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Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer G5 with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP Trunk Emulation – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Empirix Hammer G5 test system with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP trunk emulation. The Hammer G5 is a VoIP test platform consisting of a collection of applications used to configure the system, create, schedule, and monitor tests, and create reports. In this configuration, the Hammer G5 emulates SIP trunks that interface to SIP Enablement Services. The Hammer G5 originates and terminates calls through the Avaya SIP telephony network. While the call is active, it can send DTMF tones and voice media, and provide voice quality metrics. Call progress can also be monitored, and at the completion of the test, test reports can be generated. The Hammer G5 can be used to verify the functionality, reliability and quality of VoIP services and applications running through the Avaya SIP telephony network.

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 to integrate the Empirix Hammer G5 test system with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP trunk emulation. The Hammer G5 is a VoIP test platform consisting of a collection of applications used to configure the system, create, schedule, and monitor tests, and create reports. In this configuration, the Hammer G5 emulates SIP trunks that interface to SIP Enablement Services.   The Hammer G5 originates and terminates calls through the Avaya SIP telephony network. While the call is active, it can send DTMF tones and voice media, and provide voice quality metrics. Call progress can also be monitored, and at the completion of the test, test reports can be generated. The Hammer G5 can be used to verify the functionality, reliability and quality of VoIP services and applications running through the Avaya SIP telephony network.

The following set of Hammer G5 applications were used during the compliance testing:

- **Hammer Configurator** used to configure and manage the system.
- **Hammer TestBuilder** used to create and run test scripts.
- **Hammer System Monitor** used to monitor call progress.
- **Hammer Call Summary Monitor** used to monitor call completion and to create reports.

A solution using SIP endpoint emulation on the Empirix Hammer G5 is described in [4] *Application Notes for Empirix Hammer G5 with Avaya Aura™ Communication Manager and Avaya Aura™ Communication Manager using SIP Endpoint Emulation*.

1.1 Interoperability Compliance Testing

The interoperability compliance testing focused on verifying that the Hammer G5 can register with SIP Enablement Services as SIP endpoints, establish calls, send voice media, provide voice quality metrics, and generate reports. The following features and functionality were covered:

- SIP endpoint registration with SIP Enablement Services.
- Originating and terminating calls through the Avaya SIP telephony network.
- Support of G.711, G.723, and G.729 codecs.
- Support of direct IP-to-IP media (also known as "Shuffling" which allows IP endpoints to send audio RTP packets directly to each other without using media resources on the Avaya Media Gateway). Calls with Shuffling disabled was also verified.
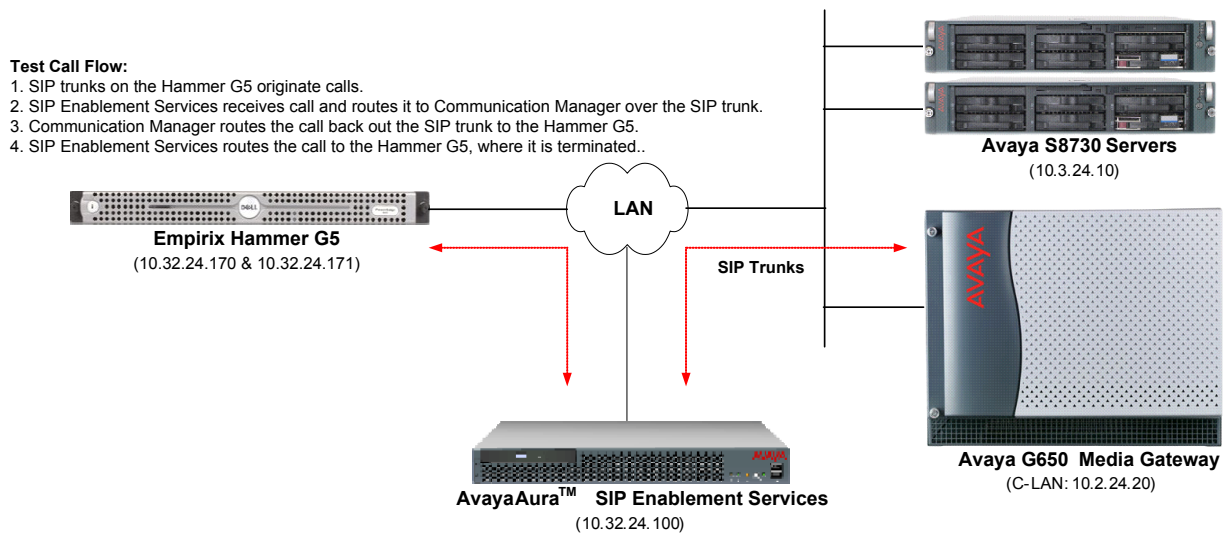- Generating voice quality metrics with Shuffling disabled.

1.2 Support

Technical support on the Empirix Hammer G5 can be obtained via phone, website, or email.

- **Phone:** (781) 266-3202
- **Web:** http://www.empirix.com/support/maintenance.asp
- **Email:** support@support.com

2 Reference Configuration

The network diagram below illustrates the test configuration. In this configuration, Avaya Aura™ SIP Enablement Services receives calls from the Hammer G5, which emulates SIP trunks. The call is routed through the Avaya SIP telephony network, including SIP Enablement Services and Communication Manager. The call is eventually routed back to the Hammer G5, where it is terminated. While the call is established, the Hammer G5 sends voice media (i.e., RTP traffic) using an audio recording. This allows voice quality metrics to be provided at the end of each call. The Hammer G5 applications running on the Hammer G5 server were used to configure the system, create and monitor the tests, and view the test reports.

Test Call Flow:
1. SIP trunks on the Hammer G5 originate calls.
2. SIP Enablement Services receives call and routes it to Communication Manager over the SIP trunk.
3. Communication Manager routes the call back out the SIP trunk to the Hammer G5.
4. SIP Enablement Services routes the call to the Hammer G5, where it is terminated..

Empirix Hammer G5
(10.32.24.170 & 10.32.24.171)

LAN

SIP Trunks

Avaya S8730 Servers
(10.3.24.10)

Avaya G650 Media Gateway
(C-LAN: 10.2.24.20)

AvayaAura™ SIP Enablement Services
(10.32.24.100)

Empirix Hammer G5 with Avaya SIP Telephony Network

3 Equipment and Software Validated

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

Equipment	Software
Avaya S8730 Servers	Avaya Aura™ Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 1 (Patch 17959)
Avaya G650 Media Gateway ▪ TN799DP C-LAN Board ▪ TN2302AP Media Processor Board	 HW01 FW031 HW03 FW094
Avaya Aura™ SIP Enablement Services	5.2.1 (SES-5.2.1.0-016.4)
Empirix Hammer G5 running on Microsoft Windows 2003 Server	1.5.8.295

4 Configure Avaya Aura^{TM} Communication Manager

This section provides the procedures for configuring Avaya Aura^{TM} Communication Manager. The procedures include the following areas:

- ▪ Administer IP Node Names
- ▪ Administer IP Codec Set
- ▪ Administer IP Network Region
- ▪ Administer SIP Trunk Group to SIP Enablement Services
- ▪ Administer SIP Stations

Avaya Aura^{TM} Communication Manager is configured through the System Access Terminal (SAT).

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4 of 44
Hammer-SIP-TRK

4.1 Administer IP Node Names

In the **IP Node Names** form, assign a **Name** and **IP Address** and for the C-LAN board in the Avaya G650 Media Gateway and SIP Enablement Services. The names will be used when configuring the SIP trunk in Communication Manager.

```
change node-names ip                                            Page   1 of   2
                              IP NODE NAMES
    Name              IP Address
Gateway001        10.32.24.1
clancrm           10.32.24.20
default           0.0.0.0
medprocrm         10.32.24.21
procr             0.0.0.0
ses               10.32.24.100
 ( 6 of 6   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

4.2 Administer IP Codec Set

In the **IP Codec Set** form, specify the audio codec required by the Hammer test. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region form covered in the next section. The Hammer G5 supports G.711, G.729AB, and G.723 codecs. For the compliance test, the IP codec set form was configured as shown below.

```
change ip-codec-set 1                                           Page   1 of   2

                         IP Codec Set

    Codec Set: 1

    Audio          Silence      Frames   Packet
    Codec          Suppression  Per Pkt  Size(ms)
 1: G.711MU          n            2         20
 2: G.729AB          n            2         20
 3: G.723-6.3K       n            1         30
```

4.3 Administer IP Network Region

In the **IP Network Region** form, configure the codec set to be used for the Hammer calls and specify whether **IP-IP Direct Audio** (Shuffling) is required. Shuffling allows audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. If voice quality metrics are to be generated by the test, disable Shuffling; otherwise, the audio traffic will not leave the Hammer G5 and the voice quality metrics will not yield accurate results. Shuffling can also be disabled at the signaling group level. The **Authoritative Domain** for this configuration is *avaya.com*.

```
change ip-network-region 1                                   Page   1 of  19
                             IP NETWORK REGION
  Region: 1
Location:              Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                  Intra-region IP-IP Direct Audio: no
     Codec Set: 1                 Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                        IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS                     RTCP Reporting Enabled? y
 Call Control PHB Value: 34       RTCP MONITOR SERVER PARAMETERS
       Audio PHB Value: 46         Use Default Server Parameters? y
       Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
       Audio 802.1p Priority: 6
       Video 802.1p Priority: 5     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

4.4 Administer SIP Trunk

Prior to configuring a SIP trunk group to SIP Enablement Services, a SIP signaling group must be configured. Configure the **Signaling Group** form as shown below. The **Near-end Node Name** and **Far-end Node Name** are the C-LAN board and SIP Enablement Services, respectively.

```
add signaling-group 10                                        Page   1 of   1
                              SIGNALING GROUP

 Group Number: 10                     Group Type: sip
                              Transport Method: tls
   IMS Enabled? n

    Near-end Node Name: clancrm              Far-end Node Name: ses
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                        Far-end Network Region: 1
 Far-end Domain: avaya.com
                                          Bypass If IP Threshold Exceeded? n
 Incoming Dialog Loopbacks: eliminate              RFC 3389 Comfort Noise? n
          DTMF over IP: rtp-payload       Direct IP-IP Audio Connections? y
 Session Establishment Timer(min): 3              IP Audio Hairpinning? n
          Enable Layer 3 Test? n                Direct IP-IP Early Media? n
 H.323 Station Outgoing Direct Media? n       Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. All calls from the Hammer G5 will be routed over this trunk group. Each call will use two trunk members one for the incoming call and another one for the outgoing call, which routes the call back to the Hammer G5.

```
add trunk-group 10                                          Page   1 of  21
                              TRUNK GROUP

 Group Number: 10                    Group Type: sip        CDR Reports: y
   Group Name: To SES                     COR: 1      TN: 1       TAC: 1010
    Direction: two-way      Outgoing Display? n
 Dial Access? n                                      Night Service:
 Queue Length: 0
 Service Type: tie                   Auth Code? n

                                                    Signaling Group: 10
                                                  Number of Members: 60
```

5 Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of Avaya Aura™ SIP Enablement Services. SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter *http://<ip-addr>/admin* as the URL in an Internet browser, where *<ip-addr>* is the IP address of SIP Enablement Services. Log in with the appropriate credentials and then navigate to **Administration→SIP Enablement Services** from the next screen. The main screen is displayed below.

From the left pane of the Administration web interface, expand the **Server Configuration** item and select **System Properties**. In the **System Properties** screen, enter the domain name assigned to the Avaya SIP-based network and the SIP License Host. For the **SIP License Host** field, enter the fully qualified domain name or the IP address of the SIP Enablement Services server that is running the WebLM application and has the associated license file installed. This entry should always correspond to the localhost unless the WebLM server is not co-resident with this server. After configuring the **System Properties** screen, click the **Update** button.

After setting up the domain in the **System Properties** screen, create a host entry for SIP Enablement Services. The following example shows the **Edit Host** screen since the host had already been configured. Enter the IP address of SIP Enablement Services in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, configure the **Host Type** field. In this example, the host server was configured as an *SES combined home-edge*. The default values for the other fields may be used as shown below. Click the **Update** button.

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya S8730 Servers since a SIP trunk is required between Communication Manager and SIP Enablement Services. In the **Communication Manager Interface** screen, enter the following information:

- A descriptive name in the **Communication Manager Server Interface Name** field (e.g., *devcon13*).
- Select the home server in the **Host** field.
- Select *TLS* (Transport Link Security) for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field.

Refer to [3] for additional information on configuring the remaining fields.

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Incoming calls originating from the Hammer G5 and arriving at SIP Enablement Services are routed to Communication Manager for termination services. Calls to be routed to Communication Manager are specified in a **Communication Manager Server Address Map**. The Uniform Resource Identifier (URI) of an incoming INVITE message is compared to the pattern configured in the address map, and if there is a match, the call is routed to Communication Manager. The URI usually takes the form of `sip:user@domain`, where `domain` can be a domain name or an IP address. In this example, `user` is actually the number that was dialed by the Hammer G5. An example of a URI would be `sip:825200@10.32.24.20`.

To configure a **Communication Manager Server Address Map**, select **Communication Manager Servers** in the left pane of the Administration web interface. This will display the **List Communication Manager Servers** screen. Click on the **Map** link associated with the appropriate server to display the **List Communication Manager Server Address Map** screen and click on the **Add Map In New Group** link. The screen below is displayed. Provide a descriptive name in the **Name** field and enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the pattern specification matches a URI that begins with `sip:8252` followed by two more digits. The following screen shows a map entry that was already configured. A new entry would be added by clicking the **Add** button.

After the **Communication Manager Server Address Map** is added, the first **Communication Manager Server Contact** is created automatically. For the address map added above, the following contact was created:

```
sip:$(user)@10.32.24.20:5061;transport=tls
```
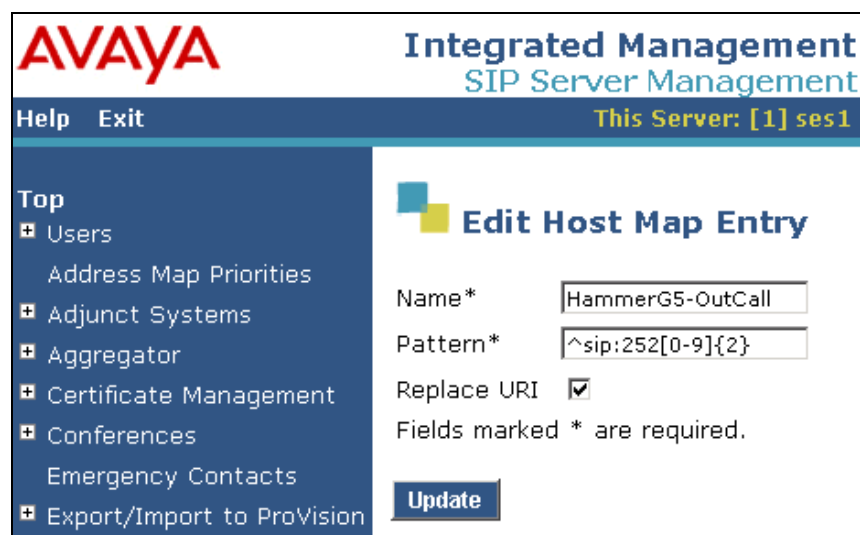
The contact specifies the IP address of the C-LAN board in the Avaya G650 Media Gateway and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for `$(user)`.

After configuring the address map, the **List Communication Manager Server Address Map** screen appears as shown below.

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After incoming calls are processed by Communication Manager, the calls are routed back to SIP Enablement Services, which will route the calls back to the Hammer G5, where the calls are terminated. SIP Enablement Services requires a host address map.

To access the **Add Host Address Map** screen, select the **Hosts** link in the left pane of the Administration web interface and then click on the **Map** link associated with the appropriate host (e.g., 10.32.24.100). The **List Host Address Map** screen is displayed. From this screen, click the **Add Map In New Group** link to display the screen shown below. Configure a descriptive **name** for the map and specify an appropriate **pattern** for the call type. In this example, the pattern is used to route calls starting with "252". By default, the **Replace URI** checkbox is selected. Click the **Add** button.

From the **List Host Address Map**, click on the **Add Another Contact** link associated with the address map added above. In this screen, the **Contact** field specifies the destination for the call and it is configured as:

```
sip:$(user)@10.32.24.171:5060;transport=udp
```

The contact specifies the IP address of the Hammer G5 and the transport protocol used to send SIP signaling messages. The user in the original request URI is substituted for $(user). Click the **Add** button when completed.

After configuring the host address maps, the **List Host Address Map** screen appears as shown below.

The last step is to configure the Hammer G5 as a trusted host on SIP Enablement Services. As a trusted host, SIP Enablement Services will not issue SIP authentication challenges for incoming requests from the Hammer G5. Specify the IP address of the Hammer G5 in the **IP Address** field and set the **Host** field to the IP address of SIP Enablement Services. A descriptive comment can be provided in the **Comment** field. Two entries were added for the Hammer G5, one for the SIP trunk interface that originates calls and another one for the SIP trunk interface that terminates the calls.

6 Configure Empirix Hammer G5

This section provides the procedures for configuring the Empirix Hammer G5. The procedures fall into the following areas:

- Configure IP addresses to be used by each Hammer G5 channel.
- Configure the SIP trunk interface using the **Hammer Configurator**.
- Create and run the test script using the **Hammer TestBuilder**.

Monitoring test call progress is covered in the **Verify Empirix Hammer G5** section (Section 0).

Note: Refer to [5] for instructions on installing the Hammer G5 software.

6.1 Configure IP Addresses on Hammer G5 Server

The Hammer G5 server needs to be configured with the IP addresses that it will need for each channel. During the compliance test, 20 SIP trunk channels were used, but they were arranged into two groups – one for outgoing calls and one for incoming calls. Each group required a unique IP address. Only two IP addresses were required, one per group. The IP address used for the SIP trunks handling outgoing calls was 10.32.24.170. The IP address used for the SIP trunk handling incoming calls was 10.32.24.171. These IP addresses were configured in the **Advanced TCP/IP Settings** under Network Connections in Windows 2003 Server.
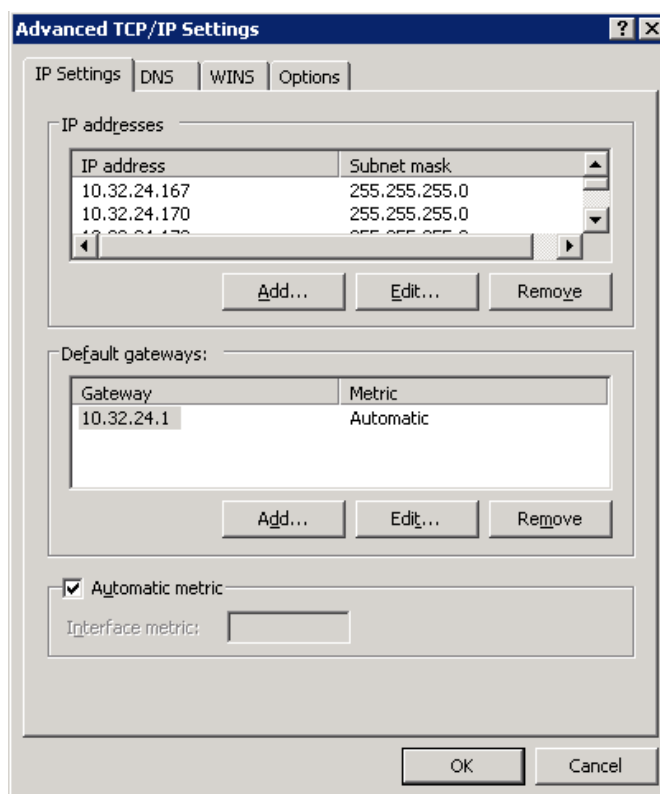
6.2 Configure SIP Endpoint Interface

The Empirix Hammer G5 is configured through the **Hammer Configurator**, a graphical user interface, residing on the Hammer G5 server. From the Hammer G5 server, run the **Hammer Configurator**. The initial window is displayed as follows.
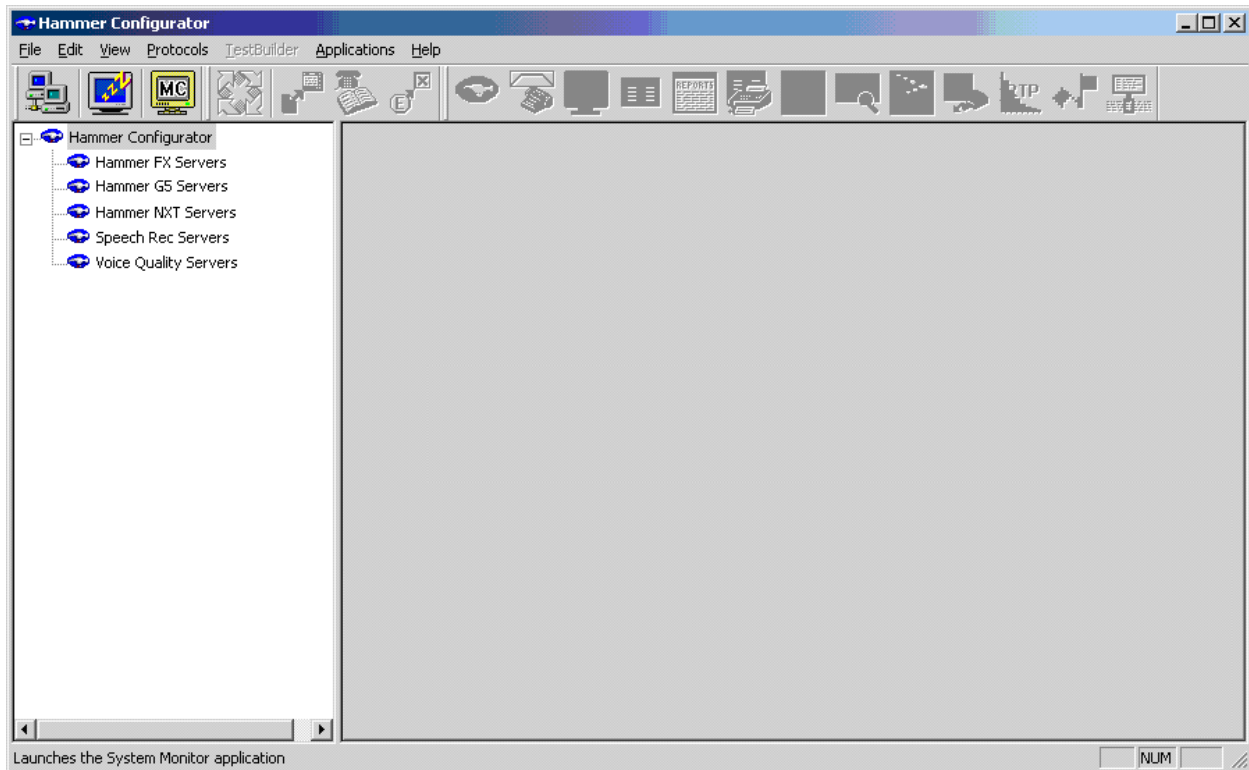
The Hammer G5 must be in **Master Controller Mode**. Verify that it is in Master Controller Mode by clicking the [MC] icon. If the [MC] icon is not displayed, the Hammer G5 is already in Master Controller Mode.

Next, select Hammer G5 Servers item in the left pane and then click on **Add a new system**. The **Add/Remove Servers** window will be displayed.
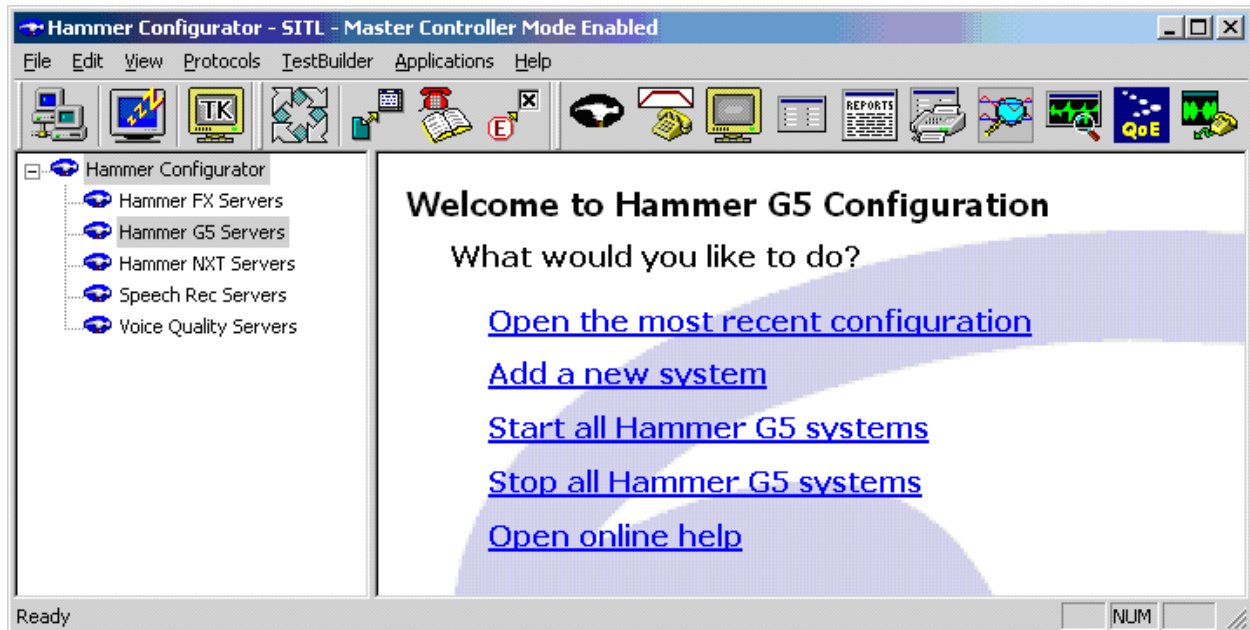
In the **Add/Remove Servers** window, specify a descriptive **Server Name** and the click the **Add** button. The server name will be moved to the **Current Servers** section. Click **OK**.
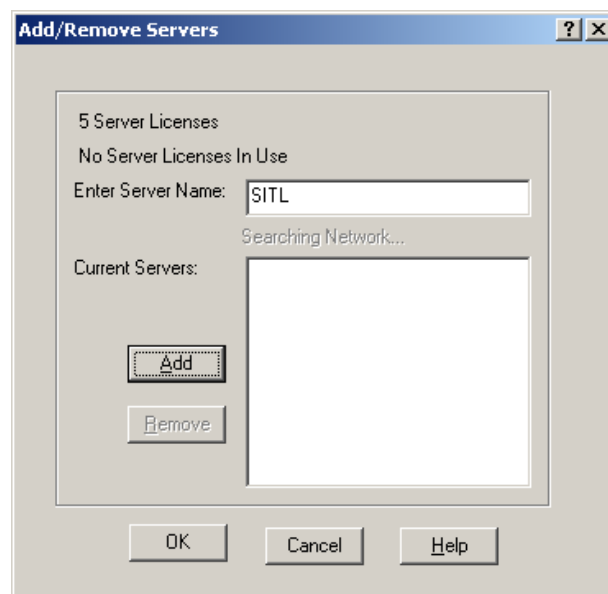
Once the server is added to the system, the server name will appear in the left pane of the **Hammer Configurator**. Expand the server name (e.g., *SITL*) in the left pane and click on *IP Channels Configuration*. The following window will be displayed. Select *Avaya SIP* for the **Signaling Project** and then click **New**.
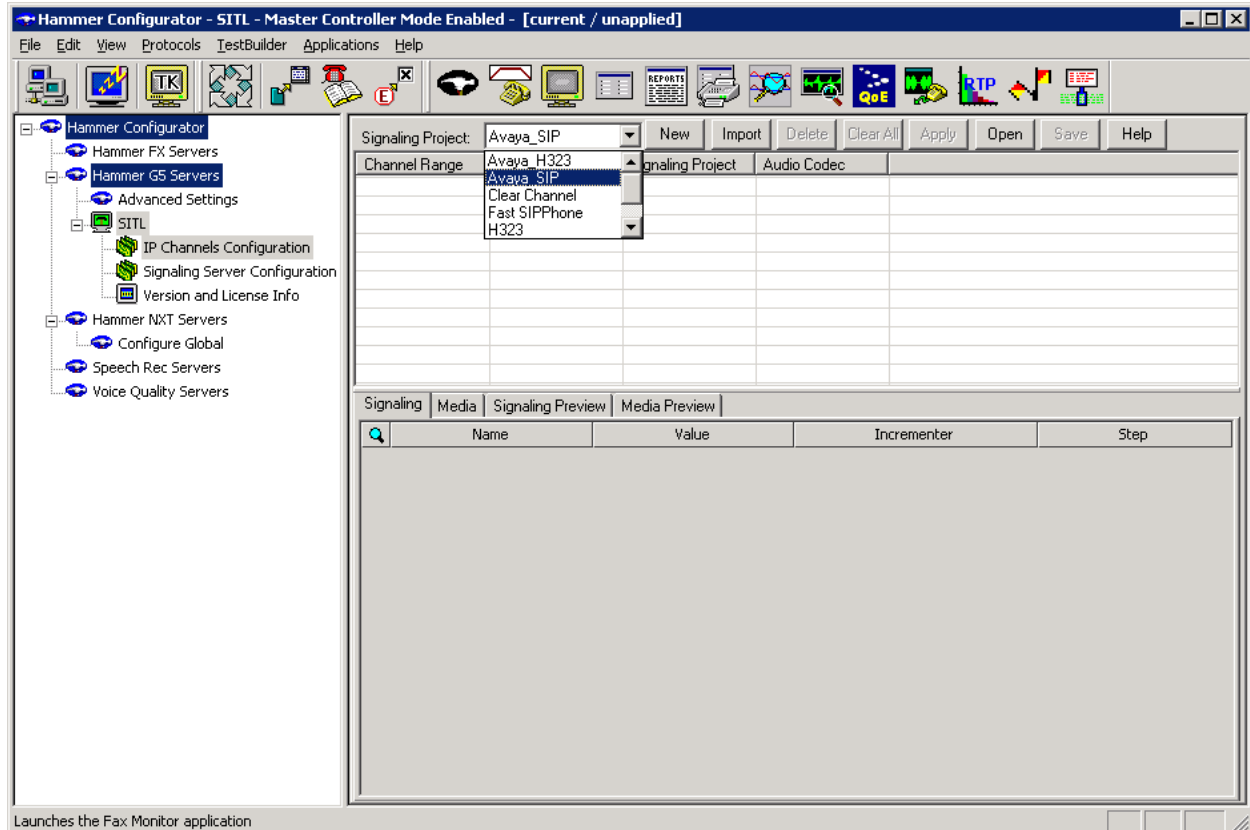
The first line in the grid that is highlighted in the figure below corresponds to the 10 originating channels. To set the number of channels in the group, click on the **Channel Range** cell in the grid and enter the number "10". The following fields in the **Signaling** tab should then be set as follows:

- **State Machine** should be set to *Avaya SIP Trunk*.
- **Trunk ANI** can be set to any digits.
- **Network Connection** should be set to the appropriate network interface on the server.
- **Local Trunk IP** should be set to the IP address of the group (e.g., 10.32.24.170)
- **Subnet Mask** should be set to the network mask (e.g., 255.255.255.0).
- **Avaya SES IP Address** should be set to the IP address of SIP Enablement Services (e.g., 10.32.24.100).
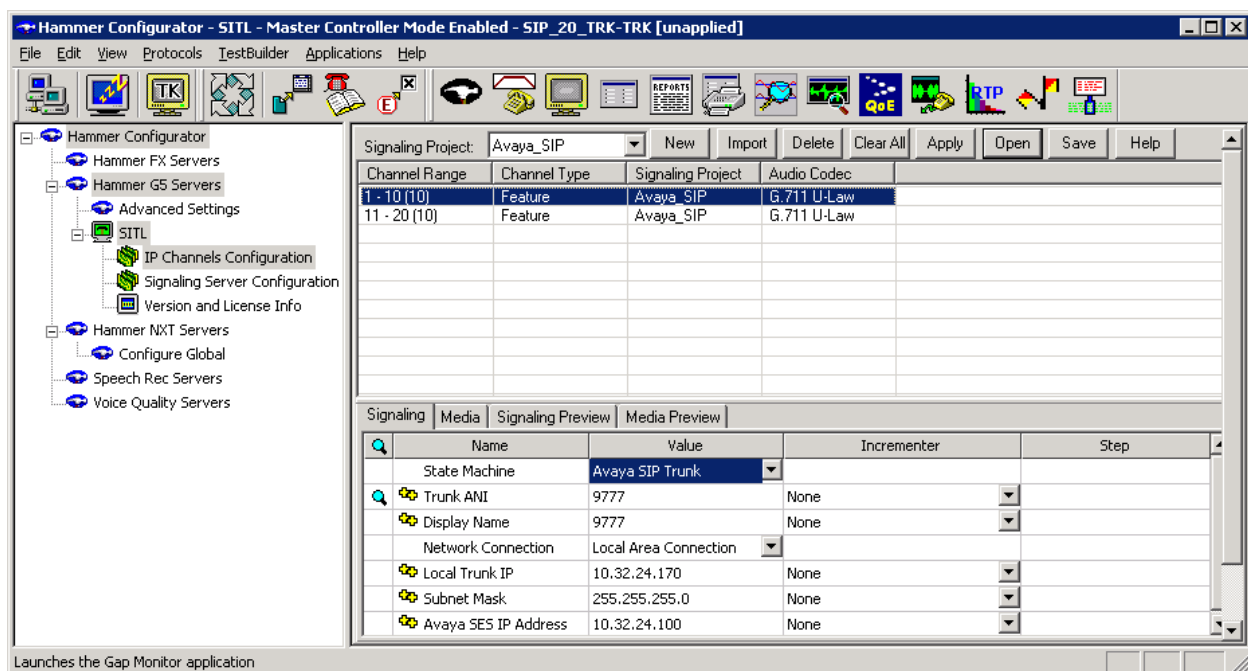
In the **Media** tab of the 10 originating channels, configure the fields as follows:

- **Audio Codec** should be set to the appropriate codec for the test. G711, G729, and G.723 codecs are supported.
- **Frequency [ms]** should be set to the appropriate value for the specified codec. It should match the Packet Size [ms] field in the **IP Codec Set** form on Communication Manager for the specified codec.
- **Network Connection** should specify the appropriate network interface.
- **Source IP Address** should be set to the IP address of the first channel in the group.
- **Media Profile** should be set to one that specifies the codec configured in the **Audio Codec** field.
- The remaining fields should be configured as shown.

The second line in the grid that is highlighted in the figure below corresponds to the second group of channels that will terminate the calls. Set the **Channel Range** cell to the number of channels in this group. The configuration of the **Signaling** tab is similar to the one for the group of originating channels with the exception that the **Trunk ANI** and **Local Trunk IP** fields will be different. This group of channels will be assigned IP address 10.32.24.171.
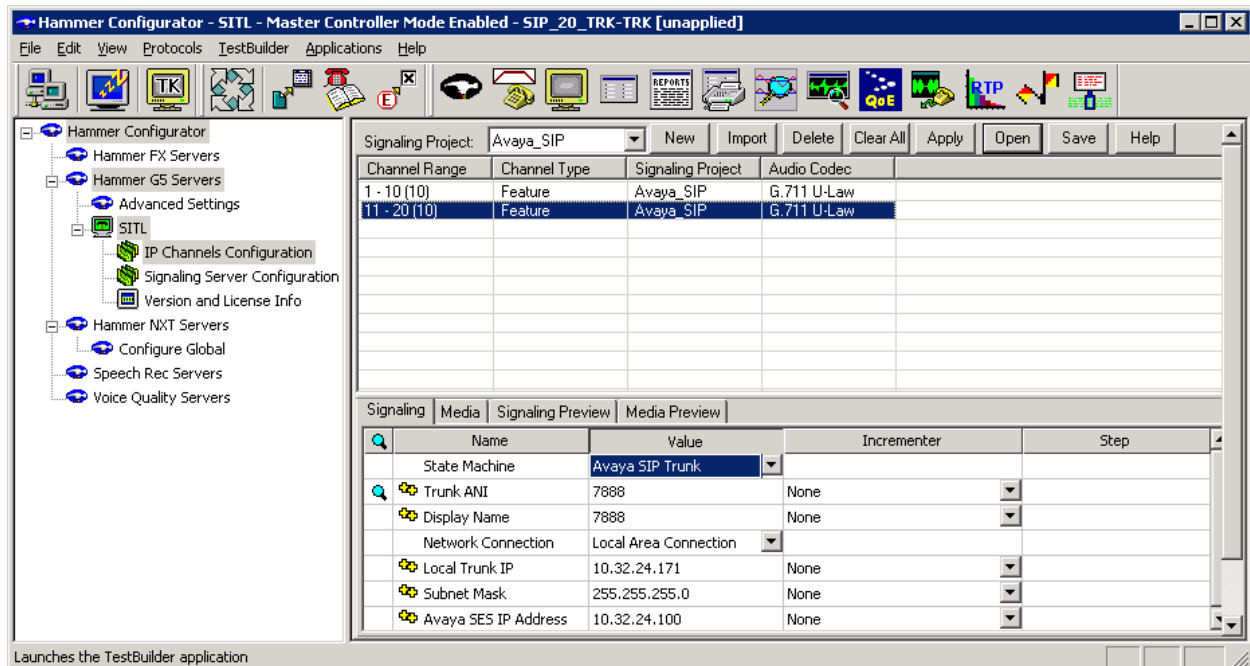
The **Media** tab for the group of terminating channels is shown below. The configuration is similar to the one for the group of originating channels except for the **Source IP Address** field.

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24 of 44
Hammer-SIP-TRK
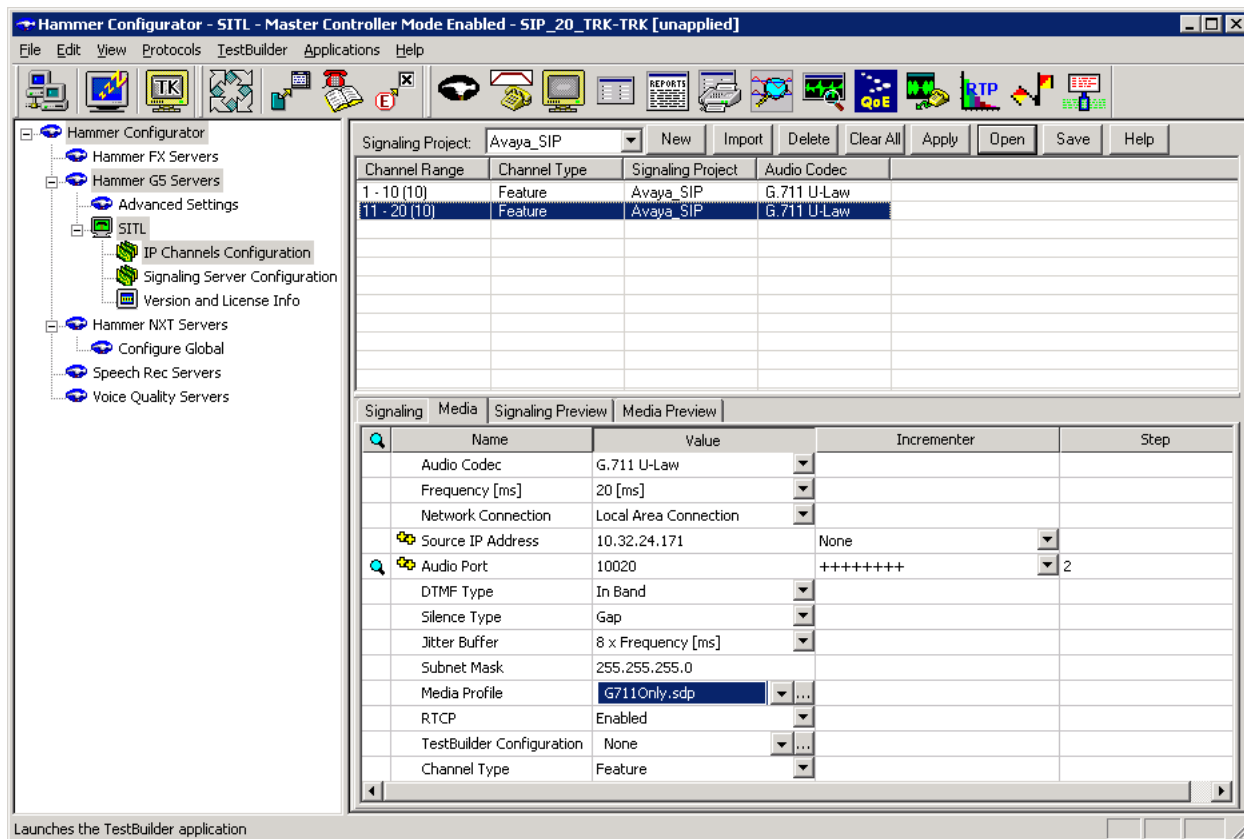
The following windows show the configuration of the **Media Profile** used in the **Media** tab above for the two channel groups. To access this window, click on the ellipses button (…) in the **Media** tab. By default, there are **Media Profiles** for each audio codec type so this configuration is not required since the profiles are available out-of-box. It is presented for informational purposes. Click on the **Audio Description** button to view the codecs that will be advertised by the Hammer G5 when placing a call.

The following window shows the codecs selected for this profile. This **Media Profile** was already created and named G711Only.sdp. It specifies G.711U and RFC 2833. When done, click **OK** to return to the previous window. Additional media profiles can be created and saved by selecting the desired codecs in this window and then clicking the **Save** button in the previous window.

This completes the configuration of the SIP trunk interface. This configuration should be saved by clicking the **Save** button on the **Hammer Configurator** window.

The configuration needs to be applied to the server for the changes to take effect. Click on the **Apply** button in the **Hammer Configurator** window. The following window is displayed as the configuration is being applied to the server.
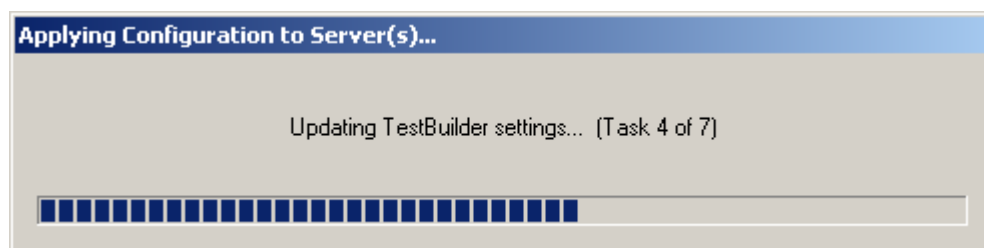

Check that the system has been started by clicking on the server name (e.g., SITL) in the left pane of the **Hammer Configurator**. If the current status is "System Is Stopped", click the **Start system** button to start the system. When the system is started, it should appear as shown below and should also specify which configuration has been applied. The configuration performed above was saved as "SIP_20_TRK-TRK".
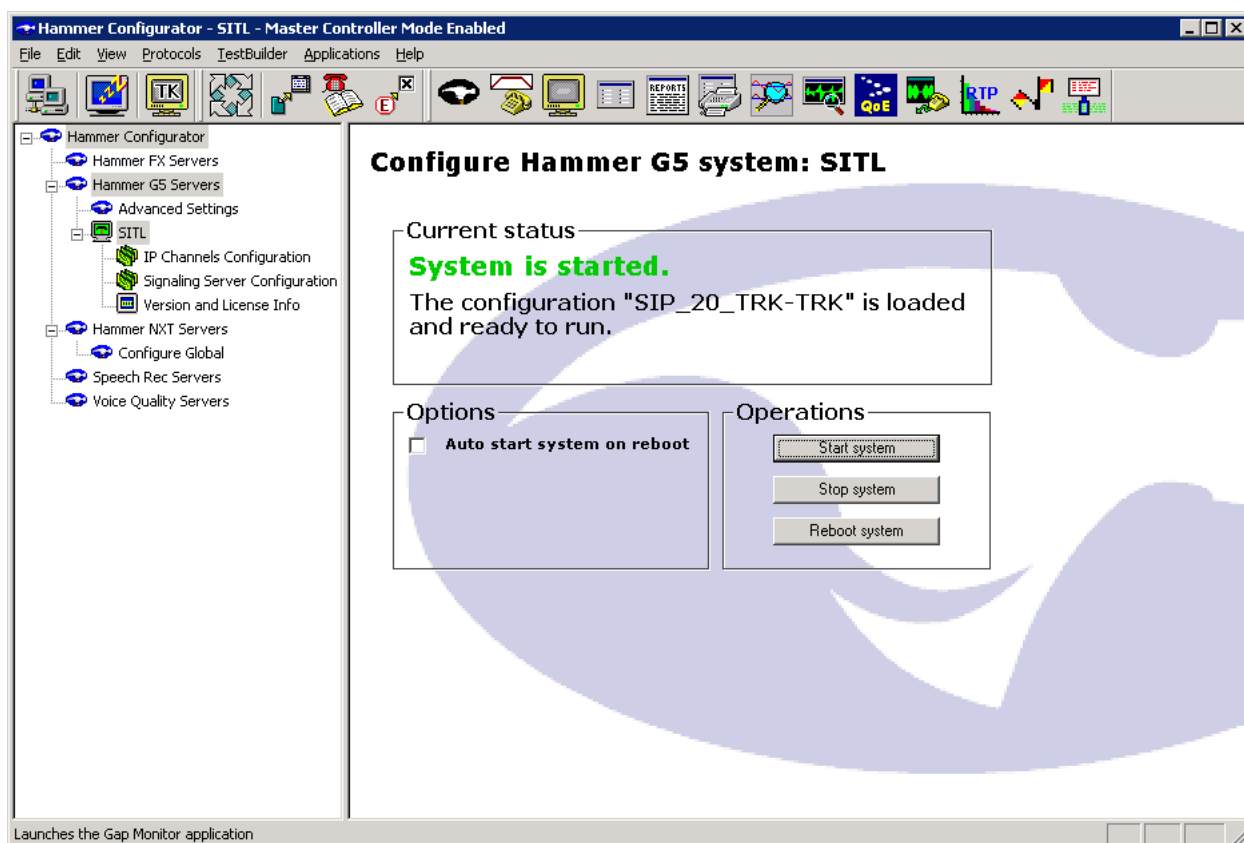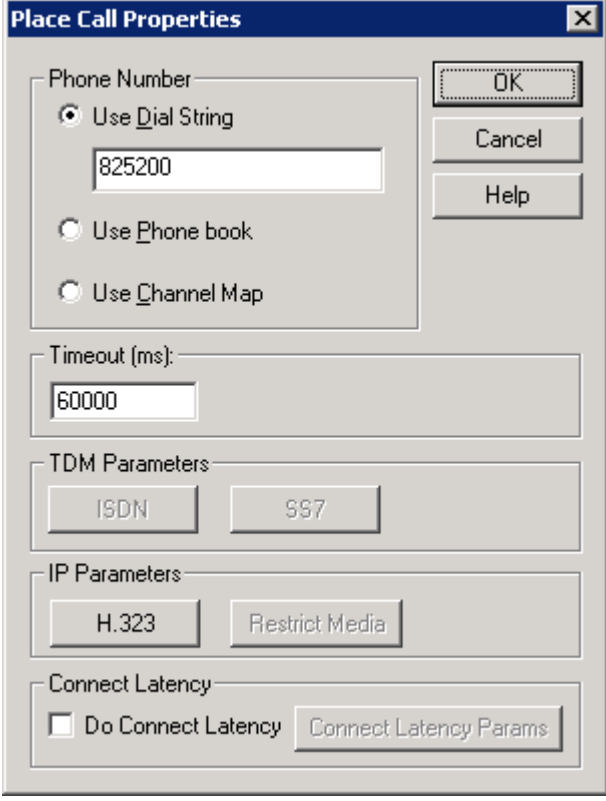
6.3 Configure and Run the Test Script

This section covers the configuration of a test script. The sample test script covered here establishes a VoIP call between two SIP trunks on the Hammer G5, followed by the originating side playing an audio prompt to the far-end so that voice quality metrics (e.g., PESQ score) can be obtained. The test script is configured with the **Hammer TestBuilder** application and is displayed in a ladder diagram shown below. Each action in the test script can be configured by double clicking on it.   For detailed instructions on how to create the test script, refer to [6]. Each test step is examined below as a brief tutorial, but test scripts will vary depending on the customer's requirements.
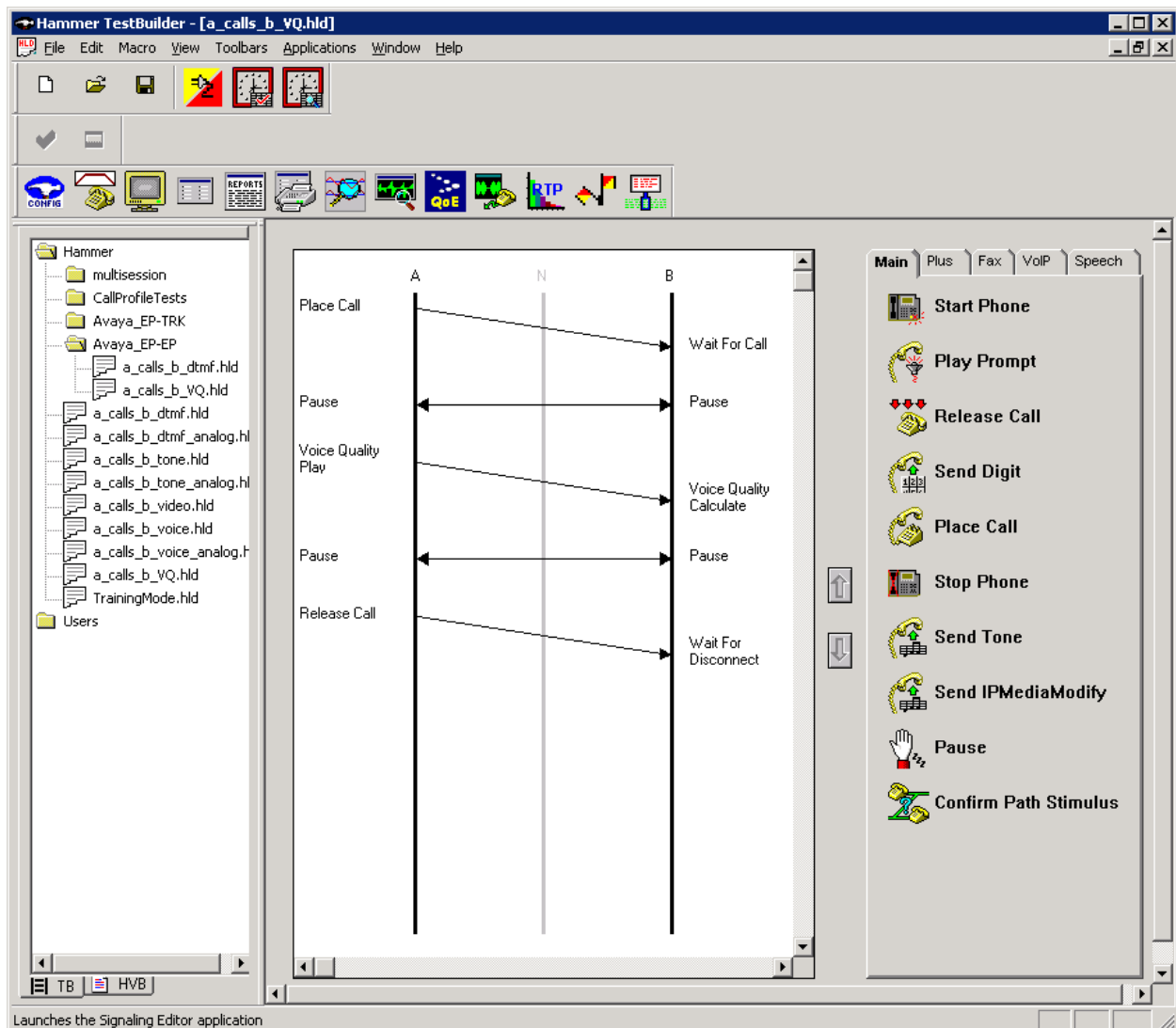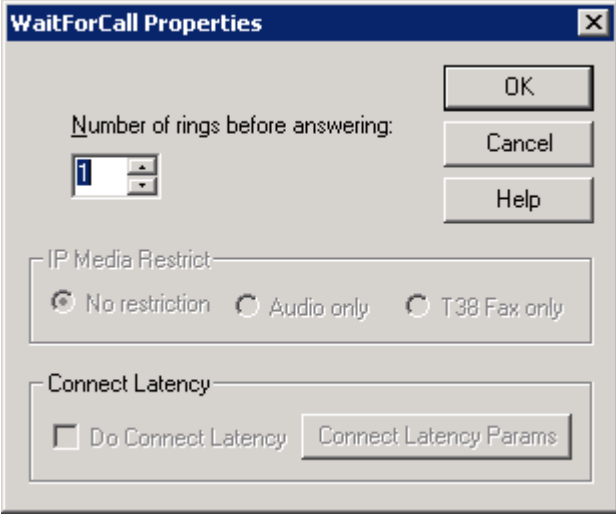
In the sample test script configured above, the A-side (originating SIP trunk) places a call to the B-side (terminating SIP trunk) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. The **Place Call Properties** is configured to dial "825200" for all calls. The '8' corresponds to the AAR access code on Communication Manager and "25200" is the dialed number that is used to route the call.
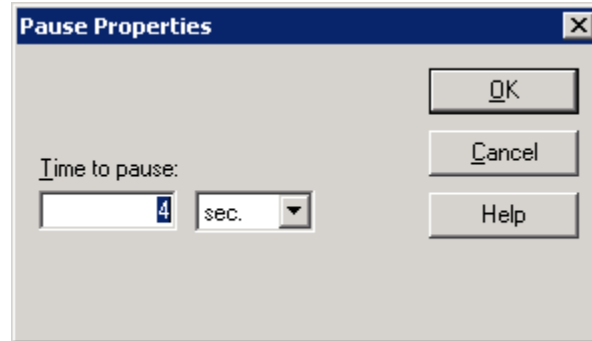

The B-side then waits for one ring before answering the call as configured in the **Wait for Call** action.

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After the call is established, the A-side pauses for 4 seconds before proceeding.
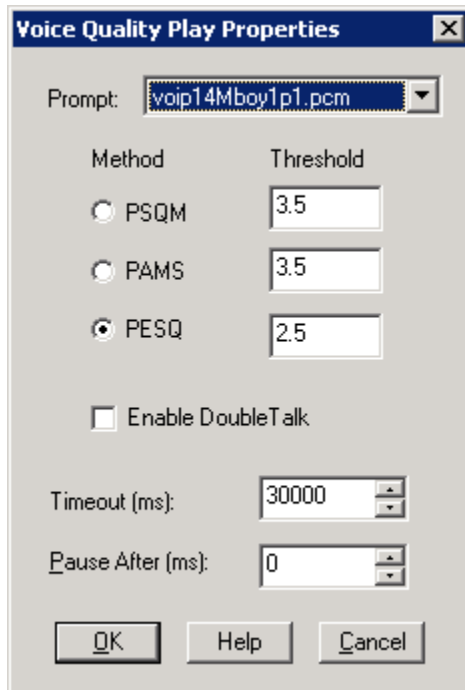

After the call is established, the B-side pauses for 2 seconds before proceeding. The B-side waits for less time because it needs to start listening for the audio prompt before the A-side starts to play it.
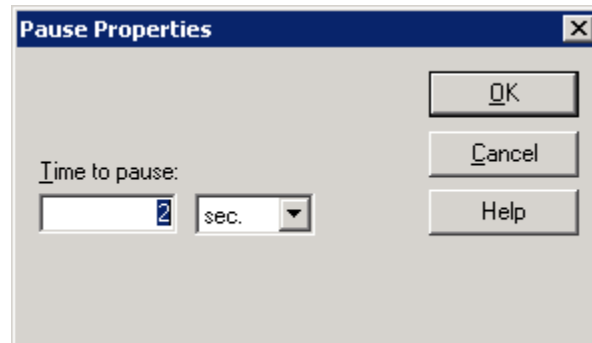
Important Note: In order for the call to succeed, the B-side needs to start listening for the audio prompt before the A-side starts to play it. During the compliance test, the B-side started to listen for the prompt about 2 seconds before the A-side started to play it. This can be controlled by the length of the **Pause** step described above. However, the B-side should not start listening too early (e.g., 5 seconds before the A-side starts playing the prompt or the call will most likely fail).
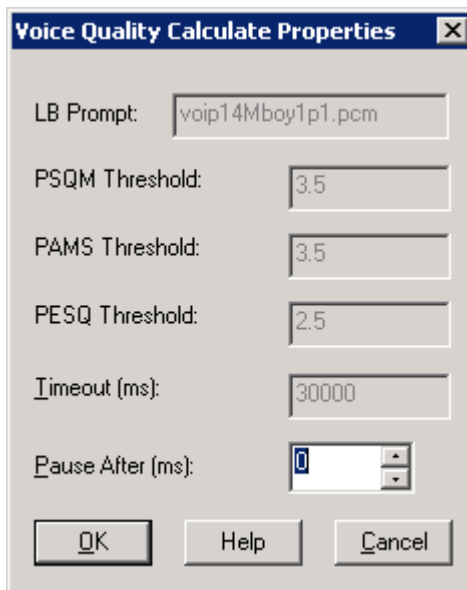
After the pause by the A-side, it starts to play the audio prompt specified in the **Prompt** field of the **Voice Quality Play Properties** window. The action properties also specify that a PESQ score should be provided.

The B-side starts to listen to prompt to calculate the voice quality score.

After the audio prompt is played the voice quality score is calculated, the A-side pauses for 3 seconds as shown below before releasing the call.
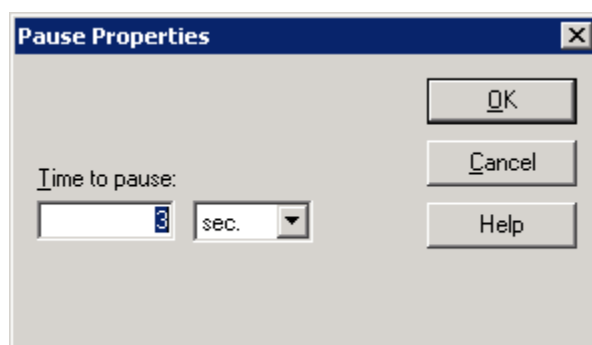
The B-side pauses for 1 second before releasing the call.
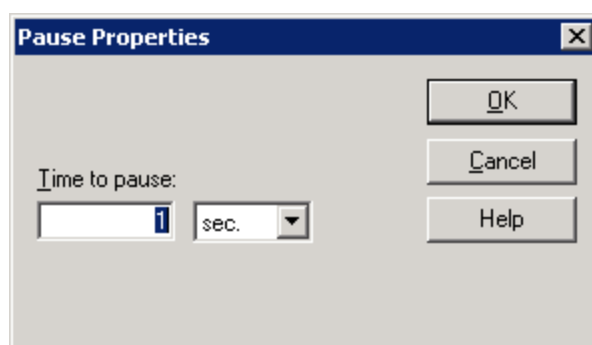
The last step of the test script is to release the call.
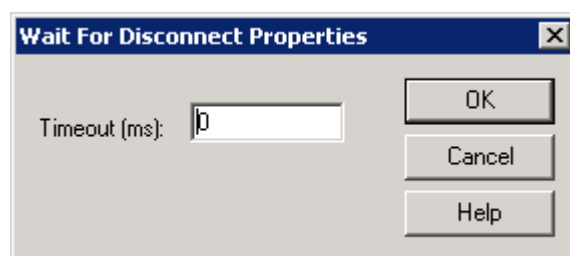
Once the test script is configured, save it from the **Hammer TestBuilder** application. Anytime a change is made to the test script, it needs to be saved before it can be run. In this example, the test script was saved as "a_calls_b_VQ.hld".

To run the test, right-mouse click on the test script in the left pane of the **Hammer TestBuilder** window and navigate to **Schedule→Edit & Run**. To re-run the test, the user can simply select **Schedule→Run**, if no changes are required.
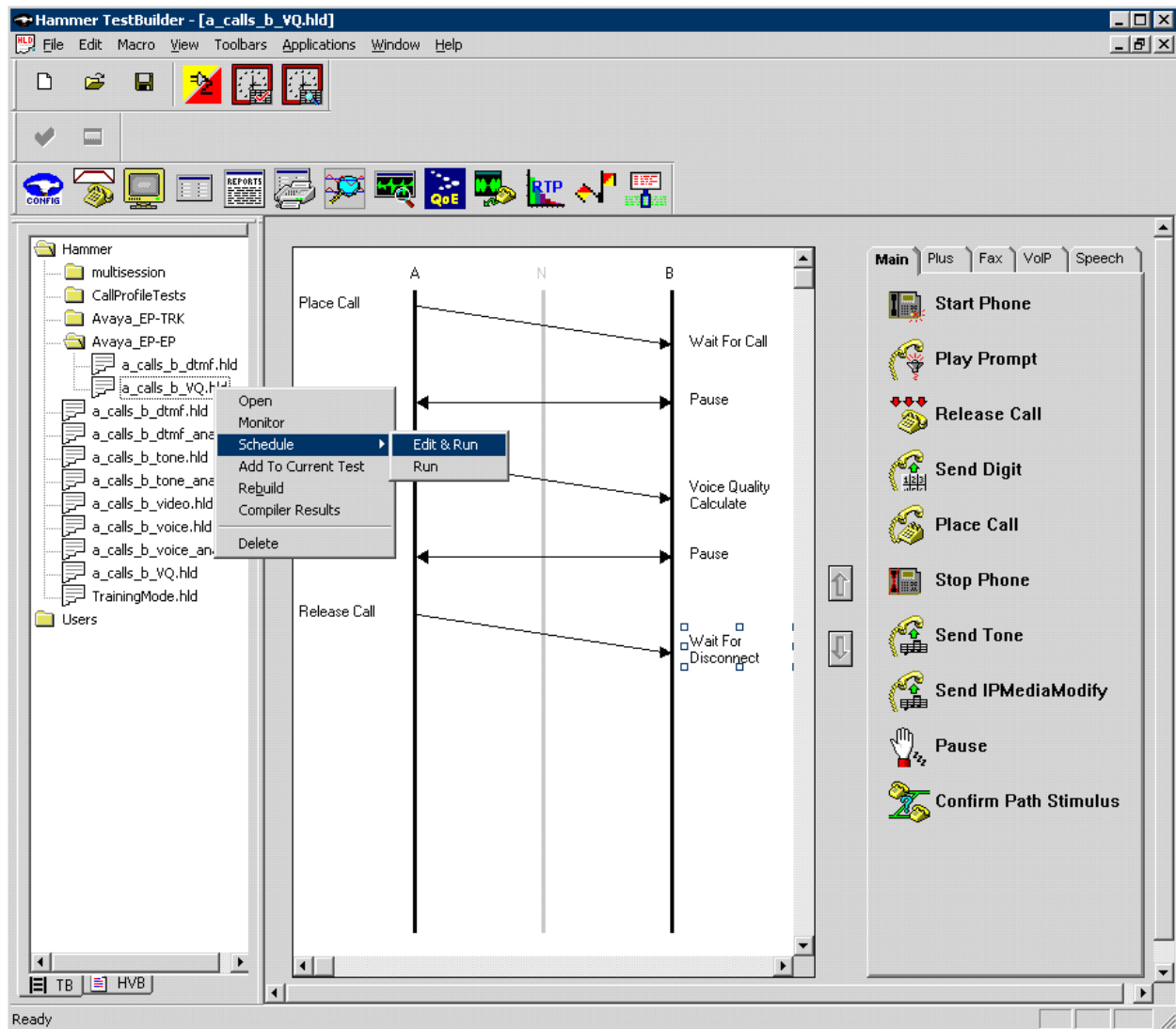
In the **Properties** window, set the **Loop Count** to the appropriate value to control the number of times the test should run. Setting this field to "-1" will allow the test to run forever. Setting this field to a specific number will run the test for the many iterations and then stop. The **Guard Time (ms)** field specifies how long to wait before the test is run again on the same channel. The **Stagger** section allows the user to specify how long to wait before the test is run on the next channel. Next, click on the ellipses button (…) in the **Channels** section to assign channels to the **A-Side** and **B-Side**.

Important Note: The **Guard Time** and **Stagger** parameters should be carefully considered for every test. A test script could fail because the configuration under test cannot handle the load generated by the Hammer G5. These parameters can slow down the test to a rate that can be reasonably handled by the test configuration.

In the **Channel Browser**, select the channels in **Channel Group 0** and assign it to the **A-Side** by clicking on the appropriate right arrow.
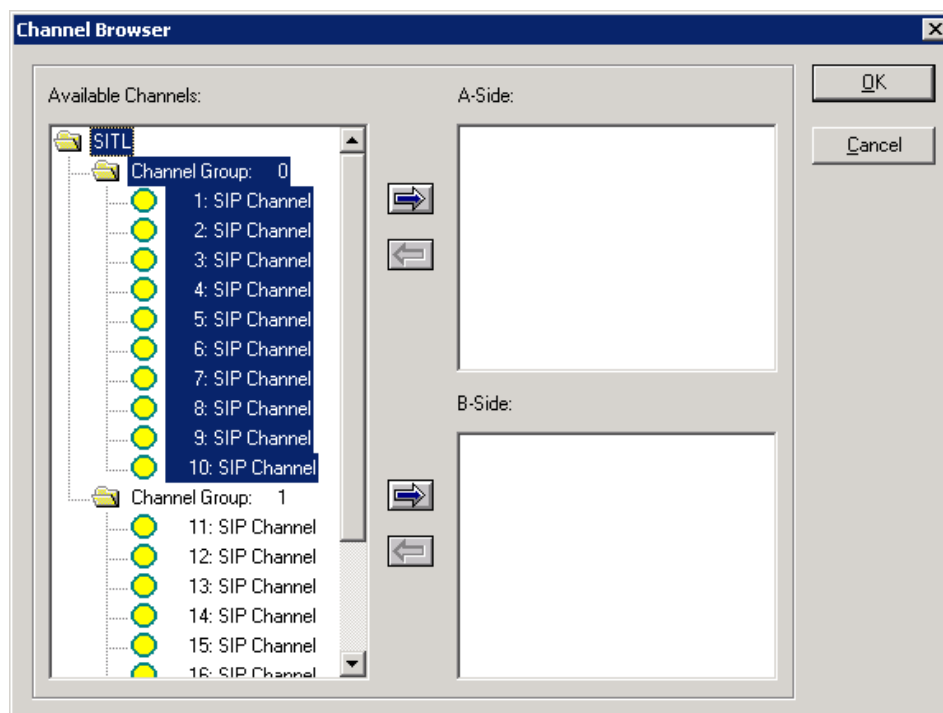
Select the channels in **Channel Group 1** and assign it to the **B-Side** by clicking on the appropriate right arrow.
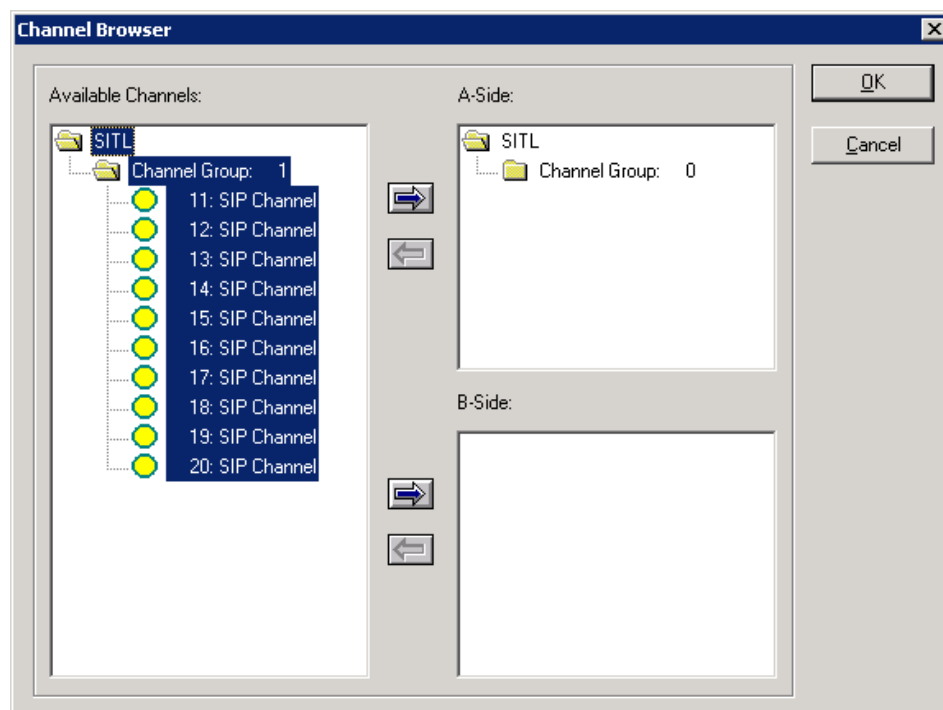
After assigning the channels to the **A-Side** and **B-Side**, the **Channel Browser** window should appear as follows. Click **OK** to return to the **Properties** window.
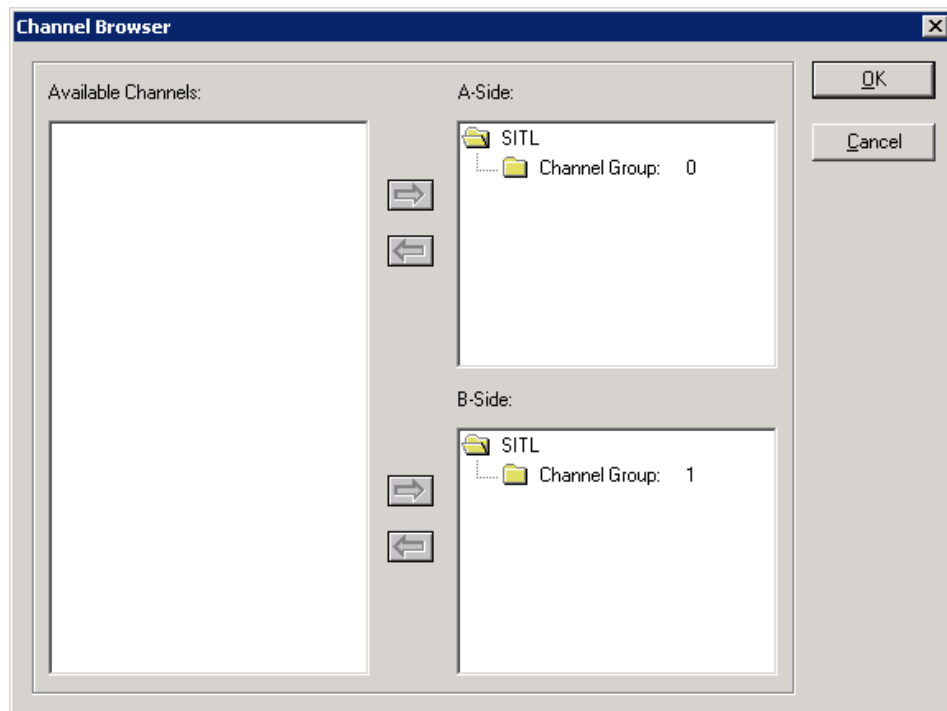
In the **Properties** window, click **OK** to start the test. Monitoring the test progress is covered in the **Verify Empirix Hammer G5** section (Section 8.2).

Helpful Tip: Once a test starts to run, if an error is encountered on a particular channel, the test will stop running on that channel unless the Hammer G5 is configured to continue running. To allow a test to continue running after an error occurs, modify the **Error Settings** by clicking on the ⊠ icon on the **Hammer Configurator**. Click on the **HC** checkbox for each error type. This will allow the test to hang up and continue when an error occurs.

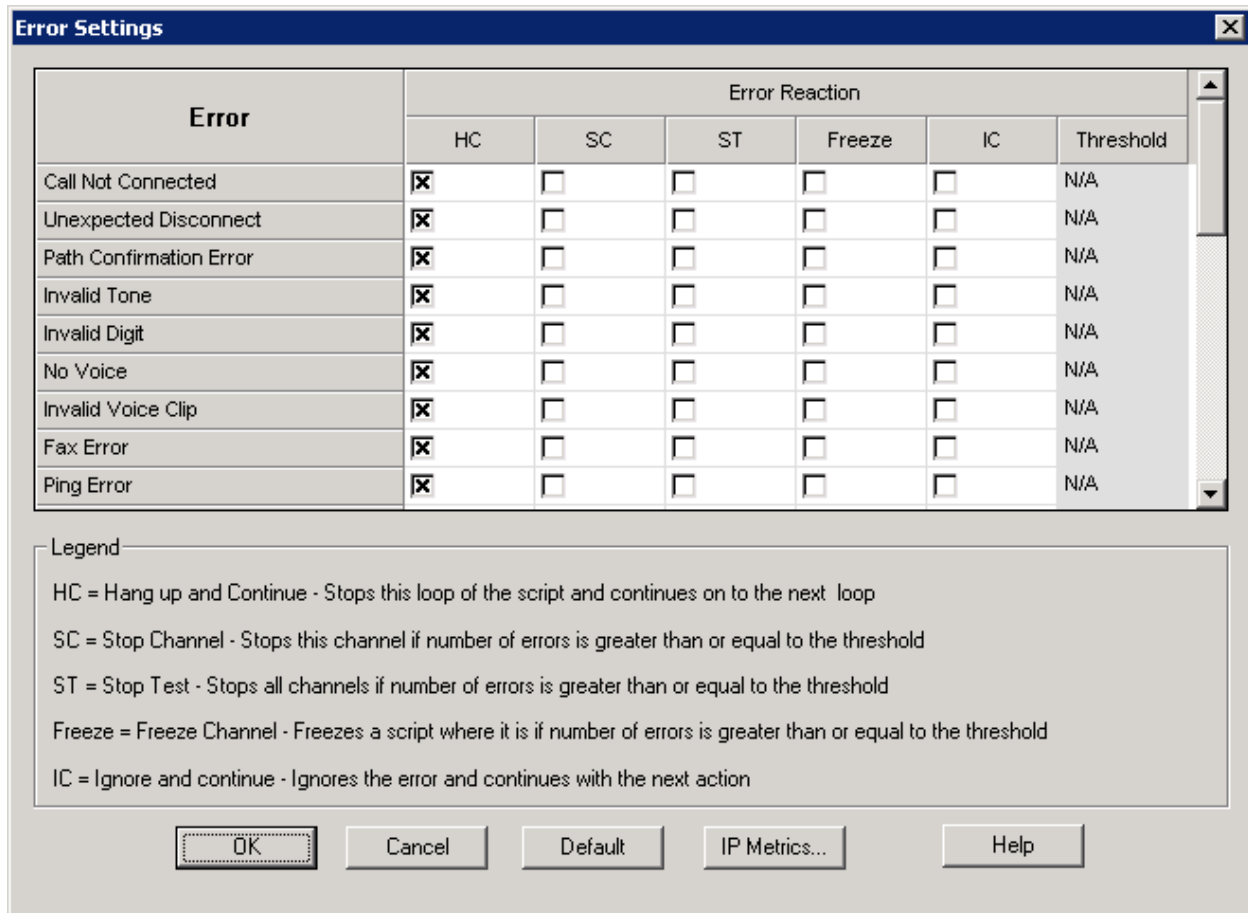| Error	Error Reaction					
	HC	SC	ST	Freeze	IC	Threshold
Call Not Connected	☒	☐	☐	☐	☐	N/A
Unexpected Disconnect	☒	☐	☐	☐	☐	N/A
Path Confirmation Error	☒	☐	☐	☐	☐	N/A
Invalid Tone	☒	☐	☐	☐	☐	N/A
Invalid Digit	☒	☐	☐	☐	☐	N/A
No Voice	☒	☐	☐	☐	☐	N/A
Invalid Voice Clip	☒	☐	☐	☐	☐	N/A
Fax Error	☒	☐	☐	☐	☐	N/A
Ping Error	☒	☐	☐	☐	☐	N/A

Legend

HC = Hang up and Continue - Stops this loop of the script and continues on to the next loop

SC = Stop Channel - Stops this channel if number of errors is greater than or equal to the threshold

ST = Stop Test - Stops all channels if number of errors is greater than or equal to the threshold

Freeze = Freeze Channel - Freezes a script where it is if number of errors is greater than or equal to the threshold

IC = Ignore and continue - Ignores the error and continues with the next action

[OK]   [Cancel]   [Default]   [IP Metrics...]   [Help]

7 General Test Approach and Test Results

Interoperability compliance testing covered feature and serviceability testing. The feature testing was conducted by originating and terminating calls using the SIP trunk channels on the Hammer G5 and establishing the calls through the Avaya SIP telephony network. The interoperability compliance testing focused on the following areas:

- Generation of moderate call load from the Hammer G5 to Communication Manager via the SIP trunk interface with the call being terminated on the Hammer G5.
- Support of the G.711MU, G.729AB and G.723-6.3K audio codecs.
- Voice quality as measured by PESQ scores and path confirmation.
- Support of non-direct audio, and direct audio with media shuffling.
- Recovery from adverse conditions during a load test.

The compliance test also covered monitoring various reports on the Hammer G5 during and after the test runs, and checking the status of various SIP resources on Communication Manager.

The serviceability testing focused on verifying the ability of the Hammer G5 to recover from adverse conditions, such as disconnecting the Ethernet cable. Disconnecting the Ethernet cable would obviously cause errors in the test run, but the test would continue to run if configured to do so, otherwise the test would stop running after encountering errors.

All test cases passed. Empirix Hammer G5 was successfully compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services using SIP trunk emulation.

8 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura™ Communication Manager, Aura™ SIP Enablement Services, and Empirix Hammer G5.

8.1 Verify Avaya Aura™ Communication Manager

When the Hammer G5 is running a test script, the **status trunk** command may be used to view the active call status. The trunk that is being monitored here is the trunk to SIP Enablement Services. This command should specify the trunk group and trunk member used for the call be specified.

```
status trunk 10/1                                           Page   1 of   3
                              TRUNK STATUS

 Trunk Group/Member: 0010/001            Service State: in-service/active
              Port: T00037             Maintenance Busy? no
 Signaling Group ID: 10


   IGAR Connection? no


   Connected Ports: T00082
```

Page 2 of the **status trunk** command indicates the codec being used for the call and whether the call is shuffled.

```
status trunk 10/1                                           Page   2 of   3
                           CALL CONTROL SIGNALING


Near-end Signaling Loc: 01A0417
  Signaling    IP Address                          Port
   Near-end:  10.32.24.20                          : 5061
    Far-end:  10.32.24.100                         : 5061
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:         H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct    Authentication Type: None
    Near-end Audio Loc:                    Codec Type: G.711MU
   Audio      IP Address                          Port
  Near-end:  10.32.24.175                         : 10030
   Far-end:  10.32.24.165                         : 10010

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:         Video Far-end Codec:
```

8.2 Verify Empirix Hammer G5

Call progress can be monitored in the **Hammer System Monitor**. The call log for an originating channel may be logged to the left window and the call log for a terminating channel may be logged to the right window. This call monitor is also useful for checking that the terminating channel is listening prior to the originating channel playing the prompt when a voice quality score is required.
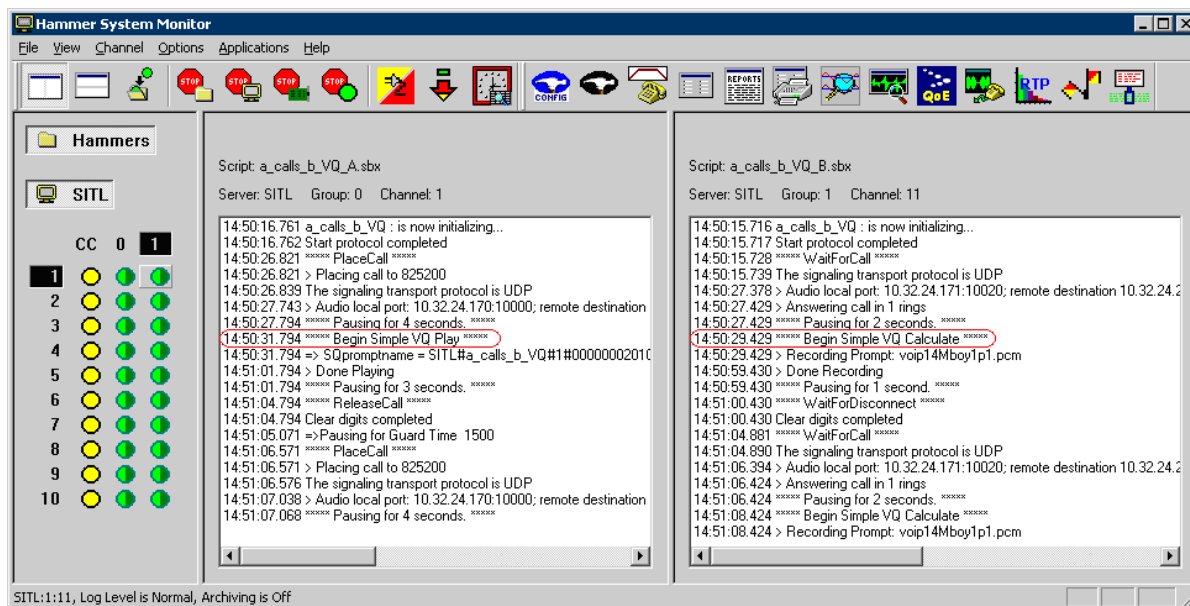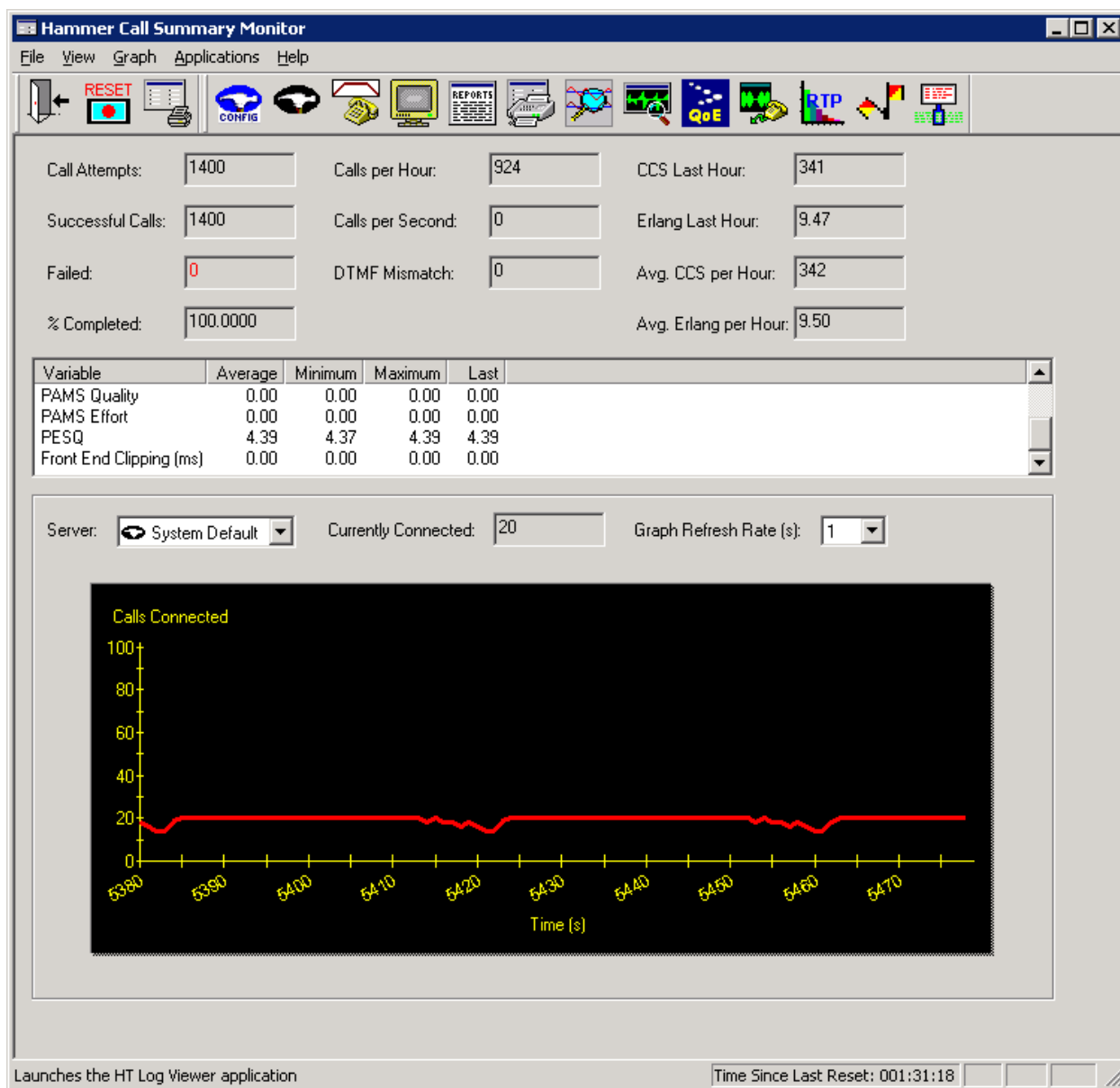

The **System Monitor** can be used to verify that a call was shuffled. This is noted by the Hammer G5 displaying the local and remote destinations in the call log.
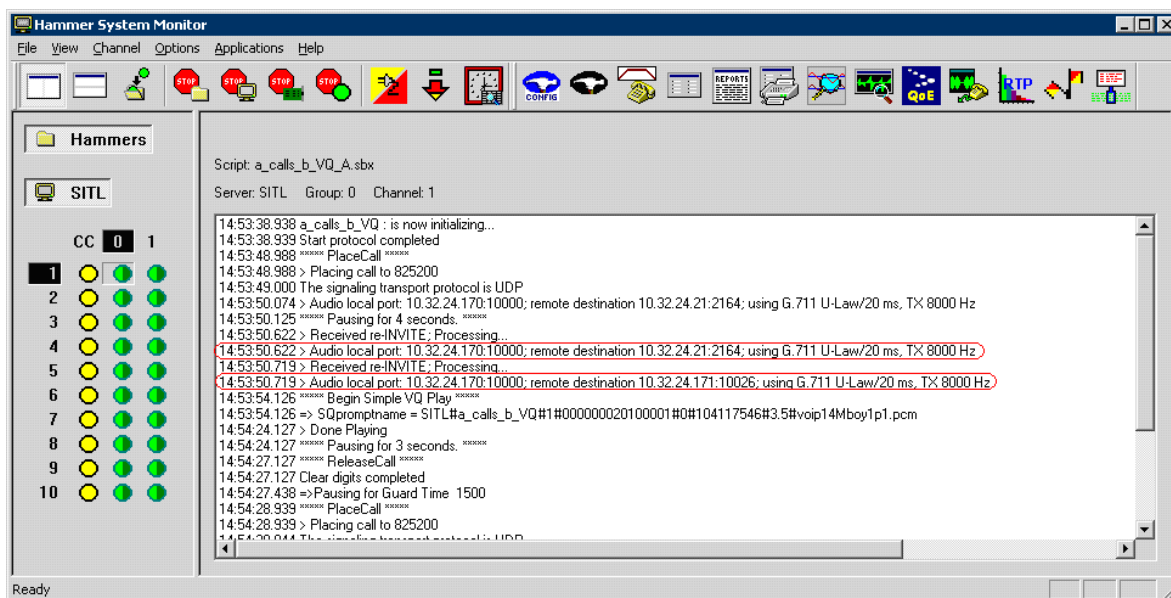
The **Hammer Call Summary Monitor** may be used to get a test status overview, including the number of call attempts, number of failed calls, PESQ scores, amongst other useful metrics.

Hammer Reports can be generated by clicking on the **Reports** icon in the **Hammer Call Summary Monitor**. Various reports are available, such as the **VQ Summary** report shown below, which displays the PESQ score for the calls on each channel.
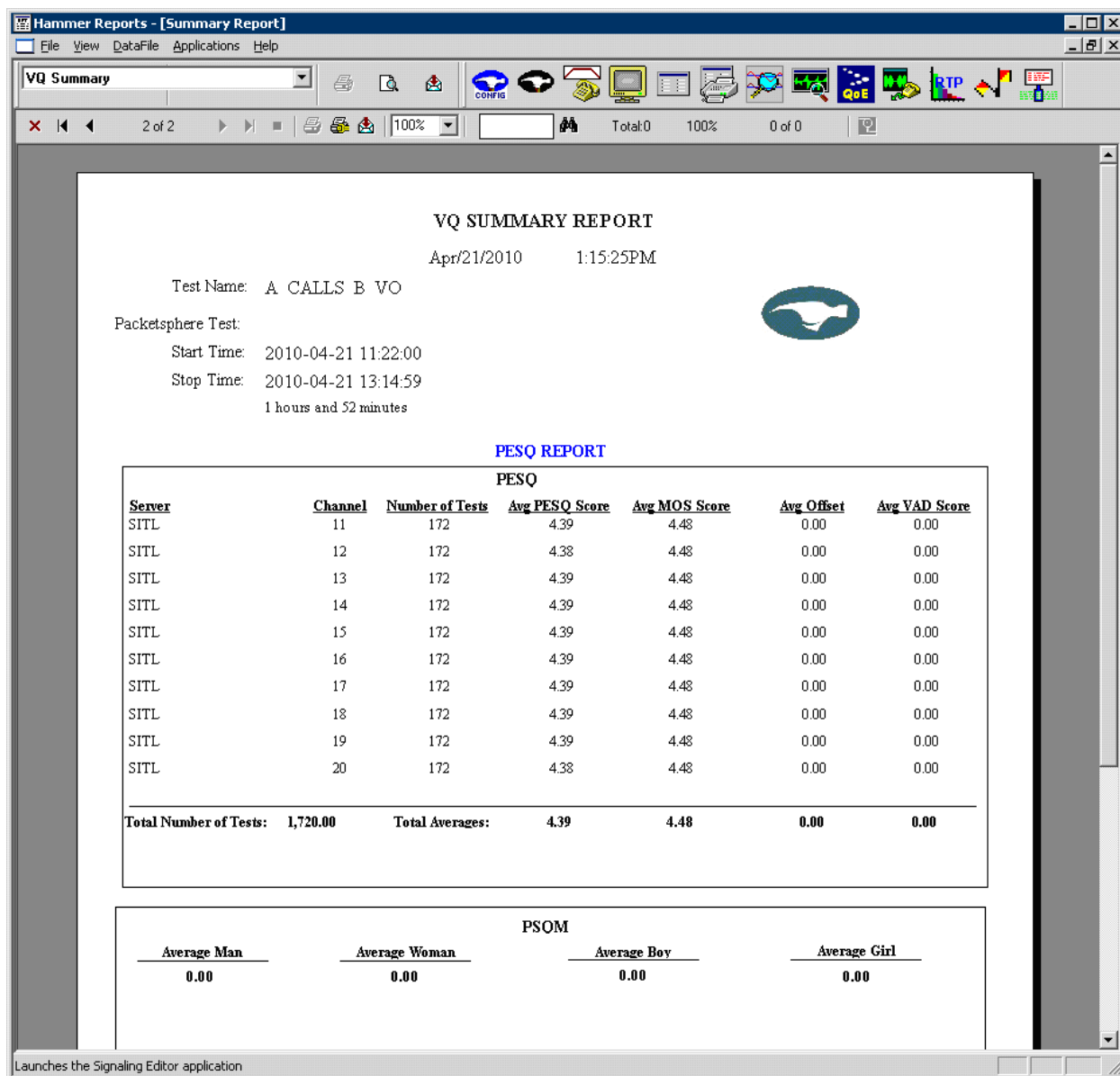
9 Conclusion

These Application Notes describe the configuration steps required to integrate the Empirix Hammer G5 with an Avaya SIP telephony network using SIP trunk emulation. The Hammer G5 was able to successfully establish calls, generate voice quality metrics, monitor the calls, and generate reports. All feature and serviceability test cases were completed successfully.

10 References

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

[1] *Administering Avaya Aura^{TM} Communication Manager*, Document 03-300509, Issue 5.0, Release 5.2, May 2009, available at http://support.avaya.com.

[2] *SIP Support in Avaya Aura^{TM} Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206, available at http://support.avaya.com.

[3] *Installing, Administering, Maintaining, and Upgrading Avaya Aura^{TM} SIP Enablement Services*, May 2009, Issue 7, Document Number 03-600768, available at http://support.avaya.com.

[4] *Application Notes for Empirix Hammer G5 with Avaya Aura™ Communication Manager Avaya Aura™ SIP Enablement Services using SIP Endpoint Emulation*, Issue 1.0, available at http://www.avaya.com.

[5] *Installation Guide Hammer G5^{TM},* Revision B, June 2009, available from Empirix.

[6] *Getting Started Guide Hammer G5^{TM},* Document Number 565-005-62, Revision E, June 2009, available from Empirix.

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Solution & Interoperability Test Lab Application Notes
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44 of 44
Hammer-SIP-TRK