



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

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# **Application Notes for Configuring BTS CommsWare Console 1.0 with Avaya® Communication Manager R8.1.2 and Avaya Aura® Session Manager R8.1.2 - Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps for provisioning BTS CommsWare Console 1.0 to interoperate with Avaya Aura® Communication R8.1.2 and Avaya Aura® Session Manager R8.1.2.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 describe the configuration steps for provisioning BTS CommsWare Console to interoperate with Avaya Aura® Communication Manager R8.1.2 and Avaya Aura® Session Manager R8.1.2.

CommsWare Console client from BTS is a screen-based Operator/Attendant console application running on a Desktop with a separate Avaya H.323 or SIP IP Deskphone for voice. It provides operators with extended call-handling functionality including call routing, call queuing, directory, search, attended and unattended transfers, camp-on and call parking (up to 6 calls per Operator). The CommsWare Console Client application works with BTS Call Director and BTS PhoneWare Directory to provide a Console solution. The Call Director connects to Communication Manager R8.1.2 using a SIP Trunk via an Entity Link on the Session Manager using UDP. An incoming call to an operator queue will be routed over the SIP Trunk to Call Director. The Call Director will then deliver the incoming call to an operator console queue. A login operator will answer the queue call from CommsWare Console client and the voice will be bridged onto the operator's Avaya Deskphone. Similarly, the outbound call is initiated from the CommsWare Console client where outbound call is made by Call Director via the SIP Trunk and the voice is bridged onto the operator's Avaya Deskphone.

**Note:** BTS supply, install and configure their solution for the end customer directly or through qualified partners. In line with BTS's request the configuration of the BTS solution is not required to be part of these Application Notes. Certain information from these Application Notes will be highlighted as a requirement for the BTS solution.

## 2. General Test Approach and Test Results

The interoperability compliance testing focused on verifying that CommsWare Console and all features behaved as expected. Various call scenarios were performed to simulate calls as would be observed on a customer premise. See **Figure 1** for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests. All calls destined for the CommsWare Console are routed to the Call Director over SIP trunks using Session Manager to route the calls.

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 the CommsWare Console did not include use of any specific encryption features as requested by BTS Holdings PLC.

## 2.1. Interoperability Compliance Testing

The interoperability compliance testing focuses on various testing scenarios to verify the use of CommsWare Console with the Avaya solution. In addition, serviceability tests were also performed to assess the reliability and accuracy of the joint solution. The testing focused on the following types of calls:

- Verification of connectivity between Communication Manager and CommsWare Console via Session Manager
- CommsWare Console Agent logs in/out
- Outbound calls to Communication Manager extensions and PSTN from the CommsWare Console
- DTMF and SIP Options keep alive
- Shuffling and non-shuffling scenarios
- G.711 Mu-law and G.711 A-law codec
- Inbound calls to the CommsWare Console queue number from Communication Manager and PSTN
- Hold and Transfer/Conference (both blind and supervised) between the Communication Manager and CommsWare Console
- Hold and Transfer/Conference (both blind and supervised) between PSTN and CommsWare Console
- CLID preservation for basic and transferred calls
- CommsWare Console Agent Parks/Unparks calls
- Greeting heard when calling queue
- CommsWare Console Agent Camps on calls
- All the above with physical console using Avaya SIP and H.323 Deskphones

Testing also includes serviceability test where the effect of call flow occurs when a LAN failure occurs in Session Manager, CommsWare server (including Call Director) and CommsWare Console.

## 2.2. Test Results

All test cases passed except for the following observations.

- For internal call transfer to a CommsWare Console (using Avaya SIP endpoint) as transferring party, the transfer-to party is not showing the calling party's extension/name.
- Loss of connectivity to Session Manager does not affect voice calls already setup by Call Director to console.

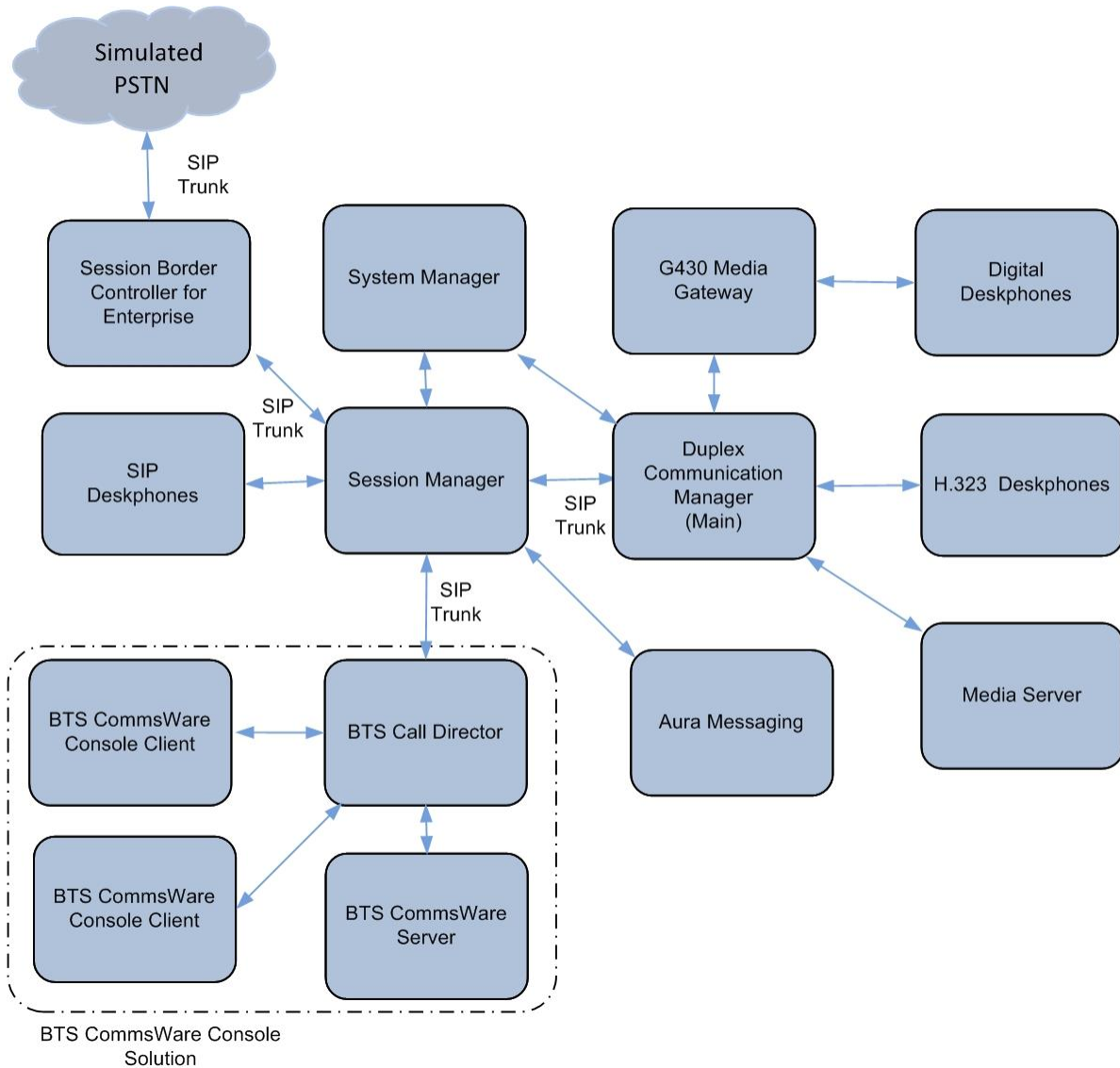
## 2.3. Support

Technical support for the BTS CommsWare solution can be obtained as follows.

Email: support@bts.co.uk  
Web: www.bts.co.uk  
Phone: +44 208 4019111

### 3. Reference Configuration

**Figure 1** shows the setup for compliance testing CommsWare Console with Communication Manager and Session Manager over SIP trunks to pass callers from Communication Manager and/or Session Manager to the CommsWare Console Operators.



**Figure 1: Connection of BTS CommsWare Console with Avaya Aura® Communication Manager R8.1.2 and Avaya Aura® Session Manager R8.1.2**

## 4. Equipment and Software Validated

The following equipment and software were used for the compliance test.

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on a virtual server	8.1.2.0.0.890.26095
Avaya G430 Media Gateway <ul style="list-style-type: none"><li>• MGP</li></ul>	41.16.0
Avaya Aura® System Manager running on a virtual server	8.1.2.0.0611588
Avaya Aura® Session Manager running on a virtual server	8.1.2.1.812101
Avaya Aura® Media Server running on a virtual server	8.0.2.93
Avaya Aura® Messaging running on a virtual server	7.1.532.002-1.263670
Avaya 9600 Series Deskphone <ul style="list-style-type: none"><li>• 9641G</li><li>• 9608</li></ul>	6.8304 (H.323) 7.1.9.0.8 (SIP)
Avaya J100 Series Deskphone <ul style="list-style-type: none"><li>• J179</li><li>• J169</li></ul>	6.8304 (H.323) 4.0.4.0.10 (SIP)
Avaya 1400 Series Digital Deskphone <ul style="list-style-type: none"><li>• 1408</li></ul>	R4 SP10
BTS Call Director	140
BTS CommsWare Server	1.07
BTS CommsWare Client	1.0.69

## 5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing and with SIP trunks in place to Session Manager. For further information on the configuration of Communication Manager, please see reference [2] in **Section 10** of these Application Notes. All the configuration changes in Communication Manager are performed through the System Access Terminal (SAT) interface.

### 5.1. Configure a Dial Plan for calls to the BTS Call Director

Enter “change dialplan analysis” command to utilise an unused extension. In this compliance test, extension beginning with **Dialed String 7** with **Total Length 5** is added.

```
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	
0	1	attd							
1	5	ext							
2	5	ext							
22	4	ext							
32	4	ext							
33	2	fac							
4	6	ext							
5	4	ext							
6	5	ext							
<b>7</b>	<b>5</b>	<b>ext</b>							
8	1	fac							
9	1	fac							
*	3	fac							
#	3	dac							

A matching pattern of **71xxx** is created on the uniform dial plan using “change uniform-dialplan 7” command **Len 5** and routing through the **aar** (private network).

```
change uniform-dialplan 7                                 Page 1 of 2
```

UNIFORM DIAL PLAN TABLE						
Percent Full: 0						
Matching Pattern	Len	Del	Insert Digits	Net	Conv	Node Num
<b>71</b>	<b>5</b>	<b>0</b>		<b>aar</b>		<b>n</b>

Check the existing route pattern for routing to the SIP Trunk Group **7** that connects to the Session Manager i.e. **6**.

```

display route-pattern 6                                     Page 1 of 4
      Pattern Number: 6      Pattern Name: non-IMS to SM
  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: 7      0      0      n user
  
```

Using “change aar analysis 71”, route the extension **71xxx** to the SIP Trunk group to the Session Manager with **Min** and **Max Dialed String** as length **5** through **aar** on **Route Pattern 6**.

```

change aar analysis 71                                     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
71      5 5      6      aar      n
      n
  
```



## 6. Configure Avaya Aura® Session Manager

In order to make changes in Session Manager a web session is established to System Manager. Log into System Manager by opening a web browser and navigating to <http://<System Manager IP Address>/SMGR>. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On** highlighted below.

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, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:   
Password:

[Change Password](#)

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

Once logged in, click on **Elements** → **Routing** shown below.

**AVAYA** Aura® System Manager 8.1

Users | Elements | Services | Widgets | Shortcuts | Search | admin

Home | **Routing**

**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

Step 8: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"
- (Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 10: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

Step 11: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

## 6.1. Domains and Locations

**Note:** It is assumed that a domain and a location have already been setup for an existing system which will not be highlighted here.

## 6.2. Adding BTS Call Director as a SIP Entity

Click on **SIP Entities** in the left column and select **New** in the right window (not shown). Enter a suitable **Name** for the new SIP Entity and the **IP Address** of the BTS Call Director. Enter the correct **Time Zone** and **Location** and click on **Commit**.

### SIP Entity Details

Commit Cancel

**General**

\* **Name:**

\* **FQDN or IP Address:**

**Type:**

**Notes:**

**Location:**

**Time Zone:**

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

**Minimum TLS Version:**

**Credential name:**

**Securable:**

**Call Detail Recording:**

### 6.3. Adding the BTS Call Director Entity Link

A UDP Entity link was added for the BTS Call Director. Click on **Entity Links** in the left column and select **New** in the main window (not shown). Enter a suitable **Name** for the Entity Link and select the **Session Manager** SIP Entity for **SIP Entity 1** and the newly created BTS Call Director Entity called **BTS Call Director** for **SIP Entity 2**. Ensure that **UDP** is selected for the **Protocol** and that **Port 5060** is used. Click on **Commit** once finished to save the new Entity Link.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	DNS Override	Connection Policy
* SM1_To_BTS	* sm1	UDP	* 5060	* BTS Call Director	* 5060	<input type="checkbox"/>	trusted

### 6.4. Adding the BTS Call Director Routing Policy

Click on **Routing Policies** in the left window and select **New** in the main window (not shown). Enter a suitable **Name** for the Routing Policy and click on **Select** under **SIP Entity as Destination**, highlighted below.

**General**

\* Name: To\_BTS

Disabled:

\* Retries: 0

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
------	--------------------	------	-------

Select the **BTS Call Director** SIP Entity as shown below and click on **Select**.

**SIP Entities** Select Cancel

---

**SIP Entities**

11 Items Filter: Enable

Name	FQDN or IP Address	Type	Notes
<input type="radio"/> AAEP-MPP	10.1.10.83	Voice Portal	
<input type="radio"/> AA Messaging	10.1.10.63	Other	AAM on DL360G7
<input type="radio"/> Avaya-CE	10.1.10.20	Avaya Breeze	
<input checked="" type="radio"/> <b>BTS Call Director</b>	10.1.10.126	SIP Trunk	
<input type="radio"/> CM8-Duplex	10.1.10.230	CM	
<input type="radio"/> g450-CM	10.1.60.18	CM	
<input type="radio"/> IP500v2	10.1.30.10	SIP Trunk	
<input type="radio"/> IPSE Expansion	10.1.30.151	SIP Trunk	
<input type="radio"/> IPSE Primary	10.1.10.121	SIP Trunk	
<input type="radio"/> presence	10.1.10.20	Presence Services	
<input type="radio"/> SBCE	10.1.10.65	SIP Trunk	SBCE

Select : None

The selected destination is now shown, click on **Commit** to save this.

**Routing Policy Details** Commit Cancel

---

**General**

\* Name:

Disabled:

\* Retries:

Notes:

---

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type
BTS Call Director	10.1.10.126	SIP Trunk

## 6.5. Adding a Dial Pattern for the BTS Call Director

Select **Dial Patterns** in the left window and select **New** in the main window (not shown). Enter the required digits for the Pattern, in the example below 71xxx is used. 71 is entered as the **Pattern** and the **Min** and **Max** digit length of **5** is used thus giving 71xxx. Click on **Add** under **Originating Locations, Origination Dial Pattern Sets, and Routing Policies** in order to select this Routing Policy.

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

**General**

\* **Pattern:**

\* **Min:**

\* **Max:**

**Emergency Call:**

**SIP Domain:**

**Notes:**

**Originating Locations, Origination Dial Pattern Sets, and Routing Policies**

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
--------------------------	-----------------------------	----------------------------	-----------------------------------	------------------------------------	---------------------	------	-------------------------	----------------------------	----------------------

Select the Originating Location and select the newly created routing policy for the BTS Call Director created in **Section 6.4** for **Routing Policies**.

**Originating Location**

Apply The Selected Routing Policies to All Originating Locations

1 Item		Filter: Enable
<input checked="" type="checkbox"/>	Name	Notes
<input checked="" type="checkbox"/>	Location1	

Select : All, None

**Origination Dial Pattern Sets**

0 Items		Filter: Enable
Name	Notes	

**Routing Policies**

10 Items					Filter: Enable
<input type="checkbox"/>	Name	Disabled	Destination	Notes	
<input type="checkbox"/>	To-aaep-mpp	<input type="checkbox"/>	AAEP-MPP		
<input type="checkbox"/>	To AAM	<input type="checkbox"/>	AA Messaging		
<input checked="" type="checkbox"/>	To_BTS	<input type="checkbox"/>	BTS Call Director		
<input type="checkbox"/>	To-CM-duplex	<input type="checkbox"/>	CM8-Duplex		
<input type="checkbox"/>	To_CM_Duplex_Trunk8	<input type="checkbox"/>	CM8-Duplex	InBound calls to CM via Trunk 8	
<input type="checkbox"/>	To-CM-Site6	<input type="checkbox"/>	g450-CM		
<input type="checkbox"/>	To-IP500v2	<input type="checkbox"/>	IP500v2		
<input type="checkbox"/>	To-IPSE Exp	<input type="checkbox"/>	IPSE Expansion		
<input type="checkbox"/>	To-IPSE Primary	<input type="checkbox"/>	IPSE Primary		
<input type="checkbox"/>	To_Service_Provider_via_SBCE	<input type="checkbox"/>	SBCE	For outbound calls to SIP via SBCE	

Select : All, None

Select Cancel

With the Routing Policy selected click on **Commit** to finish adding the **Dial Pattern**.

[Help ?](#)

**Dial Pattern Details**

**General**

\* Pattern:

\* Min:

\* Max:

Emergency Call:

SIP Domain:

Notes:

**Originating Locations, Origination Dial Pattern Sets, and Routing Policies**

1 Item		Filter: Enable							
<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Origination Dial Pattern Set Name	Origination Dial Pattern Set Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Location1				To_BTS	0	<input type="checkbox"/>	BTS Call Director	

Select : All, None

## 7. Configure BTS Call Director Server and CommsWare Server

As stated in **Section 1**, BTS does not require the configuration of BTS Call Director Server or CommsWare server to be part of these Application Notes. However, do note that queue numbers 71001-71004 were configured for the operators on the Call Director and two extensions were configured as consoles.

## 8. Verification Steps

The following step can be taken to ensure that the connection between CommsWare Console and Session Manager is configured correctly.

### 8.1. Verify Link to Avaya Aura® Session Manager

Log into System Manager as done previously in **Section 6**, and navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring Status Summary**. The following screen appear as below. Select the **BTS Call Director** SIP Entity.

**SIP Entity Link Monitoring Status Summary**

This page provides a summary of Session Manager SIP entity link monitoring status.

---

**SIP Entities Status for All Monitoring Session Manager Instances**

Run Monitor **As of 4:47 PM**

---

2 Items Filter: Enable

	Session Manager	Type	Monitored Entities					Deny	Total
			Down	Partially Up	Up	Not Monitored			
<input type="checkbox"/>	<a href="#">sm1</a>	Core	3	0	8	0	0	11	
<input type="checkbox"/>	<a href="#">sm2</a>	Core	0	0	5	0	0	5	

Select : All, None

---

**All Monitored SIP Entities**

Run Monitor

---

13 Items Filter: Enable

<input type="checkbox"/>	SIP Entity Name
<input type="checkbox"/>	<a href="#">sm2</a>
<input type="checkbox"/>	<a href="#">AA Messaging</a>
<input type="checkbox"/>	<a href="#">sm1</a>
<input type="checkbox"/>	<a href="#">Avaya-CE</a>
<input type="checkbox"/>	<a href="#">presence</a>
<input type="checkbox"/>	<a href="#">AAEP-MPP</a>
<input type="checkbox"/>	<a href="#">g450-CM</a>
<input type="checkbox"/>	<a href="#">CM8-Duplex</a>
<input type="checkbox"/>	<a href="#">SBCE</a>
<input type="checkbox"/>	<a href="#">IP500v2</a>
<input checked="" type="checkbox"/>	<a href="#">BTS Call Director</a>
<input type="checkbox"/>	<a href="#">IPSE Expansion</a>



Note that the SIP Entity **BTS Call Director**, shows **Link Status UP** and **Reason Code 200 OK**.

**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: **BTS Call Director**

Summary View

1 Item Filter: Enable

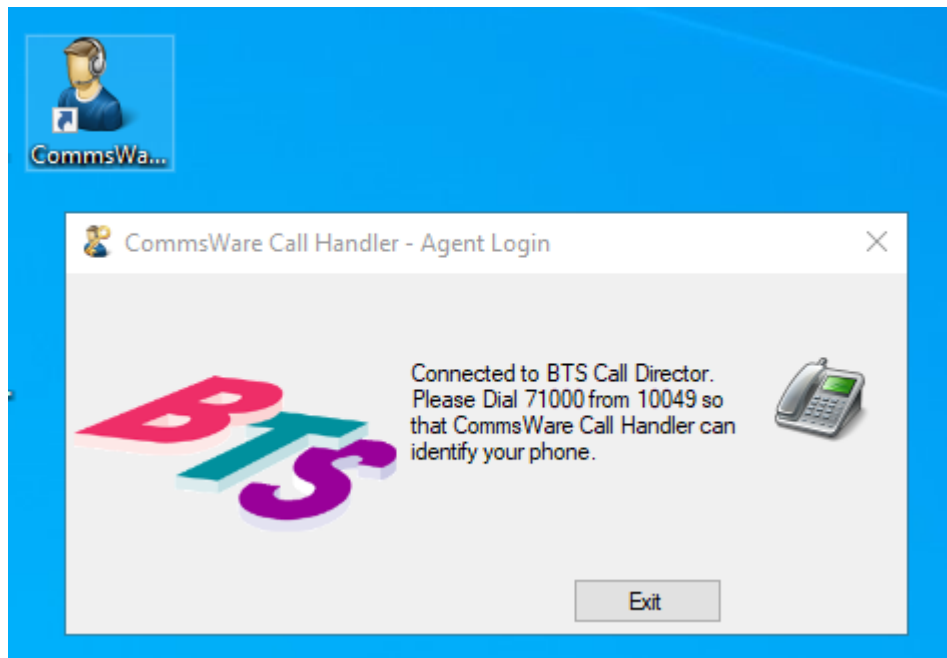
	Session Manager Name	Session Manager IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	<a href="#">sm1</a>	IPv4	10.1.10.126	5060	UDP	FALSE	UP	200 OK	UP

Select : None

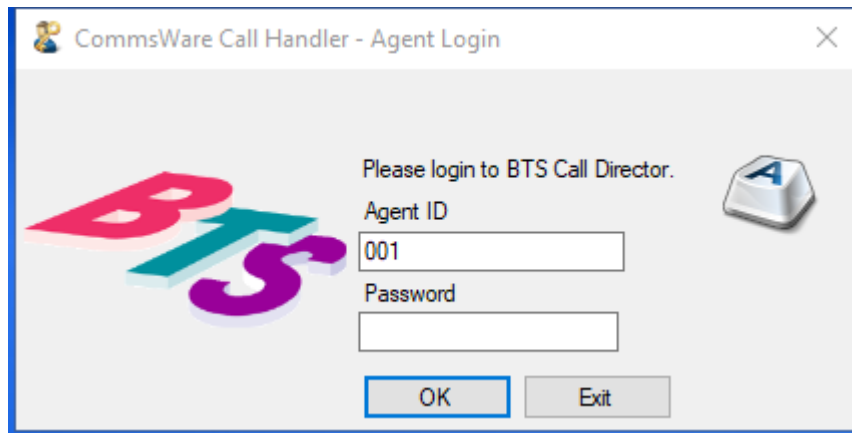
## 8.2. Verify BTS CommsWare Console

If the CommsWare Console is operational and connected correctly to the BTS Call Director and the Avaya solution the following steps should be possible.

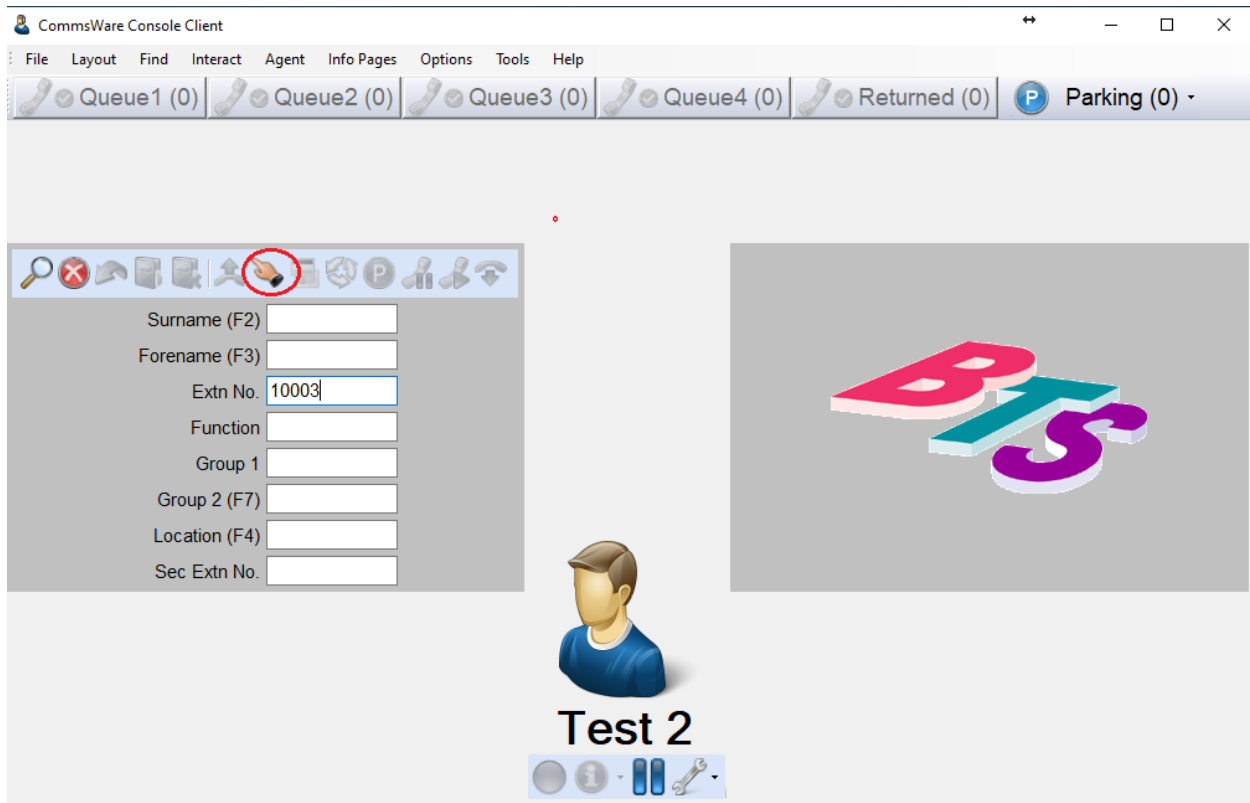
On the console client PC, click the **CommsWare Call Handler** shortcut and a message will be displayed asking to log in via an Avaya phone associated with the CommsWare Console. In this compliance testing, an Avaya H.323 and SIP Deskphone are used as physical console.



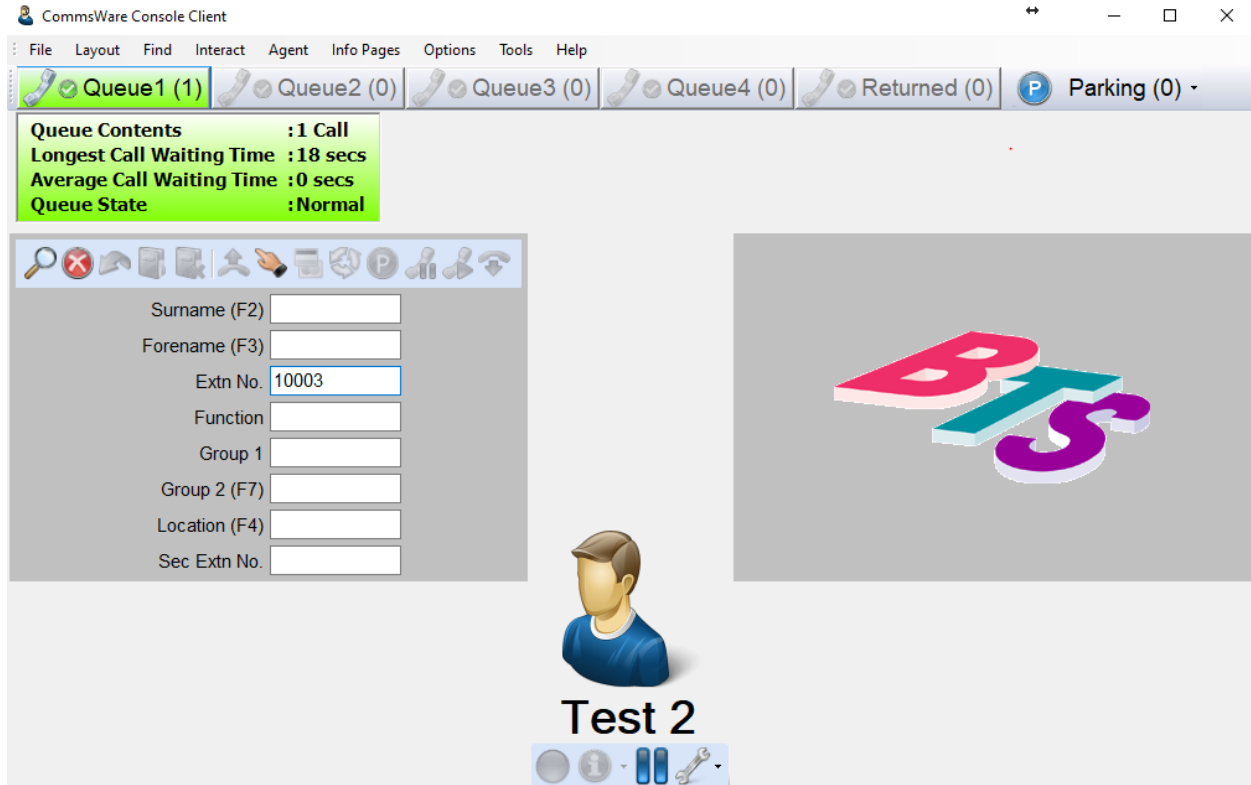
Make a call from the Avaya Deskphone to the BTS Call Director using the number indicated on the prompt message earlier and the following screen should be displayed asking for the BTS agent login details. Logged in with appropriate credentials.



Outgoing calls can be made by entering the appropriate **Extn No.** (e.g., an internal extension below) as shown below and clicking on the icon highlighted.



Incoming calls are made by calling to the queue number (for compliance testing a number of queues from 71001-71004 were added). These queue numbers are configured on the Call Director. The call was made to 71001 as shown below and this call can be answered by selecting the queue flashing.



## 9. Conclusion

These Application Notes describe the configuration steps required for BTS CommsWare Console 1.0 to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager R8.1.2. Please refer to **Section 2.2** for test results and observations.

## 10. Additional References

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

- [1] *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 8.1.x, Issue 5, Jun 2020.
- [2] *Administering Avaya Aura Communication Manager*, Release 8.1.x, Issue 6, Mar 2020
- [3] *Deploying Avaya Aura® Session Manager and Avaya Aura® Branch Session Manager in Virtual Appliance*, Release 8.1.x, Issue 3, Mar 2020.
- [4] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 6, Aug 2020.

Product documentation for BTS CommsWare Console can be requested from BTS or may be downloaded from <http://www.bts.co.uk>

- [1] *CommsWare Console Client Getting Started Guide*, 2020.
- [2] *Installing BTS CommsWare Console Client*, 2020.

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