



## **Avaya Solution & Interoperability Test Lab**

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# **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>

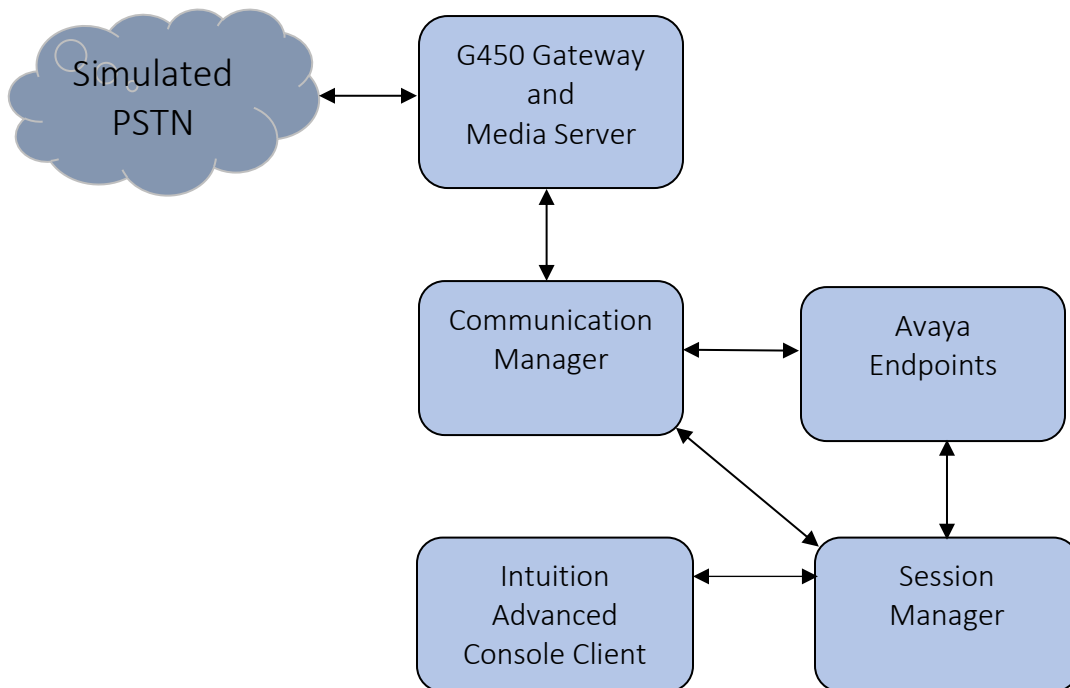
Enghouse Interactive can also be contacted as follows.

Phone: +44 870 220 2205

E-mail: [UKSupport@Enghouse.com](mailto: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:

<b>Equipment/Software</b>	<b>Release/Version</b>
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
<b>procr</b>	<b>10.64.110.213</b>
<b>sm81</b>	<b>10.64.110.212</b>

## 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
  Send UCID? y                               Modify Tandem Calling Number: no

  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:
3:
4:
5:
6:

  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 7
5 Total Administered: 2
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
```

AAR DIGIT ANALYSIS TABLE							Page	1	of	2
Location: all							Percent Full: 0			
Dialed	Total	Route	Call	Node	ANI					
String	Min	Max	Pattern	Type	Num	Reqd				
<b>51</b>	<b>5</b>	<b>5</b>	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.

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

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

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

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

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

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access may be reported to the appropriate law enforcement agency.

User ID:

Password:

[Change Password](#)

**Supported Browsers:** Internet Explorer 11.x or Firefox 65.0, 66.0 and 67.0.



## 6.2. Administer Domain

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

**Administration of Session Manager Routing Policies**

A Routing Policy consists of routing elements such as "Domains", "Locations", "SIP Entities", etc.

The recommended order of routing element administration (that means the overall routing workflow) is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Conditions" (if Flexible Routing or Regular Expression Adaptations are in use)
- Step 4: Create "Adaptations"
- Step 5: Create "SIP Entities"
  - SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
  - Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
  - Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
- Step 6: Create the "Entity Links"
  - Between Session Managers
  - Between Session Managers and "other SIP Entities"
- Step 7: Create "Time Ranges"
  - Align with the tariff information received from the Service Providers

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.

**Domain Management**

1 Item Filter: Enable

<input type="checkbox"/>	Name	Type	Notes
<input type="checkbox"/>	avaya.com	sip	

Select : All, None

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

### Domain Management

1 Item Filter: Enable

Name	Type	Notes
* avaya.com	sip	

### 6.3. Administer Locations

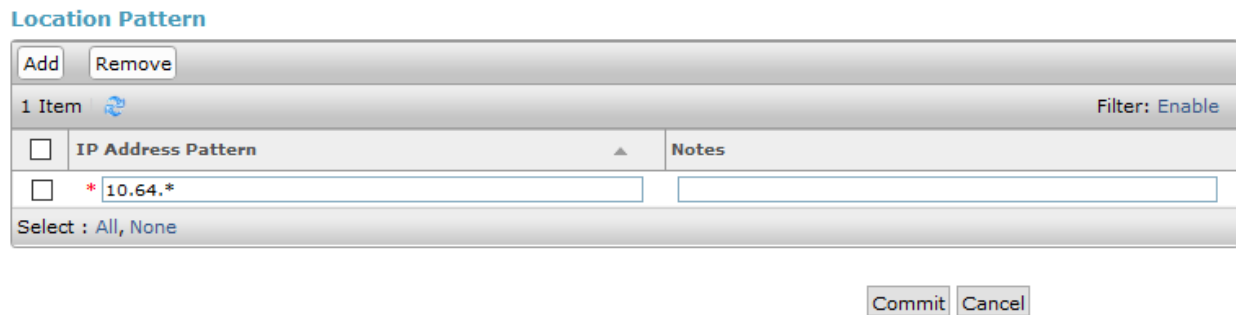
Select **Routing** → **Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a new location for Intuition Advanced Console.

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.



The screenshot shows the 'Location Details' screen with the 'General' sub-section selected. On the left is a navigation pane with 'Locations' highlighted. The main area contains a 'Name' field with the value 'DevConnect' and an empty 'Notes' field. 'Commit' and 'Cancel' buttons are at the top right, along with a 'Help ?' link.

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.



The screenshot shows the 'Location Pattern' sub-section. It features an 'Add' button and a 'Remove' button. Below is a table with one item: 'IP Address Pattern' with a checkbox and a value of '\*10.64.\*'. A 'Notes' column is also present. At the bottom, there are 'Commit' and 'Cancel' buttons.

IP Address Pattern	Notes
<input type="checkbox"/> *10.64.*	

## 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:

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

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 2<sup>nd</sup> page to view the rest of the adaptation.

**Adaptation Details** Commit Cancel [Help ?](#)

**General**

\* **Adaptation Name:**

\* **Module Name:**

**Module Parameter Type:**

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	fromto	true
<input type="checkbox"/>	iodstd	avaya.com
<input type="checkbox"/>	iosrcd	avaya.com

Select : All, None Page 1 of 2

**Adaptation Details** Commit Cancel [Help](#)

**General**

\* **Adaptation Name:**

\* **Module Name:**

**Module Parameter Type:**

<input type="checkbox"/>	Name	Value
<input type="checkbox"/>	iodstd	10.64.110.230
<input type="checkbox"/>	iosrcd	10.64.110.212

Select : All, None Page 2 of 2

## 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.

The screenshot shows the 'SIP Entity Details' configuration interface. On the left, a navigation pane is visible with 'Routing' selected. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. At the top right of the main area are 'Commit' and 'Cancel' buttons. The configuration fields are as follows:

- Name:** trio
- \* FQDN or IP Address:** 10.64.110.230
- Type:** SIP Trunk
- Notes:** (empty)
- Adaptation:** trio
- Location:** DevConnect
- Time Zone:** America/Denver

A 'Help ?' link is located in the top right corner of the main area.


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.

#### Entity Links

Override Port & Transport with DNS SRV:

Add Remove							
1 Item 							Filter: Enable
<input type="checkbox"/>	Name ▲	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* sm81_trio_5060_TCP	sm81	TCP ▼	* 5060	trio	* 5060	trusted

Select : All, None

## 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.

The screenshot shows the 'SIP Entity Details' configuration page. On the left is a navigation menu with 'Routing' selected. The main content area is titled 'SIP Entity Details' and has a 'General' tab. The fields are as follows:

- Name:** cm81
- FQDN or IP Address:** 10.64.110.213
- Type:** CM
- Notes:** (empty)
- Adaptation:** (empty)
- Location:** DevConnect
- Time Zone:** America/Denver

Buttons for 'Commit', 'Cancel', and 'Help ?' are located in the top right corner.

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”

**Entity Links**

Override Port & Transport with DNS SRV:

Add		Remove						
1 Item							Filter: Enable	
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	
<input type="checkbox"/>	* sm81_cm81_5061_TLS	sm81	TLS	* 5061	cm81	* 5061	trusted	

Select : All, None



## 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.

**Routing Policy Details** Commit Cancel [Help ?](#)

**General**

\* Name:

Disabled:

\* Retries:

Notes:

**SIP Entity as Destination**

Select

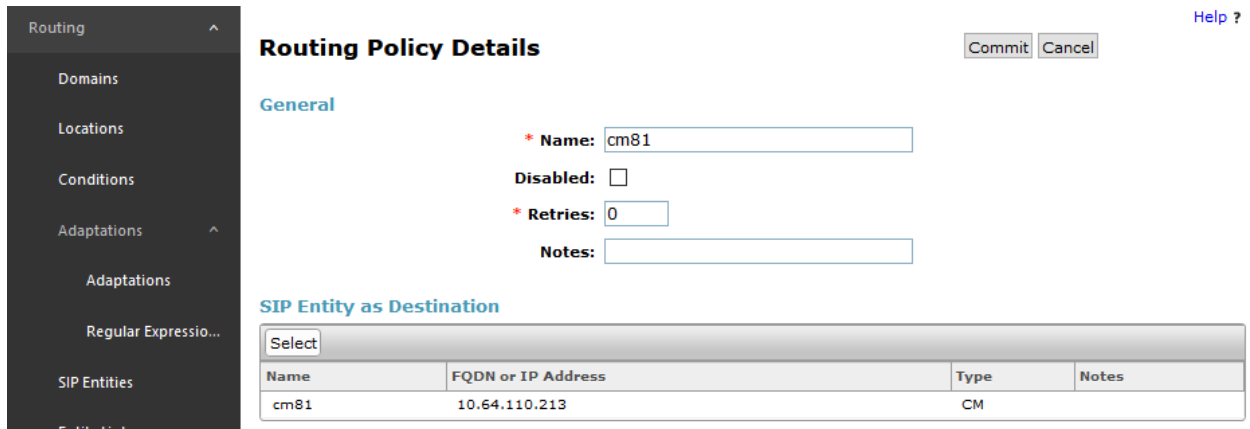
Name	FQDN or IP Address	Type	Notes
trio	10.64.110.230	SIP Trunk	

## 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.



**Routing Policy Details** Commit Cancel [Help ?](#)

**General**

\* Name:

Disabled:

\* Retries:

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
cm81	10.64.110.213	CM	

## 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.

**Dial Pattern Details** Commit Cancel [Help ?](#)

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations and Routing Policies**

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		trio	0	<input type="checkbox"/>	trio	

Select : All, None

## 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.

**Dial Pattern Details** Commit Cancel [Help ?](#)

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations and Routing Policies**

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name <small>▲</small>	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		cm81	0	<input type="checkbox"/>	cm81	

Select : All, None

- Routing ^
- Domains
- Locations
- Conditions
- Adaptations ^
- Adaptations
- Regular Expressio...
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns ^

## Dial Pattern Details

[Help ?](#)

### General

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**  ▾

**Notes:**

### Originating Locations and Routing Policies

<input type="button" value="Add"/>		<input type="button" value="Remove"/>					
1 Item		Filter: <input type="button" value="Enable"/>					
<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		cm81	0	<input type="checkbox"/>	cm81	

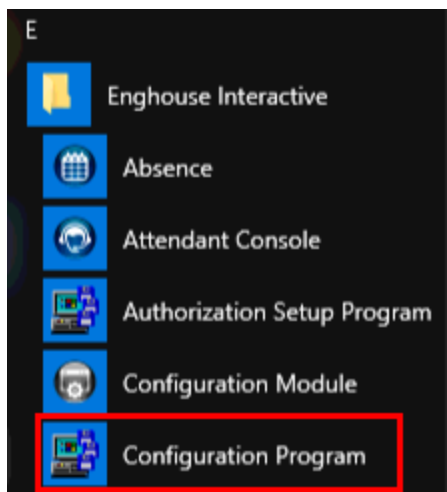
Select : All, None

## 7. Configure Intuition Advanced Console

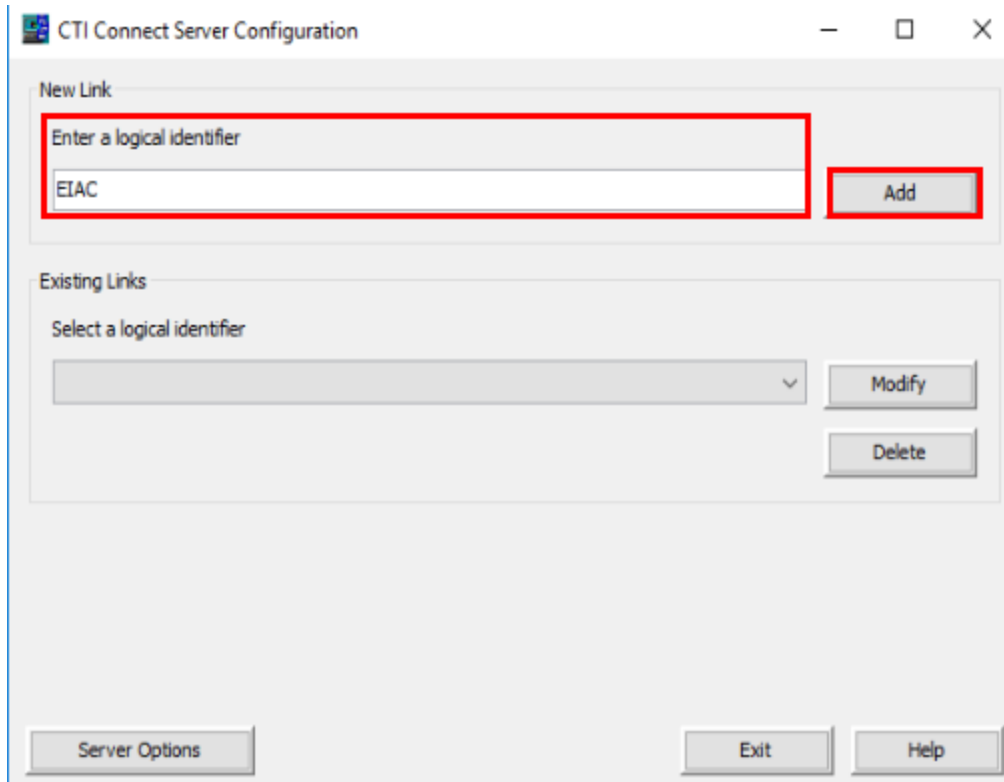
This section shows how to configure Intuition Advanced Console to successfully connect to Session Manager. The installation of the IAC software is assumed to be completed and the IAC services are up and running. The steps to configure SIP Trunks are as follows:

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

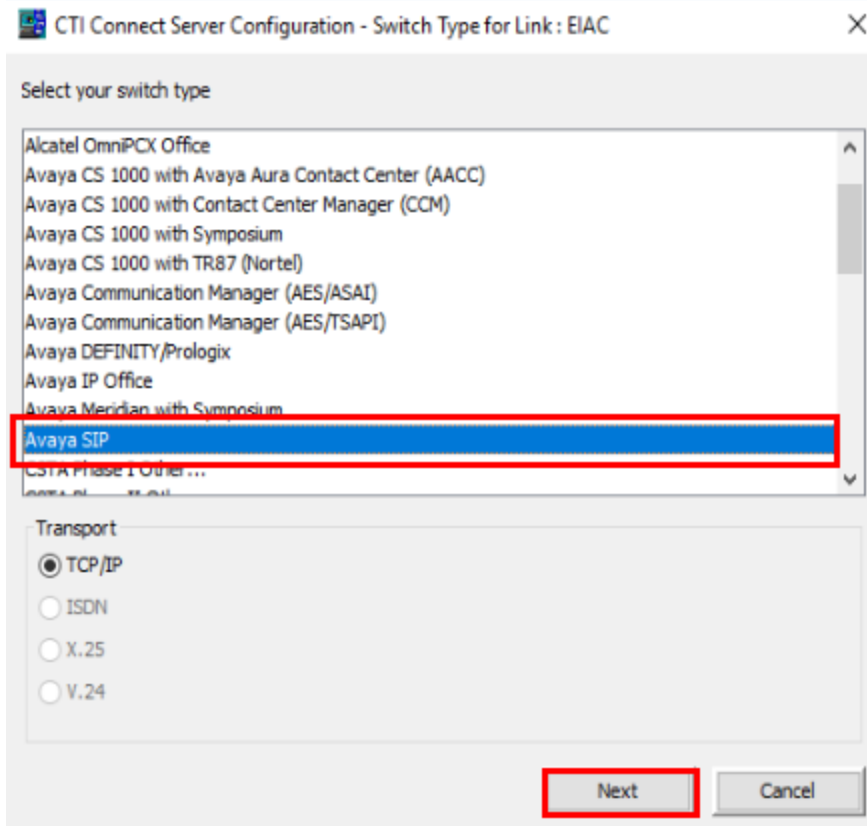
### 7.1. Intuition Advanced Console to use SIP Trunks



- Launch the “Configuration Program”



- Enter "EIAC" as logical identifier and click **Add**



- Select **Avaya SIP**



CTI Connect Server Configuration - Configuring Link: EIAC

⌵
✕

<p><b>Transport</b></p> <p>Switch IP Address: localhost</p> <p>Port Number: 7777</p> <p>Local IP Address (optional):</p>	<p><b>Protocol Specific</b></p> <p><input checked="" type="checkbox"/> ACSE</p> <p><input checked="" type="checkbox"/> Device Query</p> <p><input checked="" type="checkbox"/> Persistent Agent Connection</p> <p>Agent Connection ID: 0</p> <p>Agent Connection Name: CTI Connect</p> <p>Avaya Sess. Man. Address: 10.64.110.212</p> <p>Route Point Format: ^5\d{4}\$</p> <p>Failover IP Address (optional):</p> <p>Monitoring Interval (secs.): 30</p> <p>SIP Profile: external</p> <p>Transfer Type: Re-INVITE</p> <p>Conference Termination: Final Member</p> <p>Transport: TCP</p> <p>Default Route Dest. (optional):</p> <p>Route Request Timeout (ms): 10000</p> <p>Sounds Path: ouse interactive\media gateway\sounds</p>
<p><b>Common</b></p> <p><input checked="" type="checkbox"/> Auto Start Link</p> <p><input type="checkbox"/> Auto Restart Monitors</p> <p>Timestamp: Server</p> <p>Call Information Manager: localhost</p>	
<p><b>Device Level Authorization</b></p> <p>Authorization: Off</p>	

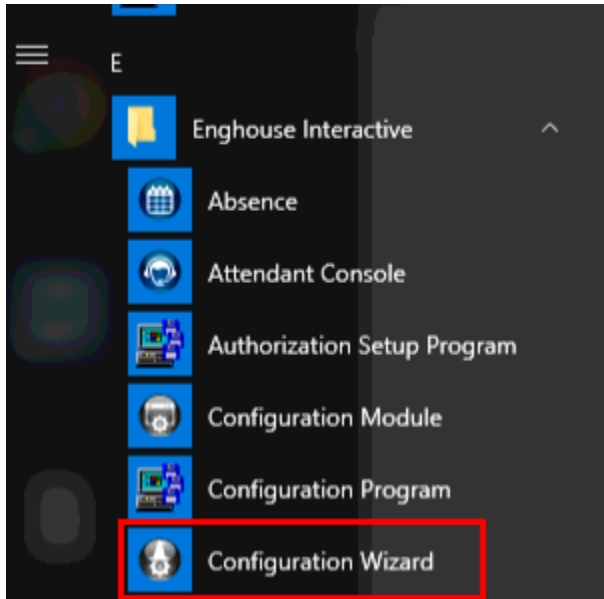
Advanced
Trace
Save
Cancel

- 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

EIAC - Configuration Wizard - Intuition Advanced Console - Item 1 of 5 ×

Press Escape to exit Help

<b>General</b>	
Phone DN	<input type="text" value="70105"/>
Server Reconnection Interval	<input type="text" value="5"/>
<b>Primary</b>	
Server IP	<input type="text" value="127.0.0.1"/>
Server Port	<input type="text" value="59152"/>
Routing DN	<input type="text" value="59999"/>
<b>Backup</b>	
Server IP	<input type="text" value="127.0.0.1"/>
Server Port	<input type="text" value="59152"/>
Routing DN	<input type="text" value="8900"/>

### Configuring the Enghouse Intuition Advanced Console Server Connections

Use the first Wizard page to configure on the client the IP addresses and ports of the primary and backup Enghouse Intuition Advanced Console servers, and the server reconnection interval.

1. Under **General**:
  - a. Type the **Phone DN**. This is the DN of the telephony device to connect to.

**Help:**

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

Database connection method

Primary

Configure data source:

Server name: WIN-UFLD705NG19\ENGHOUSE

Database name: EIAC

Use Windows authentication?

Use existing data source

EIAC

10 Retry interval (s)

Enable backup EIAC database? (optional licensed feature)

Enable Combined Operator Statistics? (optional licensed feature)

Cancel < Back **Next >** Help

Press F1 to enter Help

## Configuring the Database Connection

Use the **Database connection method** page to configure the computer's connection to the Enghouse Intuition Advanced Console database and, if licensed, EIAC backup databases and the Combined Operator Statistics database.

To configure the EIAC database connection:

1. Select the **Primary** tab.
2. Select **Configure data source**.

Help: < Back Next >

- Leave default and click **Next**

EIAC system login

Standard

Integrated

Cancel < Back **Next >** Help

Press F1 to enter Help

## Configuring the Enghouse Intuition Advanced Console System Login

Use the **Enghouse Intuition Advanced Console system login** page to configure the type of system login.

1. Select either:
  - o **Standard** - log in to the application using your Enghouse Intuition Advanced Console-specific login credentials.

Help: < Back Next >

- Leave default and click **Next**

Number plan  Include + in dialable characters

Press F1 to enter Help

### Configuring the Attendant Console Server Dialable Characters

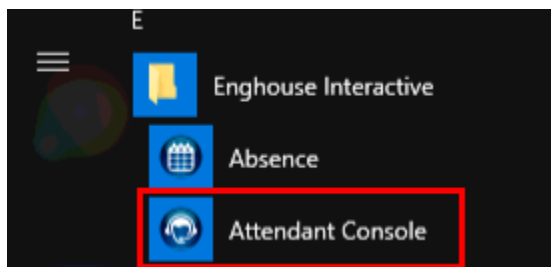
Use the **Attendant Console Server** page to configure the Enghouse Intuition Advanced Console dialable characters to include the + (plus) character. The dialable characters are configured on the Console Server and then passed to the client applications: Attendant Console, Person, and Click-2-Dial.

1. Set **Include + in dialable characters** to

Cancel < Back **Next >** Help

Help: < Back Next >

- Leave default and click **Next**



- Launch the attendant console

Please provide your login name and password.

Database: EIAC

Username: op1

Password:

OK Cancel

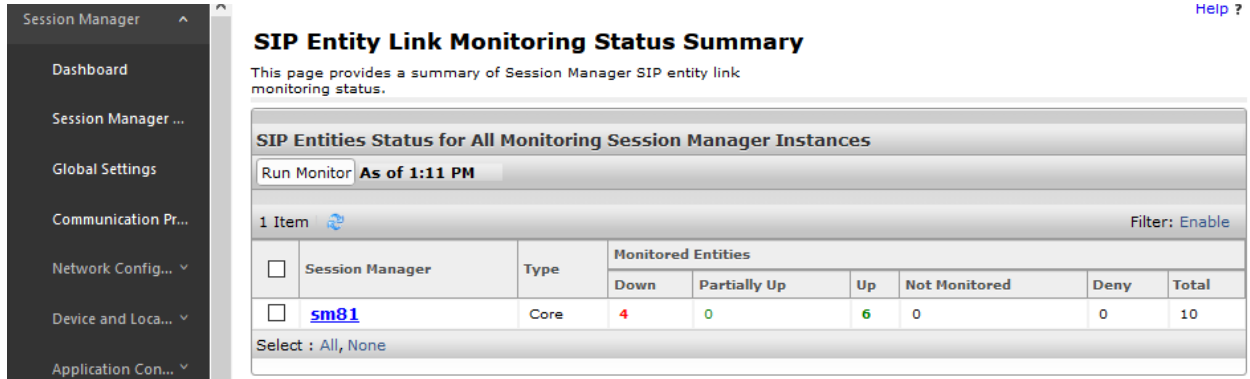
- Enter your credentials and click **OK**

## 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**.



The screenshot shows the 'SIP Entity Link Monitoring Status Summary' page. On the left is a navigation menu with items like 'Dashboard', 'Session Manager ...', 'Global Settings', 'Communication Pr...', 'Network Config...', 'Device and Loca...', and 'Application Con...'. The main content area has a title 'SIP Entity Link Monitoring Status Summary' and a subtitle 'This page provides a summary of Session Manager SIP entity link monitoring status.' Below this is a section titled 'SIP Entities Status for All Monitoring Session Manager Instances' with a 'Run Monitor' button and a timestamp 'As of 1:11 PM'. A table shows monitoring data for one item, 'sm81', with columns for 'Down', 'Partially Up', 'Up', 'Not Monitored', 'Deny', and 'Total'. A 'Filter: Enable' link is also present.

	Session Manager	Type	Monitored Entities					
			Down	Partially Up	Up	Not Monitored	Deny	Total
<input type="checkbox"/>	<a href="#">sm81</a>	Core	4	0	6	0	0	10

Select : All, None

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
<input type="radio"/>	<a href="#">sm81</a>	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

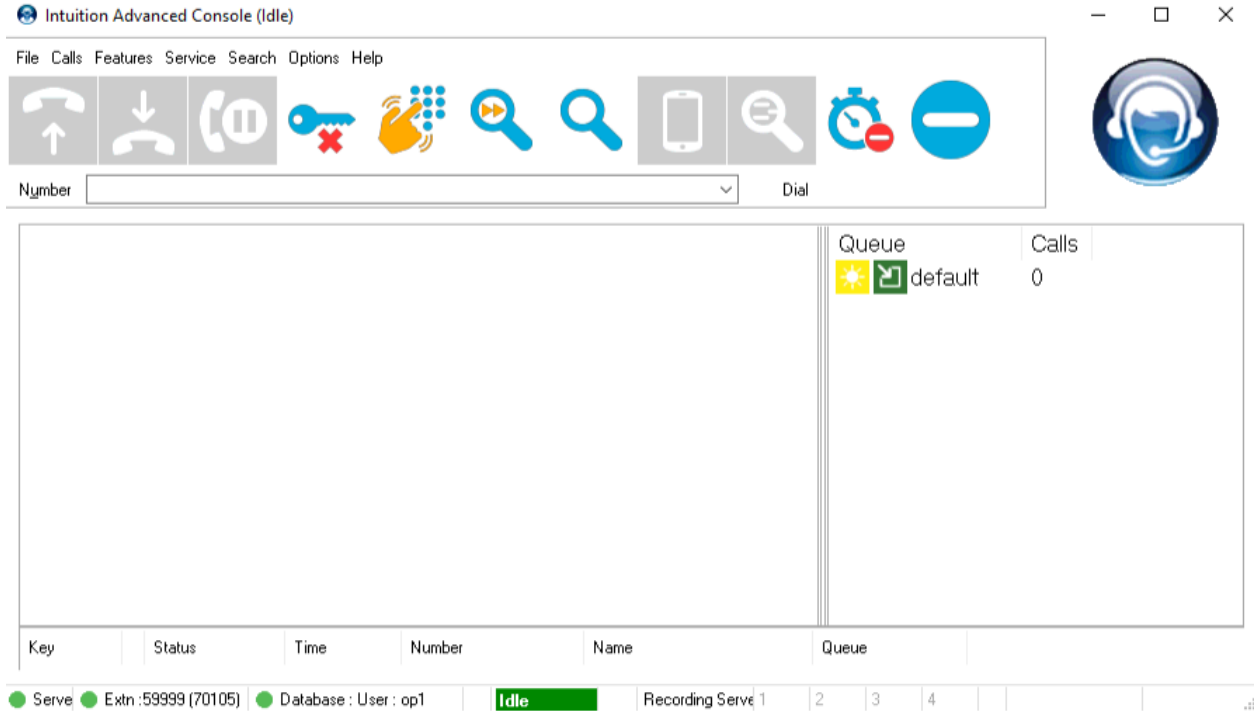
1 Item Filter: Enable

	Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	<a href="#">sm81</a>	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.



Intuition Advanced Console (Idle)

File Calls Features Service Search Options Help

Number  Dial

Queue	Calls
  default	0

Key Status Time Number Name Queue

● Serve ● Extn : 59999 (70105) ● Database : User : op1 **Idle** Recording Serve 1 2 3 4

## 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>



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