



Application Notes for Configuring QuesCom 300 IP/GSM Gateway with Avaya SIP Enablement Services and Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration using a QuesCom 300 IP/GSM gateway, Avaya Communication Manager, and Avaya SIP Enablement Services. The QuesCom 300 IP/GSM is an IP-GSM gateway, supporting outgoing and incoming Global System for Mobile communications GSM calls. All GSM calls made from Avaya Communications Manager will be routed via the Avaya SIP Enablement Services (SES) server to the QuesCom 300 IP/GSM gateway out to the GSM network. The QuesCom 300 IP/GSM can also receive calls from the GSM network and route the calls back to Avaya Communication Manager via Avaya SIP Enablement services.

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 a compliance-tested configuration using a QuesCom 300 IP/GSM gateway, Avaya SIP Enablement Services (SES) 4.0 and Avaya Communication Manager 4.0.1.

The QuesCom 300 IP/GSM is an IP-GSM gateway, supporting outgoing and incoming GSM calls. All GSM calls made from Avaya Communications Manager will be routed via the Avaya SES to the QuesCom 300 IP/GSM gateway to the GSM network. The QuesCom 300 IP/GSM can also receive calls from the GSM network and route the calls back to Avaya Communication Manager via the Avaya SES. The QuesCom 300 IP/GSM can provide a backup route for the PSTN and also be backed up by the PSTN. This can be configured in Avaya Communication Manager using Automatic Route Selection (ARS). These Application Notes focus on a configuration where a SIP trunk connects Avaya SES and the QuesCom 300 IP/GSM.

Avaya Communication Manager runs on the Avaya S8500 Server; the solution described herein is also extensible to other Avaya Servers and Media Gateways. The Avaya G650 Media Gateway is connected to the PSTN via an E1 ISDN-PRI line. The Avaya SES server is networked with Avaya Communication Manager and the QuesCom 300 via SIP trunking. The QuesCom 300 IP/GSM in turn connects to the GSM network via Subscriber Identity Module (SIM) cards that reside on GSM boards inserted in the QuesCom 300 IP/GSM.

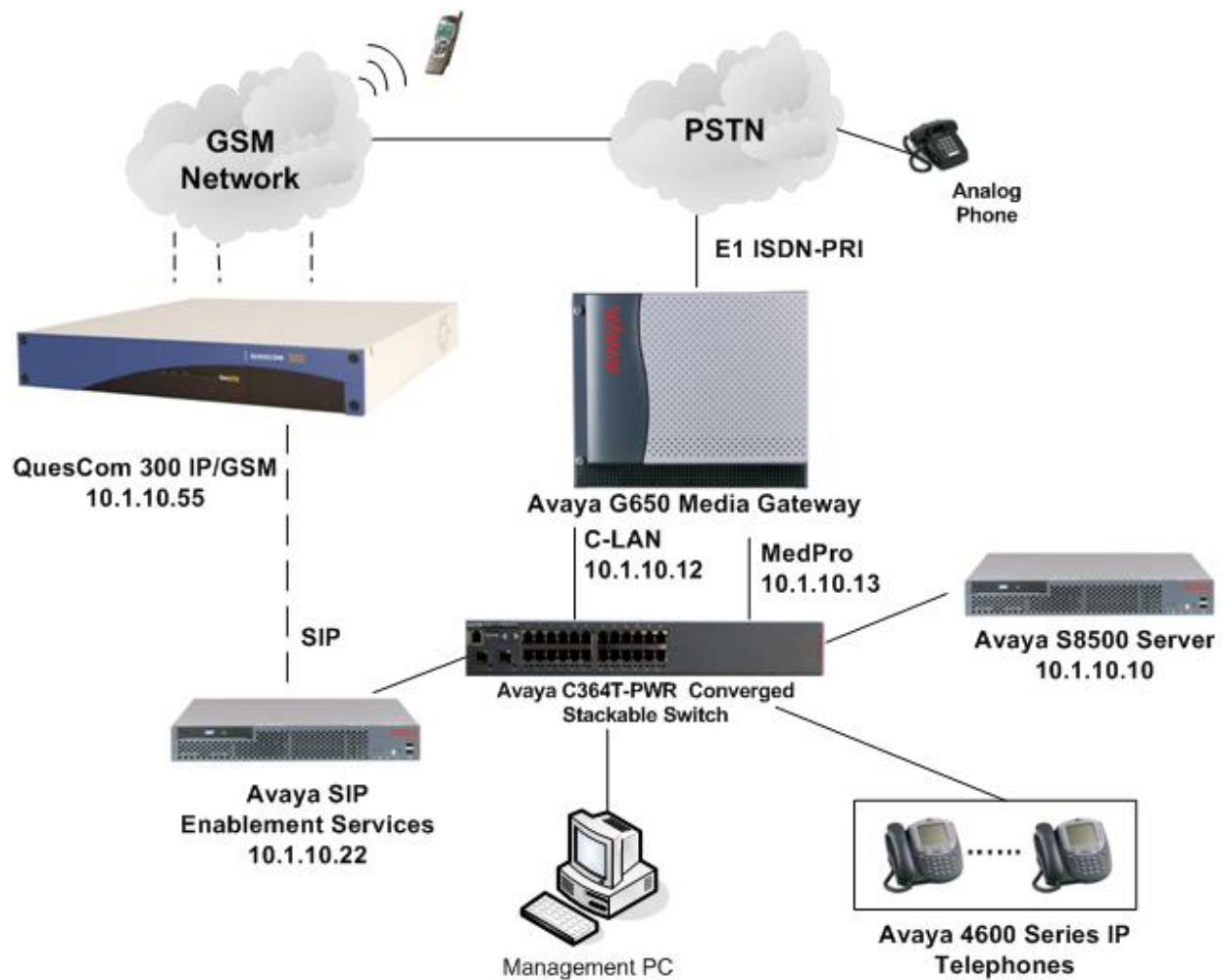


Figure 1: Avaya Communication Manager and Avaya SIP Enablement Services with QuesCom 300 IP/GSM

2. Equipment and Software Validated

Equipment	Software
Avaya SIP Enablement Services configured as a home/edge server.	4.0 (33.6)
Avaya S8500 Server running Avaya Communication Manager	4.0.1 (R014x.00.0.731.2)
Avaya G650 Media Gateway C-LAN TN799DP Medpro TN2302AP	HW 1, FW24 HW 20, FW116
Avaya C364T-PWR Converged Stackable Switch	4.3.12
Avaya 46XX Series IP Telephones (H.323)	2.8
QuesCom 300 IP/GSM	IAD05.00B030P000

3. Configure Avaya Communication Manager

Basic configuration of Avaya Communication Manager and Avaya SES are beyond the scope of these Application Notes. See Section 10 for Avaya documentation references. The steps are performed from the System Access Terminal (SAT) interface.

3.1. SIP Trunks

The steps in this section verify that there are sufficient number of SIP trunks between Avaya Communication Manager and Avaya SES.

Step	Description																																																																								
1.	<p>Use the display system-parameters customer-options command to verify that sufficient SIP trunk capacity exists. On Page 2, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.</p> <table><tr><td>display system-parameters customer-options</td><td>Page</td><td>2 of</td><td>10</td></tr><tr><td colspan="4">OPTIONAL FEATURES</td></tr><tr><td>IP PORT CAPACITIES</td><td></td><td></td><td>USED</td></tr><tr><td>Maximum Administered H.323 Trunks:</td><td>800</td><td></td><td>90</td></tr><tr><td>Maximum Concurrently Registered IP Stations:</td><td>2400</td><td></td><td>5</td></tr><tr><td>Maximum Administered Remote Office Trunks:</td><td>0</td><td></td><td>0</td></tr><tr><td>Maximum Concurrently Registered Remote Office Stations:</td><td>0</td><td></td><td>0</td></tr><tr><td>Maximum Concurrently Registered IP eCons:</td><td>20</td><td></td><td>0</td></tr><tr><td>Max Concur Registered Unauthenticated H.323 Stations:</td><td>2400</td><td></td><td>0</td></tr><tr><td>Maximum Video Capable H.323 Stations:</td><td>2400</td><td></td><td>0</td></tr><tr><td>Maximum Video Capable IP Softphones:</td><td>2400</td><td></td><td>0</td></tr><tr><td>Maximum Administered SIP Trunks:</td><td>800</td><td></td><td>35</td></tr><tr><td>Maximum Number of DS1 Boards with Echo Cancellation:</td><td>0</td><td></td><td>0</td></tr><tr><td>Maximum TN2501 VAL Boards:</td><td>10</td><td></td><td>0</td></tr><tr><td>Maximum Media Gateway VAL Sources:</td><td>250</td><td></td><td>0</td></tr><tr><td>Maximum TN2602 Boards with 80 VoIP Channels:</td><td>128</td><td></td><td>0</td></tr><tr><td>Maximum TN2602 Boards with 320 VoIP Channels:</td><td>128</td><td></td><td>1</td></tr><tr><td>Maximum Number of Expanded Meet-me Conference Ports:</td><td>300</td><td></td><td>0</td></tr></table>	display system-parameters customer-options	Page	2 of	10	OPTIONAL FEATURES				IP PORT CAPACITIES			USED	Maximum Administered H.323 Trunks:	800		90	Maximum Concurrently Registered IP Stations:	2400		5	Maximum Administered Remote Office Trunks:	0		0	Maximum Concurrently Registered Remote Office Stations:	0		0	Maximum Concurrently Registered IP eCons:	20		0	Max Concur Registered Unauthenticated H.323 Stations:	2400		0	Maximum Video Capable H.323 Stations:	2400		0	Maximum Video Capable IP Softphones:	2400		0	Maximum Administered SIP Trunks:	800		35	Maximum Number of DS1 Boards with Echo Cancellation:	0		0	Maximum TN2501 VAL Boards:	10		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
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Step	Description
2.	<p>Enter the display trunk-group n command, where “n” is the pre-configured SIP trunk group number between Avaya Communication Manager and Avaya SES. The number of ports configured should be coordinated with the number of SIM cards available in the QuesCom 300 gateway.</p> <pre> display trunk-group 30 TRUNK GROUP Group Number: 30 Group Type: sip CDR Reports: y Group Name: SIP TRUNK COR: 1 TN: 1 TAC: 730 Direction: two-way Outgoing Display? n Dial Access? n Night Service: Queue Length: 0 Service Type: tie Auth Code? n Signaling Group: 30 Number of Members: 5 </pre>
3.	<p>On the GROUP MEMBER ASSIGNMENTS screen (Page 5 of the trunk-group form). Verify the following group member assignments.</p> <pre> display trunk-group 30 TRUNK GROUP Administered Members (min/max): 1/5 GROUP MEMBER ASSIGNMENTS Total Administered Members: 5 Port Name 1: T00001 SIP TRUNK 2: T00002 SIP TRUNK 3: T00003 SIP TRUNK 4: T00004 SIP TRUNK 5: T00005 SIP TRUNK 6: </pre>

3.2. PSTN E1 ISDN-PRI

This section displays the PSTN E1 ISDN-PRI configuration on Avaya Communication Manager in the sample configuration of **Figure 1**. See Section 10 for Avaya documentation references.

Step	Description
1.	<p>Enter display ds1 <board location> to display the PSTN DS1 Circuit Pack configuration.</p> <pre> display ds1 01A12 DS1 CIRCUIT PACK Location: 01A12 Name: PRI to BT Bit Rate: 2.048 Line Coding: hdb3 Signaling Mode: isdn-pri Connect: network TN-C7 Long Timers? n Country Protocol: etsi Interworking Message: PROgress Protocol Version: b Interface Companding: alaw CRC? y Idle Code: 01010100 DCP/Analog Bearer Capability: 3.1kHz T303 Timer(sec): 4 Slip Detection? n Near-end CSU Type: other </pre>
2.	<p>Enter display trunk-group <number> to display the PSTN trunk-group configuration.</p> <pre> display trunk-group 19 Page 1 of 22 TRUNK GROUP Group Number: 19 Group Type: isdn CDR Reports: y Group Name: PRI to BT COR: 1 TN: 1 TAC: 719 Direction: two-way Outgoing Display? n Carrier Medium: PRI/BRI Dial Access? y Busy Threshold: 255 Night Service: Queue Length: 0 Service Type: public-ntwrk Auth Code? n TestCall ITC: rest Far End Test Line No: TestCall BCC: 4 </pre> <pre> display trunk-group 19 Page 2 of 22 Group Type: isdn TRUNK PARAMETERS Codeset to Send Display: 6 Codeset to Send National IEs: 6 Max Message Size to Send: 260 Charge Advice: none Supplementary Service Protocol: a Digit Handling (in/out): enbloc/overlap Trunk Hunt: cyclical QSIG Value-Added? n Digital Loss Group: 13 Incoming Calling Number - Delete: Insert: Format: Bit Rate: 1200 Synchronization: async Duplex: full Disconnect Supervision - In? y Out? n Answer Supervision Timeout: 0 </pre>

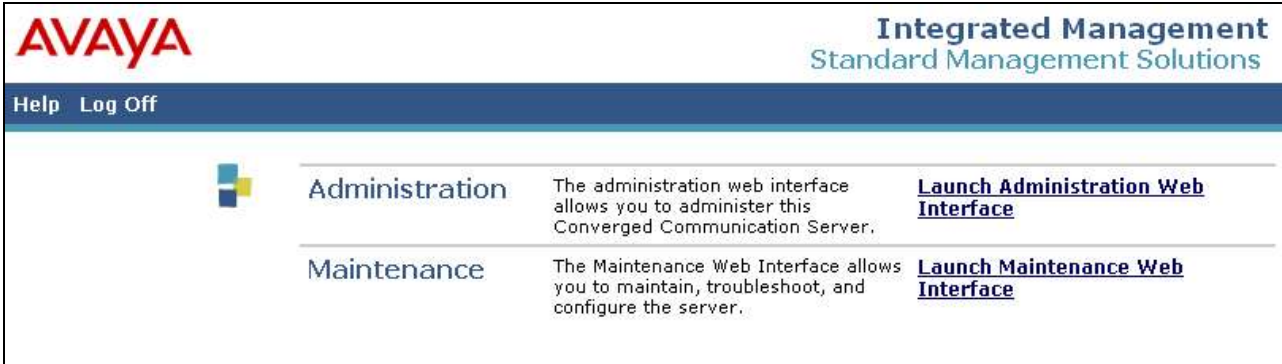
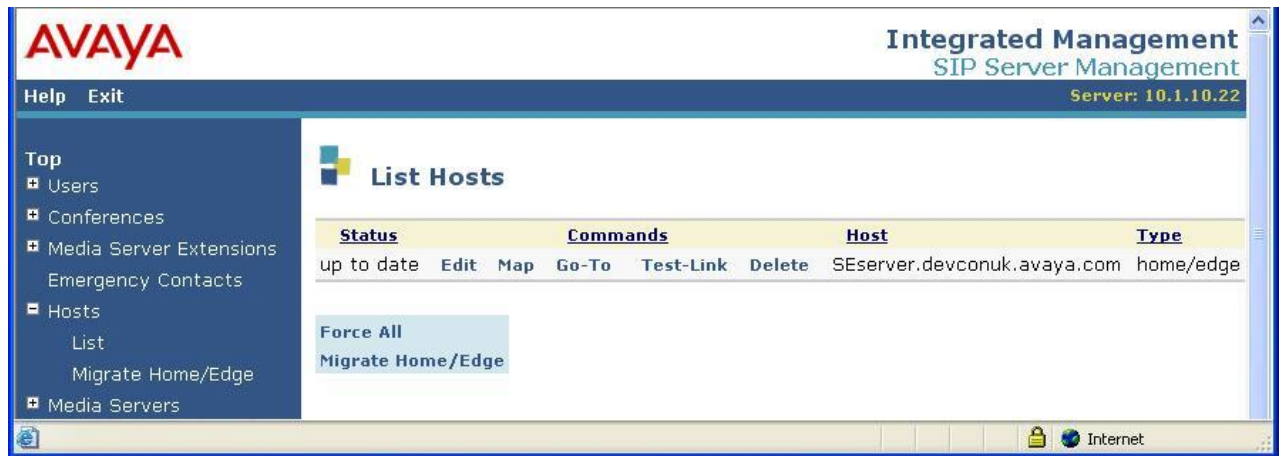
Step	Description																														
	<div><div>display trunk-group 19Page3 of 22</div><div>TRUNK FEATURES</div><div>ACA Assignment? nMeasured: bothWideband Support? nMaintenance Tests? yData Restriction? nNCA-TSC Trunk Member:Send Name: nSend Calling Number: ySend EMU Visitor CPN? nUsed for DCS? nSuppress # Outpulsing? yFormat: publicOutgoing Channel ID Encoding: preferredUII IE Treatment: sharedMaximum Size of UII IE Contents: 128Replace Restricted Numbers? nReplace Unavailable Numbers? nSend Connected Number: yHold/Unhold Notifications? yModify Tandem Calling Number? nBSR Reply-best DISC Cause Value: 31Dsl Echo Cancellation? nSend UII IE? ySend UCID? nSend Codeset 6/7 LAI IE? yApply Local Ringback? nUS NI Delayed Calling Name Update? nNetwork (Japan) Needs Connect Before Disconnect? n</div></div>																														
	<div><div>display trunk-group 19Page6 of 22</div><div>TRUNK GROUP</div><div>Administered Members (min/max): 1/5</div><div>GROUP MEMBER ASSIGNMENTSTotal Administered Members: 5</div><div><table><thead><tr><th>Port</th><th>Code Sfx</th><th>Name</th><th>Night</th><th>Sig Grp</th></tr></thead><tbody><tr><td>1: 01A1201</td><td>TN2464</td><td>C</td><td></td><td>19</td></tr><tr><td>2: 01A1202</td><td>TN2464</td><td>C</td><td></td><td>19</td></tr><tr><td>3: 01A1203</td><td>TN2464</td><td>C</td><td></td><td>19</td></tr><tr><td>4: 01A1204</td><td>TN2464</td><td>C</td><td></td><td>19</td></tr><tr><td>5: 01A1205</td><td>TN2464</td><td>C</td><td></td><td>19</td></tr></tbody></table></div></div>	Port	Code Sfx	Name	Night	Sig Grp	1: 01A1201	TN2464	C		19	2: 01A1202	TN2464	C		19	3: 01A1203	TN2464	C		19	4: 01A1204	TN2464	C		19	5: 01A1205	TN2464	C		19
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4: 01A1204	TN2464	C		19																											
5: 01A1205	TN2464	C		19																											
3.	<div><div>Enter display signaling-group <number> to display the PSTN signaling-group configuration.</div><div><div>display signaling-group 19Page1 of 5</div><div>SIGNALING GROUP</div><div>Group Number: 19Group Type: isdn-priAssociated Signaling? yMax number of NCA TSC: 5Primary D-Channel: 01A1216Max number of CA TSC: 5Trunk Group for NCA TSC: 19Trunk Group for Channel Selection: 19X-Mobility/Wireless Type: NONESupplementary Service Protocol: a</div></div></div>																														

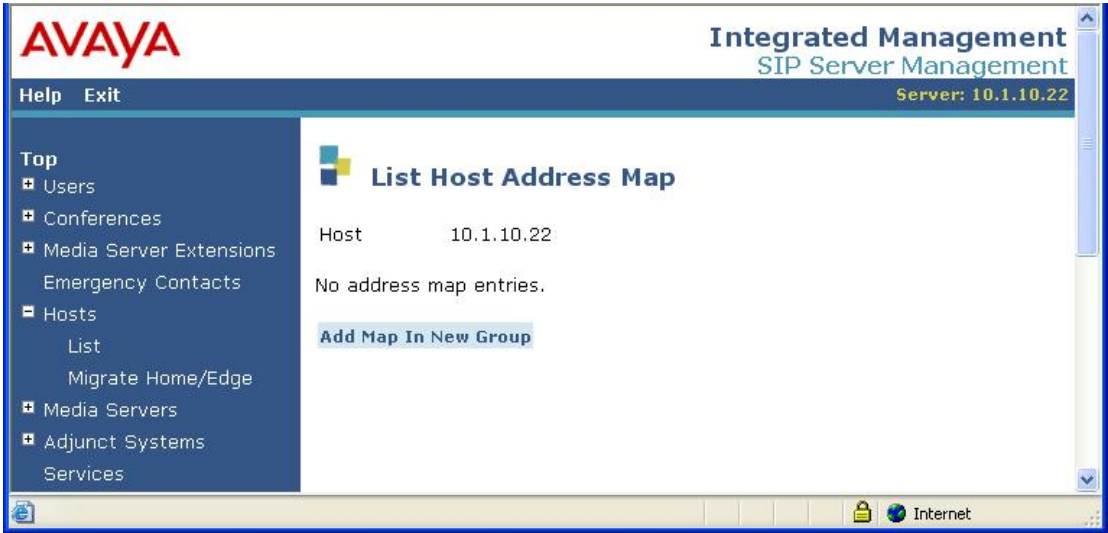
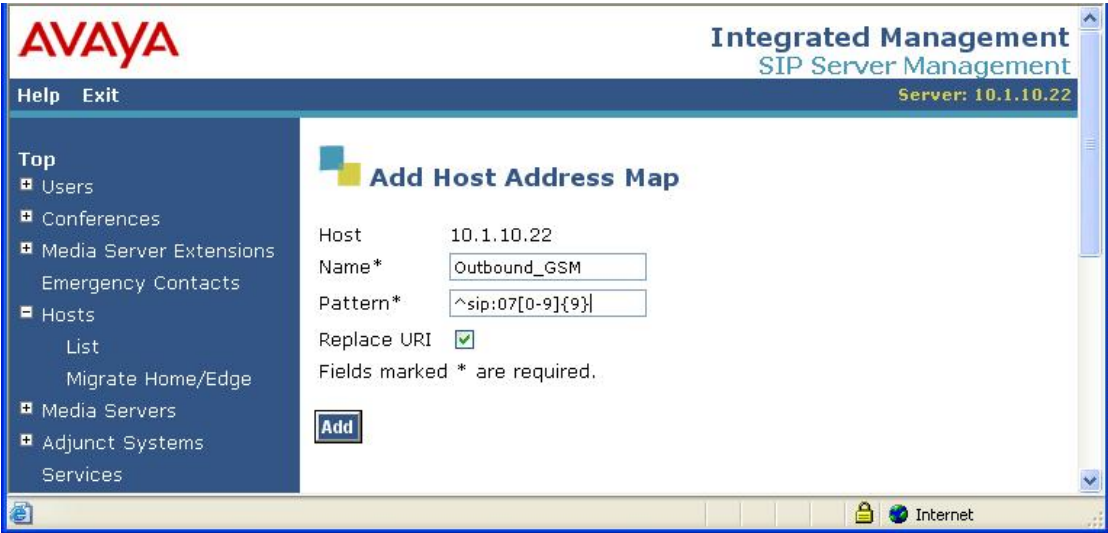
3.3. ARS Tables and Route Patterns


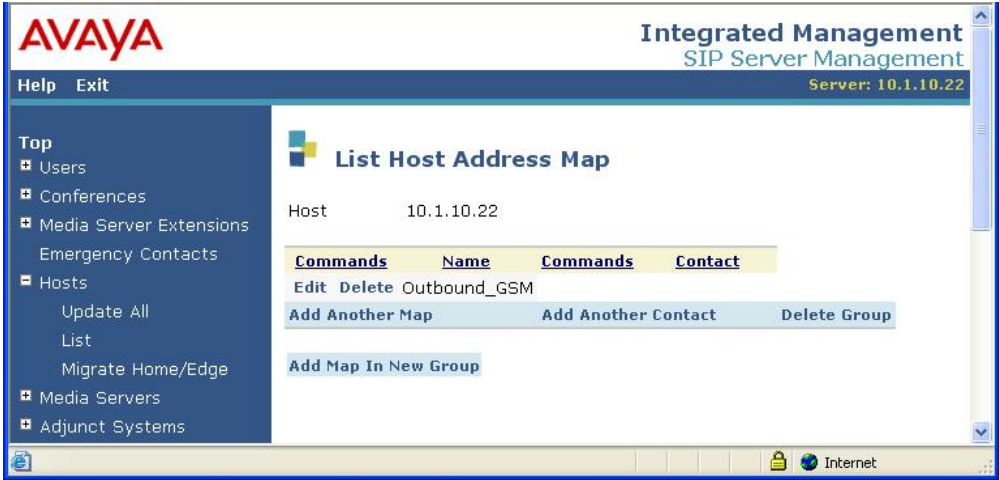
When placing outbound calls to the public network, stations on Avaya Communication Manager must first dial the ARS Feature Access Code (FAC) before dialing an external number. The single digit “9” was used as the ARS FAC in the compliance-tested configuration (not shown).


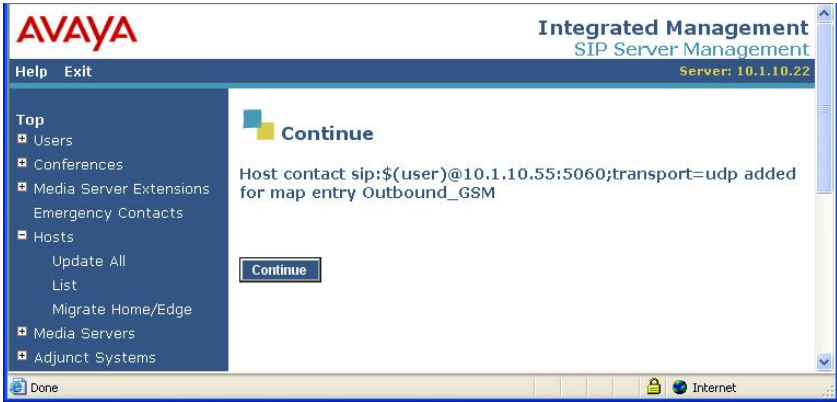
Step	Description																																																																																																																																																																																																																				
1.	<p>Enter the change ars analysis 0 command. Configure Dialed String entries according to customer requirements. In the example below, the entries match dialed numbers as follows:</p> <ul style="list-style-type: none">The “079” Dialed String matches 11-digit dialed numbers that begin with 079, and routes calls to Route Pattern 79.																																																																																																																																																																																																																				
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2.	<p>Enter the change route-pattern n command, where “n” is the route pattern that processes dialed numbers configured in Step 1. Add two routing preference entries as follows:</p> <p>1) First Routing Preference – SIP IP trunk to QuesCom 300</p> <ul style="list-style-type: none">Grp No – enter the trunk group number routed to the QuesCom 300 gateway (Section 3.1, Step 3)FRL - assign a Facility Restriction Level to this routing preference.LAR - set Look Ahead Routing to “next” to rehunt within the next routing preference if calls are rejected. LAR allows Avaya Communication Manager to re-attempt the call on another channel if the call is rejected with certain SIP response codes. <p>2) Second Routing Preference – PSTN E1 ISDN-PRI</p> <ul style="list-style-type: none">Grp No – enter the trunk group that contains trunk members from the PSTN E1 ISDN-PRI (Section 3.2, Step 2).FRL - assign a Facility Restriction Level to this routing preference.																																																																																																																																																																																																																				
	<table><tr><td colspan="12">change route-pattern 79</td><td>Page</td><td>1 of</td><td>3</td></tr><tr><td colspan="12">Pattern Number: 79</td><td colspan="3">Pattern Name: Quescom SIP</td></tr><tr><td colspan="12">SCCAN? n</td><td colspan="3">Secure SIP? n</td></tr><tr><td>Grp</td><td>FRL</td><td>NPA</td><td>Pfx</td><td>Hop</td><td>Toll</td><td>No.</td><td>Inserted</td><td colspan="4"></td><td>DCS/</td><td>IXC</td></tr><tr><td>No</td><td></td><td></td><td>Mrk</td><td>Lmt</td><td>List</td><td>Del</td><td>Digits</td><td colspan="4"></td><td>QSIG</td><td></td></tr><tr><td colspan="7"></td><td>Dgts</td><td colspan="4"></td><td>Intw</td><td></td></tr><tr><td>1:</td><td>30</td><td>0</td><td colspan="5"></td><td colspan="4"></td><td>n</td><td>user</td></tr><tr><td>2:</td><td>19</td><td>0</td><td colspan="5"></td><td colspan="4"></td><td>n</td><td>user</td></tr><tr><td>3:</td><td colspan="5"></td><td colspan="5"></td><td>n</td><td>user</td></tr><tr><td colspan="2">BCC</td><td>VALUE</td><td>TSC</td><td>CA-TSC</td><td colspan="2">ITC</td><td>BCIE</td><td colspan="2">Service/Feature</td><td>PARM</td><td>No.</td><td>Numbering</td><td>LAR</td></tr><tr><td colspan="2">0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>W</td><td colspan="2">Request</td><td colspan="2"></td><td>Dgts</td><td>Format</td><td></td></tr><tr><td colspan="12"></td><td>Subaddress</td><td></td></tr><tr><td>1:</td><td>y</td><td>y</td><td>y</td><td>y</td><td>y</td><td>n</td><td>n</td><td colspan="2">rest</td><td colspan="2"></td><td></td><td>next</td></tr><tr><td>2:</td><td>y</td><td>y</td><td>y</td><td>y</td><td>y</td><td>n</td><td>n</td><td colspan="2">rest</td><td colspan="2"></td><td></td><td>none</td></tr><tr><td>3:</td><td>y</td><td>y</td><td>y</td><td>y</td><td>y</td><td>n</td><td>n</td><td colspan="2">rest</td><td colspan="2"></td><td></td><td>none</td></tr></table>	change route-pattern 79												Page	1 of	3	Pattern Number: 79												Pattern Name: Quescom SIP			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:	30	0										n	user	2:	19	0										n	user	3:											n	user	BCC		VALUE	TSC	CA-TSC	ITC		BCIE	Service/Feature		PARM	No.	Numbering	LAR	0		1	2	3	4	W	Request				Dgts	Format														Subaddress		1:	y	y	y	y	y	n	n	rest					next	2:	y	y	y	y	y	n	n	rest					none	3:	y	y	y	y	y	n	n	rest					none
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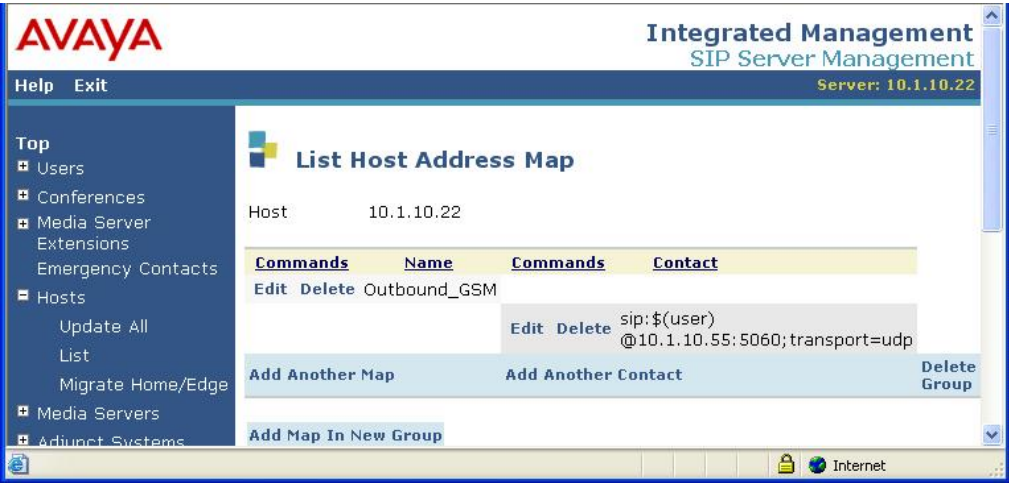
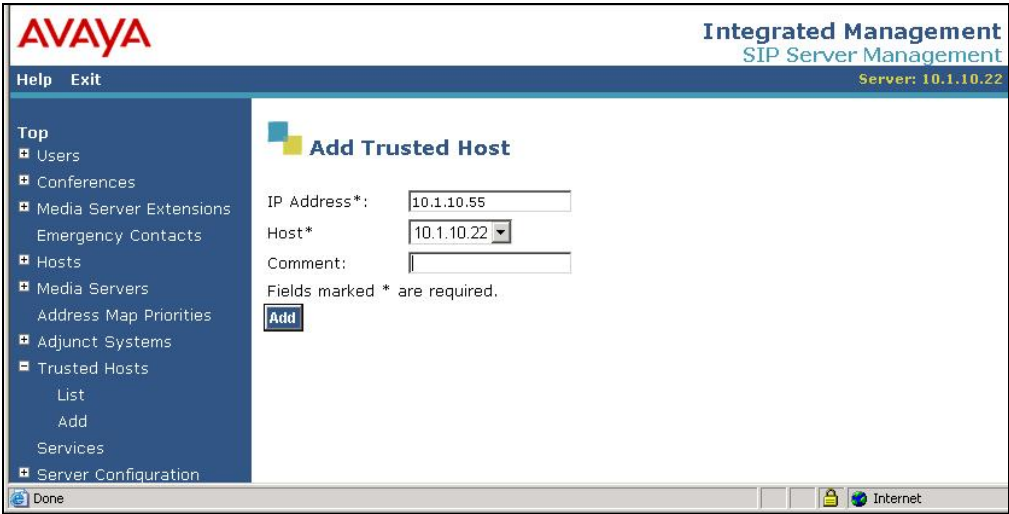
4. Configure Avaya SIP Enablement Services

Step	Description
1.	<p>Launch a web browser, enter <a href="https://<IP address of SES server>/admin">https://<IP address of SES server>/admin in the URL, and log in with the appropriate credentials. Click on the Launch Administration Web Interface link upon successful login.</p> 
2.	<p>Outbound calls are first directed by Avaya Communication Manager routing decisions to the SIP trunk group (Section 3.2). These calls are then subject to further routing decisions determined by the Host Address Maps in Avaya SES. Click Hosts → List and then Edit.</p> 

Step	Description
3.	<p>Click the Add Map In New Group link.</p> 
4.	<p>In the Add Host Address Map screen, configure the following.</p> <ul style="list-style-type: none"> • Name – Enter a descriptive name for the map. • Pattern – Specify an appropriate pattern for the call type. In this example, the pattern used is “^sip:07[0-9]{9}”. Any number 11 digits long beginning with 07 will use this host address map. • Replace URI – Leave the Replace URI checkbox selected. <p>Click the Add button.</p> 

Step	Description
5.	<p>Click the Continue button.</p>  <p>The screenshot shows the Avaya Integrated Management SIP Server Management web interface. The left sidebar contains a navigation menu with options: Top, Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts, and Update All. The main content area displays a 'Continue' button and the message 'Host address map Outbound_GSM added.' Below the message is another 'Continue' button. The status bar at the bottom shows 'Done' and 'Internet'.</p>
6.	<p>The next step is to enter the contact address for the QuesCom 300. Click on the Add Another Contact link associated with the address map added in Step 7.</p>  <p>The screenshot shows the Avaya Integrated Management SIP Server Management web interface. The left sidebar contains a navigation menu with options: Top, Users, Conferences, Media Server Extensions, Emergency Contacts, Hosts, Update All, List, Migrate Home/Edge, Media Servers, and Adjunct Systems. The main content area displays the 'List Host Address Map' page. It shows a table with columns: Commands, Name, Commands, and Contact. The table contains one row: Edit, Delete, Outbound_GSM. Below the table are three buttons: 'Add Another Map', 'Add Another Contact', and 'Delete Group'. The status bar at the bottom shows 'Internet'.</p>

Step	Description
7.	<p>In the Add Host Contact screen, the Contact field specifies the destination for the call and it is entered as: “sip:\$(user)@10.1.10.55:5060;transport=udp”, where 10.1.10.55 is the IP address of the QuesCom 300 in this configuration. The user part in the original request URI is inserted in place of the “\$(user)” string before the message is sent to the QuesCom 300. Click the Add button when completed.</p> 
8.	<p>Click the Continue button.</p> 

Step	Description
9.	<p>After making changes within Avaya SES, it is necessary to commit the database changes using the Update link that appears when changes are pending. Perform this step by clicking on the Update link or Hosts→ Update All.</p> 
10.	<p>Administer the QuesCom 300 gateway as a trusted host. To configure a trusted host, click on Trusted Hosts → Add. Enter the IP address of the QuesCom gateway in the IP Address field and click on Add.</p>  <p>After configuring the trusted host, the administrator must go back to the SES administration web interface, and click on the Update link in the bottom left pane as shown in Step 9 for the changes to take effect.</p>

5. Configure the QuesCom 300 IP/GSM

This section describes the steps for configuring the QuesCom 300 IP/GSM gateway.

5.1. QuesCom Server Configuration

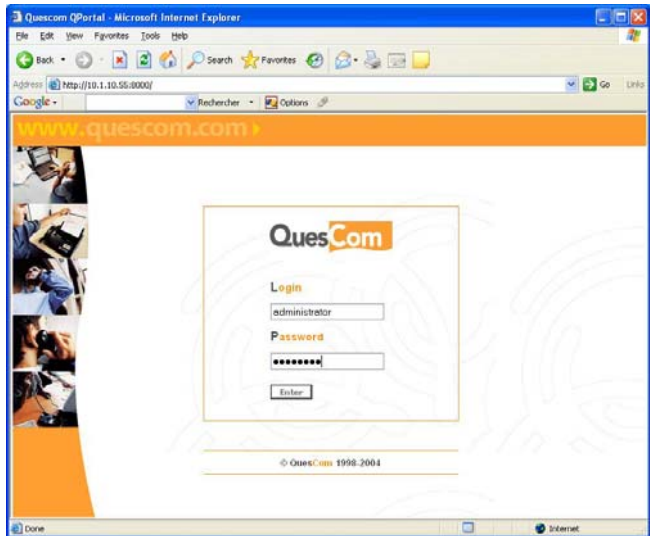

This section includes the necessary configuration steps to allow the QuesCom 300 IP/GSM gateway to make outbound calls to the GSM network once connected to the Avaya SES.

Step	Description
1.	<p>After the initial installation of the QuesCom server, telnet into the QuesCom server from the management PC shown in figure 1, using the default IP address “192.168.1.1”. Log in using the appropriate username and password.</p> <pre>C:\> telnet 192.168.1.1 login: administrator Password: ***** Enterprise Series, Serial# Q300-A1-00010016, Version IAD05.00B030P000 BIOS Version 6.00 PG from 06/29/2004 Security Patch SP002 Copyright (c) 1998-2007 QuesCom S.A.</pre> <p>At the prompt, type the following command gwconfig /setup.</p> <pre>X:\>gwconfig /setup Application has been registered to the QCFGSvc QCFGSvc Version 5.00.000.006 Copyright (c) 1998-2007 QuesCom S.A.</pre> <p>Enter “1” for English.</p> <pre>Enter the Gateway Administration language [1]: 1 English 2 French 3 German > 1 GWconfig language: English</pre> <p>Enter a name for the QuesCom 300 gateway.</p> <pre>Setting up Gateway components... Enter the Gateway network name [Q300-00010016]: Q300 Gateway Network Name: Q300</pre> <p>Enter IP address, subnet mask and default gateway for the QuesCom gateway.</p> <pre>Enter the Gateway IP address [192.168.1.1]: 10.1.10.55 The Gateway IP address: 10.1.10.55 Enter the Gateway subnet mask [255.0.0.0]: 255.255.255.0 The Gateway subnet mask: 255.255.255.0</pre>

Step	Description
	<div data-bbox="277 226 1528 380"> <p>Enter the default Gateway [10.1.10.1]: 10.1.10.1 The default Gateway: 10.1.10.1</p> </div> <div data-bbox="277 415 721 451"> <p>Enter “N” for the following option</p> </div> <div data-bbox="277 451 1528 497"> <p>Do you want to activate conferencing? [Y/N]: N</p> </div> <div data-bbox="277 533 1000 569"> <p>Enter “0” for the server to operate in Stand-Alone mode.</p> </div> <div data-bbox="277 569 1528 722"> <pre> Enter the 'Call Server' mode [0]: 0 Stand-Alone mode 1 Relay mode > 0 Call Server mode: Stand-Alone </pre> </div> <div data-bbox="277 758 1045 793"> <p>Enter Company Name. This can be any alphanumeric name.</p> </div> <div data-bbox="277 793 1528 871"> <p>Enter Company Name []: Avaya Company Name: Avaya</p> </div> <div data-bbox="277 907 737 942"> <p>Enter “2” to select the SIP protocol.</p> </div> <div data-bbox="277 942 1528 1241"> <p>Do you want to activate SIP or H.323 connectivity now? [Y/N]: Y</p> <p>Declare VOIP Gateway/Softswitch which will be allowed to send calls to the QuesCom gateway</p> <pre> 0 Skip to next step/Do it later 1 H.323 (no registration) 2 SIP (no registration) > 2 </pre> </div> <div data-bbox="277 1276 873 1312"> <p>Enter the IP address and name for Avaya SES.</p> </div> <div data-bbox="277 1312 1528 1499"> <p>Enter the IP Address of the SIP Proxy: 10.1.10.22 SIP Proxy IP Address: 10.1.10.22</p> <p>Enter the name of the SIP Proxy: SES SIP Proxy name: SES</p> </div> <div data-bbox="277 1535 1198 1570"> <p>Enter “0” to configure the incoming calls to the Quescom gateway later.</p> </div> <div data-bbox="277 1570 1528 1793"> <p>Declare VOIP Gateway/Softswitch which will be allowed to send calls to the QuesCom gateway</p> <pre> 0 Skip to next step/Do it later 1 H.323 (no registration) 2 SIP (no registration) > 0 </pre> </div>

Step	Description
	<p>Enter “0” for the following Voice Box option.</p> <div data-bbox="277 264 1528 411"> <p>Do you want to use the 'Voice Box' service [0]?</p> <p>0 No</p> <p>1 Yes</p> <p>> 0</p> </div> <p>Configure the time zone and daylight saving settings.</p> <div data-bbox="277 485 1528 600"> <p>Enter Time Zone number (0 to skip / L to view the list): 27</p> <p>Do you want to enable saving the TimeZone DayLight Information? [Y/N]: Y</p> </div> <p>Verify the selected parameters press any key to continue and enter “1” to confirm the setup.</p> <div data-bbox="277 674 1528 1440"> <p>Selected parameters for Quick setup mode are:</p> <p>Gateway Network Name: Q300</p> <p>The Gateway IP address: 10.1.10.55</p> <p>The Gateway subnet mask: 255.255.255.0</p> <p>The default Gateway: 10.1.10.1</p> <p>Press any key to continue...</p> <p>Gateway's serial number: Q300-A1-00010016</p> <p>IVR language country: ENG - English</p> <p>Email language country: ENG - English</p> <p>Country Tones: United Kingdom</p> <p>Country Numbering: United Kingdom</p> <p>Call Server mode: Stand-Alone</p> <p>Company Name: Avaya</p> <p>Do you confirm this setup [1]:</p> <p>0 No (to exit, and GWconfig /setup command can be re-entered)</p> <p>1 Yes(to continue the setup and restart the QuesCom Gateway)</p> <p>> 1</p> </div> <div data-bbox="277 1482 1528 1682"> <p>Setting up QPortal Application...</p> <p>Please wait...</p> <p>Rebooting system...</p> <p>Warning: Do not restart the Gateway, update process in progress...</p> <p>Please, wait up to 3 minutes.</p> </div>

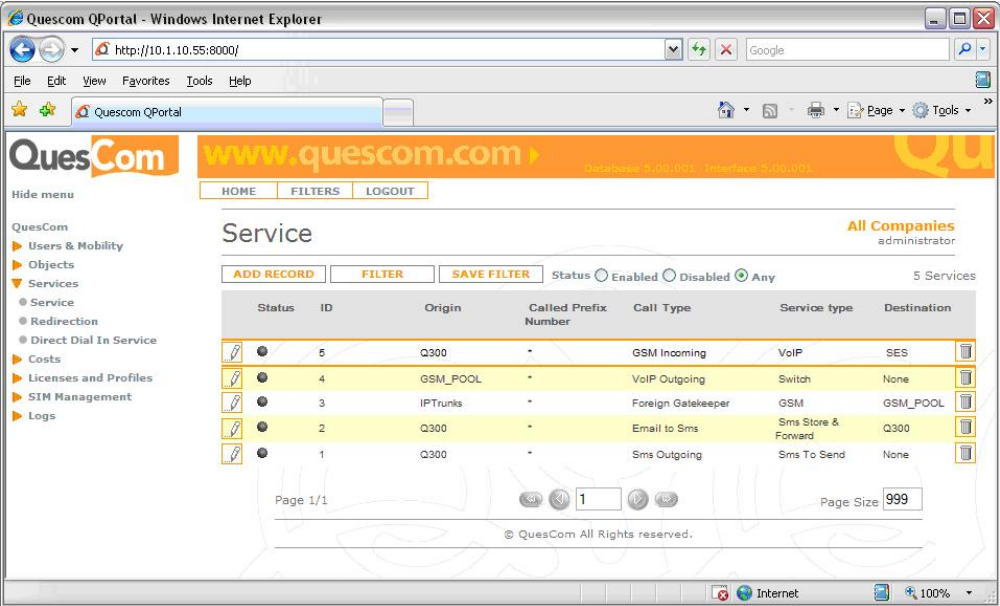
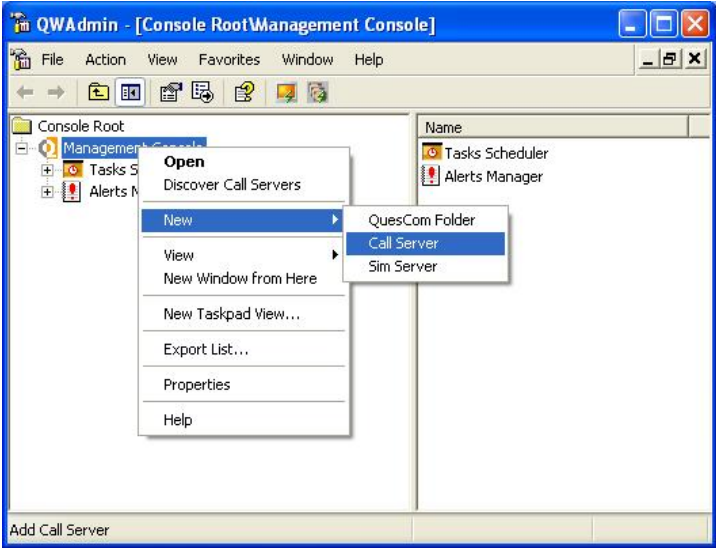
5.2. QuesCom Routing Configuration

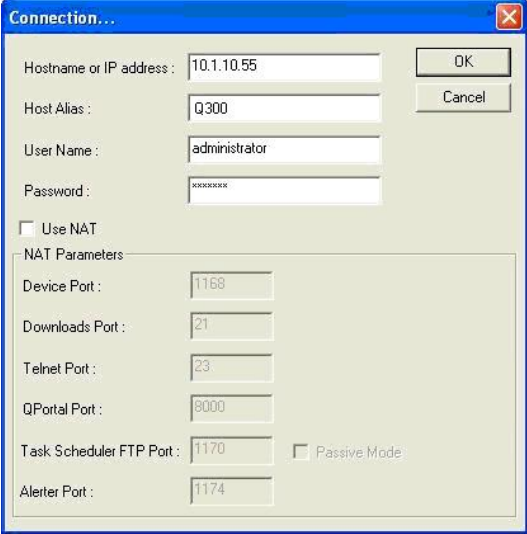
Step	Description																														
1.	<p>Open a web browser from the management PC and enter the following URL <code>http://<QuesCom 300 IPaddress:8000></code>. For this configuration “http://10.1.10.55:8000” was entered. Log in using the appropriate user name and password.</p> <div></div>																														
2.	<p>Click on Services → Service. Four entries are present by default. ID “3” is created by default and is routing for outbound calls from Avaya Communication Manager via the Avaya SES to the QuesCom 300 gateway. ID “4” is routing of outbound calls from the QuesCom 300 gateway to the GSM network. Service IDs “1” and “2” are also created by default, and are related to SMS (Short Message Service) that were not tested during compliance testing.</p> <div></div> <table><tr><th>ID</th><th>Origin</th><th>Called Prefix Number</th><th>Call Type</th><th>Service type</th><th>Destination</th></tr><tr><td>4</td><td>GSM_POOL</td><td>*</td><td>VoIP Outgoing</td><td>Switch</td><td>None</td></tr><tr><td>3</td><td>Avaya</td><td>*</td><td>Foreign Gatekeeper</td><td>VoIP</td><td>GSM_POOL</td></tr><tr><td>2</td><td>Q400</td><td>*</td><td>Email to Sms</td><td>Sms Store & Forward</td><td>Q400</td></tr><tr><td>1</td><td>Q400</td><td>*</td><td>Sms Outgoing</td><td>Sms To Send</td><td>None</td></tr></table>	ID	Origin	Called Prefix Number	Call Type	Service type	Destination	4	GSM_POOL	*	VoIP Outgoing	Switch	None	3	Avaya	*	Foreign Gatekeeper	VoIP	GSM_POOL	2	Q400	*	Email to Sms	Sms Store & Forward	Q400	1	Q400	*	Sms Outgoing	Sms To Send	None
ID	Origin	Called Prefix Number	Call Type	Service type	Destination																										
4	GSM_POOL	*	VoIP Outgoing	Switch	None																										
3	Avaya	*	Foreign Gatekeeper	VoIP	GSM_POOL																										
2	Q400	*	Email to Sms	Sms Store & Forward	Q400																										
1	Q400	*	Sms Outgoing	Sms To Send	None																										

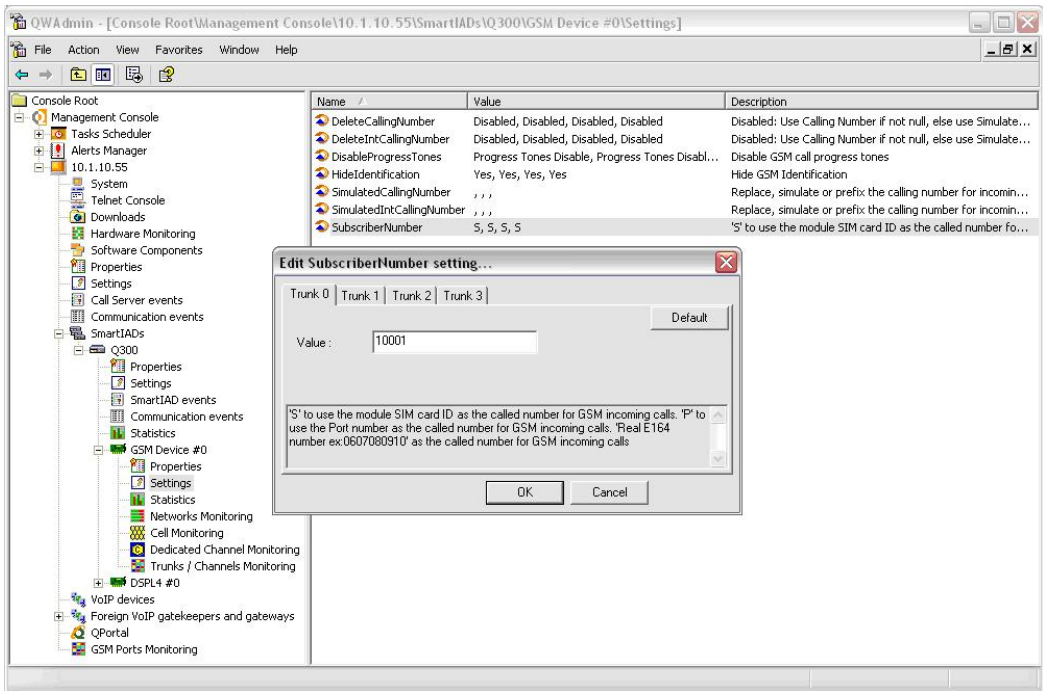
Step	Description
3.	<p>Routing of inbound calls to the QuesCom 300 gateway from the GSM network is created by clicking on ADD RECORD button on the main Service screen shown in Step 6. On the Service screen, configure the following as shown below.</p> <ul style="list-style-type: none"> • Origin Type – select radio button “Device” • Origin – select “Q300(SmartAD)” • Called Prefix Number – enter “*” • Call Type – select “GSM Incoming” • Service type – select “VoIP” • Destination Type – select radio button “Foreign GK” • Device – select “SES”, which was configured during the initial configuration in Section 5.1, Step 1. <p>The other parameters can be left with default values. Click on Save.</p>

The screenshot displays the QuesCom Service configuration window. The interface includes a top navigation bar with 'HOME', 'FILTERS', and 'LOGOUT' buttons, and a 'SAVE' button. A left sidebar lists navigation options like 'Users & Mobility', 'Objects', 'Services', 'Costs', 'Licenses and Profiles', 'SIM Management', and 'Logs'. The main configuration area is divided into several sections:

- Origin:** Includes radio buttons for 'Device' (selected), 'Foreign GK', and 'CTI'. Below are dropdowns for 'Origin' (Q300 (SmartAD)), 'Called Prefix Number' (*), and 'Call Type' (GSM Incoming). There are also checkboxes for 'Enabled for' H323, QGP, and SIP.
- Destination:** Includes radio buttons for 'IP Address (H323)', 'Foreign GK' (selected), 'CTI', and 'Device'. Below are fields for 'IP Address', 'Device' (SES), 'Balancing mode' (None, Bal., Cycling, Sim), and 'Called number lookup' (Local, Nquire, External).
- Service associated:** Includes a 'Service disabled' checkbox, 'Service type' (VoIP), 'Authentication Type' (None), 'Called Number Type' (ISDN), and 'Voice Fax Mode' (Switch, VoIP, CTI Application).
- Call Server Operations:** Includes dropdowns for 'LCR Support' (No), 'CDR Support' (No), and 'Cost Support' (No).
- VoIP Service:** Includes a 'Law Transcoding' checkbox (checked) and a 'Quality of service(HEX)' dropdown (Min Delay).
- Backup Mode:** Includes checkboxes for 'Enabled for' DSP, RSVP, Relay, and Other.
- Fax /Voice Service:** Includes dropdowns for 'Voice Fax Type' (Store & Forward), 'Store & Forward Type' (FAX TO EMAIL), 'Called Number', 'Notify Receipt Type' (None), 'Notify receipt to', and 'Send To'.

Step	Description
4.	<p>The inbound call route pattern added in Step 3 is displayed on the main Service screen by clicking on Services → Service.</p> 
5.	<p>From the management PC shown in Figure 1, launch the QesCom 300 QWA management console by clicking Start → Programs → QesCom → QesCom Management Console. Right click on Management Console and click New → Call Server.</p> 

Step	Description
6.	<p>In the Connection dialog, configure the following and click OK:</p> <ul style="list-style-type: none"> • Hostname or IP address – enter the IP address of the QuesCom 300 gateway • Host Alias – enter a descriptive name for the QuesCom 300 gateway • User Name and Password 

Step	Description
7.	<p>Expand the Management Console tree by clicking on Q300 (10.1.10.55) → SmartIADs → Q300 → GSM Device #0 → Settings → SubscriberNumber. In the Edit SubscriberNumber setting dialog box, click on the Trunk 0 tab each trunk is associated with a SIM card. Enter an Avaya Communication station that incoming calls will be routed to in the Value field. For the convenience of compliance testing, the calls were routed to a station on Avaya Communication Manager for all incoming trunks. Replicate this field for all 4 Trunks. Click OK</p> <p>Right click on Q300 under SmartIADs and click on Save configuration, then right click back on Q300 and click on Stop(not shown). Right click Q300 and click on Start and wait for the SIM cards to register(not shown).</p> 

6. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying the routing of inbound/outbound calls to/from the QuesCom 300.

6.1. General Test Approach

The general approach was to place inbound and outbound calls through the QuesCom 300 and verify successful call completion. The main objectives were to verify that:

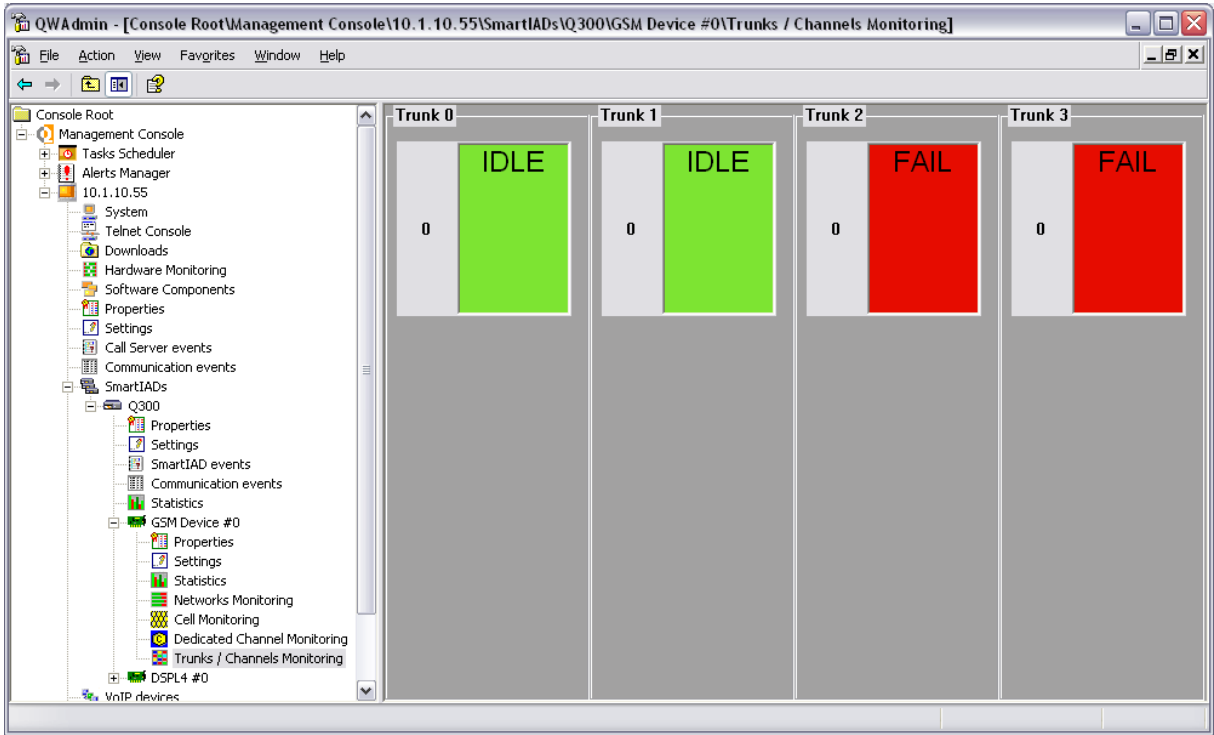
- When internal extensions place outbound calls to GSM numbers, the calls are routed to the QuesCom 300, and the QuesCom 300 decides on the least cost routing and routes the call to the GSM network.
- Inbound calls from the GSM network to the QuesCom 300 are successfully forwarded to Avaya SES using both direct routing (mapping of a SIM card phone number to an Avaya Communication Manager extension) and post-dialing (SIM card answers an inbound call and upon a prompt, the external caller enters an Avaya Communication Manager extension).
- Transfers and conferences between Avaya Communication Manager stations complete properly on outbound and inbound calls routed through the QuesCom 300.

6.2. Test Results

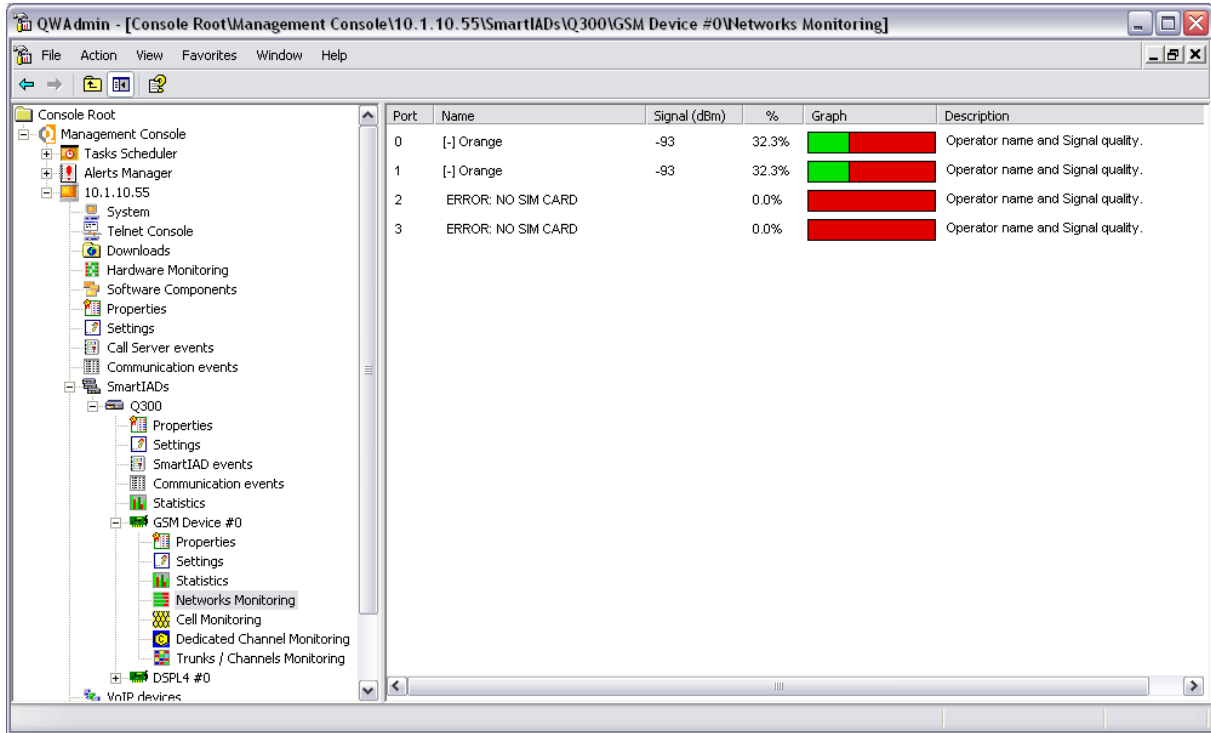
The test objectives of Section 6.1 were verified. For serviceability testing, outbound and inbound calls routed through the QuesCom 300 complete successfully after recovering from failures such as Ethernet cable disconnects, and resets of Avaya Communication Manager, Avaya SES and the QuesCom 300.

7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Communication Manager and QuesCom 300.

Step	Description
1.	<p>From the SAT, enter the command status signaling-group s, where s is the number of a signaling group configured in Section 3.2 and verify that the Group State is “in service”.</p> <p>From the SAT, enter the command status trunk-group t, where t is the number of a trunk group configured in Section 3.2, and verify that the Service States of all trunks are “in-service/idle” or “in-service/active”.</p>
2.	<p>Expand the Management Console tree by clicking on Q300 (10.1.10.55) → SmartIADs → Q300 → GSM Device #0 → Trunks/Channels Monitoring. Ensure the Trunks configured are the colour green with IDLE.</p>  <p>The screenshot shows the QWAdmin Management Console interface. The left pane displays a tree view of the system hierarchy, with 'Trunks / Channels Monitoring' selected under 'GSM Device #0'. The right pane shows four trunk status boxes: Trunk 0 (IDLE, green), Trunk 1 (IDLE, green), Trunk 2 (FAIL, red), and Trunk 3 (FAIL, red). Each box also displays a count of '0'.</p>

Step	Description
3.	Expand the Management Console tree by clicking on Q300(10.1.10.55) → SmartIADs → Q300 → GSM Device #0 → Networks Monitoring . Ensure the Signal(dBm) is above -90.



Port	Name	Signal (dBm)	%	Graph	Description
0	[-] Orange	-93	32.3%		Operator name and Signal quality.
1	[-] Orange	-93	32.3%		Operator name and Signal quality.
2	ERROR: NO SIM CARD		0.0%		Operator name and Signal quality.
3	ERROR: NO SIM CARD		0.0%		Operator name and Signal quality.

8. Support

Technical support from QuesCom can be requested in any of the following three ways.

- The corporate QuesCom Reporting Tool (QRT) account on the QuesCom web site at <http://support.quescom.com> and follow instructions.
- The Support Line number. +33 820203846 (France) Voice Message is available during off days and non-working time.
- Sending an email to support@quescom.com

9. Conclusion

These Application Notes describe the configuration steps required for QuesCom IP/GSM 300 version IAD05.00B030P000 to successfully interoperate with Avaya Communication Manager 4.0.1 and Avaya SES 4.0. All feature functionality and serviceability test cases were completed successfully.

10. Additional References

This section references the Avaya and QuesCom IP/GSM 300 product documentation that are relevant to these Application Notes.

The following Avaya Documents are available at <http://support.avaya.com>

- *Documentation for Avaya Communication Manager (4.0), Media Gateways and Servers*, Document ID 03-300151, Issue 6, February 2007, available at: <http://support.avaya.com>.
- *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 series Media Server*, Document ID 555-245-206, Issue 7, May 2007.

The following documents can be requested from QuesCom by sending an e-mail to support@quescom.com.

- Getting Started with QuesCom 300 IP/GSM: GS-Q300IPGSM300-V01.pdf
- QuesCom 300 IP/GSM Administrator Guide: AG-Q300IPGSM300-V01.pdf
- How to configure GSM Incoming calls to a remote Gatekeeper: Configuring GSM incoming calls.pdf

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