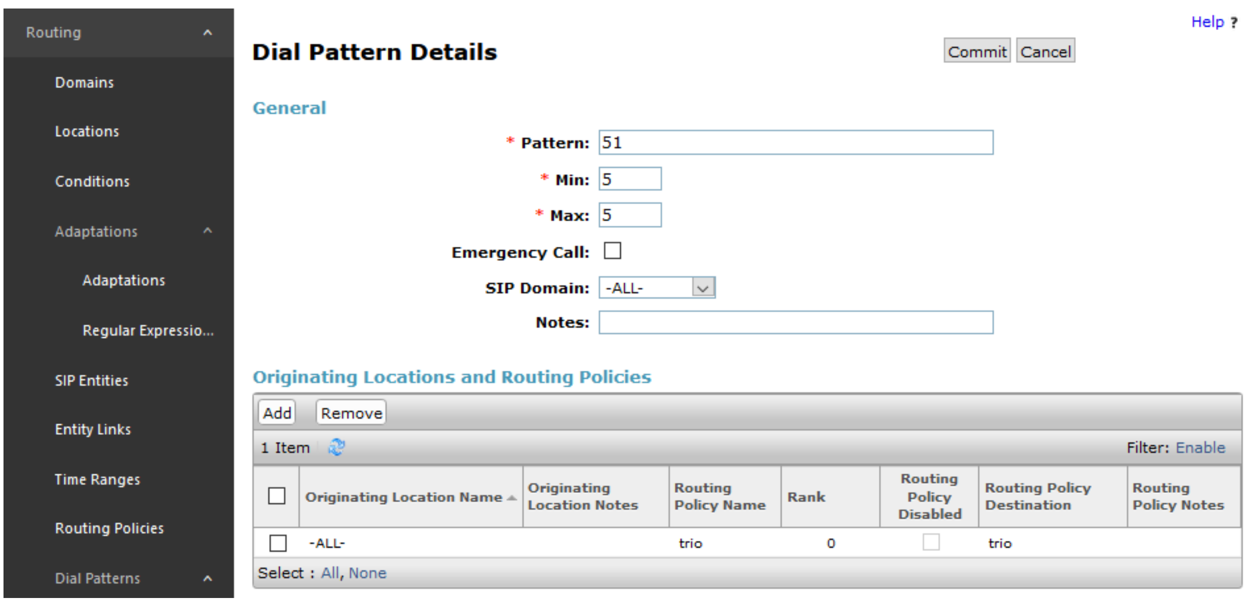
Add a new dial pattern for Intuition Advanced Console and Communication Manager.

6.7.1. Dial Pattern for Intuition Advanced Console

Select **Routing → Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Intuition Advanced Console. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "51".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select "ALL"

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Intuition Advanced Console. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in all locations. The Intuition Advanced Console routing policy from **Section 6.6.1** was selected as shown below.

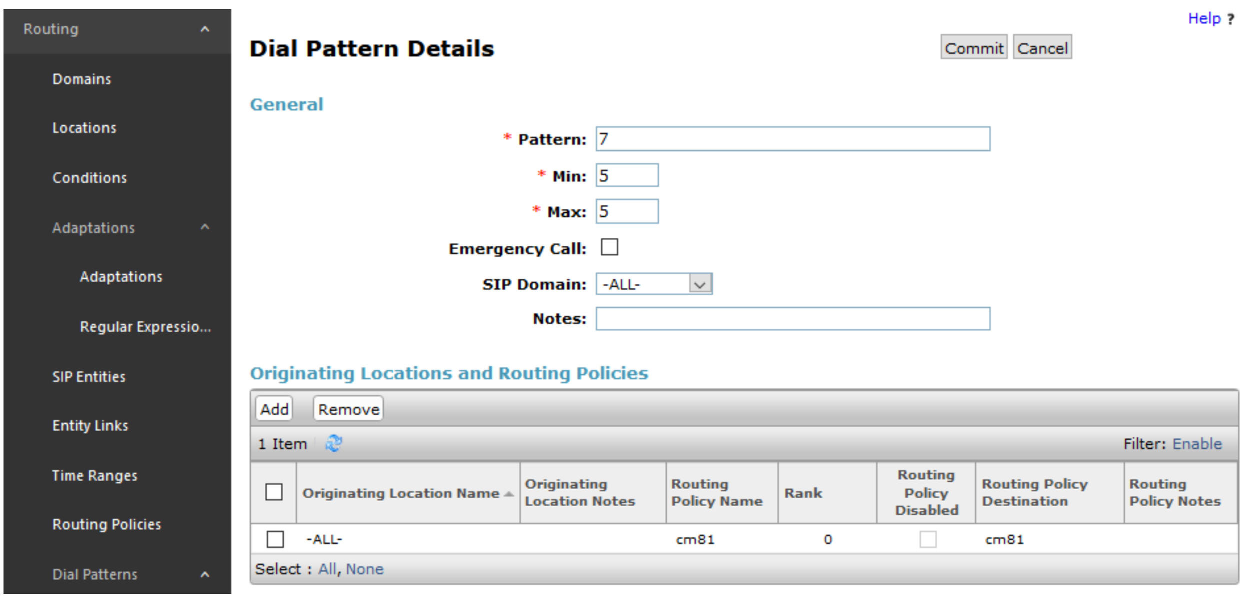


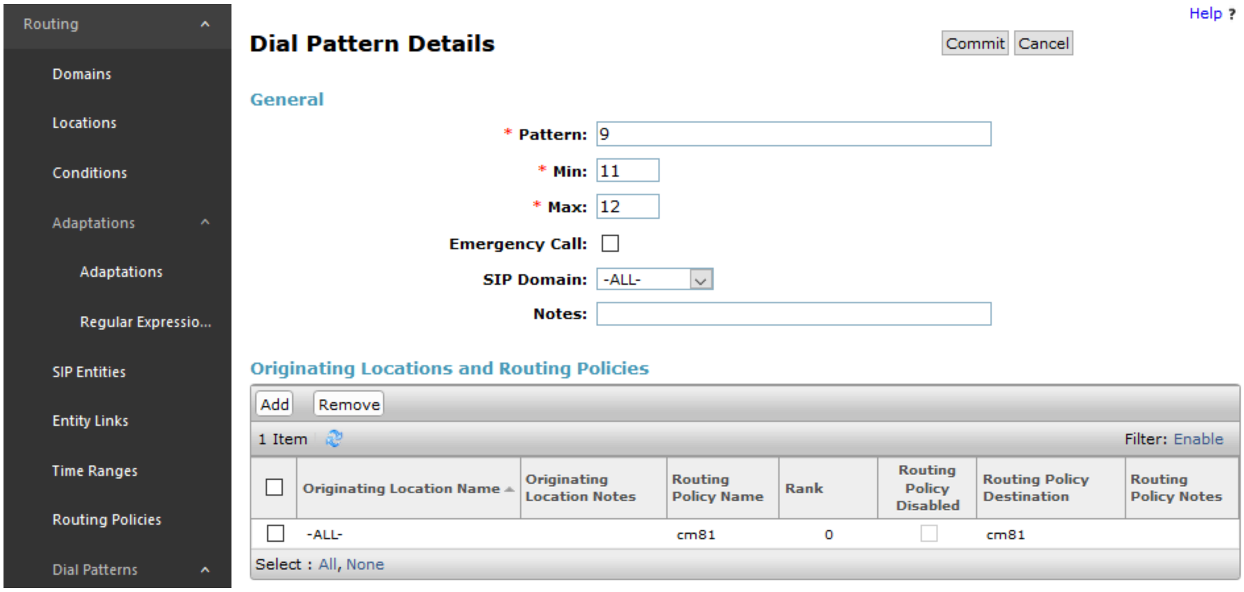
6.7.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Communication Manager. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "7" and "9".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select "ALL"

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for call originations from all locations. The Communication Manager routing policy from **Section 6.6.2** was selected as shown below.



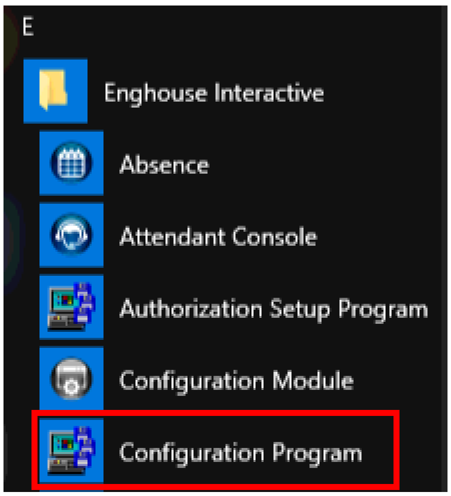

Routing
Domains
Locations
Conditions
Adaptations
Adaptations
Regular Expressio...
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Help ?
Dial Pattern Details
Commit
Cancel
General
* Pattern: 9
* Min: 11
* Max: 12
Emergency Call:
SIP Domain: -ALL-
Notes:
Originating Locations and Routing Policies
Add
Remove
1 Item
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
-ALL-
cm81
0
cm81
Select : All, None

7. Configure Intuition Advanced Console

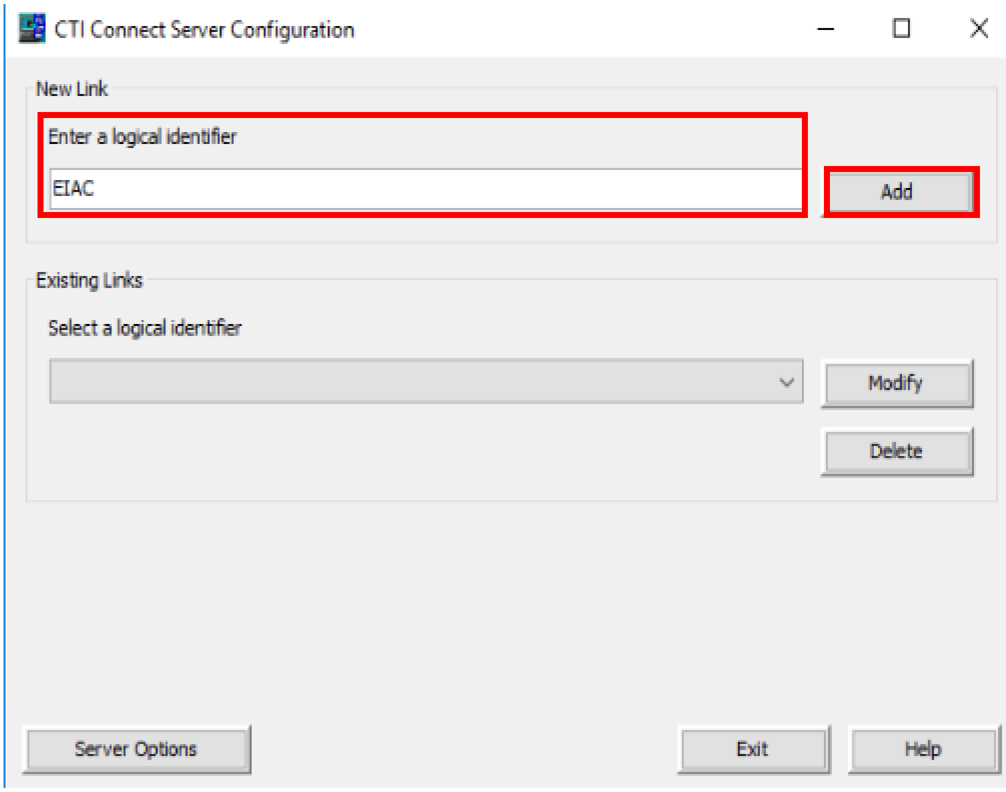
This section shows how to configure Intuition Advanced Console to successfully connect to Session Manager. The installation of the IAC software is assumed to be completed and the IAC services are up and running. The steps to configure SIP Trunks are as follows:

- Configure IAC to use SIP Trunks
- Configure Absence
- Configure Intuition Advanced Console Attendant

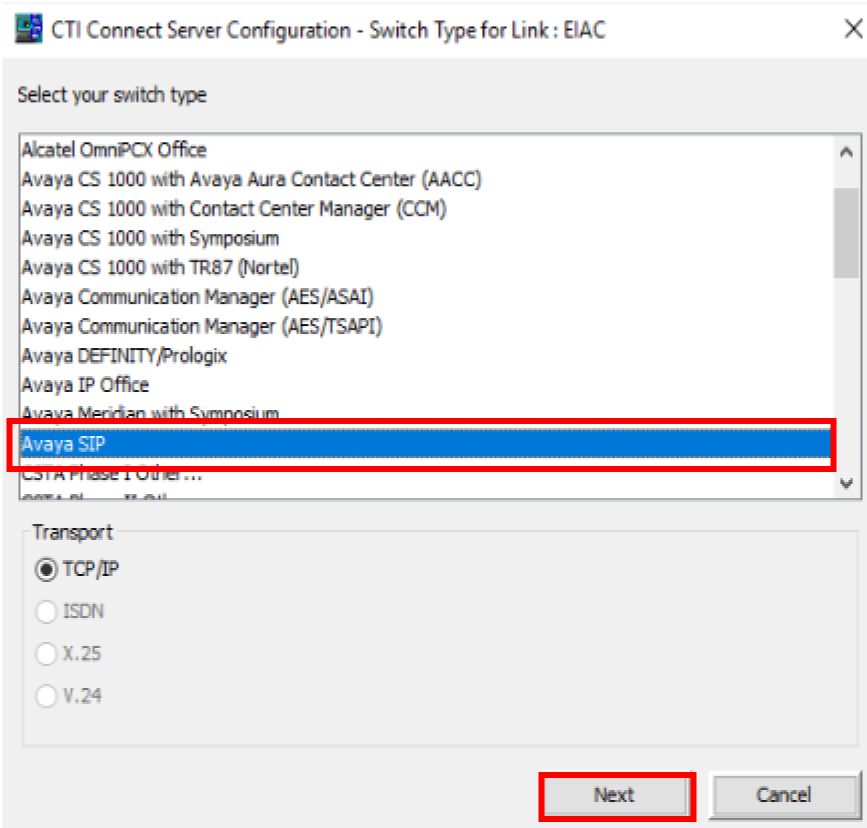
7.1. Intuition Advanced Console to use SIP Trunks



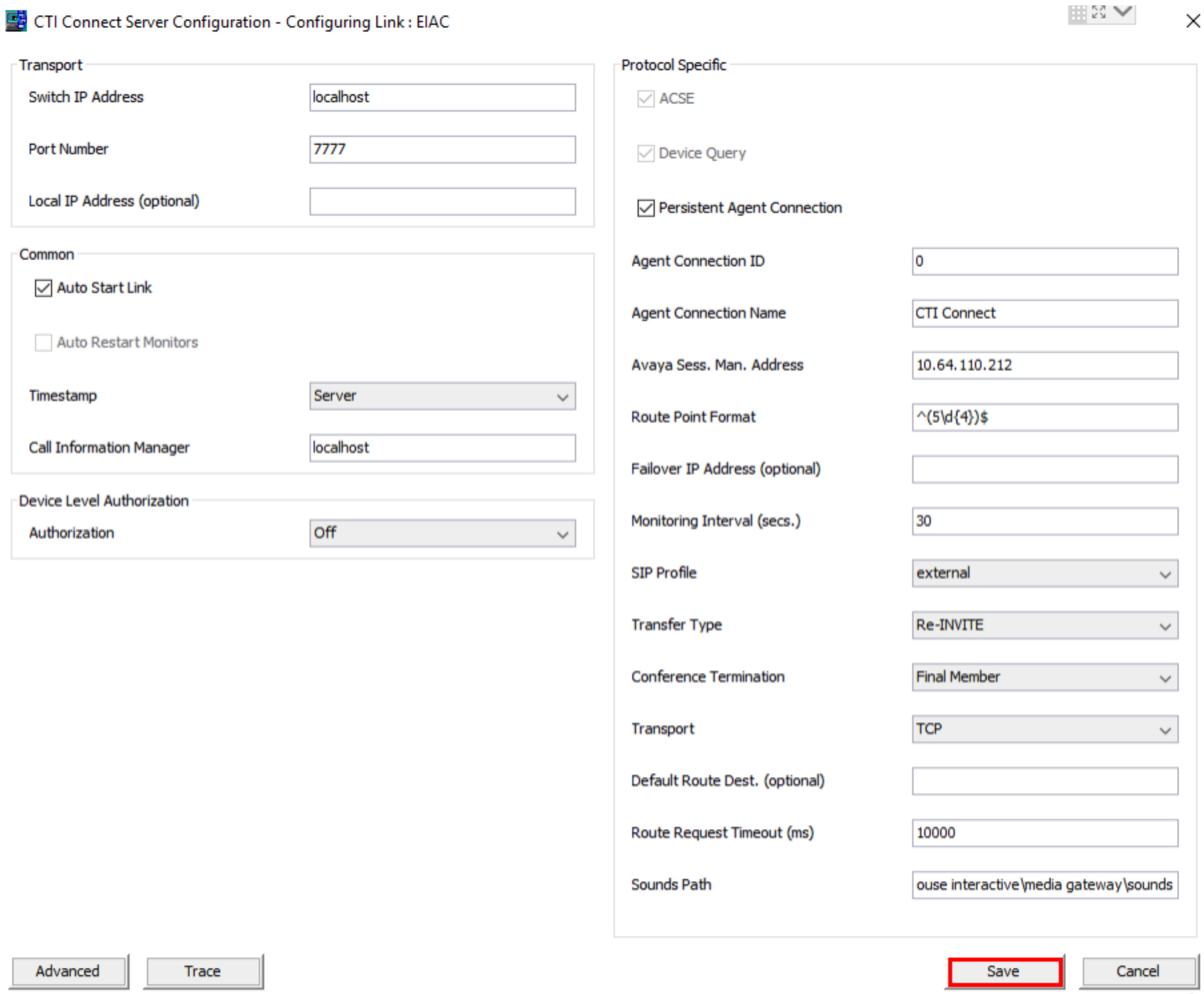
- Launch the "Configuration Program"



- Enter “EIAC” as logical identifier and click **Add**



- Select **Avaya SIP**



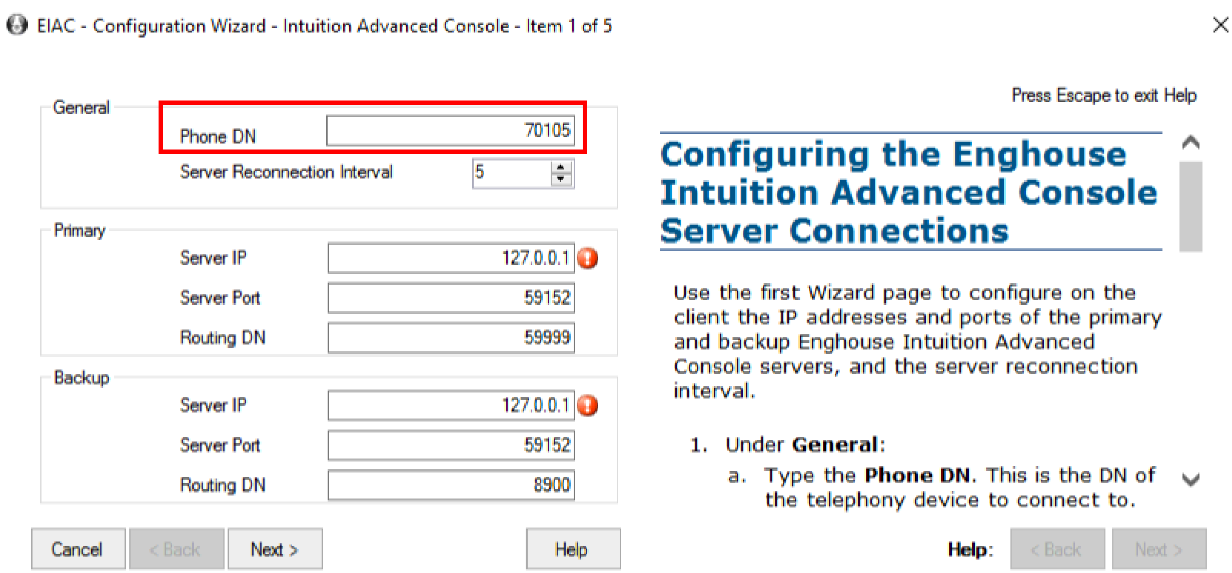
- Enter parameters according to table below, leave all other configuration as default, click save when done.

Port Number	7777
Avaya Sess. Man. Address	Session Manager IP

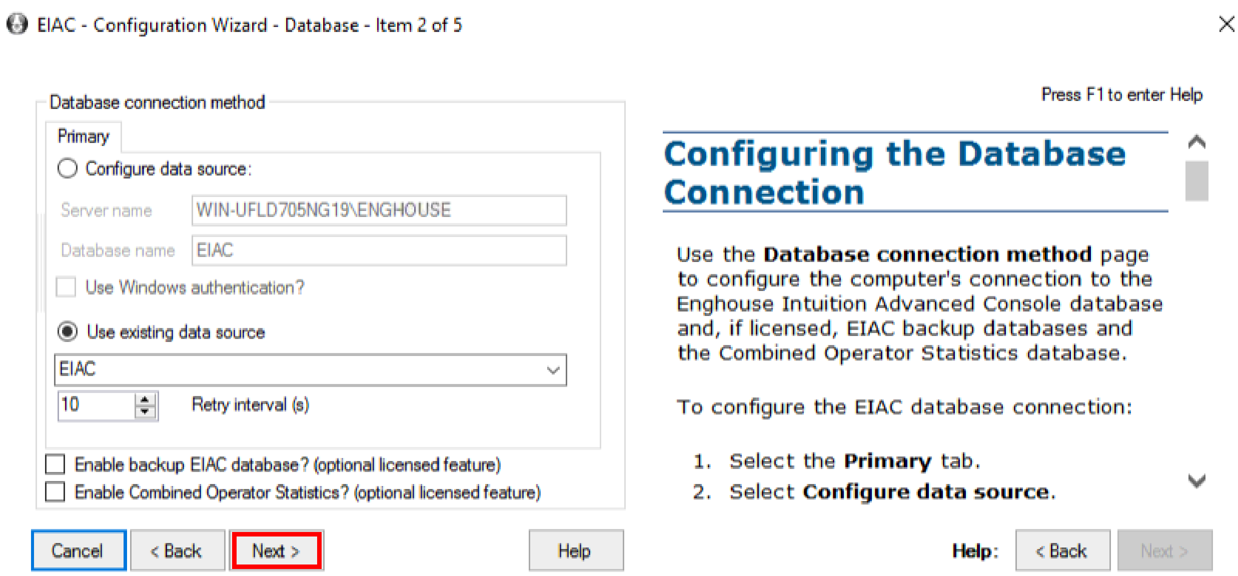
7.2. Configure Attendant Console

The attendant uses a regular Communication Manager telephone to make and receive calls, which are directed to the telephone by the IAC server.

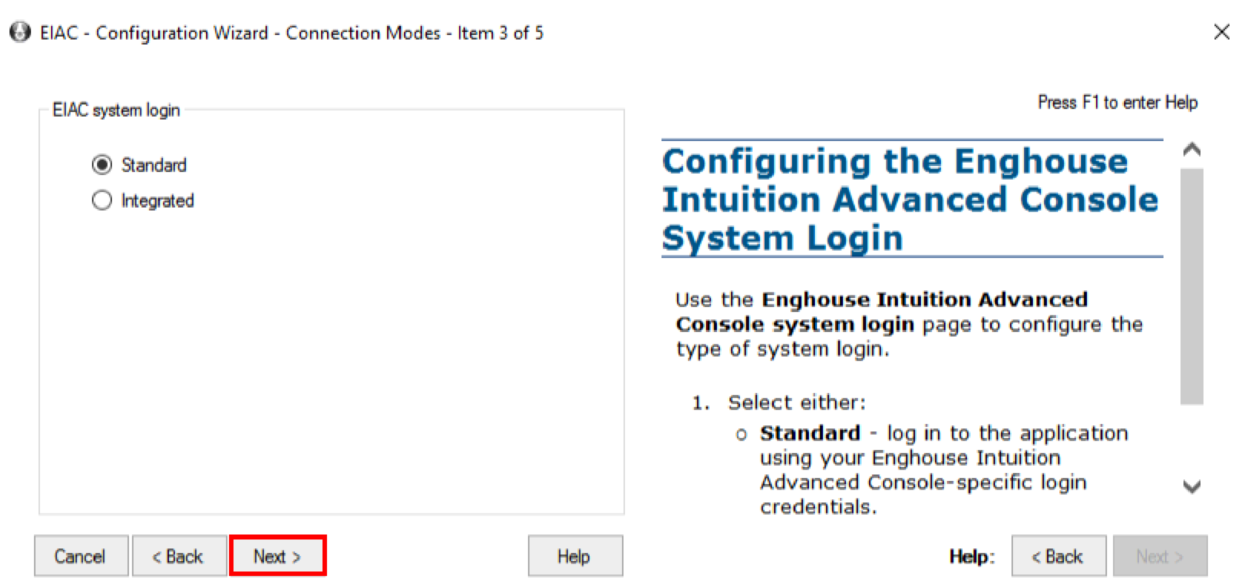
- Launch Configuration Wizard



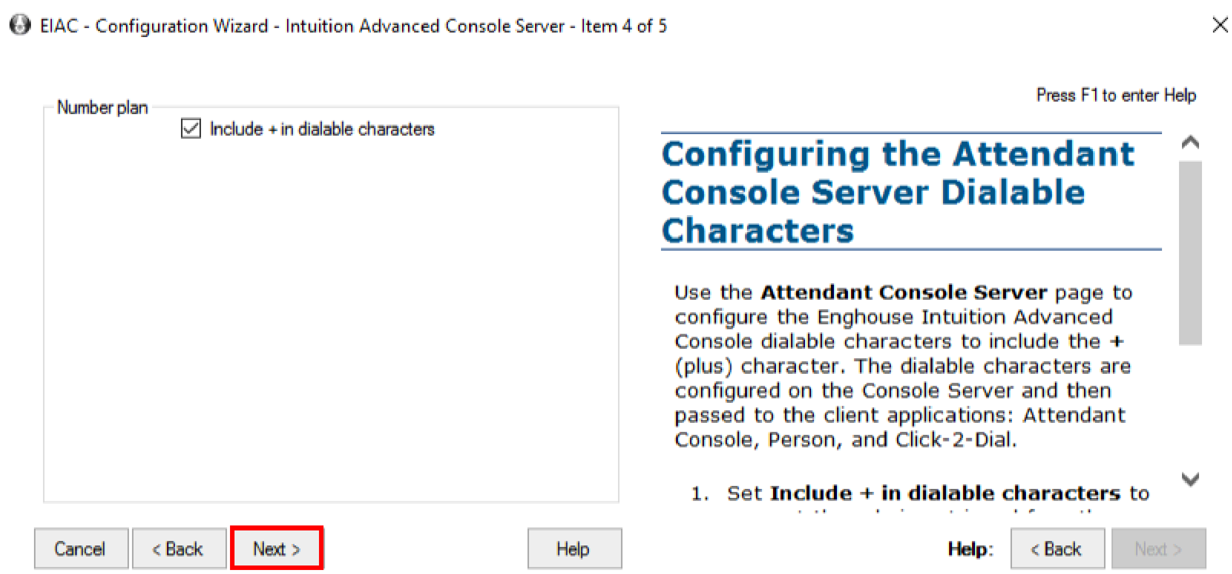
- Enter the attendant **Phone DN** and click **Next**



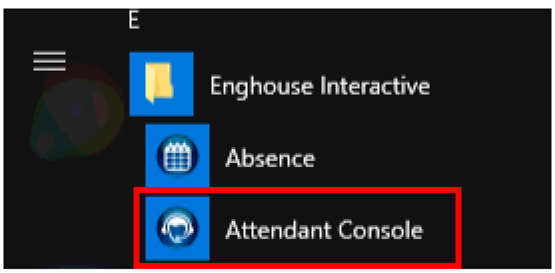
- Leave default and click **Next**



- Leave default and click **Next**



- Leave default and click **Next**



- Launch the attendant console

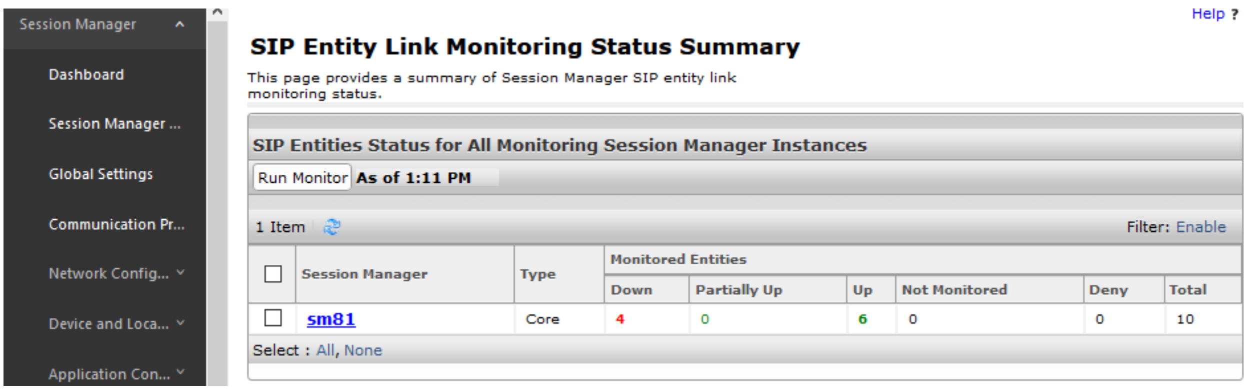
- Enter your credentials and click **OK**

8. Verification Steps

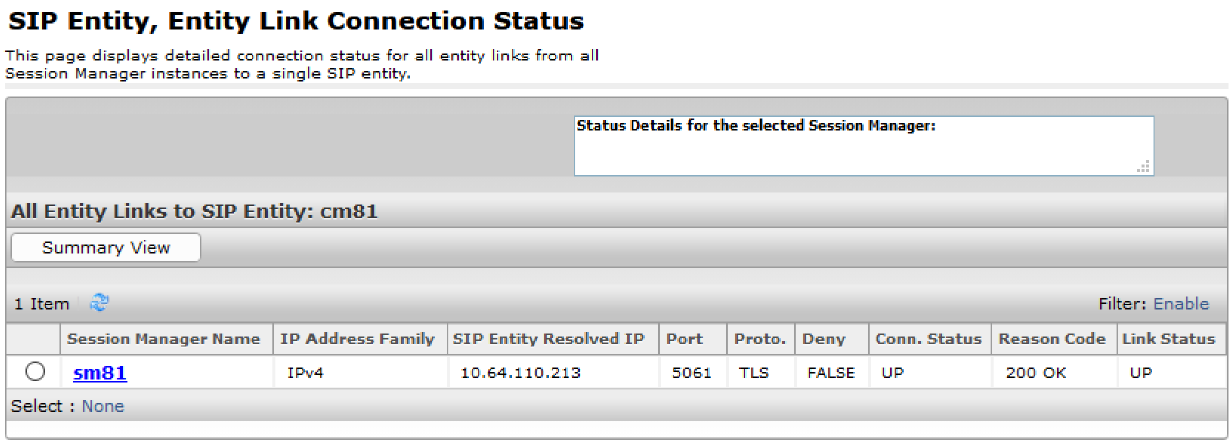
This section provides the tests that can be performed to verify correct configuration of the Avaya and Intuition Advanced Console solution.

8.1. Verify Session Manager

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager → System Status → SIP Entity Monitoring**.



Verify that the state of the Session Manager links to Communication Manager and Intuition Advanced Console by selecting the SIP Entity names.



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

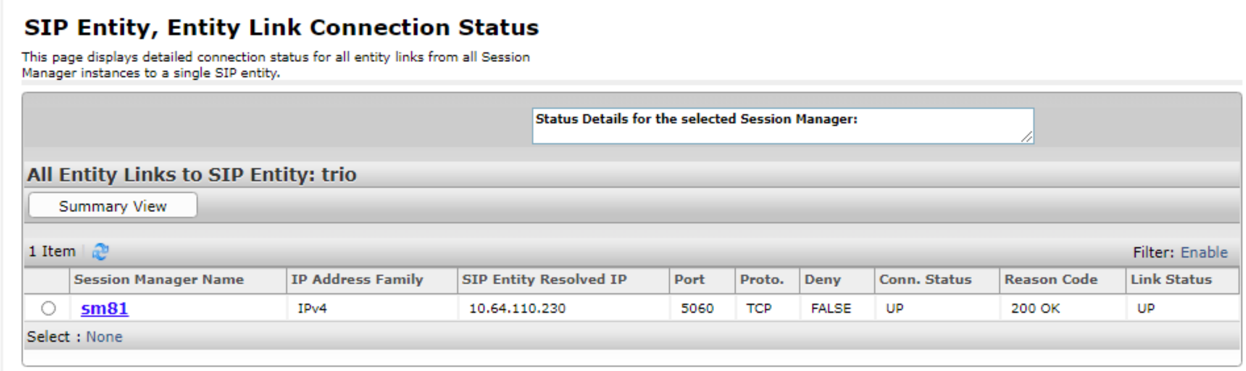
All Entity Links to SIP Entity: cm81

Summary View

1 Item Filter: Enable

	Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	sm81	IPv4	10.64.110.213	5061	TLS	FALSE	UP	200 OK	UP

Select : None



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

All Entity Links to SIP Entity: trio

Summary View

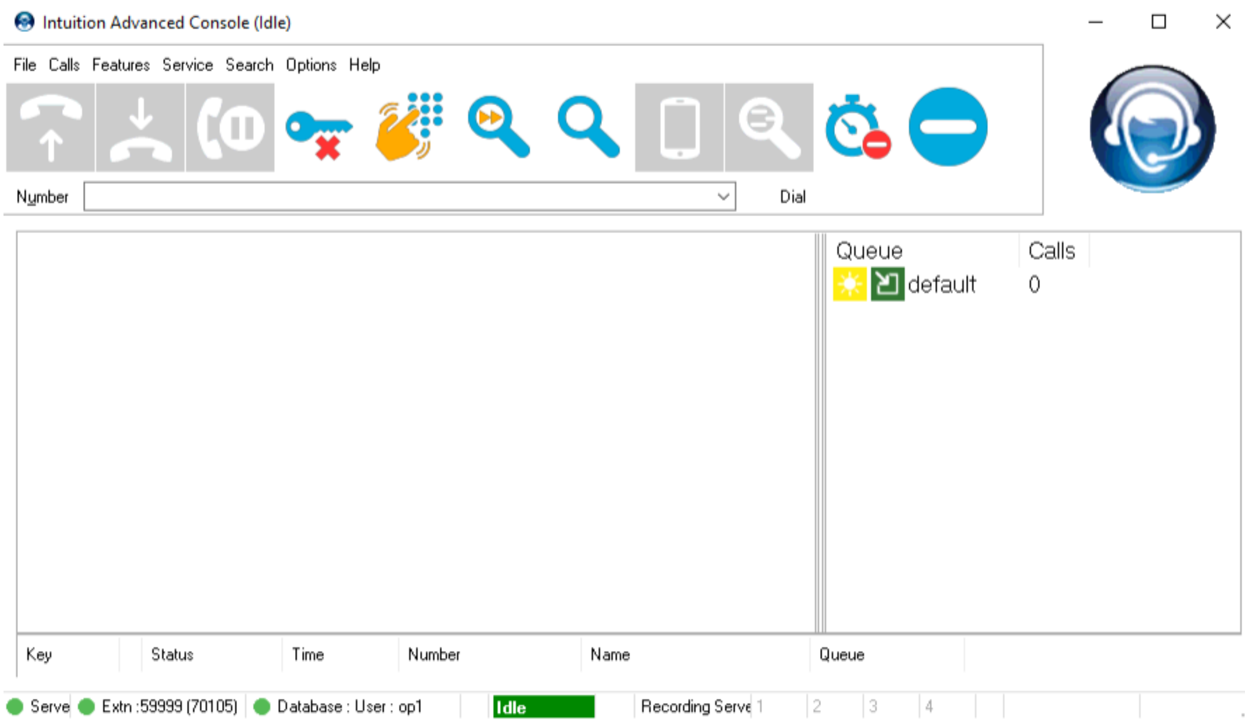
1 Item Filter: Enable

	Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	sm81	IPv4	10.64.110.230	5060	TCP	FALSE	UP	200 OK	UP

Select : None

8.2. Verify Attendant Console

To verify that IAC is connected to Communication Manager via Session Manager, log in to the attendant console.



9. Conclusion

These Application Notes describe the procedures required to configure Intuition Advanced Console from Enghouse Interactive to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks.

All feature functionality test cases described in **Section 2.1** were passed with the observations pointed out in **Section 2.2**.

10. Additional References

This section references the product documentation relevant to these Application Notes.

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020
[2] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 6, August 2020

Product Documentation for Enghouse Interactive can be obtained in the installed software or at: http://enghouseinteractive.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.