



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

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# **Application Notes for Amtelco Genesis Intelligent Series with Avaya Aura® Session Manager – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Amtelco Genesis Intelligent Series to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Amtelco Genesis Intelligent Series is a SIP-based solution that provides operator users with phone and call controls.

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 required for Amtelco Genesis Intelligent Series (Genesis) to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunks. Genesis is a SIP-based solution that provides operator users with phone and call controls.

The Genesis solution consists of the Genesis Telephony Server, Intelligent Series Server, Intelligent Series Supervisor, and Intelligent Series Soft Agent. Operators have desktops running the Intelligent Series Soft Agent application, with dedicated audio connections via SIP with the Genesis Telephony Server.

In the compliance testing, calls from internal and external callers were routed over SIP trunks via Session Manager to Genesis for operator functions. Genesis tracked the operator states and routed calls to available operators, and populated answering operator desktops with pertinent call information such as calling and called numbers. All call controls were performed from the operator desktops.

The unsupervised transfer feature was accomplished by Genesis via use of SIP REFER, and the supervised transfer and supervised conference features were accomplished by Genesis via merge/unmerge of respective audio connections.

## 2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually with necessary operator actions such as hold and transfer performed from the operator desktops.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to the Genesis servers and/or clients.

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.

## 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included inbound, outbound, internal, external, G.711, outbound DTMF, hold/resume, drop, display, transfer, supervised conference, multiple calls, and multiple operators.

The serviceability testing focused on verifying the ability of Genesis to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to the Genesis servers and/or clients.

## 2.2. Test Results

All test cases were executed and verified. The following were observations on Genesis from the compliance testing.

- When the desktop running the Soft Agent application experiences a network disruption while on an active call, the Soft Agent application may disappear from the screen. A user can recover the application when the network connection is restored by using Windows task manager to manually end the Soft Agent process and then re-launch the application. For additional help reach out to Amtelco Support.
- Genesis returned 404 Not Found for OPTIONS messages from Session Manager, and this was displayed on the SIP Entity connection status screen on Session Manager. This did not appear to have any other negative impact.

## 2.3. Support

Technical support on Genesis can be obtained through the following:

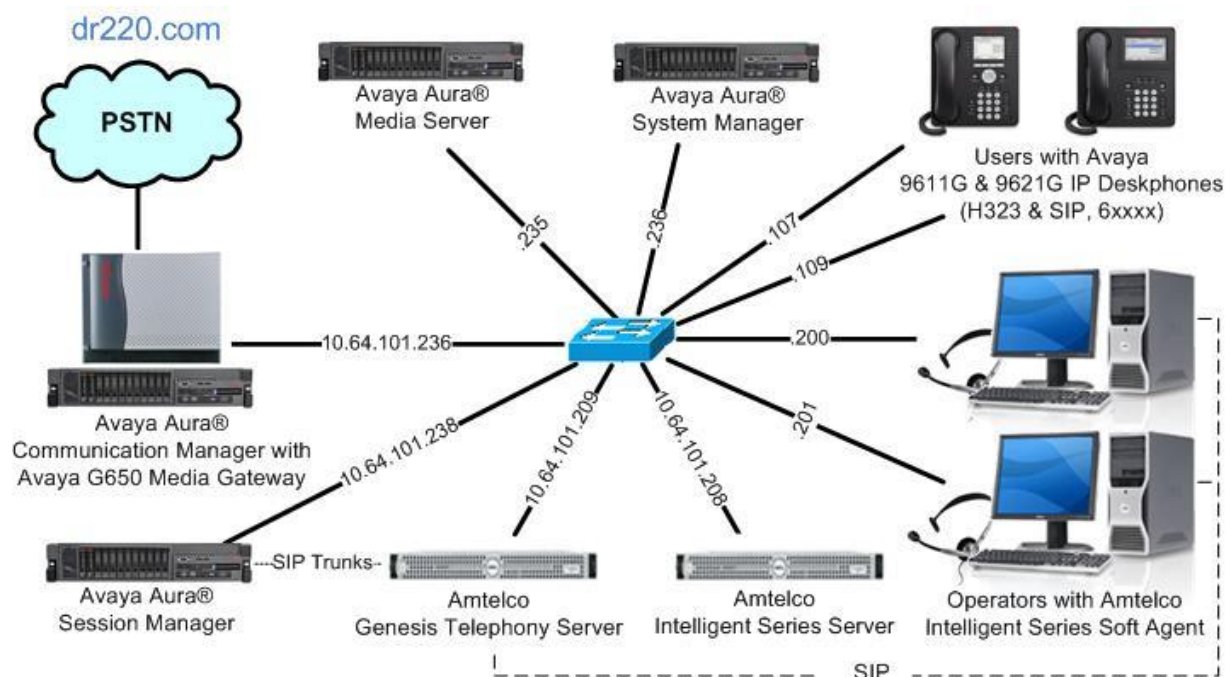
- **Phone:** (800) 553-7679
- **Email:** [service@amtelco.com](mailto:service@amtelco.com)
- **Web:** [www.amtelco.com/Welcome.htm](http://www.amtelco.com/Welcome.htm)

### 3. Reference Configuration

As shown in **Figure 1**, operators have desktops running the Intelligent Series Soft Agent application, and dedicated SIP connections with the Genesis Telephony Server as part of log in. The Intelligent Series Supervisor was running on the supervisor desktop.

SIP trunks were used between the Genesis Telephony Server and Session Manager. A five digit Uniform Dial Plan was used to facilitate dialing with Genesis. Calls to extensions 52xxx were routed over the SIP trunks to Genesis. In particular, internal users on Communication Manager will dial 52222 to reach Genesis, and calls from external users will be routed with digits 52000 to Genesis.

The detailed administration of connectivity between Communication Manager and Session Manager are not the focus of these Application Notes and will not be described.



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in Virtual Environment	7.0.1.1 (7.0.1.1.0.441.23169)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	7.7.0.334
Avaya Aura® Session Manager in Virtual Environment	7.0.1.1 (7.0.1.1.70114)
Avaya Aura® System Manager in Virtual Environment	7.0.1.1 (7.0.1.1.065378)
Avaya 9611GIP Deskphones (H.323)	6.6302
Avaya 9621G IP Deskphone (SIP)	7.0.1.2.9
Amtelco Genesis Telephony Server on Debian <ul style="list-style-type: none"><li>Asterisk</li></ul>	3.1.2 Build 1703241247 8.6 14.2.1
Amtelco Intelligent Series Server on Microsoft Windows Server 2008 R2 Enterprise <ul style="list-style-type: none"><li>Microsoft SQL Server 2014</li></ul>	5.0.6291.16498 SP1 12.0.2000.8
Amtelco Intelligent Series Supervisor on Microsoft Windows 10 Pro	5.0.6263.7
Amtelco Intelligent Series Soft Agent on Microsoft Windows 10 Pro	5.0.6263.09

## 5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- 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 uniform dial plan
- Administer AAR analysis
- Administer ISDN trunk group
- Administer tandem calling party number

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with Genesis.

### 5.1. Verify License

Log into the System Access Terminal (SAT) 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.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

display system-parameters customer-options		Page 2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	12000	10
Maximum Concurrently Registered IP Stations:	18000	4
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:	41000	0
Maximum Video Capable IP Softphones:	18000	0
<b>Maximum Administered SIP Trunks:</b>	<b>24000</b>	<b>30</b>
Maximum Administered Ad-hoc Video Conferencing Ports:	24000	0

## 5.2. Administer System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class of Restriction or Class of Service levels. Refer to [1] for more details.

```
change system-parameters features                               Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y
      Music/Tone on Hold: music Type: ext 40001
      Music (or Silence) on Transferred Trunk Calls? Call-wait
      DID/Tie/ISDN/SIP Intercept Treatment: attendant
      Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
      Automatic Circuit Assurance (ACA) Enabled? n
```

### 5.3. Administer SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “52”. 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 52		Page 1 of 21	
TRUNK GROUP			
Group Number: 52	Group Type: sip	CDR Reports: y	
Group Name: SIP Trunks to Genesis	COR: 1	TN: 1	TAC: 1052
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:		
	Number of Members: 0		

Navigate to **Page 3**, and enter “private” for **Numbering Format**.

add trunk-group 52		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UUI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
		Hold/Unhold Notifications? y	
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			



## 5.4. Administer SIP Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “52”. 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” in this case.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration with Genesis.
- **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 Genesis.
- **Far-end Domain:** The applicable domain name for the network.

For **Direct IP-IP Audio Connections**, enter “n” since Genesis does not support media shuffling.

add signaling-group 52		Page 1 of 2
SIGNALING GROUP		
Group Number: 52	<b>Group Type: sip</b>	
IMS Enabled? n	<b>Transport Method: tls</b>	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: Others	
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y		
<b>Near-end Node Name: procr</b>	<b>Far-end Node Name: sm7-sig</b>	
<b>Near-end Listen Port: 5052</b>	<b>Far-end Listen Port: 5052</b>	
	<b>Far-end Network Region: 5</b>	
<b>Far-end Domain: dr220.com</b>	Far-end Secondary Node Name:	
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	<b>Direct IP-IP Audio Connections? n</b>	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
	Alternate Route Timer(sec): 6	

## 5.5. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.3**. 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 5.4**.
- **Number of Members:** The desired number of members, in this case “10”.

```
change trunk-group 52                                     Page 1 of 21

                                TRUNK GROUP

Group Number: 52                      Group Type: sip          CDR Reports: y
  Group Name: SIP Trunks to Genesis    COR: 1                TN: 1          TAC: 1052
  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: 52
                                      Number of Members: 10
```

## 5.6. 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 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “no” 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 Genesis.

change ip-network-region 5		Page 1 of 20	
IP NETWORK REGION			
Region: 5			
Location:		Authoritative Domain: dr220.com	
Name: Genesis		Stub Network Region: n	
MEDIA PARAMETERS		Intra-region IP-IP Direct Audio: no	
Codec Set: 5		Inter-region IP-IP Direct Audio: no	
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			

Navigate to **Page 4**, and specify this codec set to be used for calls with the network region used by the Avaya endpoints and with the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and trunk to the PSTN.

change ip-network-region 5										Page 4 of 20			
Source Region: 5 Inter Network Region Connection Management										I	M		
										G	A	t	
dst codec direct	WAN-BW-limits			Video		Intervening			Dyn	A	G	c	
rgn set	WAN	Units	Total	Norm	Prio	Shr	Regions			CAC	R	L	e
1	5	y	NoLimit								n	t	
2													
3													
4													
5	5										all		
6													

## 5.7. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Genesis supports the G.711 and G.729 codec variants, with G.729 requiring special license on Genesis. The compliance testing only covered the G.711 codec.

change ip-codec-set 5				Page	1 of	2
IP Codec Set						
Codec Set: 5						
Audio		Silence	Frames	Packet		
Codec		Suppression	Per Pkt	Size (ms)		
1:	<b>G.711MU</b>	n	2	20		
2:						
3:						
4:						
5:						

## 5.8. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an available route pattern number to be used to reach Genesis, in this case “52”. 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 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

change route-pattern 52												Page	1 of	3						
Pattern Number: 52												Pattern Name: Genesis								
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:	52	0										n	user							
2:											n	user								
3:											n	user								
4:											n	user								
5:											n	user								
6:											n	user								
BCC VALUE												TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
0		1	2	M	4	W	Request							Dgts	Format					
												Subaddress								
1:	y	y	y	y	y	n	n	rest					none							

## 5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to Genesis. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 52 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	6	52		5	Total Administered: 1
					Maximum Entries: 540

## 5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 52xxx to Genesis. Note that other routing methods may be used. Use the “change uniform-dialplan 0” command, and add an entry to specify the use of AAR for routing of digits 52xxx, as shown below.

change uniform-dialplan 0					Page 1 of 2
UNIFORM DIAL PLAN TABLE					
					Percent Full: 0
Matching			Insert	Node	
Pattern	Len	Del	Digits	Net Conv	Num
52	5	0		aar	n

## 5.11. Administer AAR Analysis

Use the “change aar analysis 0” command, and add an entry to specify how to route calls to 52xxx. In the example shown below, calls with digits 52xxx will be routed as an AAR call using route pattern “52” from **Section 5.8**.

change aar analysis 0					Page 1 of 2
AAR DIGIT ANALYSIS TABLE					
Location: all					Percent Full: 2
	Dialed	Total	Route	Call	Node
	String	Min	Max	Type	Num
52		5	5	52	unku
					ANI
					Reqd
					n

## 5.12. Administer ISDN Trunk Group

Use the “change trunk-group n” command, where “n” is the existing ISDN trunk group number used to reach the PSTN if applicable, in this case “13”.

Navigate to **Page 3**. For **Modify Tandem Calling Number**, enter “tandem-cpn-form” to allow calling party numbers from Genesis to be modified.

change trunk-group 13			Page	3	of	21
TRUNK FEATURES						
ACA Assignment? n		Measured: none	Wideband Support? n			
			Maintenance Tests? y			
		Data Restriction? n	NCA-TSC Trunk Member:			
		Send Name: y	Send Calling Number: y			
Used for DCS? n			Send EMU Visitor CPN? n			
Suppress # Outpulsing? n		Format: natl-pub				
Outgoing Channel ID Encoding: preferred		UII IE Treatment: service-provider				
				Replace Restricted Numbers? n		
				Replace Unavailable Numbers? n		
				Send Connected Number: y		
Network Call Redirection: none				Hold/Unhold Notifications? n		
Send UII IE? y		<b>Modify Tandem Calling Number: tandem-cpn-form</b>				
Send UCID? n						
Send Codeset 6/7 LAI IE? y		Dsl Echo Cancellation? n				
Apply Local Ringback? n		US NI Delayed Calling Name Update? n				
Show ANSWERED BY on Display? Y		Invoke ID for USNI Calling Name: variable				
		Network (Japan) Needs Connect Before Disconnect? n				

## 5.13. Administer Tandem Calling Party Number

Use the “change tandem-calling-party-num” command, to define the calling party number to send to the PSTN for tandem calls from Genesis.

In the example shown below, all calls originating from a 5-digit extension beginning with 52 and routed to trunk group 13 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case “pub-unk”.

change tandem-calling-party-num					Page	1 of	8
CALLING PARTY NUMBER CONVERSION							
FOR TANDEM CALLS							
CPN		Trk		Number			
Len	Prefix	Grp(s)	Delete	Insert	Format		
5	6	13		30353	pub-unk		
<b>5</b>	<b>52</b>	<b>13</b>		<b>30353</b>	<b>pub-unk</b>		

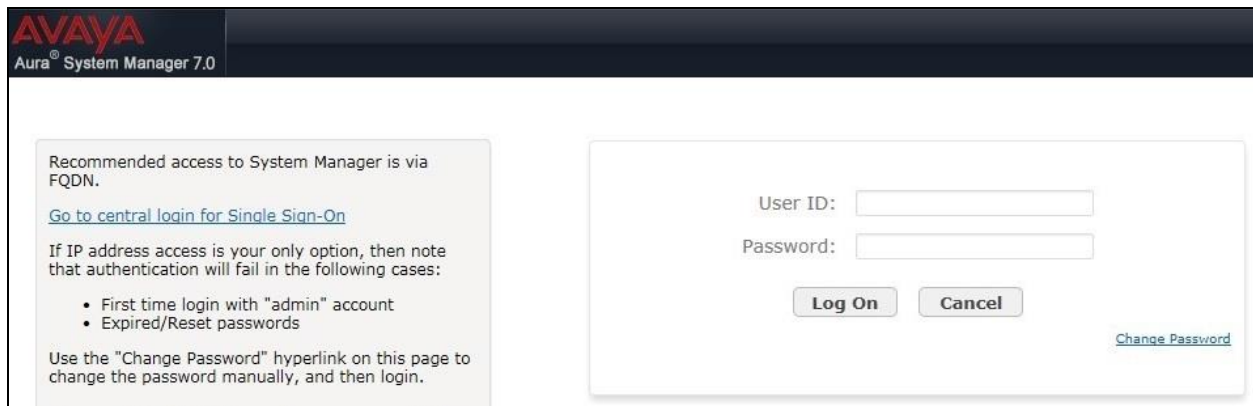
## 6. Configure Avaya Aura® Session Manager

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

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

### 6.1. Launch System Manager

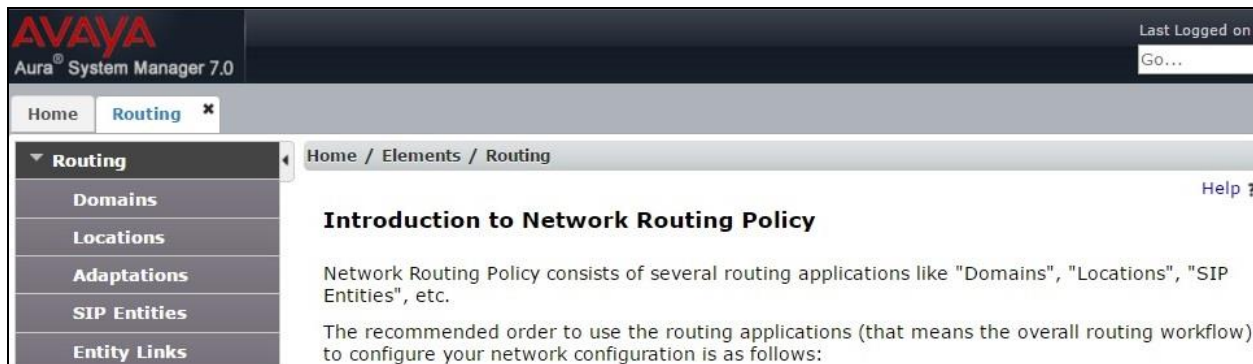
Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



The screenshot shows the Avaya Aura System Manager 7.0 login interface. The header includes the Avaya logo and 'Aura® System Manager 7.0'. The main content area is divided into two sections. The left section contains a message: 'Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On'. Below this, it states: 'If IP address access is your only option, then note that authentication will fail in the following cases:'. A bulleted list follows: '• First time login with "admin" account' and '• Expired/Reset passwords'. It concludes with: 'Use the "Change Password" hyperlink on this page to change the password manually, and then login.' The right section is a login form with fields for 'User ID:' and 'Password:', 'Log On' and 'Cancel' buttons, and a 'Change Password' link.

### 6.2. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Genesis.



The screenshot shows the Avaya Aura System Manager 7.0 interface after navigating to the Routing section. The header includes the Avaya logo, 'Aura® System Manager 7.0', and a 'Last Logged on' field with a 'Go...' button. The main navigation pane on the left shows 'Home' and 'Routing' (selected). Under 'Routing', there are links for 'Domains', 'Locations', 'Adaptations', 'SIP Entities', and 'Entity Links'. The main content area displays the 'Introduction to Network Routing Policy' screen. It includes a breadcrumb trail: 'Home / Elements / Routing'. The title is 'Introduction to Network Routing Policy'. The text states: 'Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.' and 'The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:'. A 'Help ?' link is visible in the top right corner.

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.

AVAYA  
Aura® System Manager 7.0

Last Logged  
Go...

Home Routing x

Home / Elements / Routing / Locations

### Location Details

Commit

#### General

\* Name: Genesis-Loc

Notes: Amtelco Genesis

#### Dial Plan Transparency in Survivable Mode

Enabled: ☐

Listed Directory Number:

Associated CM SIP Entity:

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of the Genesis Telephony Server in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

#### Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

\* Latency before Overall Alarm Trigger: 5 Minutes

\* Latency before Multimedia Alarm Trigger: 5 Minutes

#### Location Pattern

Add Remove

1 Item Filter: Enable

	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.64.101.209	Amtelco Genesis

Select : All, None

Commit Cancel



## 6.3. Administer SIP Entities

Add two new SIP entities, one for Genesis and one for the new SIP trunks with Communication Manager.

### 6.3.1. SIP Entity for Genesis

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

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 the Genesis Telephony Server.
- **Type:** “SIP Trunk”
- **Notes:** Any desired notes.
- **Location:** Select the Genesis location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at April 3, 2017 10:10 AM  
Go...

Home Routing

Home / Elements / Routing / SIP Entities

### SIP Entity Details

Commit Cancel

#### General

\* Name:

\* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

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

Credential name:

Securable: ☐

Call Detail Recording:

#### Loop Detection

Loop Detection Mode:

Loop Count Threshold:

Loop Detection Interval (in msec):

#### SIP Link Monitoring

SIP Link Monitoring:

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, in this case “DR-SM7”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The Genesis entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that Genesis can support UDP and TCP, and the compliance testing used the UDP protocol.

### Entity Links

Override Port & Transport with DNS ☐ SRV: ☐

Add Remove

1 Item
Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* SM7-Genesis	DR-SM7	UDP	* 5060	Genesis	* 5060	trusted

Select : All, None

### SIP Responses to an OPTIONS Request

Add Remove

0 Items
Filter: Enable

	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--	-------------------------------	---------------------	-------

Commit Cancel

### 6.3.2. 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 Genesis.

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”
- **Notes:** Any desired notes.
- **Adaptation:** Select the applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at April  
GO...

Home Routing

Home / Elements / Routing / SIP Entities

### SIP Entity Details

Commit Cancel

#### General

\* Name: DR-CM7-5052

\* FQDN or IP Address: 10.64.101.236

Type: CM

Notes: CM7 Port 5052 (Amtelco Genesis)

Adaptation: DR-CM7-Adaptation

Location: DR-Loc

Time Zone: America/New\_York

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

Credential name:

Securable: ☐

Call Detail Recording: none

#### Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

#### SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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, in this case “DR-SM7”.
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group far-end listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group near-end listen port number from **Section 5.4**.
- **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"/>	* SM7-CM7-5052	DR-SM7 ▼	TLS ▼	* 5052	DR-CM7-5052 ▼	* 5052	trusted ▼

Select : All, None

### SIP Responses to an OPTIONS Request

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

## 6.4. Administer Routing Policies

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

### 6.4.1. Routing Policy for Genesis

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

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the Genesis entity name from **Section 6.3.1**. The screen below shows the result of the selection.

AVAYA  
Aura® System Manager 7.0

Last Logged on  
Go...

Home Routing

Home / Elements / Routing / Routing Policies

### Routing Policy Details

Commit

#### General

\* Name: To-Genesis

Disabled: ☐

\* Retries: 0

Notes: Amtelco Genesis

#### SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Genesis	10.64.101.209	SIP Trunk	Amtelco Genesis

## 6.4.2. 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**. Enter optional **Notes**, 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 6.3.2**. The screen below shows the result of the selection.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on  
Go...

Home Routing

Home / Elements / Routing / Routing Policies

### Routing Policy Details

Commit

#### General

\* Name: To-CM7-5052

Disabled: ☐

\* Retries: 0

Notes: To CM7 from Amtelco Genesis

#### SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
DR-CM7-5052	10.64.101.236	CM	CM7 Port 5052 (Amtelco Genesis)

## 6.5. Administer Dial Patterns

Add a new dial pattern for Genesis, and update existing dial patterns for Communication Manager.

### 6.5.1. Dial Pattern for Genesis

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 Genesis. 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 “52”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select the applicable domain, in this case “dr220.com”.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Genesis. In the compliance testing, the entry allowed for call originations from Communication Manager endpoints in locations “DR-Loc” and “NJ-Loc”. The Genesis routing policy from **Section 6.4.1** was selected as shown below.

**AVAYA**  
Aura® System Manager 7.0

Home / Elements / Routing / Dial Patterns

### Dial Pattern Details

**General**

\* Pattern: 52

\* Min: 5

\* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: dr220.com ▼

Notes: Amtelco Genesis

**Originating Locations and Routing Policies**

Add Remove

2 Items

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Destination
<input type="checkbox"/>	DR-Loc	TLT DR Network	To-Genesis	0	<input type="checkbox"/>	Genesis
<input type="checkbox"/>	NJ-Loc	TLT NJ Network	To-Genesis	0	<input type="checkbox"/>	Genesis



## 6.5.2. Dial Pattern for Communication Manager

Select **Routing** → **Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern “6” (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from Genesis. In the compliance testing, the new policy allowed for call origination from the Genesis location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.4.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, operators on Genesis manually added the prefix “9” for outbound calls to the PSTN, and therefore the existing dial pattern for “9” was also changed (not shown below).

**AVAYA**  
Aura® System Manager 7.0

Home / Elements / Routing / Dial Patterns

### Dial Pattern Details

Commit Cancel

**General**

\* Pattern: 9

\* Min: 11

\* Max: 11

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes: To CM7 PSTN

**Originating Locations and Routing Policies**

Add Remove

3 Items

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination
<input type="checkbox"/>	DR-Loc	TLT DR Network	To-CM7	0	<input type="checkbox"/>	DR-CM7
<input type="checkbox"/>	Genesis-Loc	Amtelco Genesis	To-CM7-5052	0	<input type="checkbox"/>	DR-CM7-5052
<input type="checkbox"/>	NJ-Loc	TLT NJ Network	To-CM7	0	<input type="checkbox"/>	DR-CM7

Select : All, None



## 7. Configure Amtelco Genesis Intelligent Series

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

- Launch web interface
- Obtain application name
- Administer trunks
- Administer routes
- Administer agents
- Administer access control lists
- Launch Intelligent Series Supervisor
- Administer IS system
- Administer IS client
- Administer IS agent
- Restart IS service
- Launch Intelligent Series Soft Agent
- Administer setup

The configuration of Genesis is typically performed by Amtelco technicians. The procedural steps are presented in these Application Notes for informational purposes.

### 7.1. Launch Web Interface

From a PC, launch an Internet browser window and access the Genesis web-based interface by using the URL “http://<ip-address:5080>/Admin/Application/Index”, where “ip-address” is the IP address of the Genesis Telephony Server.

### 7.2. Obtain Application Name

The **Applications** screen below is displayed in the right pane. Make a note of the application **Name**, in this case “IS”, which is created as part of installation. The name will be used in later sections.

The screenshot shows the Genesis web interface. At the top, there's a header with the word "Genesis". Below it, there's a navigation bar with tabs: "Administration", "Diagnostics", "Licenses", and "About". The "Administration" tab is selected. On the left side, there's a sidebar menu with items: "Applications", "Agents", "Emergency Agents", "SIP Options", "Trunks", "Routes", "Call Types", "Class Of Service", and "Music On Hold". The "Applications" item is selected. The main content area is titled "Applications". It contains a "Create New" button and a table with two columns: "Name" and "Description". The "Name" column header is circled in red. The table has one row with the value "IS" in the "Name" column and "Intelligent Series Server" in the "Description" column. Below the table, there's a pagination bar that says "Page 1 of 1" and "First Previous Next Last".

## 7.3. Administer Trunks

Select **Trunks** in the left pane, followed by **Create New SIP Trunk** (not shown) in the updated right pane, to display the **Trunk Information** screen below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **Application:** Select the application name from **Section 7.2**.
- **Maximum Channels:** Enter the number of trunk members from **Section 5.5**.
- **Extension:** The routing extension digits from **Section 3** for calls from PSTN.
- **Host:** IP address of the Session Manager signaling interface.
- **Port:** The Genesis SIP entity port number from **Section 6.3.1**.
- **UserName:** The routing extension digits from **Section 3** for calls from PSTN.
- **Destination IP:** IP address of the Session Manager signaling interface.

# Genesis

**Administration** | Diagnostics | Licenses | About

Applications  
Agents  
Emergency Agents  
SIP Options  
Trunks  
Routes  
Call Types  
Class Of Service  
Music On Hold

## Trunk Information

**Name** SM Trunks

**Application** IS

**Maximum Channels** 10

## SIP Service Provider Settings

**Extension** 52000

**Direction** In/Out

**Host** 10.64.101.238

**Port** 5060

**Register** ☐

**UserName** 52000

**Secret**

**DtmfMode** RFC2833

**Nat** ☐

**Qualify** ☐

**CustomSettings**

## Transfer

**Destination IP** 10.64.101.238

**Hangup After Blind Transfer** ☐

**Hangup After Blind Transfer Delay (Seconds)** 0

Save Cancel

## 7.4. Administer Routes

Select **Routes** in the left pane, followed by **Create New Route** (not shown) in the updated right pane, to display the **Route Information** screen below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Number:** An available route number.
- **Name:** A descriptive name.

In the **Route Trunks** sub-section, select the trunk from **Section 7.3** under **Available** and move to **Selected**, as shown below.

The screenshot displays the 'Genesis' application interface. At the top, there's a navigation bar with tabs: 'Administration' (selected), 'Diagnostics', 'Licenses', and 'About'. On the left side, a vertical menu lists various system components: 'Applications', 'Agents', 'Emergency Agents', 'SIP Options', 'Trunks', 'Routes' (highlighted), 'Call Types', 'Class Of Service', and 'Music On Hold'. The main content area is titled 'Route Information' and contains several input fields: 'Number' with the value '0', 'Name' with the value 'SM Route', and a 'Hunt' checkbox which is unchecked. Below these fields is the 'Route Trunks' section, which is divided into two panes: 'Available' and 'Selected'. The 'Available' pane is currently empty. The 'Selected' pane contains a single entry, 'SM Trunks'. Between the two panes are two buttons with right and left arrows for moving items. At the bottom of the 'Route Trunks' section, there are 'Save' and 'Cancel' buttons.

## 7.5. Administer Agents

Select **Agents** in the left pane, to display the **Agents** screen. One agent is needed for each operator user, and by default the first agent is automatically created, as shown below. To create additional agents, select **Create New**.

The screenshot shows the Genesis web interface. The 'Administration' tab is selected in the top navigation bar. In the left sidebar, 'Agents' is selected. The main content area is titled 'Agents' and contains two buttons: 'Create New' and 'Modify Range'. Below these buttons is a table with the header 'Application Agent Number'. The table contains one row with the value '1' in the 'Agent Number' column. Below the table, there are links for 'Edit' and 'Delete'. At the bottom right, it says 'Page 1 of 1' and 'First Previous Next Last'.

The **Create a new agent** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Agent Number:** An available agent number.
- **Password:** A desired password.
- **Application:** Select the application name from **Section 7.2**.
- **Transport:** “udp”

The screenshot shows the 'Create a new agent' screen in the Genesis web interface. The 'Administration' tab is selected. In the left sidebar, 'Agents' is selected. The main content area is titled 'Create a new agent'. It contains the following fields: 'Agent Number' (text input with value '2'), 'Password' (password input with masked characters '....'), 'Application' (dropdown menu with value 'IS'), 'Custom Settings' (text area), and 'Transport' (dropdown menu with value 'udp'). Below these fields is a section titled 'Access Control Lists'. It contains two columns: 'Available' and 'Selected'. Each column has a list box and a button to move items between them.

## 7.6. Administer Access Control Lists

Select **SIP Options** in the left pane, followed by **Access Control Lists** in the updated right pane, to display the screen below. Make certain **SIP Type** is set to “SIP”, as shown below.

Select **Access Control Lists**.

The screenshot shows the Genesis Administration interface. The left sidebar contains a list of navigation items: Applications, Agents, Emergency Agents, SIP Options, Trunks, Routes, Call Types, Class Of Service, and Music On Hold. The main content area is titled 'SIP Settings' and includes a list of links: General, Access Control Lists (highlighted with a red box), PJSIP Settings, Address of Record List, Authentication Records, Domain Aliases, Global, Registrations, System, and Transports. Below this, the 'Active SIP Type' section shows a dropdown menu set to 'SIP' (highlighted with a red box) and a note: 'Changing type requires a restart'. There are 'Save' and 'Cancel' buttons at the bottom.

The **Access Control List Information** screen is displayed. Enter a desired **Name**, and create a **permit** entry for each network subnet from **Section 3**, and create a generic **deny** entry as shown below.

The screenshot shows the 'Access Control List Information' screen in the Genesis Administration interface. The left sidebar is the same as in the previous screenshot. The main content area has a title 'Access Control List Information'. It includes a 'Name' field with the value 'Primary'. Below this is a 'Custom Settings' text area containing the following text: permit=192.168.200.0/24, permit=10.64.101.0/24, permit=10.64.125.0/24, and deny=0.0.0.0/0.0.0.0. There are 'Save' and 'Cancel' buttons at the bottom.

## 7.7. Launch Intelligent Series Supervisor

From the supervisor PC, double-click on the Intelligent Series Supervisor shortcut icon shown below, which was created as part of Intelligent Series Supervisor installation.

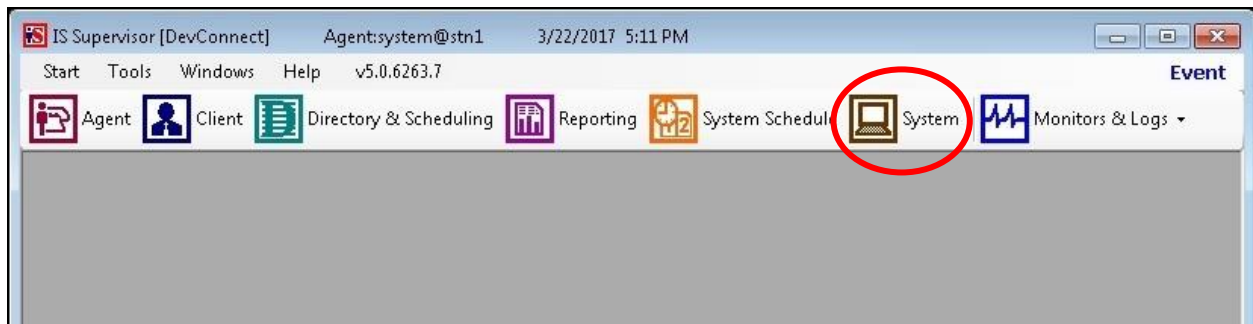


The **Supervisor Login** screen is displayed. Log in using the appropriate credentials.



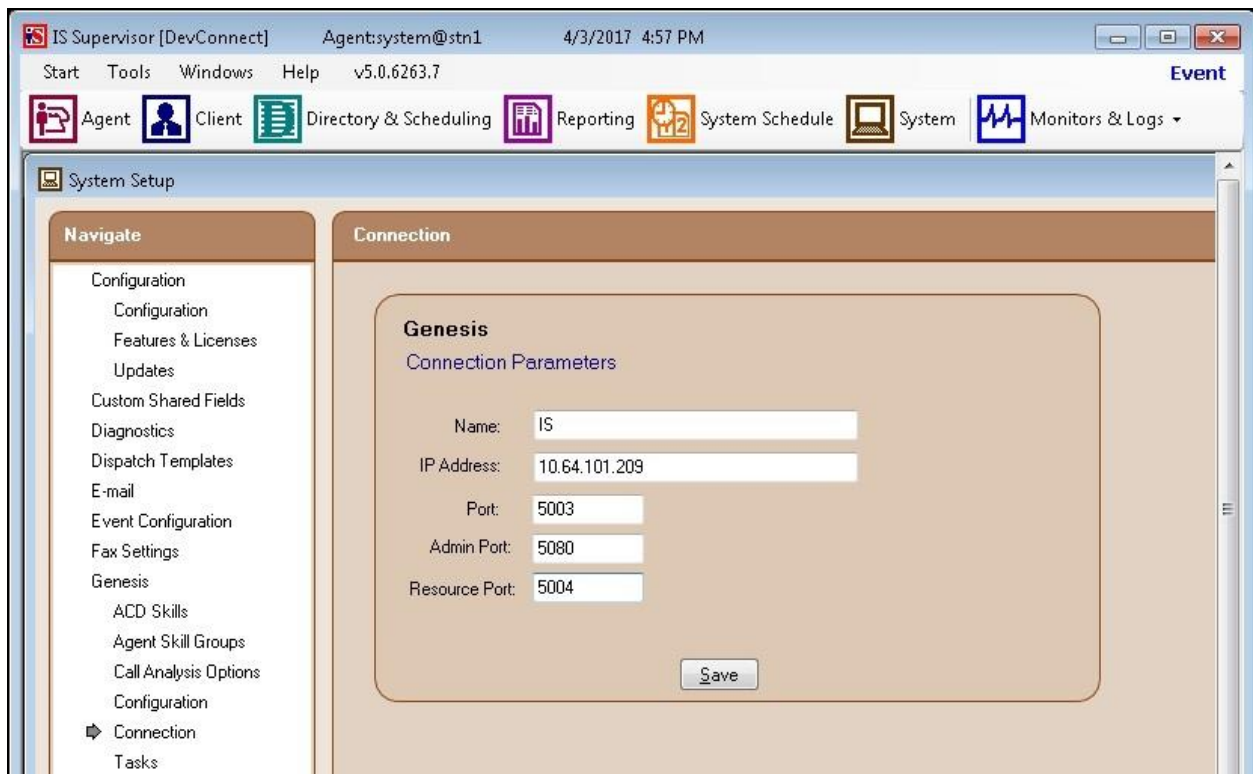
## 7.8. Administer IS System

The **IS Supervisor** screen is displayed. Select **System** from the top of the screen.



The screen is updated with **System Setup** displayed in the lower pane. Select **Genesis** → **Connection** from the left pane, to display the **Connection** screen in the right pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** Enter the application name from **Section 7.2**.
- **IP Address:** IP address of the Genesis Telephony Server.
- **Port:** “5003”
- **Admin Port:** “5080”
- **Resource Port:** “5004”



Select **Genesis** → **Telephony** from the left pane, to display the **Telephony** screen in the right pane. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Caller ID:** The desired calling party extension to use for outbound calls.
- **Caller Name:** The desired calling party name to use for outbound calls.

The screenshot shows the IS Supervisor [DevConnect] application window. The title bar includes the application name, user 'Agent:system@stn1', and date/time '4/3/2017 5:01 PM'. The menu bar has 'Start', 'Tools', 'Windows', and 'Help'. Below the menu bar is a toolbar with icons for Agent, Client, Directory & Scheduling, Reporting, System Schedule, System, and Monitors & Logs. The main window is titled 'System Setup' and is divided into two panes. The left pane, 'Navigate', contains a tree view with categories like Configuration, Updates, Custom Shared Fields, Diagnostics, Dispatch Templates, E-mail, Event Configuration, Fax Settings, Genesis, ACD Skills, Agent Skill Groups, Call Analysis Options, Configuration, Connection, Tasks, Telephony, Holiday, and IS Call Log. The right pane, 'Telephony', displays the 'Genesis' settings. The 'Telephony Settings' section includes the following fields: 'Auto Answer Repeat Interval' (0 seconds), 'Calls for ATTA' (0), 'Waits List Refresh Rate' (0 seconds (0 -100)), 'Caller ID' (52555), 'Caller Name' (Genesis), and 'Patch Time' (15 seconds). There is a checkbox for 'Hangup Patch After Patch Time Elapses' which is currently unchecked. A 'Save' button is located at the bottom of the settings section.

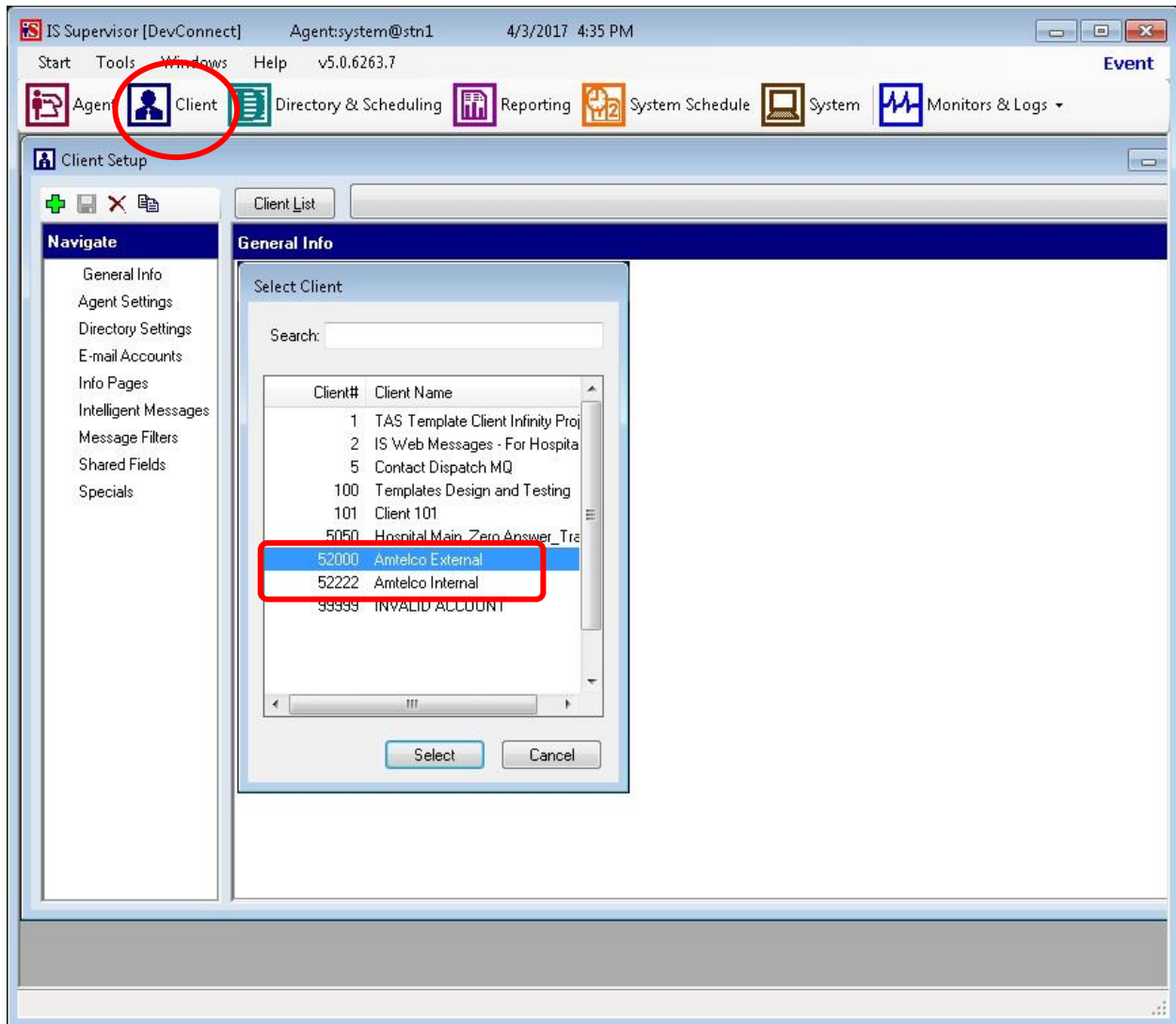
Field	Value	Unit/Range
Auto Answer Repeat Interval	0	seconds
Calls for ATTA	0	
Waits List Refresh Rate	0	seconds (0 -100)
Caller ID	52555	
Caller Name	Genesis	
Patch Time	15	seconds



## 7.9. Administer IS Client

Select **Client** from the top of the screen. The screen is updated with **Client Setup** displayed in the lower pane.

Follow reference [3] to create desired client entries to associate with called numbers for the customer network. In the compliance testing, calls from the PSTN will be routed with digits 52000 to Genesis, and calls from internal users on Communication Manager will be routed with digits 52222 to Genesis. Therefore two clients were created, as shown below.

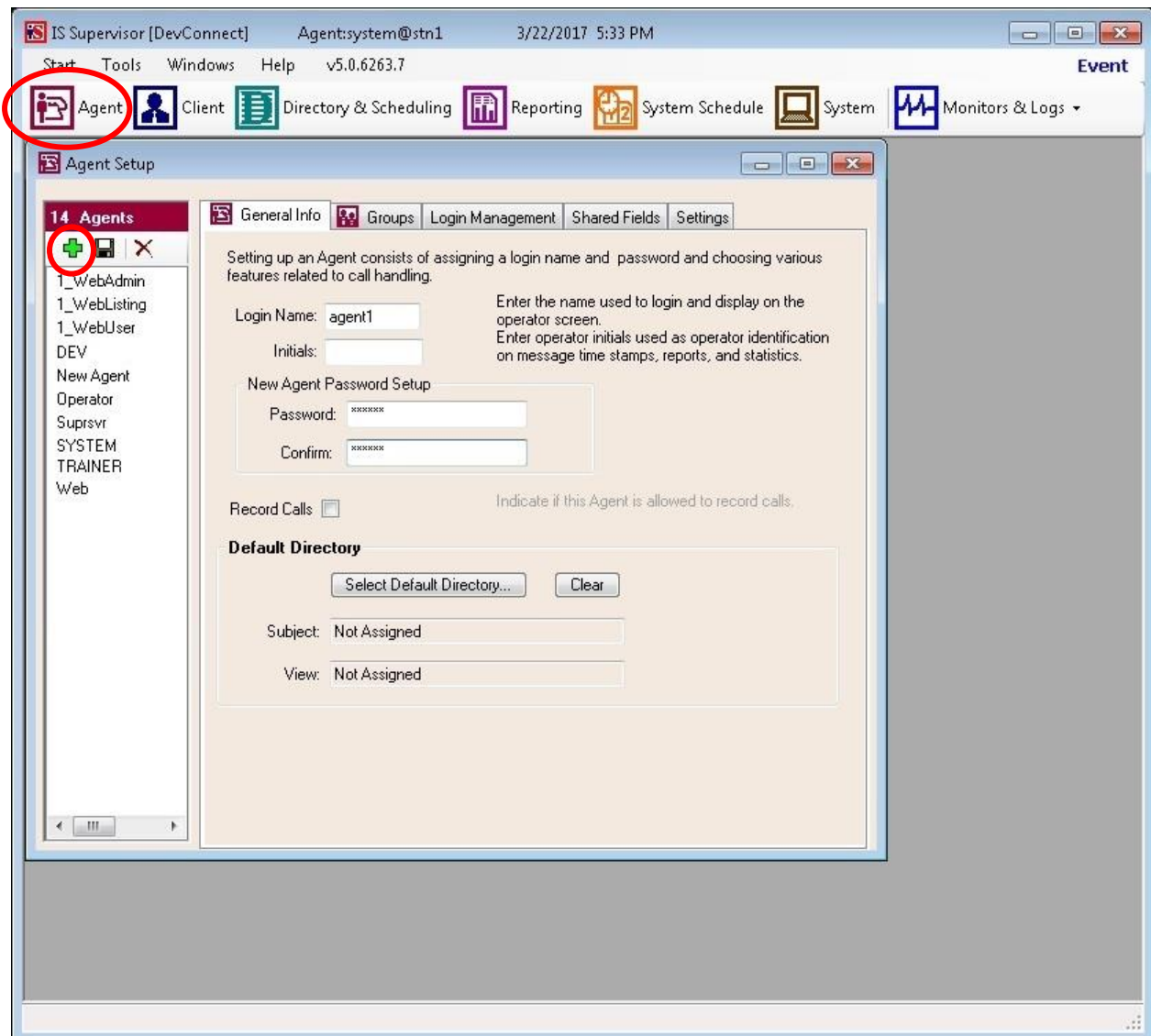


## 7.10. Administer IS Agent

Select **Agent** from the top of the screen. The screen is updated with **Agent Setup** displayed in the lower pane. Click on the **New Agent** icon in the left pane to create a new agent entry.

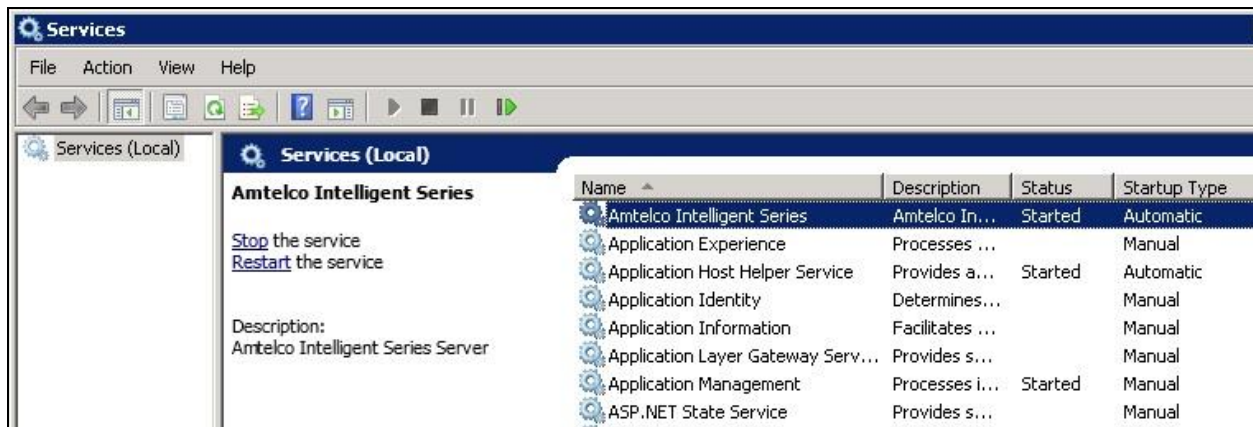
The **General Info** tab is displayed. For **Login Name**, **Password**, and **Confirm**, enter desired values. Retain the default values in the remaining fields.

One agent is needed for each operator user, and two agents were created in the compliance testing.



## 7.11. Restart IS Service

From the Intelligent Series Server, select **Start → Control Panel → Administrative Tools → Services** to display the **Services** screen. Locate and restart the **Amtelco Intelligent Series** service, as shown below.



## 7.12. Launch Intelligent Series Soft Agent

From an operator PC, double-click on the Soft Agent shortcut icon shown below, which was created as part of the Intelligent Series Soft Agent installation.



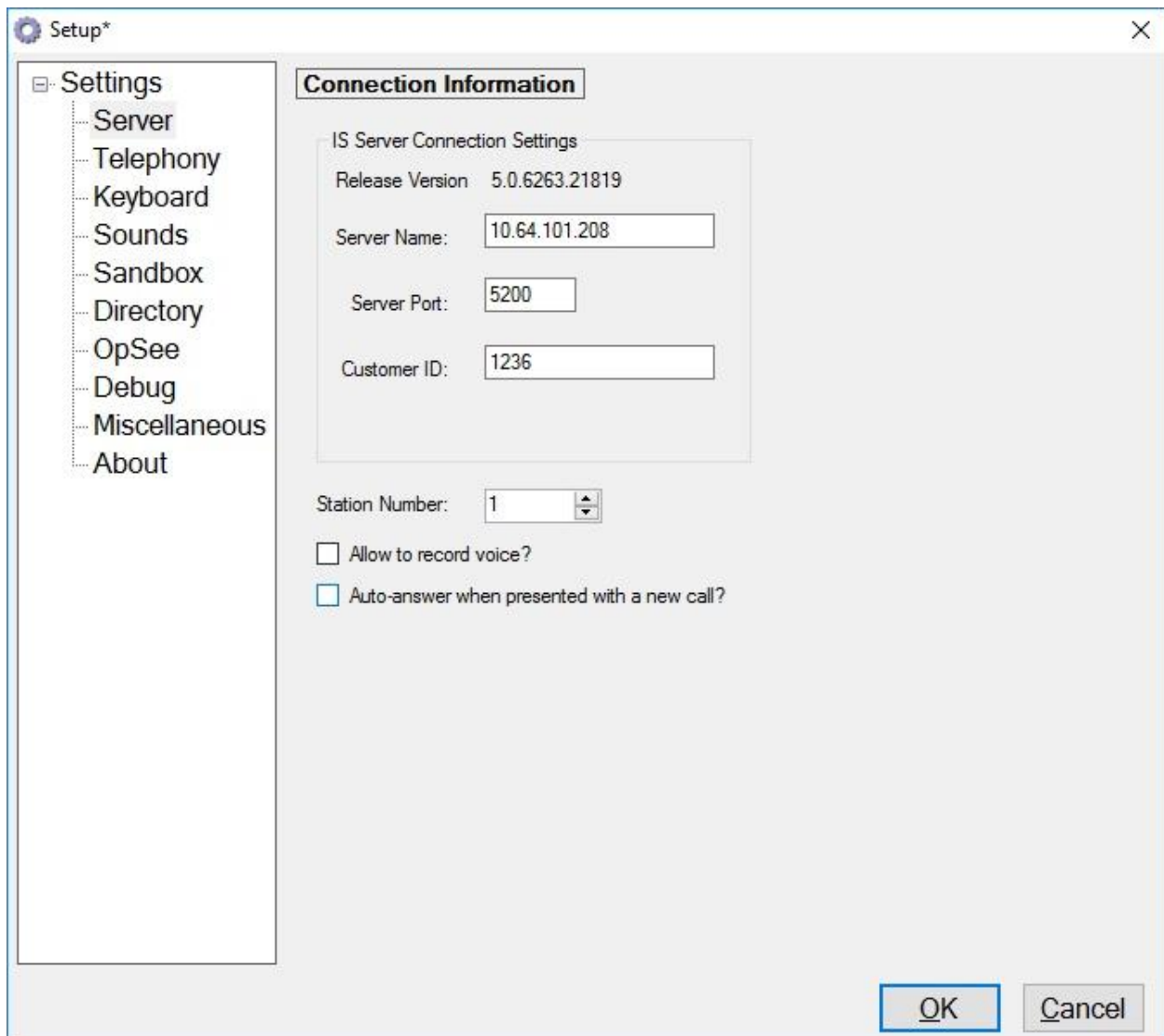
The **Soft Agent Login** screen is displayed. Press the **Ctrl** and **F12** keys together to enter setup.



## 7.13. Administer Setup

The **Setup** screen below is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Server Name:** IP address of the Intelligent Series Server.
- **Server Port:** “5200”
- **Customer ID:** The unique customer ID assigned by Amtelco, in this case “1236”.
- **Station Number:** An available station number, in this case “1”.



The screenshot shows a window titled "Setup\*" with a close button (X) in the top right corner. On the left is a tree view under "Settings" with the following items: Server, Telephony, Keyboard, Sounds, Sandbox, Directory, OpSee, Debug, Miscellaneous, and About. The "Server" item is selected. The main area is titled "Connection Information" and contains a sub-section "IS Server Connection Settings" with the following fields:

- Release Version: 5.0.6263.21819
- Server Name: 10.64.101.208
- Server Port: 5200
- Customer ID: 1236

Below these fields is a "Station Number" field with a spinner box set to "1". At the bottom of the main area are two checkboxes:

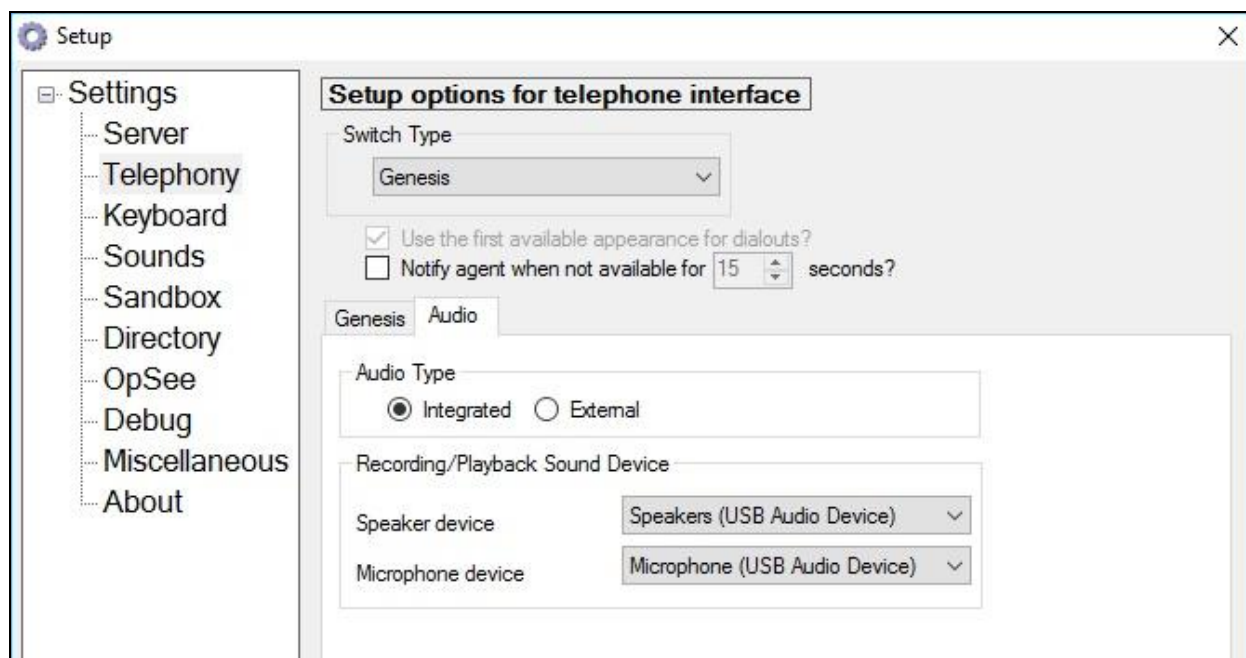
- ☐ Allow to record voice?
- ☐ Auto-answer when presented with a new call?

At the bottom right of the window are "OK" and "Cancel" buttons.

Select **Settings** → **Telephony** from the left pane, to display the screen below. For **Switch Type**, select “Genesis”. Select the desired **Number of appearances**, and enter “5060” for **Port**.

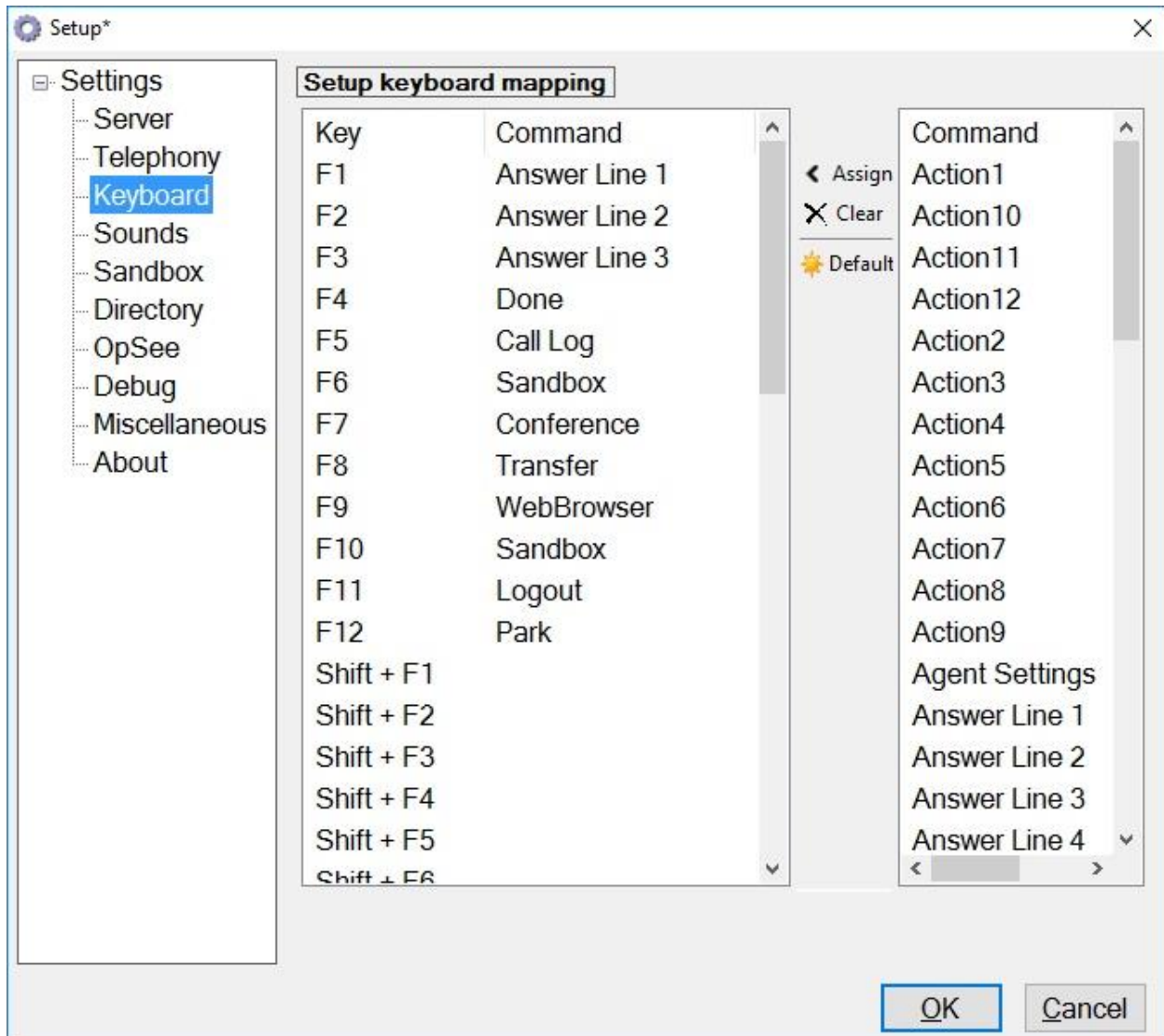


Select the **Audio** tab in the right pane, to display the screen below. For **Audio Type**, select **Integrated**. For **Speaker device** and **Microphone device**, select the applicable devices, as shown below.



Select **Settings** → **Keyboard** from the left pane, to display the screen below. Follow reference [3] to set the desired keyboard mapping for the agent. The setting used in the compliance testing is shown below.

Repeat **Section 7.12** and **Section 7.13** for each operator in **Section 3**. In the compliance testing, two operators were configured.



## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Genesis.

### 8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the “in-service/idle” state as shown below.

```
status trunk 52
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0052/001	T00146	in-service/idle	no
0052/002	T00147	in-service/idle	no
0052/003	T00148	in-service/idle	no
0052/004	T00149	in-service/idle	no
0052/005	T00150	in-service/idle	no
0052/006	T00151	in-service/idle	no
0052/007	T00152	in-service/idle	no
0052/008	T00153	in-service/idle	no
0052/009	T00154	in-service/idle	no
0052/010	T00155	in-service/idle	no

Verify the status of the SIP signaling groups by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is “in-service”, as shown below.

```
status signaling-group 52
```

STATUS SIGNALING GROUP	
Group ID:	52
Group Type:	sip
<b>Group State:</b>	<b>in-service</b>



## 8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements** → **Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the Genesis entity name from **Section 6.3.1**.

**AVAYA**  
Aura® System Manager 7.0

Last Log  
Go...

Home Session Manager

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

### 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

1 Items | Refresh

	Session Manager	Type	Monitored Entities				
			Down	Partially Up	Up	Not Monitored	Deny
<input type="checkbox"/>	<a href="#">DR-SM7</a>	Core	5	0	5	0	0
<input type="checkbox"/>							
<input type="checkbox"/>							
<input type="checkbox"/>							

Select: All, None

#### All Monitored SIP Entities

Run Monitor

10 Items (1 Selected) | Refresh

	SIP Entity Name
<input type="checkbox"/>	<a href="#">IPO2-IPOSE</a>
<input checked="" type="checkbox"/>	<a href="#">Genesis</a>



The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are “UP”, as shown below.

**AVAYA**  
Aura® System Manager 7.0

Last Logg  
Go...

Home Session Manager x

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

## 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.

All Entity Links to SIP Entity: Genesis

Status Details for the selected Session Manager:

Summary View

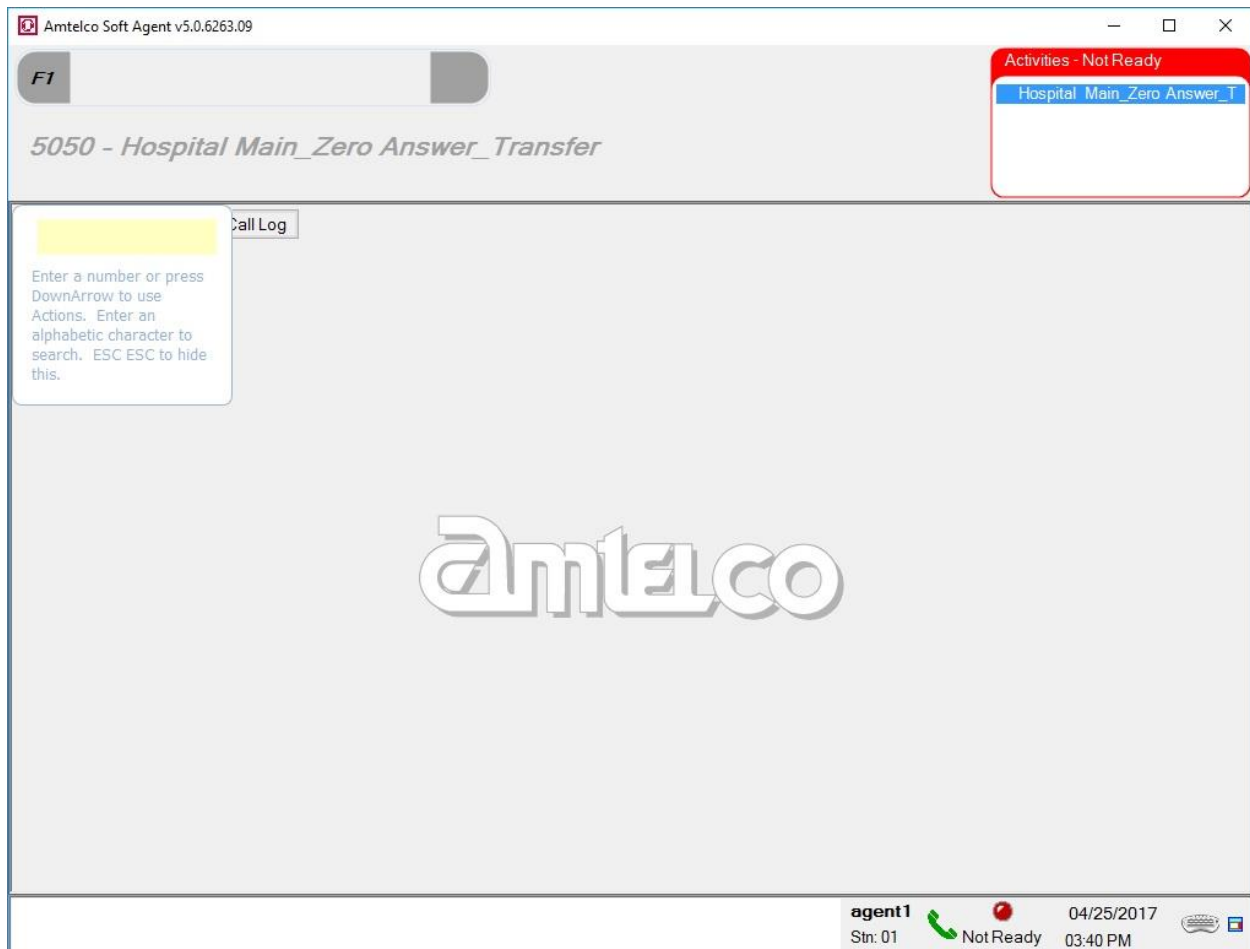
1 Items | Refresh

Session Manager	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code
<a href="#">DR-SM7</a>	10.64.101.20	5060	UDP	FALSE	UP	404 Not Found

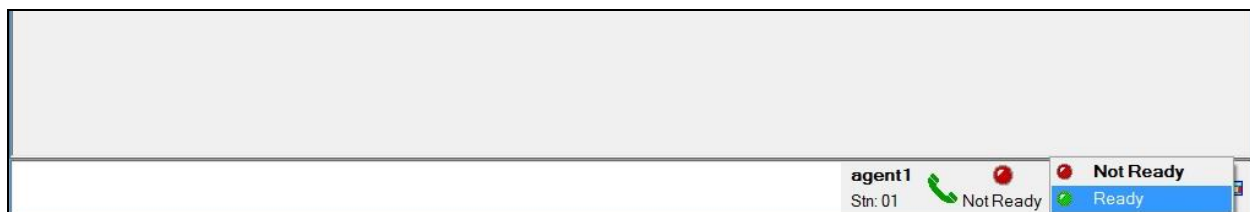
### 8.3. Verify Amtelco Genesis Intelligent Series

From the operator PC, follow the procedure in **Section 7.12** to launch the Intelligent Series Soft Agent and log in with the appropriate credentials from **Section 7.10**.

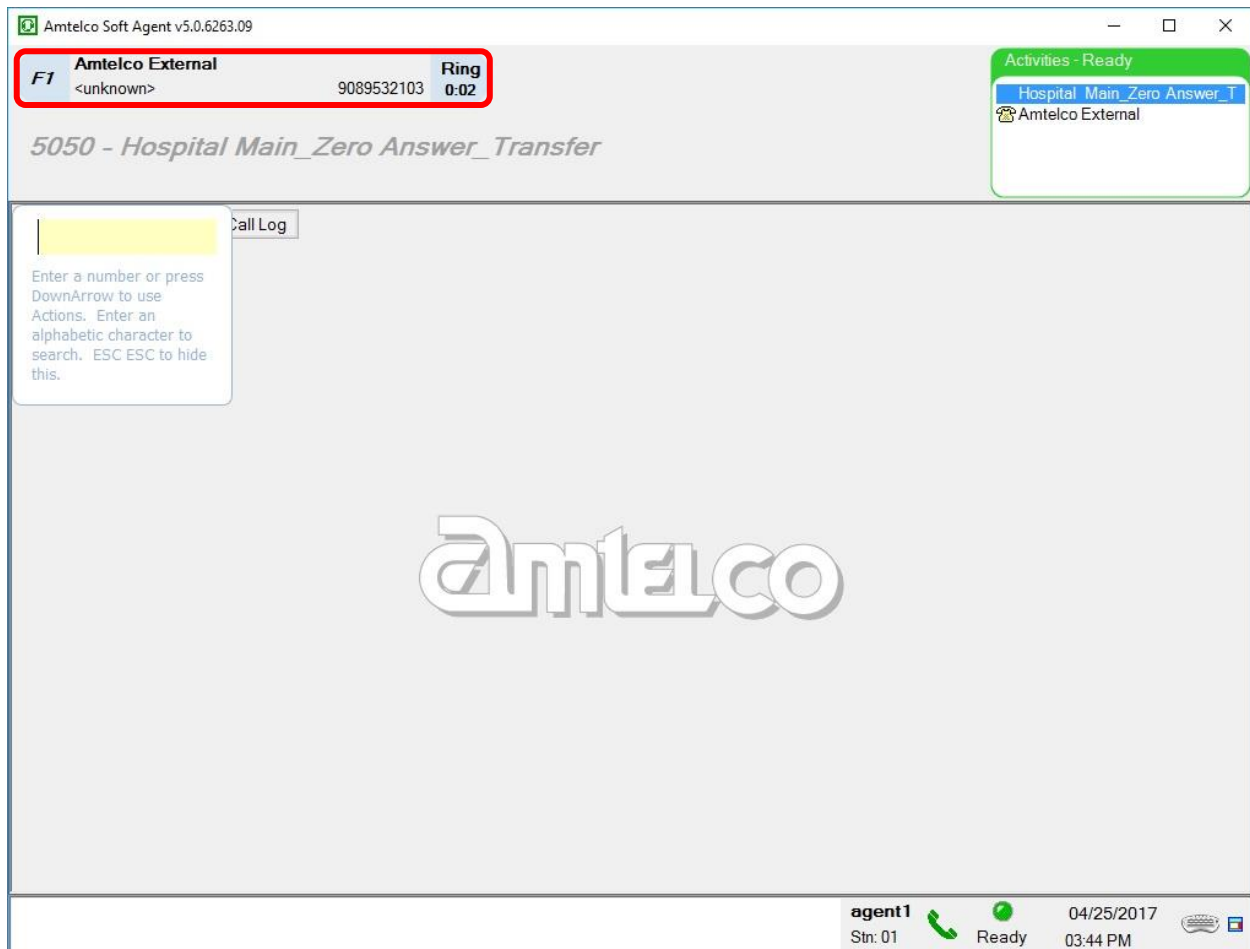
The **Amtelco Soft Agent** screen below is displayed.



In the lower right portion of the screen, right click on **Not Ready** and select **Ready**.



Make an incoming call from PSTN to reach Genesis. Verify that the call is ringing at the available operator, and that the operator screen is updated to reflect a ringing call along with the calling party number and the called client name, as shown below. In this case, the calling party number is **9089532103**, and the called client name is **Amtelco External**. Press the **F1** key or click in the applicable call line area highlighted below to answer the call.



Verify that the operator is connected to the PSTN with two-way talk paths. Also verify that the operator screen is updated to reflect the **Talk** state, as shown below.



## 9. Conclusion

These Application Notes describe the configuration steps required for Amtelco Genesis Intelligent Series to successfully interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

## 10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2.1, August 2016, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016, available at <http://support.avaya.com>.
3. *Administering Avaya Aura® System Manager for Release 7.0.1*, Release 7.0.1, Issue 4, April 2017, available at <http://support.avaya.com>.
4. *System Setup Supervisor Reference Guide*, March 2017, available at <https://service.amtelco.com/doclib/library.htm>.
5. *Soft Agent User Reference Guide*, May 2016, available at <https://service.amtelco.com/doclib/library.htm>.

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