



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

---

# **Application Notes for Datapulse Intuition Enterprise Server with Avaya Aura<sup>®</sup> Session Manager and Avaya Aura<sup>®</sup> Communication Manager - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for the Datapulse Intuition Enterprise Server to act as an attendant gateway using Avaya Aura<sup>®</sup> Session Manager to connect to Avaya Aura<sup>®</sup> Communication Manager. Calls are queued in the Datapulse Intuition Enterprise Server and then sent to the Datapulse Intuition Enterprise Attendant application where calls can be distributed as required. Datapulse Intuition Enterprise Server is a windows based application that can operate many instances of the Datapulse Intuition Attendant.

Information in these Application Notes has been obtained through interoperability compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

The purpose of this document is to describe the steps required for Datapulse Intuition Enterprise Server to interoperate successfully with Avaya Aura<sup>®</sup> Session Manager and Avaya Aura<sup>®</sup> Communication Manager. Datapulse Intuition Enterprise Server is configured to receive calls sent by Avaya Aura<sup>®</sup> Communication Manager via Avaya Aura<sup>®</sup> Session Manager. Calls are queued by the Datapulse Intuition Enterprise Server and distributed to Datapulse Intuition Enterprise Attendant application. Calls can be distributed to agents, stations and external numbers as required using either a blind or announced method of transfer. The Datapulse Enterprise Server can also route unanswered or busy extension calls to the Intuition Enterprise Attendant application. These calls are presented to the Intuition Enterprise Attendant application as busy or unanswered calls giving the operator information with regard to the intended recipient of the call and the reason the call has been redirected.

## 2. General Test Approach and Test Results

To test the interoperability of Datapulse Intuition Enterprise Server with Session Manager and Communication Manager, calls were presented and distributed to the Intuition Enterprise Attendant application using a number of scenarios designed to test the features of Datapulse Intuition Enterprise Server.

### 2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The feature testing verified Intuition Enterprise Server's ability to receive and process inbound calls. The serviceability testing introduced failure scenarios to see if Intuition Enterprise Server could recover from failures in connectivity to Session Manager and Communication Manager.

Interoperability compliance testing covered the following features and functionality

- Ability for an inbound call from different sources to be answered.
- Ability for an inbound call to be forwarded to the required number.
- Ability for an inbound call to be retrieved when intended recipient is not available.
- Ability for the Intuition Enterprise Attendant application to make calls receive and transfer calls.

### 2.2. Test Results

During testing Intuition Enterprise Server carried out all tasks with the expected outcome in each scenario.

### 2.3. Support

For technical support of Datapulse products contact the Datapulse Service Desk:

Web: <http://www.datapulse.com>

Email: [support@datapulse.com](mailto:support@datapulse.com)

Telephone: +1 800 657 1530 in North America

+44 (0) 118 972 8400 in Europe

+61 433 986 344 in Asia

+971 4 501 5600 in Middle East & Africa

### 3. Reference Configuration

The configuration used as an example in these Application Notes is shown in **Error! Reference source not found.** The diagram illustrates an Enterprise SIP environment. The configuration consists of an S8800 server running Communication Manager connected to a G650 Media Gateway, Session Manager and System Manager that allow LAN connectivity to H.323 Avaya IP handsets. The configuration also includes a Windows 2003 server running Intuition Enterprise Server and a Windows PC running Intuition Enterprise Attendant.

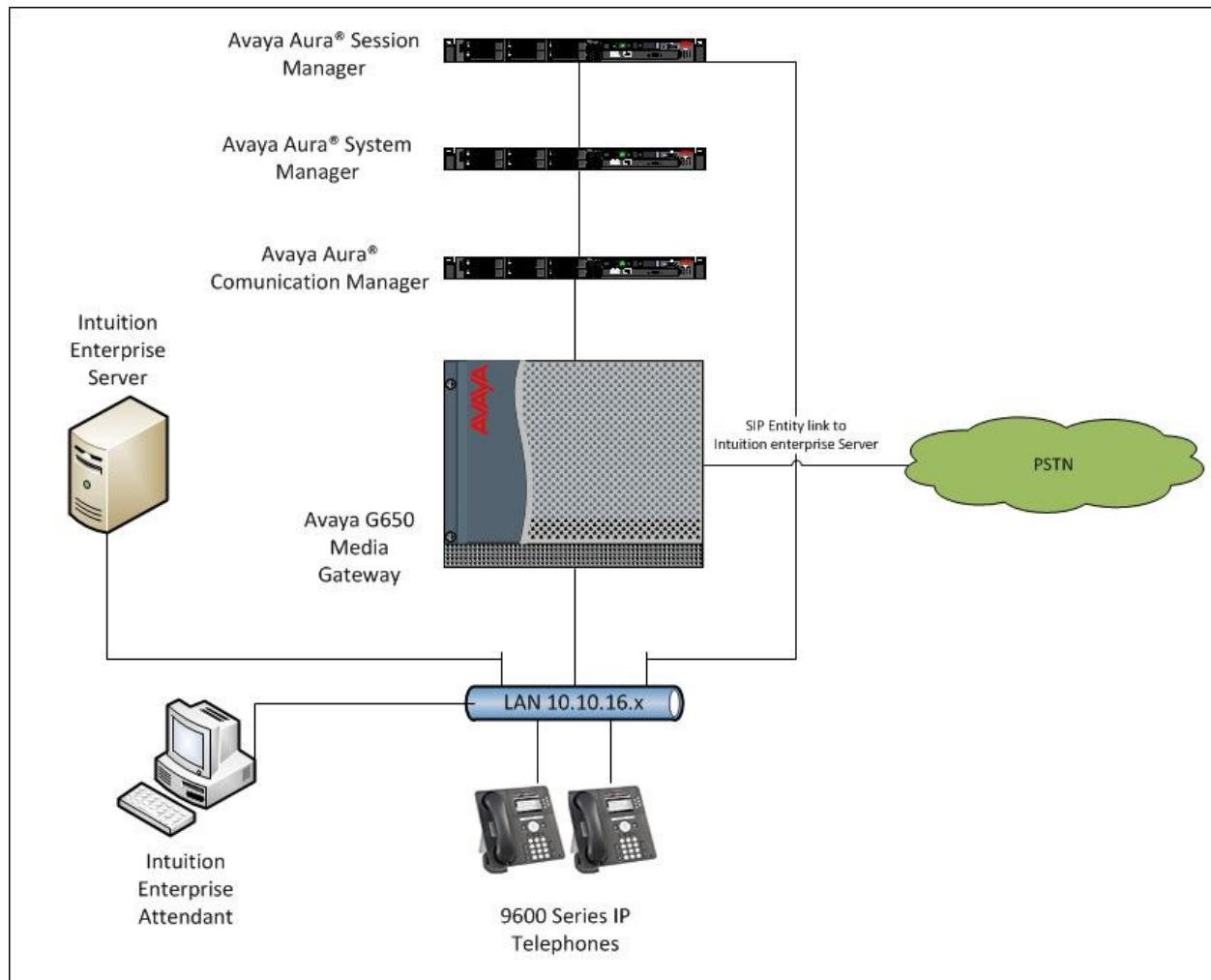


Figure 1: Intuition Enterprise Server with SIP Enterprise Configuration.

## 4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server with G650 Media Gateway	Avaya Aura <sup>®</sup> Communication Manager 6.0.1 (R016x.00.0.345.0) with Service Pack 1 (Patch 18444)
Avaya S8800 Server	Avaya Aura <sup>®</sup> Session Manager 6.1 (Build 6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura <sup>®</sup> System Manager 6.1 (Build 6.1.0.4.5072-6.1.4.113)
Avaya 9600 Series Handsets	3.1 (H.323)
Datapulse Intuition Enterprise Server	3.1
Datapulse Intuition Enterprise Attendant	3.1

## 5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring the Communication Manager to communicate with Intuition Enterprise Server via a SIP Trunk. Log in to Communication Manager with the appropriate credentials. The following steps are performed using the SAT (System Access Terminal) on Communication Manager:

- Verify system capacity
- Define the Dial Plan
- Define Node Name
- Define SIP Trunk to Avaya Aura® Session Manager
- Define Call Routing

### 5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that the **Maximum Administered SIP Trunks** allowed in 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	250	
Maximum Concurrently Registered IP Stations:		18000	2	
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	0	
Maximum Video Capable IP Softphones:		18000	5	
<b>Maximum Administered SIP Trunks:</b>		<b>24000</b>	<b>349</b>	
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0	
Maximum Number of DS1 Boards with Echo Cancellation:		522	0	
Maximum TN2501 VAL Boards:		128	0	
Maximum Media Gateway VAL Sources:		250	0	
Maximum TN2602 Boards with 80 VoIP Channels:		128	0	
Maximum TN2602 Boards with 320 VoIP Channels:		128	1	
Maximum Number of Expanded Meet-me Conference Ports:		300	0	
(NOTE: You must logoff & login to effect the permission changes.)				

## 5.2. Define Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. The number used to access Intuition enterprise server must be administered as an **aar** type. A **4** digit entry using **4** as the first number is used in this example.

change dialplan analysis						Page 1 of 12			
DIAL PLAN ANALYSIS TABLE									
Location: all						Percent Full: 2			
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	
1	4	ext							
309	4	ext							
350	4	ext							
4	4	aar							

## 5.3. Define Node Name

Use the **change node-names ip** command to add the Session Manager security module **Name** and **IP Address**. The **CLAN** IP address should also be noted for use in the **Near-end Node Name** field in **Section 5.4**.

change node-names ip		IP NODE NAMES	
Name	IP Address		
AES522	10.10.16.25		
<b>CLAN</b>	<b>10.10.16.31</b>		
CM521	10.10.16.23		
Gateway	10.10.16.1		
IPbuffer	10.10.16.184		
Intuition	10.10.16.51		
<b>SM61</b>	<b>10.10.16.201</b>		

## 5.4. Define SIP Trunk

Use the **add signaling-group** command to define the **Group Type**, **Transport Method**, **Near-end and Far-End Node Names** and **Ports** to be used by the SIP Trunk.

```
add signaling-group 6
                                SIGNALING GROUP

Group Number: 6                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: Others

Near-end Node Name: CLAN        Far-end Node Name: SM61
Near-end Listen Port: 5060      Far-end Listen Port: 5060
                                Far-end Network Region: 1

Far-end Domain:

Incoming Dialog Loopbacks: eliminate Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload          RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3 Direct IP-IP Audio Connections? y
    Enable Layer 3 Test? n          IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n Initial IP-IP Direct Media? n
                                Alternate Route Timer(sec): 6
```

Use **add trunk-group** command to define the SIP Trunk group. On **Page 1** the **Group Type** is set to **sip** and the **Service Type** is set to **tie**. The **Signaling Group** administered above is added as well as the **Number of Members** required.

```
add trunk-group 6
                                TRUNK GROUP
                                Page 1 of 21

Group Number: 6                Group Type: sip                CDR Reports: y
Group Name: Intuition Server    COR: 1                      TN: 1                TAC: 707
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: 6
                                Number of Members: 10
```

On **Page 3** set the **Numbering Format** to **private**

<b>add trunk-group 6</b>	<b>Page</b> 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
<b>Numbering Format: private</b>	
	UII Treatment: service-provider
	Replace Restricted Numbers? n
	Replace Unavailable Numbers? n
Modify Tandem Calling Number: no	
Show ANSWERED BY on Display? y	

## 5.5. Define Call Routing

Use the **change aar analysis** command to configure the **Dialed String**, **Route Pattern**, **Call Type** and **Total Min/Max** digits to be dialed.

<b>change aar analysis 4</b>	<b>Page</b> 1 of 2				
AAR DIGIT ANALYSIS TABLE					
Location: all					
Percent Full: 1					
<b>Dialed String</b>	<b>Total Min Max</b>	<b>Route Pattern</b>	<b>Call Type</b>	Node	ANI
444	4 4	6	unku	Num	Reqd
					n

Use the **change route-pattern** command to configure the trunk **Grp No** and **FRL**.

<b>change route-pattern 6</b>	<b>Page</b> 1 of 3								
Pattern Number: 7		Pattern Name: Intuition							
SCCAN? n		Secure SIP? n							
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
No		Mrk	Lmt	List	Del	Digits		QSIG	
						Dgts		Intw	
1:	6	0						n	user

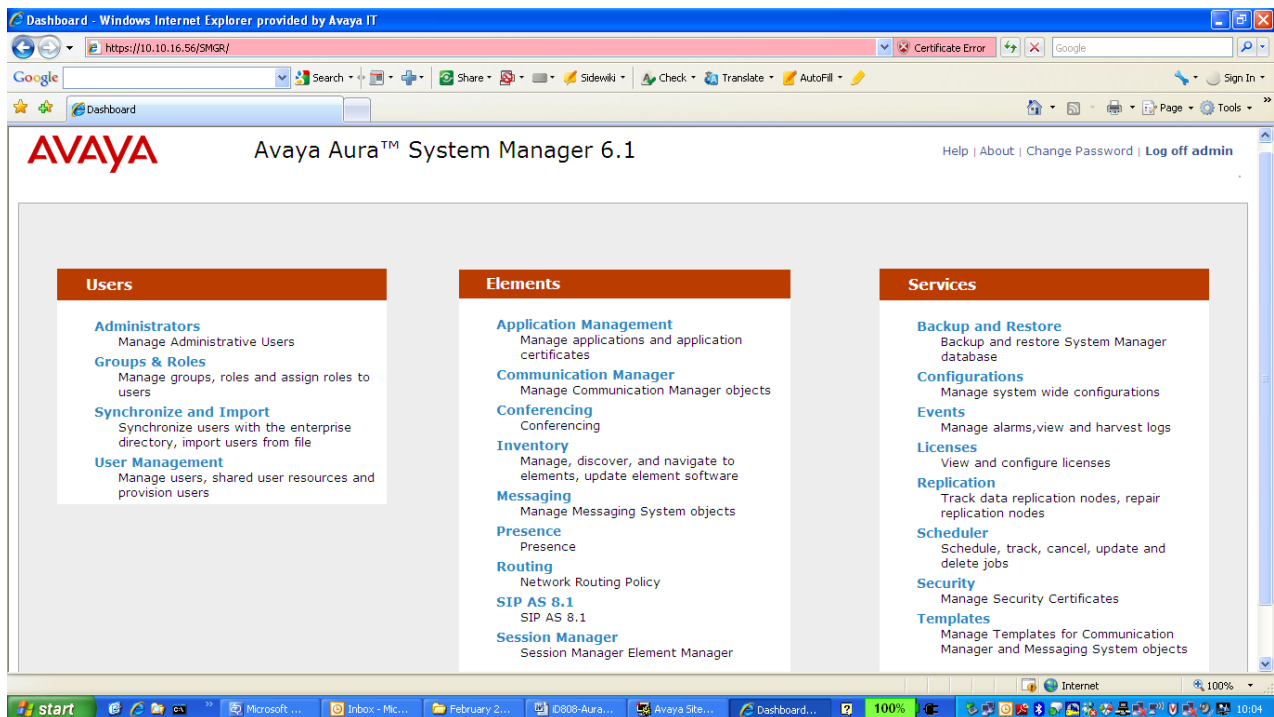


## 6. Configure Avaya Aura® Session Manager

These steps are designed to set up Session Manager to communicate with Intuition Enterprise Server as a SIP Entity and allow calls to be routed.

### 6.1. Logging in to Avaya Aura® System Manager

To access the administration web interface, enter **http://<ip-addr>/SMGR** as the URL in an Internet browser. Where <ip-addr> is the IP address of System Manager on System Platform. Log in with the appropriate credentials. The Dashboard screen is displayed, as shown below.



## 6.2. Verify System Properties

From the Dashboard of the web interface, choose Session Manager from the Elements section. Verify that a green tick shows under **Tests Passed**, **Security Module** is **Up** and **Service State** is set to **Accept New Service**.

**Session Manager Dashboard**  
This page provides the overall status and health summary of each administered Session Manager.

**Session Manager Instances**  
Service State: [Dropdown] Shutdown System: [Dropdown] As of 11:44 AM

<input type="checkbox"/>	Session Manager	Type	Alarms	Tests Pass	Security Module	Service State	Entity Monitoring	Active Call Count	Registrations	Version
<input type="checkbox"/>	61sesmgr	Core	8/0/1	✓	Up	Accept New Service	0/2	0	4	6.1.0.0.610

Select : All, None

Next go to Routing from the Elements section of the Dashboard and select Domains and check the administered domain.

**Domain Management**  
Edit New Duplicate Delete More Actions

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

Select : All, None

## 6.3. Add Location

Select Routing from the Elements section of the Dashboard and chose **Locations**. Click on the new button (not shown) and add a **Name** and the IP Address range in the format shown under **Location Pattern** → **IP Address Pattern**. Click on the **Commit** button to save.

The screenshot shows the 'Location Details' configuration page. On the left is a sidebar menu with options: Domains, Locations (selected), Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a 'Commit' button in the top right. Below the title 'Location Details', there is a note: 'Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting'. The 'General' section contains fields for 'Name' (filled with 'SessionMGR') and 'Notes'. The 'Overall Managed Bandwidth' section has 'Managed Bandwidth Units' (set to 'Kbit/sec') and 'Total Bandwidth'. The 'Per-Call Bandwidth Parameters' section has 'Default Audio Bandwidth' (set to '80 Kbit/sec'). The 'Location Pattern' section has 'Add' and 'Remove' buttons, a table with 1 item, and a 'Refresh' button. The table has columns for 'IP Address Pattern' and 'Notes'. The first row shows '10.10.16.\*' in the 'IP Address Pattern' column. A 'Filter:' label is visible on the right side of the table.

## 6.4. Create a SIP entity

From the Elements section of the Dashboard choose **Routing**. From the left hand side menu choose **SIP Entities**. Click on the **New** button (not shown) and enter the required information, **Location** of the Session Manager Manager created in **Section 6.3** and set the **Type** to Session Manager.

The screenshot shows the 'SIP Entity Details' configuration page. The sidebar menu on the left has 'Routing' selected, and the main content area has a breadcrumb trail: 'Home / Elements / Routing / SIP Entities - SIP Entity Details'. The 'Commit' button is in the top right. The 'General' section contains fields for 'Name' (filled with '61sesmgr'), 'FQDN or IP Address' (filled with '10.10.16.201'), 'Type' (set to 'Session Manager'), and 'Notes'. The 'Location' is set to 'SessionMGR'. The 'Outbound Proxy' and 'Time Zone' (set to 'Etc/GMT') are also visible. The 'Credential name' field is empty. The 'SIP Link Monitoring' section has a 'SIP Link Monitoring' checkbox (checked) and a 'Use Session Manager Configuration' dropdown menu.

Add the protocol and port information in the **Port** section shown below. The **Entity Links** section will automatically populate after the link is added in **Section 6.5** Click **Commit** to save the changes.

Entity Links

Add

Remove

2 Items

Refresh

Filter: Enable

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	61sesmgr	TCP	* 5060	Commgr	* 5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	61sesmgr	TLS	* 5061	SBC6	* 5061	<input checked="" type="checkbox"/>

Select : All, None

Port

Add

Remove

3 Items

Refresh

Filter: Enable

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

Select : All, None

\* Input Required

Commit

Cancel

Repeat above steps to add Communication Manager entity by setting the **Type** field to **CM**. Click on the **Commit** button to save.

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

SIP Entity Details

Commit

General

\* Name:

Commgr

\* FQDN or IP Address:

10.10.16.31

Type:

CM

Notes:

Adaptation:

Location:

SessionMGR

Time Zone:

Etc/GMT

Override Port & Transport with DNS SRV:

☐

\* SIP Timer B/F (in seconds):

4

Credential name:

Call Detail Recording:

none

Repeat above steps to add the Intuition Enterprise Server by setting the **Type** field to **SIP Trunk**. Click on the **Commit** button to save.

The screenshot shows a web interface for configuring SIP entities. On the left is a navigation menu with the following items: Routing (selected), Domains, Locations, Adaptations, SIP Entities (highlighted in blue), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a breadcrumb trail: Home / Elements / Routing / SIP Entities - SIP Entity Details. Below the breadcrumb is the title 'SIP Entity Details' and a 'Commit' button in the top right corner. The 'General' tab is active. The configuration fields are as follows: 'Name' is 'Intuition'; 'FQDN or IP Address' is '10.10.16.51'; 'Type' is a dropdown menu set to 'SIP Trunk'; 'Notes' is an empty text box; 'Adaptation' is a dropdown menu; 'Location' is a dropdown menu set to 'SessionMGR'; 'Time Zone' is a dropdown menu set to 'Etc/GMT'; 'Override Port & Transport with DNS SRV' is an unchecked checkbox; '\* SIP Timer B/F (in seconds):' is a text box containing '4'; 'Credential name' is an empty text box; and 'Call Detail Recording' is a dropdown menu set to 'egress'.

Routing

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

Commit

General

\* Name: Intuition

\* FQDN or IP Address: 10.10.16.51

Type: SIP Trunk

Notes:

Adaptation:

Location: SessionMGR

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

\* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: egress

## 6.5. Add a Communication Manager Entity Link

From the Routing menu choose **Entity Links**. Click on the New button (not Shown) give the link an appropriate **Name** and then choose the entities added in **Section 6.4**, the **Protocol** to use (TCP used in this example) and the **Port** used to communicate on. Click on the **Commit** button to save.

The screenshot shows the 'Entity Links' configuration page. On the left is a sidebar menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (highlighted), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a breadcrumb 'Home / Elements / Routing / Entity Links- Entity Links' and a 'Commit' button. Below this is a table with the following data:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* ToCM	* 61sesmgr	TCP	* 5060	* Commgr	* 5060	<input checked="" type="checkbox"/>	

At the bottom, there is a '\* Input Required' message and another 'Commit' button.

## 6.6. Add an Intuition Entity Link

From the **Routing** menu choose **Entity Links**. Click on the New button (not Shown) give the link an appropriate name and then choose the entities added in **Section 6.4**, the **Protocol** to use (TCP used in this example) and the **Port** used to communicate on. Click on the **Commit** button to save.

The screenshot shows the 'Entity Links' configuration page, similar to the previous one. The sidebar menu is the same. The main content area shows a table with the following data:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* to Intuition	* 61sesmgr	TCP	* 5060	* Intuition	* 5060	<input checked="" type="checkbox"/>	

At the bottom, there is a '\* Input Required' message and another 'Commit' button.

## 6.7. Add Communication Manager Managed Element

From the **Elements** section of the Dashboard choose **Inventory** and then **Manage Elements**. Click the **New** button and enter a valid **Name**, **Type** as **CM** and the **Node** as the Communication Manager Server IP address. Click on **Commit** to save.

The screenshot shows the 'Edit CM: CommsMGR' form. The left sidebar has 'Inventory' expanded with 'Manage Elements' selected. The main area has tabs for 'Application' and 'Attributes'. The 'Application' tab is active, showing a form with the following fields: 'Name' (CommsMGR), 'Type' (CM), 'Description' (empty), and 'Node' (10.10.16.47). A 'Commit' button is in the top right corner.

## 6.8. Add Routing Policy

From the **Elements** section of the dashboard choose **Routing** and then **Routing Policies**. Click on the **New** button (not shown) and add a **Name** for the policy. Click on the **Select** button under **SIP Entity as Destination**.

The screenshot shows the 'Routing Policy Details' form. The left sidebar has 'Routing' expanded with 'Routing Policies' selected. The main area has tabs for 'General' and 'SIP Entity as Destination'. The 'General' tab is active, showing a form with the following fields: 'Name' (to Intuition), 'Disabled' (checkbox), and 'Notes' (empty). A 'Commit' button is in the top right corner. Below the 'SIP Entity as Destination' tab is a 'Select' button.

Select the Intuition Enterprise Server entity as a Destination and click on the **Select** button. When routed back to the previous screen click on the **Commit** button (not show) to save.

The screenshot shows the 'SIP Entity List' table. The left sidebar has 'Routing' expanded with 'Routing Policies' selected. The main area has a 'SIP Entity List' section with a 'Select' button. Below it is a table with the following data:

Name	FQDN or IP Address	Type	Notes
61sesmgr	10.10.16.201	Session Manager	
Commgr	10.10.16.31	CM	
Intuition	10.10.16.51	SIP Trunk	
SBC5	10.10.16.122	Gateway	

Below the table is a 'Select : None' button.

## 6.9. Add Dial Pattern

Select **Dial Patterns** from the **Routing** menu and click on the **New** button (not shown). Enter the **Pattern** used for dialing the Intuition Enterprise Server and the **Min** and **Max** digits. Click on the **Add** button under **Originating Locations and Routing Policies**.

Home / Elements / Routing / Dial Patterns- Dial Pattern Details

Dial Pattern Details Commit

General

\* Pattern: 444

\* Min: 4

\* Max: 4

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add Remove

Check the **Originating Location** and **Routing policies** required and click on the **Select** button. Click on the **Commit** button when routed back to the previous page to save.

Home / Elements / Routing / Dial Patterns- Originating Location and Routing Policy List

Originating Location and Routing Policy List Select

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

1 Item Refresh Filter

<input checked="" type="checkbox"/>	Name	Notes
<input checked="" type="checkbox"/>	SessionMGR	

Select : All, None

Routing Policies

3 Items Refresh Filter

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	ToCH	<input type="checkbox"/>	Commgr	
<input checked="" type="checkbox"/>	to Intuition	<input type="checkbox"/>	Intuition	
<input type="checkbox"/>	toSBC	<input checked="" type="checkbox"/>	SBC6	

**Note:** Dial Pattern will automatically be added to the routing policies selected.

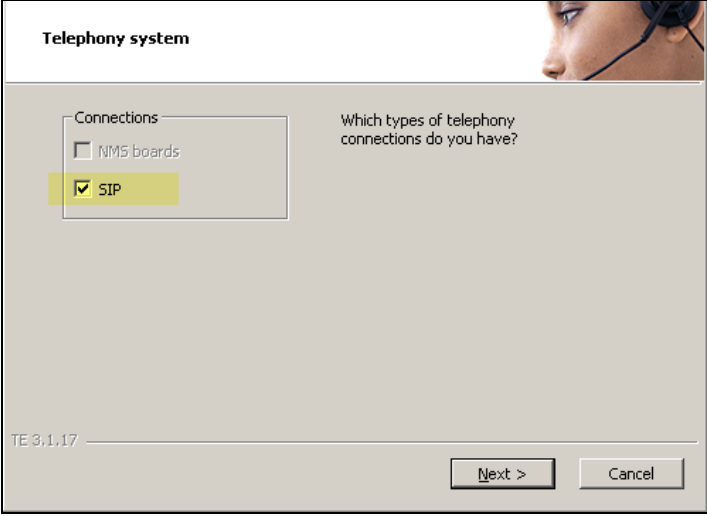


## 7. Configure Datapulse Intuition Enterprise Server

These steps are designed to set up Intuition Enterprise server to interoperate with CM and Session Manager using SIP.

### 7.1. General setup

Start the Line interface configuration program from **Programs → Intuition Enterprise** on the start menu. Make sure **SIP** is selected and click **Next**.



**Telephony system**

Connections

☐ NMS boards

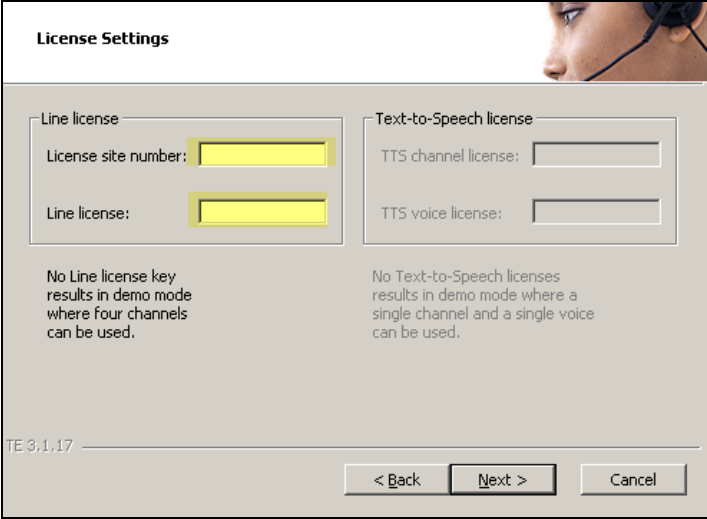
☒ SIP

Which types of telephony connections do you have?

TE 3.1.17

Next > Cancel

Enter your **License site number** and **Line License** codes and click **Next**.



**License Settings**

Line license

License site number:

Line license:

Text-to-Speech license

TTS channel license:

TTS voice license:

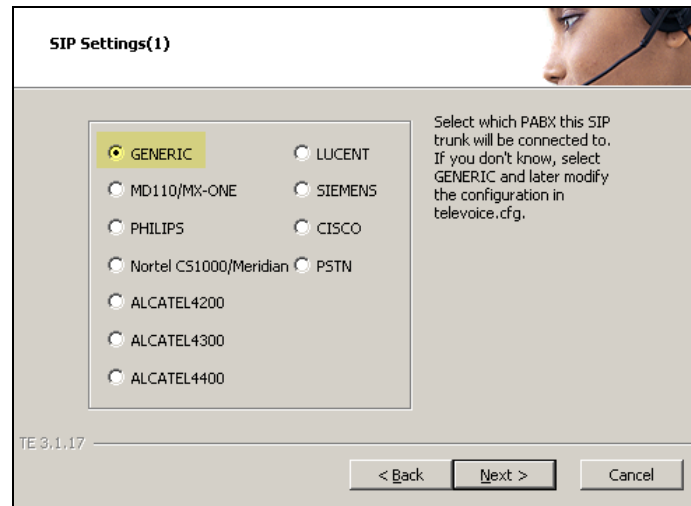
No Line license key results in demo mode where four channels can be used.

No Text-to-Speech licenses results in demo mode where a single channel and a single voice can be used.

TE 3.1.17

< Back Next > Cancel

Make sure **GENERIC** is selected and click **Next**.



**SIP Settings(1)**

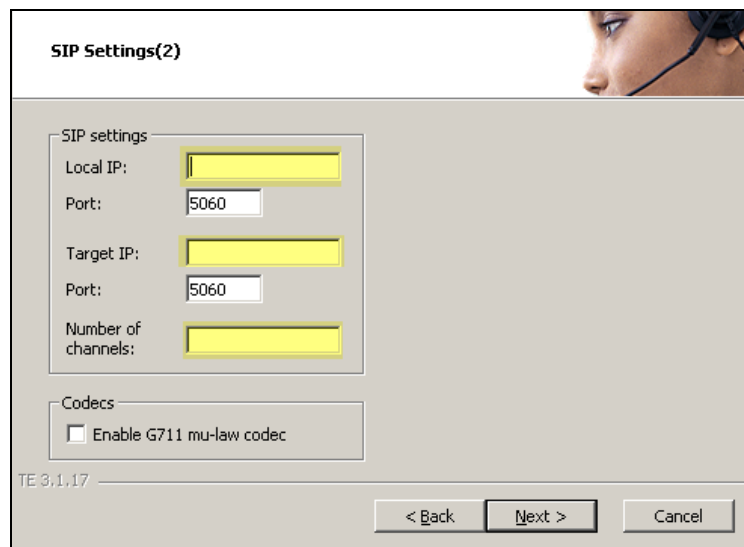
Select which PABX this SIP trunk will be connected to. If you don't know, select GENERIC and later modify the configuration in televoice.cfg.

☒ **GENERIC**      ☐ LUCENT  
☐ MD110/MX-ONE      ☐ SIEMENS  
☐ PHILIPS      ☐ CISCO  
☐ Nortel CS1000/Meridian      ☐ PSTN  
☐ ALCATEL4200  
☐ ALCATEL4300  
☐ ALCATEL4400

TE 3.1.17

< Back    Next >    Cancel

Enter your **Local IP**, the **Target IP** that relates to the Communication Manager interface from **Section 5.3** and **Number of channels** that should be activated. When done, click **Next**.



**SIP Settings(2)**

SIP settings

Local IP:

Port:

Target IP:

Port:

Number of channels:

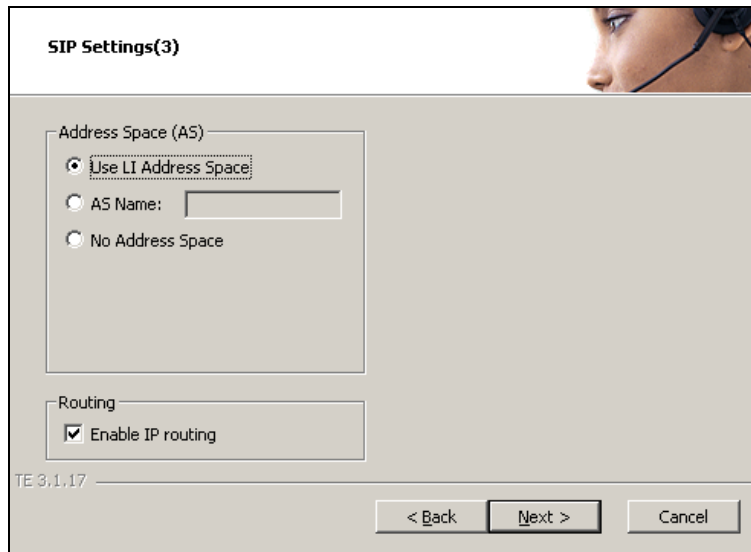
Codecs

☐ Enable G711 mu-law codec

TE 3.1.17

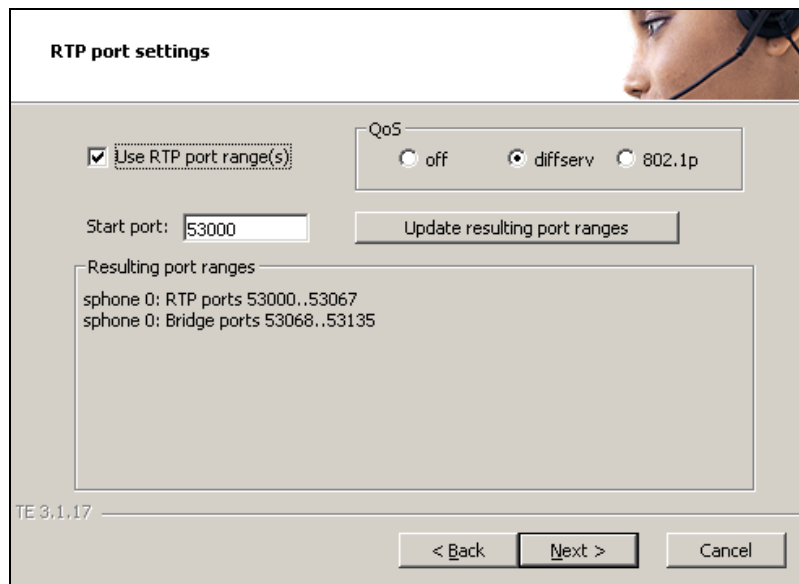
< Back    Next >    Cancel

Click **Next** to use the default settings.



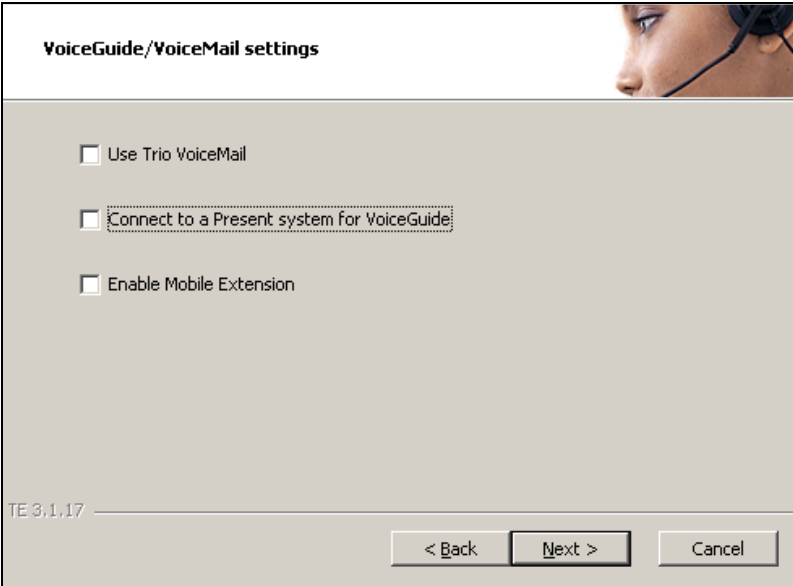
The image shows a dialog box titled "SIP Settings(3)". It has a header bar with a small profile picture of a person wearing a headset. The main area contains two sections: "Address Space (AS)" and "Routing". In the "Address Space (AS)" section, there are three radio buttons: "Use LI Address Space" (which is selected), "AS Name:" followed by an empty text box, and "No Address Space". In the "Routing" section, there is a checked checkbox labeled "Enable IP routing". At the bottom left, it says "TE 3.1.17". At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel".

Click **Next** to use the default setting.



The image shows a dialog box titled "RTP port settings". It has a header bar with a small profile picture of a person wearing a headset. The main area contains several controls: a checked checkbox labeled "Use RTP port range(s)", a "QoS" section with three radio buttons ("off", "diffserv" which is selected, and "802.1p"), a "Start port:" label followed by a text box containing "53000", and an "Update resulting port ranges" button. Below these is a section titled "Resulting port ranges" which contains two lines of text: "sphone 0: RTP ports 53000..53067" and "sphone 0: Bridge ports 53068..53135". At the bottom left, it says "TE 3.1.17". At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel".

Click **Next** to use the default setting.



**VoiceGuide/VoiceMail settings**

☐ Use Trio VoiceMail

☐ Connect to a Present system for VoiceGuide

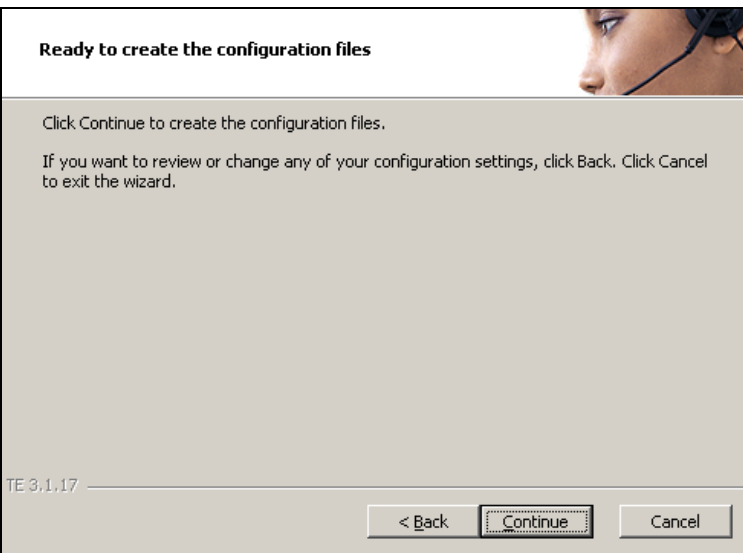
☐ Enable Mobile Extension

TE 3.1.17

< Back   Next >   Cancel

This dialog box is titled "VoiceGuide/VoiceMail settings". It contains three unchecked checkboxes: "Use Trio VoiceMail", "Connect to a Present system for VoiceGuide", and "Enable Mobile Extension". The "Connect to a Present system for VoiceGuide" checkbox is highlighted with a dashed border. At the bottom left is the text "TE 3.1.17". At the bottom right are three buttons: "< Back", "Next >", and "Cancel".

Click **Continue**.



**Ready to create the configuration files**

Click Continue to create the configuration files.

If you want to review or change any of your configuration settings, click Back. Click Cancel to exit the wizard.

TE 3.1.17

< Back   Continue   Cancel

This dialog box is titled "Ready to create the configuration files". It contains two paragraphs of text: "Click Continue to create the configuration files." and "If you want to review or change any of your configuration settings, click Back. Click Cancel to exit the wizard." At the bottom left is the text "TE 3.1.17". At the bottom right are three buttons: "< Back", "Continue", and "Cancel". The "Continue" button is highlighted with a dashed border.

## 7.2. Special configuration for Avaya Aura® Session Manager

Open C:\TE\ProgramData\LI\cfg\televoice.cfg. Find the [SIP\_1] Section.

```
[sip_1]
signallingprotocol=sip
localhost=10.10.16.51
targetHost=10.10.16.201
uriScheme=1
transferPoint=afterAnswer
usetcp = 1
```

Make sure 'uriScheme' is set to '1'

Add the row 'ucetecp=1'

## 7.3. Restart the TeleVoice service

From the start menu on the Intuition Enterprise Server open **Programs → Intuition Enterprise → Enterprise Management Center** select **TeleVoice** service and click on the **Restart** button to activate changes.

The screenshot shows the Enterprise Management Center interface. On the left is a navigation tree with 'Overview' selected, and sub-items: Servers, Services, frlabb, Parameters, and Subsystems. The main area displays the 'frlabbb' subsystem with a table of services. The 'TeleVoice' service is highlighted with a red box. At the bottom right, there are three buttons: 'Restart', 'Start', and 'Stop', with 'Restart' also highlighted by a red box.

Name	Status	Comment
MySQL	Running	MySQL Service
W3SVC	Running	World Wide Web Publishing Service Service
OAM	Running	Trio Operations And Maintenance Service
CPM	Running	Client Phone Manager Service
CC1	Running	Trio Contact Center CC1
CC1Mail	Not Active	Trio Contact Center CC1 Mail
TeleVoice	Running	Trio TeleVoice Service
CC1Line	Running	Trio Contact Center Line Interface CC1
Present	Running	Company Directory Service

## 8. Verification Steps

The following steps can be used to verify and troubleshoot the installation of the configuration tested in this document.

### 8.1. SIP Trunk Status

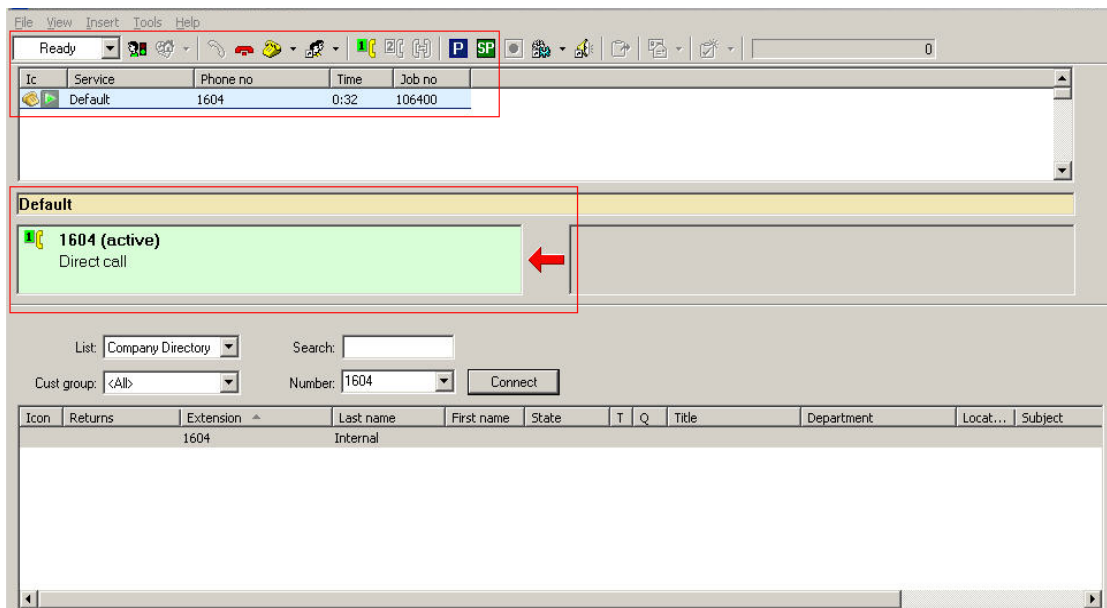
Make a call to the Intuition Enterprise server and answer the call using Intuition Enterprise Agent. Use the **status trunk** command to view **in-service active** trunk carrying the call.

```
status trunk 7
```

TRUNK GROUP STATUS				
Member	Port	Service State	Mtce Connected Ports Busy	
0007/001	T00629	in-service/idle	no	
0007/002	T00630	in-service/idle	no	
0007/003	T00631	in-service/idle	no	
<b>0007/004</b>	<b>T00632</b>	<b>in-service/active</b>	<b>no</b>	<b>S00019</b>
0007/005	T00633	in-service/idle	no	
0007/006	T00634	in-service/idle	no	
0007/007	T00635	in-service/idle	no	
0007/008	T00636	in-service/idle	no	
0007/009	T00637	in-service/idle	no	
0007/010	T00638	in-service/idle	no	

### 8.2. Active Call Information

Make a call the Intuition Enterprise Server and answer the call using Intuition Enterprise Attendant.



## 9. Conclusion

These Application Notes have described the administration steps required to use Datapulse Intuition Enterprise Server with Avaya Aura<sup>®</sup> Communication Manager and Avaya Aura<sup>®</sup> Session Manager. All features were tested successfully with the configuration detailed above.

## 10. References

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

- [1] *Administering Avaya Aura<sup>®</sup> Communication Manager*, 9<sup>th</sup> August 2010, Document Number 03-300509.
- [2] *SIP Support in Avaya Aura<sup>®</sup> Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.
- [3] *Installing and configuring Avaya Aura<sup>®</sup> Session Manager*, 5<sup>th</sup> January 2011, Document Number 03-603473.
- [4] *Session Initiation Protocol Service Examples draft-ietf-sipping-service-examples-15*, Internet-Draft, 11<sup>th</sup> July 2008, available at <http://tools.ietf.org/html/draft-ietf-sipping-service-examples-15>

---

**©2011 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).