Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 Endpoint Emulation – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Empirix Hammer IP test solution with Avaya Aura® Communication Manager using H.323 endpoint emulation. Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide a complete understanding of expected performance. Hammer IP can be used to assess and monitor network performance, reliability and quality of VoIP services in an Avaya IP telephony network. In this configuration, the Hammer IP emulates H.323 endpoints that originate and terminate calls through Avaya Aura® Communication Manager. While the call is active, Hammer IP 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. Hammer IP provides a collection of applications used to configure the system; create, schedule, and monitor tests; and create reports.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the 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 to integrate Empirix Hammer IP test system with Avaya Aura® Communication Manager using H.323 endpoint emulation. Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide a complete understanding of expected performance. Hammer IP can be used to assess and monitor network performance, reliability and quality of VoIP services in an Avaya IP telephony network. In this configuration, the Hammer IP emulates H.323 endpoints that originate and terminate calls through Avaya Aura® Communication Manager. While the call is active, Hammer IP 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. Hammer IP provides a collection of applications used to configure the system; create, schedule, and monitor tests; and create reports.

The following set of Hammer IP 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 H.323 registration status and call progress.
- **Hammer Call Summary Monitor** used to monitor call completion.

Below is a list of related Application Notes.

- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 Trunk Emulation* **[2]**
- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint Emulation* **[3]**
- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunk Emulation* **[4]**

2 General Test Approach

Interoperability compliance testing covered feature and serviceability testing. The feature testing was conducted by originating and terminating calls using H.323 endpoint channels on Hammer IP and establishing the calls through Communication Manager. The compliance test also covered monitoring various reports on Hammer IP during and after the test runs, and checking the status of various H.323 resources on Communication Manager. The serviceability testing focused on verifying the ability of Hammer IP to recover from adverse conditions, such as disconnecting the Ethernet cable and rebooting the server.

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 testing focused on verifying that the Hammer IP can register with Avaya Aura® Communication Manager as H.323 endpoints, establish calls, send voice media, and provide voice quality metrics. The following features and functionality were covered:

- H.323 endpoint registration with Avaya Aura® Communication Manager.
- Originating and terminating calls through Avaya Aura® Communication Manager.
- Support of G.711mu-law 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 and IP Audio Hairpinning disabled were also verified.
- Generating voice quality metrics with Shuffling disabled.
- DTMF support.
- Support for H.323 agent login to allow calls directly to a hunt group to be routed to an available agent, which is a Hammer IP H.323 endpoint.
- Support for Avaya H.323 Time-to-Service (TTS).
- SRTP with clear text H.323 signaling from H.323 endpoints to H.323 endpoints and H.323/SIP trunks.
- Originating calls from H.323 endpoints and terminating calls on H.323 endpoints, H.323 trunks, SIP endpoints, and SIP trunks.

Note: Performance and load testing was not the focus of the compliance test.

2.2 Test Results

All test cases passed. Empirix Hammer IP was successful in originating calls using H.323 endpoint emulation and terminating calls on channels emulating H.323 endpoints, H.323 trunks, SIP endpoints, and SIP trunks. It was observed that after reconnecting the Ethernet cable or re-establishing network connectivity, while a test script was running, required the channels to be unregistered and the configuration to be re-applied.

Important Note: The purpose of this compliance test was to verify interoperability between Empirix Hammer IP and Avaya Aura® Communication Manager using H.323 endpoint emulation. That is, the goal was to verify that Hammer IP can register with Communication Manager and establish calls. This was successfully verified. If a Hammer test encounters failed calls, there are various items to consider, including:

- The **Guard Time** and **Stagger** parameters may be set too aggressively (e.g., Hammer IP may be initiating too many calls too quickly) and the configuration under test may not be able to handle the load generated by Hammer IP. These parameters should be considered carefully for each test. It may be necessary to slow down the test to a rate that can be reasonably handled by the test configuration.

- Resources may be getting exhausted in the Avaya media gateway. These resources may include media processing resources, touch-tone receivers (TTRs), network trunks, and TDM bus resources.

Generally speaking, call failures encountered in Hammer IP are usually a result of one of the issues mentioned above.

Note: Communication Manager does not shuffle calls between a SIP trunk and H.323 trunk. This is per design. If the originating endpoint on Hammer is a SIP endpoint, note that the call arrives on Communication Manager via a SIP trunk. Therefore, a call from a SIP endpoint to an H.323 trunk is essentially a call from a SIP trunk to an H.323 trunk and the call is not shuffled.

2.3 Support

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

- **Phone:** (978) 313-7002
- **Web:** http://www.empirix.com/support/maintenance.aspx
- **Email:** supportcontract@empirix.com

3 Reference Configuration

The network diagram below illustrates the test configuration. In this configuration, Communication Manager receives calls from Hammer IP, which emulates H.323 endpoints. The call is then routed back to Hammer IP. The call can be terminated to another H.323 endpoint, H.323 trunk, SIP endpoint[1], or SIP trunk. While the call is established, Hammer IP can send DTMF and/or voice media (i.e., RTP traffic) using an audio recording. A voice quality test allows voice quality metrics to be provided at the end of each call. The Hammer IP applications running on the Hammer IP server were used to configure the system, create and monitor the tests, and view the test reports.

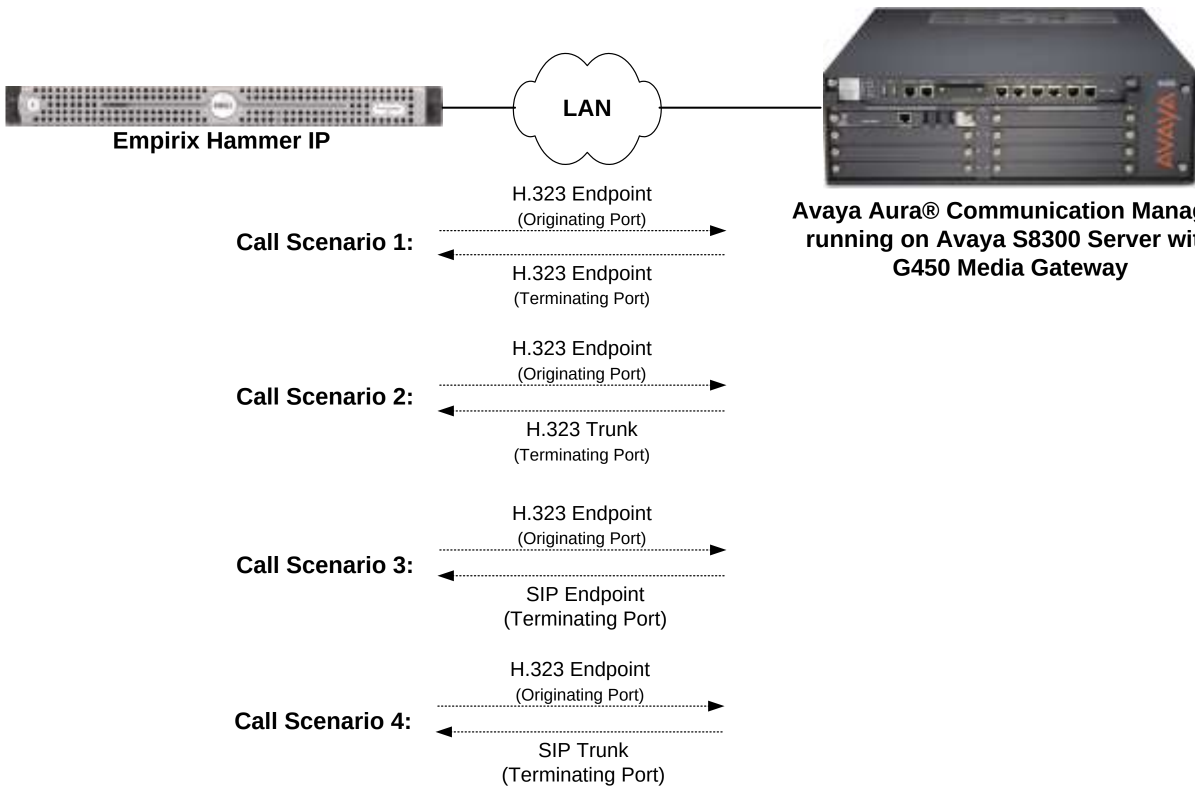


Figure 1: Empirix Hammer IP with Avaya Aura® Communication Manager

[1] To terminate the call to a SIP endpoint or SIP trunk, Avaya Aura® Session Manager is required, but is not shown in the configuration diagram.

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 running on S8300 Server	6.3.9 SP 9.1 (R016x.03.0.124.0 with Patch 22098)
Avaya G450 Media Gateway	Gateway FW 36.12.0
Avaya Aura® Session Manager running on an S8800 Server	6.3.11.0.631103
Empirix Hammer IP running on Microsoft Windows Server 2008 R2 with Dual 2.40 GHz Intel Xeon CPU and 12.0 GB of RAM	6.0.0.85

5 Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The configuration was performed using the System Access Terminal (SAT). The procedures include the following areas:

- Administer IP Codec Set
- Administer IP Network Region
- Administer H.323 Stations

5.1 Administer IP Codec Set

In the **IP Codec Set** form, specify the audio codec(s) required by the test that will be run on the Hammer IP. 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 covered in the next section. For the compliance test, G.711mu-law, G.729AB, and G.729A codecs were used. In the IP codec set form, specify the appropriate codec being used by the Hammer test. Below is the IP codec set configured for **G.711mu-law**.

```
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:
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

If SRTP is required for the test, set **Media Encryption** to *1-srtp-aescm128-hmac80*. This is the media encryption supported by Hammer IP.

```
   Media Encryption
 1: 1-srtp-aescm128-hmac80
 2:
 3:
```

5.2 Administer IP Network Region

In the **IP Network Region** form, specify the codec set to be used for Hammer calls and specify whether **IP-IP Direct Audio** (Shuffling) is required for the test. Shuffling allows audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. Note that if Shuffling is enabled, audio traffic does not egress the Hammer IP since the calls would be shuffled. In the following example, Shuffling is disabled.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devcon.com
    Name:                     Stub Network Region: n
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? n
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        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
```

On **Page 3**, set the **Near End Establishes TCP Signaling Socket** field to *y* if Communication Manager should initiate setting up the TCP signaling socket. Setting this field to *n* will allow Hammer IP to initiate setting up the socket.

```
change ip-network-region 1                                      Page   3 of  20
                               IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:    Insert:
 Maximum Number of Trunks to Use for IGAR:
 Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)    H.323 SECURITY PROFILES
 1                                   1   challenge
 2                                   2
 3                                   3
 4                                   4
 5
 6                                   Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

5.3 Administer H.323 Stations

Configure a H.323 station for each H.323 channel on the Hammer IP. Set the **Type** field to *9620* or *9630*. Set the **Port** field to *IP* and configure a descriptive **Name**. Lastly, configure the **Security Code** that will be used by the Hammer IP to register with Communication Manager. For the compliance test, 20 H.323 stations were used with extensions ranging from 41011 to 41030. The first group of 10 channels (extensions 41011 to 41020) was used to originate calls. The calls were then terminated on the remaining 10 channels (extensions 41021 to 41030). Repeat this procedure for each channel required by the Hammer test.

```
add station 41011                                             Page   1 of   5
                                     STATION

Extension: 41011                         Lock Messages? n             BCC: 0
     Type: 9620                          Security Code: 1234           TN: 1
     Port: IP                          Coverage Path 1:               COR: 1
     Name: Hammer                      Coverage Path 2:               COS: 1
                                       Hunt-to Station:             Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 41011
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
                              Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y
```

Page 2 of the station form allows Shuffling to be enabled or disabled at the station level. Shuffling can also be disabled at the IP Network Region level.

```
add station 41011                                             Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                        EC500 State: enabled
         Multimedia Mode: enhanced                  Audible Message Waiting? n
    MWI Served User Type:                        Display Client Redirection? n
             AUDIX Name:                        Select Last Used Appearance? n
                                                  Coverage After Forwarding? s
                                                    Multimedia Early Answer? n
                                            Direct IP-IP Audio Connections? y
  Emergency Location Ext: 41011        Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

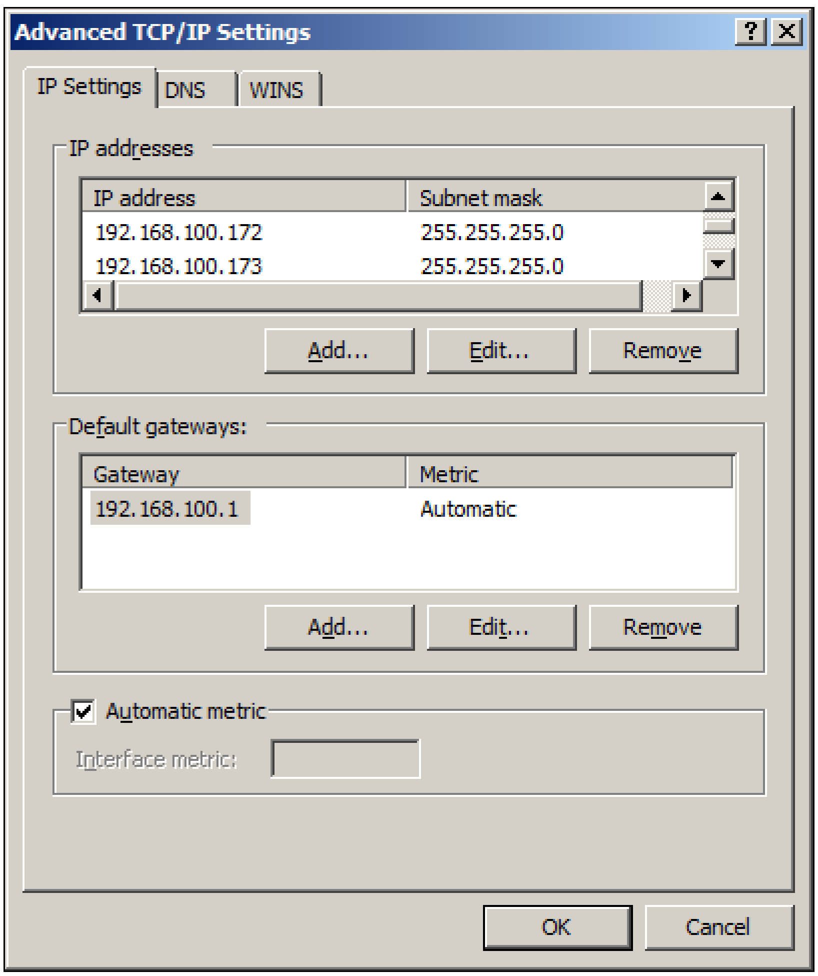
6 Configure Empirix Hammer IP

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

- Assign unique IP addresses to each Hammer IP channel.
- Configure the system, including the originating and terminating channels and the PhoneBook, using the **Hammer Configurator**.
- Save and apply the Hammer configuration and start the Hammer server.
- Create and run the test script using the **Hammer TestBuilder**.

6.1 Configure IP Addresses on Hammer IP Server

The Hammer IP server needs to be configured with IP addresses for each channel. During the compliance test, 20 H.323 endpoint channels were used. 10 channels were used to originate calls and 10 channels were used to terminate calls. This requires a block of 20 unique IP addresses, which must be contiguous. The 20 IP addresses used were from 192.168.100.171 to 192.168.100.190. These IP addresses are configured in the **Advanced TCP/IP Settings** under Network Connections in Windows Server 2008.



6.2 Configure System

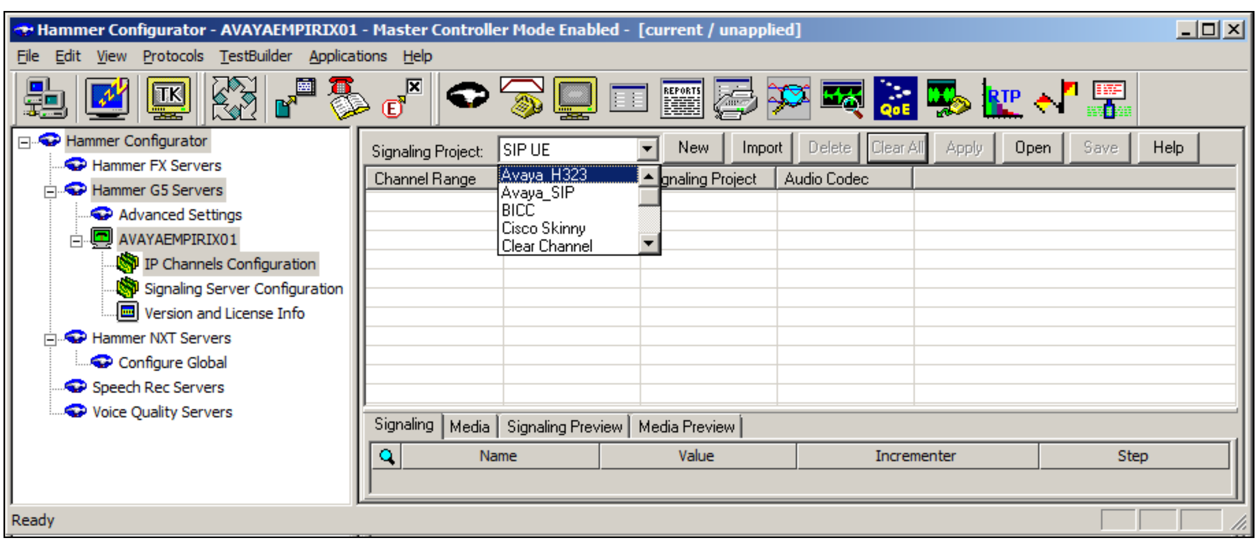
This section covers the configuration of originating and terminating channels and the PhoneBook on Hammer IP. In this configuration, the originating channels emulate H.323 endpoints (described in **Section 6.2.1**). The terminating channels can emulate H.323 endpoints, H.323 trunks, SIP endpoints, or SIP trunks. These Application Notes will explicitly describe the configuration for terminating calls to H.323 endpoints in **Section 6.2.2.1**. In addition, it will provide references to other Application Notes for configuring terminating channels as H.323 trunks, SIP endpoints, and SIP trunks in **Sections 6.2.2.2**, **6.2.2.3**, and **6.2.2.4**, respectively. Only one of those sections needs to be followed depending on the configuration desired.

6.2.1 Configure Originating Channels – H.323 Endpoints

Empirix Hammer IP is configured through the **Hammer Configurator**, a graphical user interface, residing on the Hammer IP server. From the Hammer IP server, run the **Hammer Configurator**. The following screen is displayed.

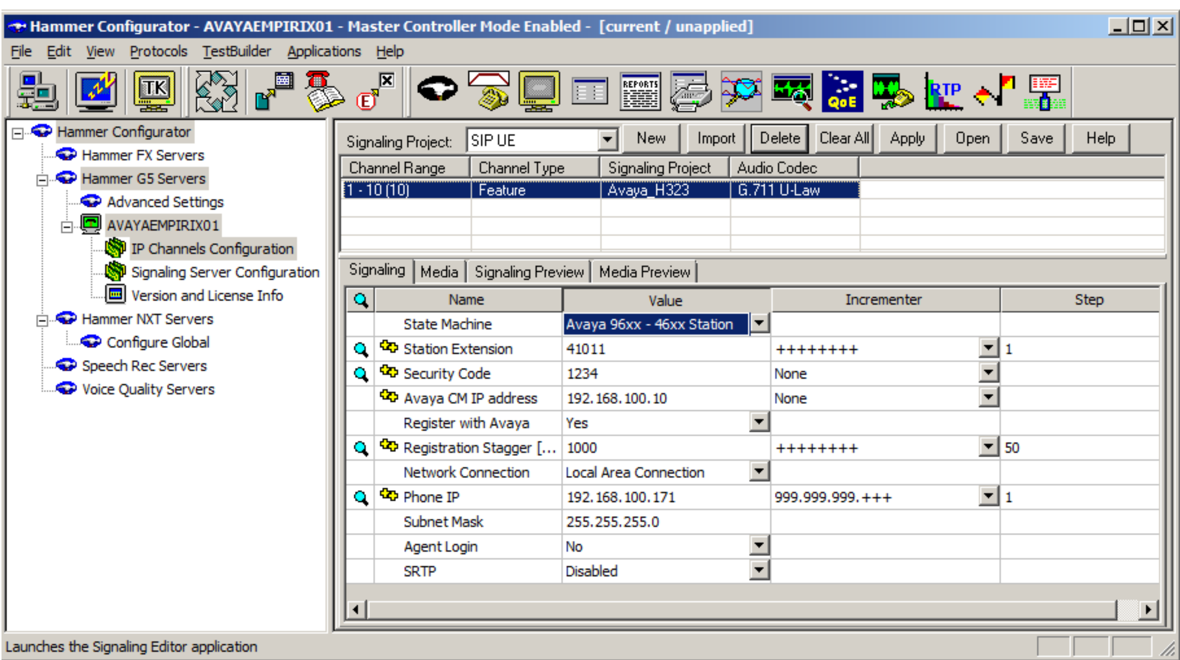
Note: It is assumed that Hammer IP is already in **Master Controller Mode**. To verify, check that the title bar of the **Hammer Configurator** indicates *Master Controller Mode Enabled* as shown below. It is also assumed that a system was already added to the configuration. In this configuration, the system name is *AVAYAEMPIRIX01*, which corresponds to the server name.

In the **Hammer Configurator**, the server name will appear in the left pane of the **Hammer Configurator**. Expand the server name (e.g., *AVAYAEMPIRIX01*) in the left pane and click on **IP Channels Configuration.** The following window will be displayed. Select *Avaya H.323* for the **Signaling Project** and then click **New**.



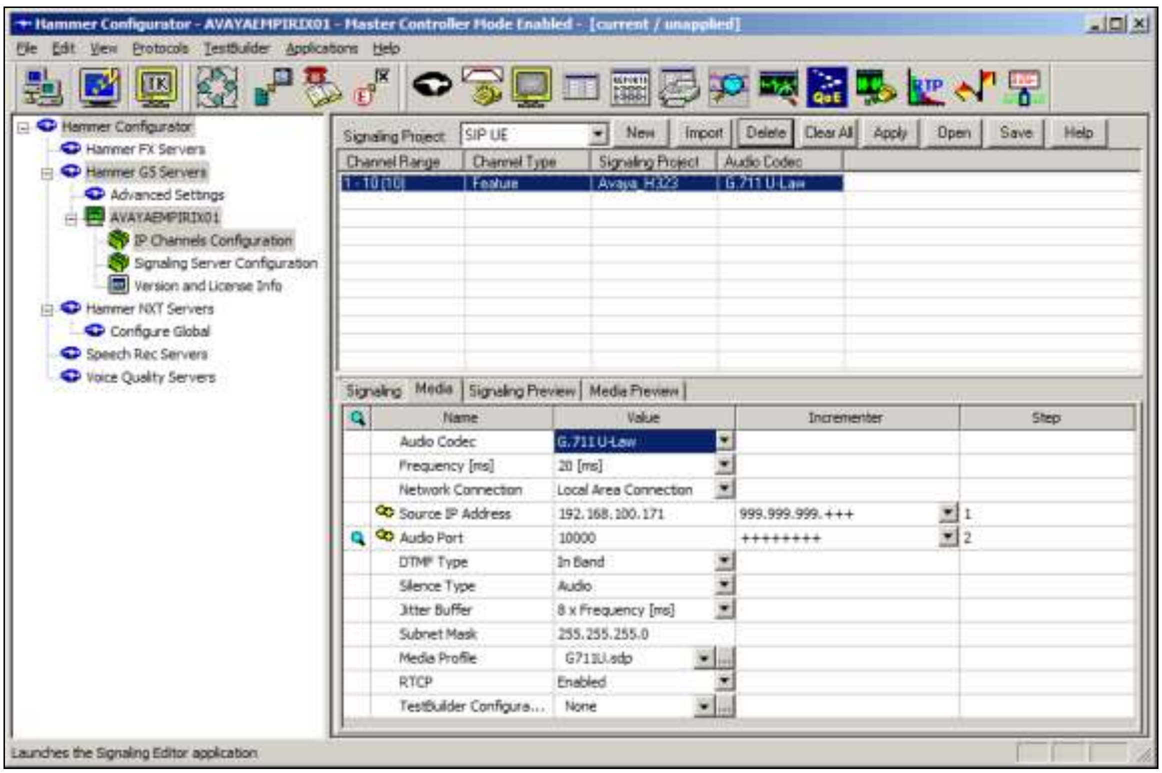
The first line in the grid that is highlighted in the figure below corresponds to the 10 originating channels. To set the number of channels in the group, click on the **Channel Range** cell in the grid and enter the number *10*. Set the **Channel Type** cell to *Feature*. The following fields in the **Signaling** tab should then be set as follows:

- **State Machine** should be set to *Avaya 96xx – 46xx Station*.
- **Station Extension** should be set to the first extension in the group (e.g., *41011*) and the **Incrementer** and **Step** fields should be set as shown so that the extension of the subsequent channels are incremented by one. This covers extensions from *41011* to *41020*.
- **Security Code** should match the one configured in the corresponding **Station** form on Communication Manager.
- **Avaya CM IP Address** should be set to the S8300 Server IP address (e.g., *192.168.100.10*). Use the C-LAN IP address if a C-LAN card is being used in a G650 Media Gateway.
- **Register with Avaya** should be set to *Yes*.
- **Network Connection** should be set to the appropriate network interface.
- **Phone IP** should be set to the IP address of the first channel in the group and the **Incrementer** and **Step** fields should be set as shown so that the last octet of the IP address is incremented by one. Note that this requires a block of contiguous IP addresses. This covers IP addresses from *192.168.100.171* to *192.168.100.180*.
- **Subnet Mask** should be set to the network mask (e.g., *255.255.255.0*).
- **Agent Login** should be set to *No* for the originating channels. However, for the terminating channels, **Agent Login** may be set to *Yes* if the terminating H.323 endpoints will act as agents in a contact center environment (i.e., agents logged into a hunt group/split). Otherwise, set this field to *No*.
- **SRTP** should be set to *Disabled* unless enabled in **Section 5.1**.



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 U-Law, G729AB, and G.729A were used during the compliance testing.
- **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. The **Incrementer** and **Step** fields should be set as shown so that the last octet of the IP address is incremented for the subsequent channels. Note that the IP addresses for the channels must be contiguous.
- **Media Profile** should be set to one that specifies the codec configured in the **Audio Codec** field. See **Appendix A** for instructions on configuring a **Media Profile**.
- The default values for the remaining fields may be used as shown.



6.2.2 Configure Terminating Channels

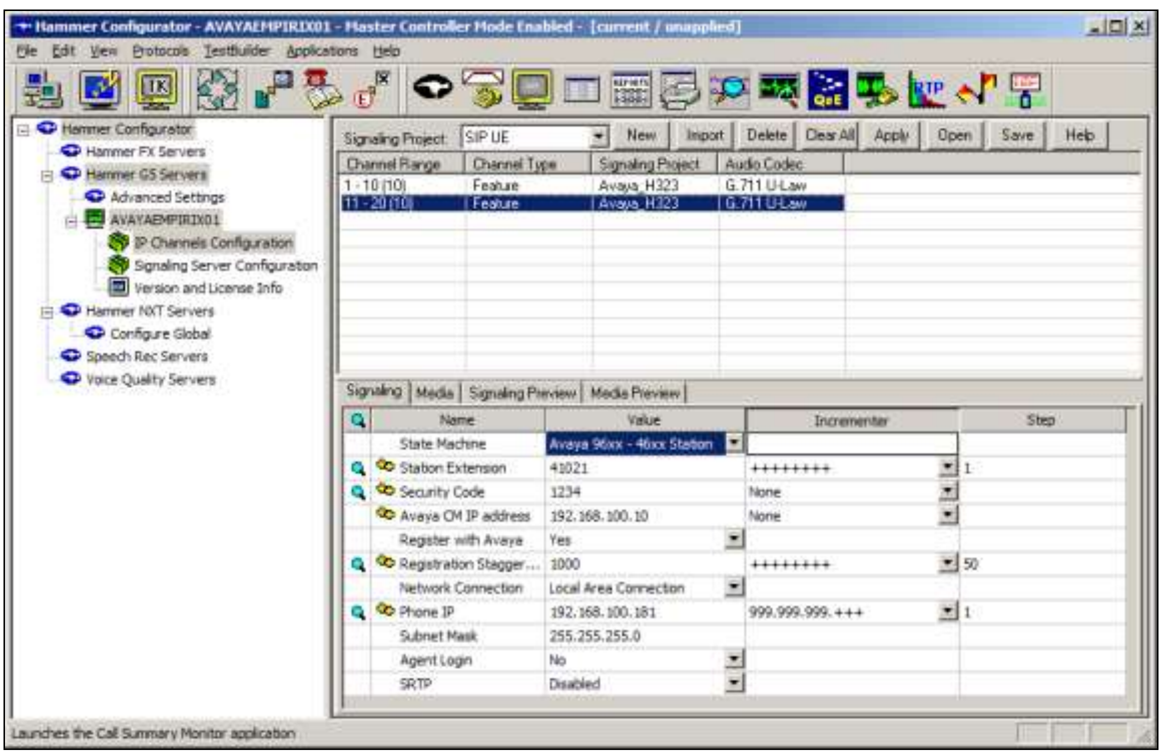
During the compliance test, the originating channels emulated H.323 endpoints with the calls terminating on H.323 endpoints, H.323 trunks, SIP endpoints, or SIP trunks. Select one of the following subsections depending on the configuration desired.

- **Section 6.2.2.1** for terminating calls on H.323 endpoints.
- **Section 6.2.2.2** for terminating calls on H.323 trunks.
- **Section 6.2.2.3** for terminating calls on SIP endpoints.
- **Section 6.2.2.4** for terminating calls on SIP trunks.

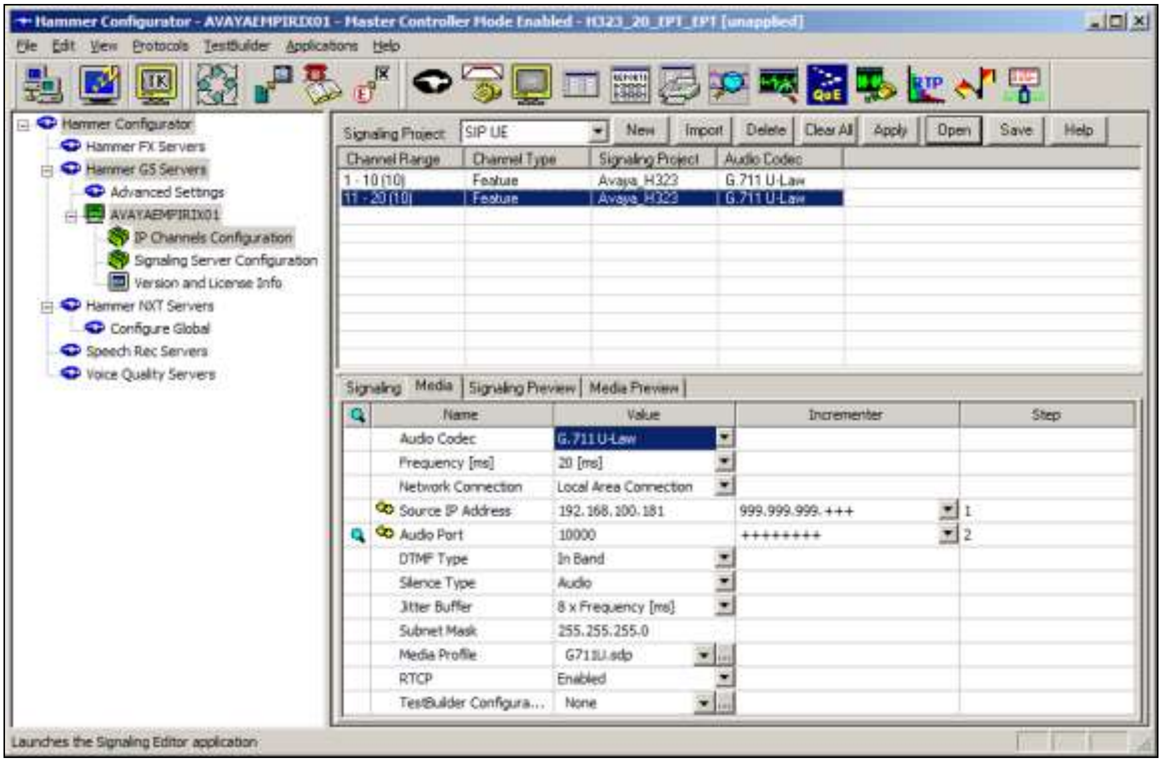
Note: Ensure that the originating and terminating channels are assigned unique IP addresses and extensions.

6.2.2.1 Configure Terminating Channels – H.323 Endpoints

The second line in the grid that is highlighted in the figure below corresponds to the second group of channels that will terminate calls. Set the **Channel Range** cell to the number of channels in this group. Set the **Channel Type** cell to *Feature*. The configuration of the **Signaling** tab is similar to the one for the group of originating channels in **Section 6.2.1** with the exception that the **Station Extension** and **Phone IP** fields will be different. This group of channels will be assigned extensions *41021* to *41030* and IP addresses from *192.168.100.181* to *192.168.100.190*. Again, the IP addresses for this group of channels must be contiguous. Also, note that **Agent Login** may be enabled for the terminating channels as mentioned above. To enable agent login, see **Appendix B**.



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.



6.2.2.2 Configure Terminating Channels – H.323 Trunks

To terminate the calls to H.323 trunks follow the instructions described in **[2]**, specifically:

- **Section 5** describes how to configure H.323 trunks and call routing on Communication Manager.
- **Section 6.2.2.1** describes how to configure terminating H.323 trunks on Hammer IP.
- **Section 6.4** describes how to specify the dialed digits when running a test script.

The configuration described in all the aforementioned sections of **[2]** must be completed for terminating calls to H.323 trunks.

6.2.2.3 Configure Terminating Channels – SIP Endpoints

To terminate the calls to SIP endpoints follow the instructions described in **[3]**, specifically:

- **Section 5** describes how to configure SIP trunk to Session Manager, SIP stations, and call routing on Communication Manager.
- **Section 6** describes how to configure Session Manager, including the SIP trunk to Communication Manager and SIP endpoints. This section needs to be configured in its entirety.
- **Section 7.2.2.1** describes how to configure terminating SIP endpoints on Hammer IP.
- **Section 7.2.3** describes how to configure the PhoneBook.
- **Section 7.4** describes how to disable the **Do Connect Latency** option (required) and how to specify the dialed digits when running a test script.

The configuration described in all the aforementioned sections of **[3]** must be completed for terminating calls to H.323 trunks.

6.2.2.4 Configure Terminating Channels – SIP Trunks

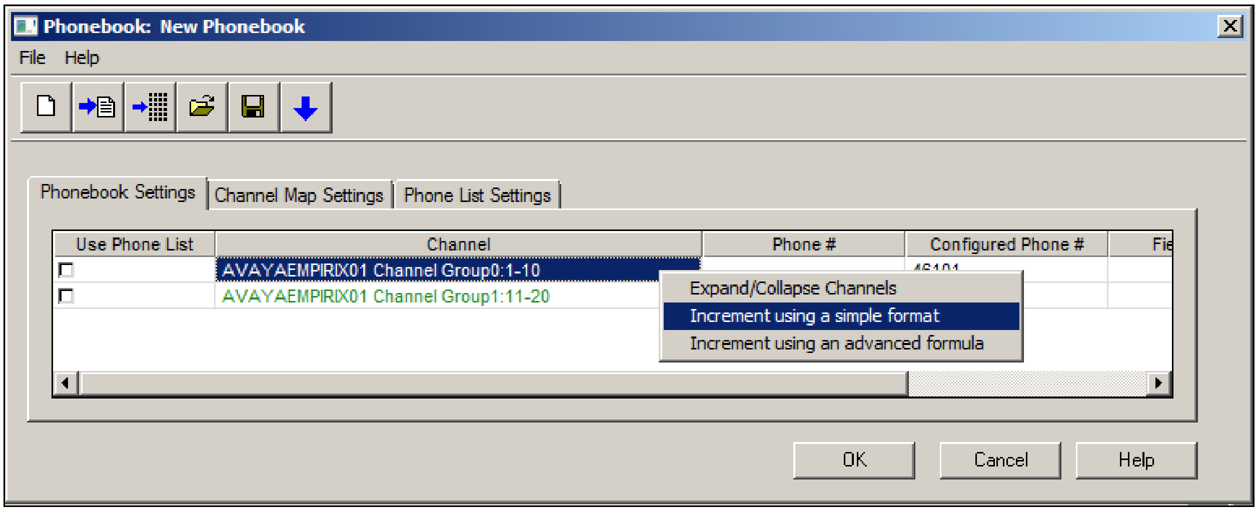
To terminate the calls to SIP trunks follow the instructions described in **[4]**, specifically:

- **Section 5** describes how to configure call routing on Communication Manager.
- **Section 6** describes how to configure SIP trunks to Hammer IP on Session Manager.
- **Section 7.2.2.1** describes how to configure terminating SIP trunks on Hammer IP.
- **Section 7.4** describes how to specify the dialed digits when running a test script.

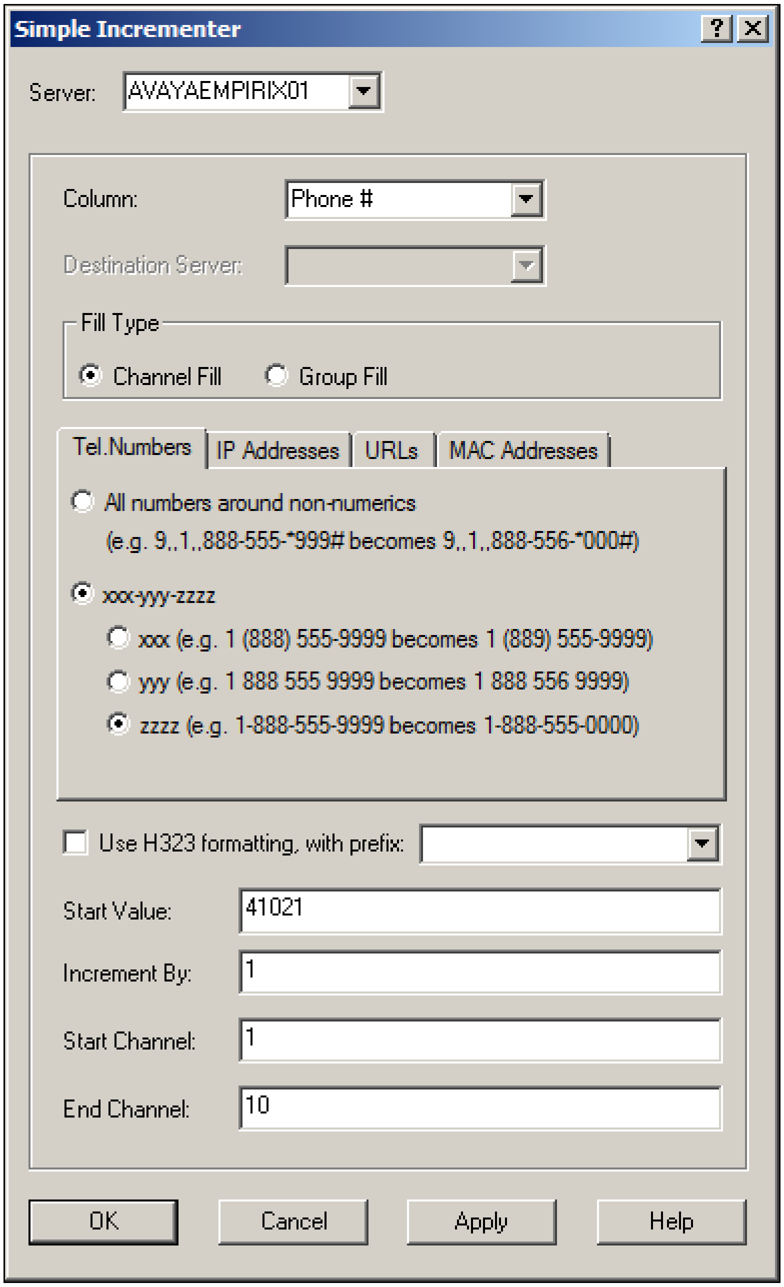
The configuration described in all the aforementioned sections of **[4]** must be completed for terminating calls to SIP trunks.

6.2.3 Configure the PhoneBook

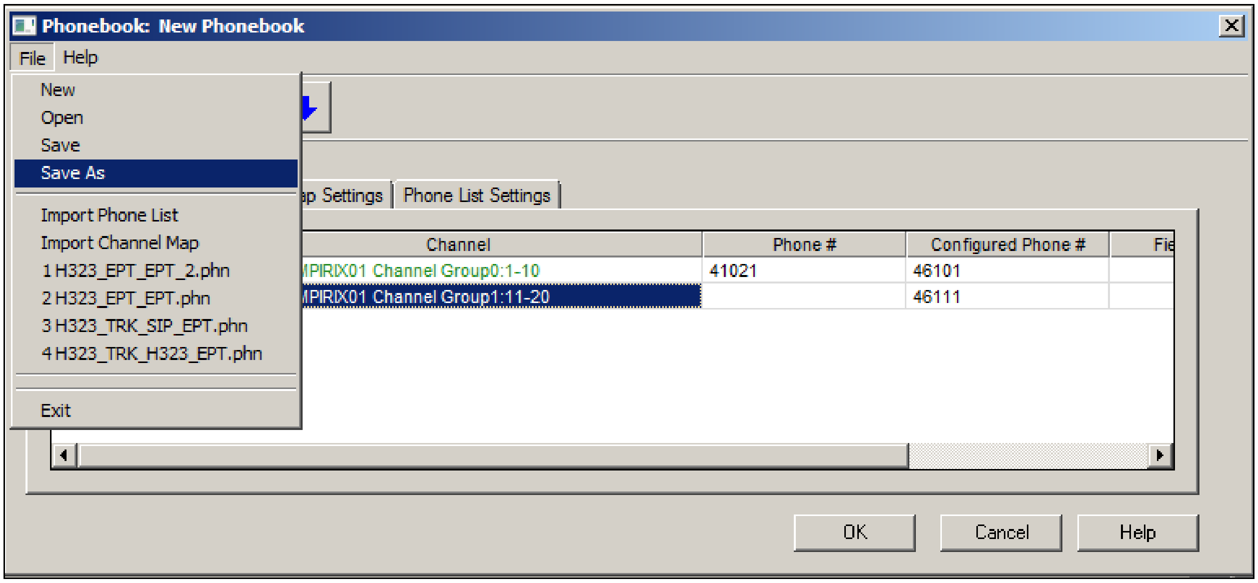
The **PhoneBook** is used to specify which number each originating channel should dial when placing a call. This section specifies the **PhoneBook** configuration when calls are being terminated on H.323 endpoints. Click on the **PhoneBook** icon in the **Hammer Configurator**. The **PhoneBook** window is displayed below. The **Channel** column is automatically displayed with the appropriate channel groups. Right-mouse click on the first line corresponding to the group of originating channels (channels 1-10) and select the **Increment using a simple format** option as shown below.



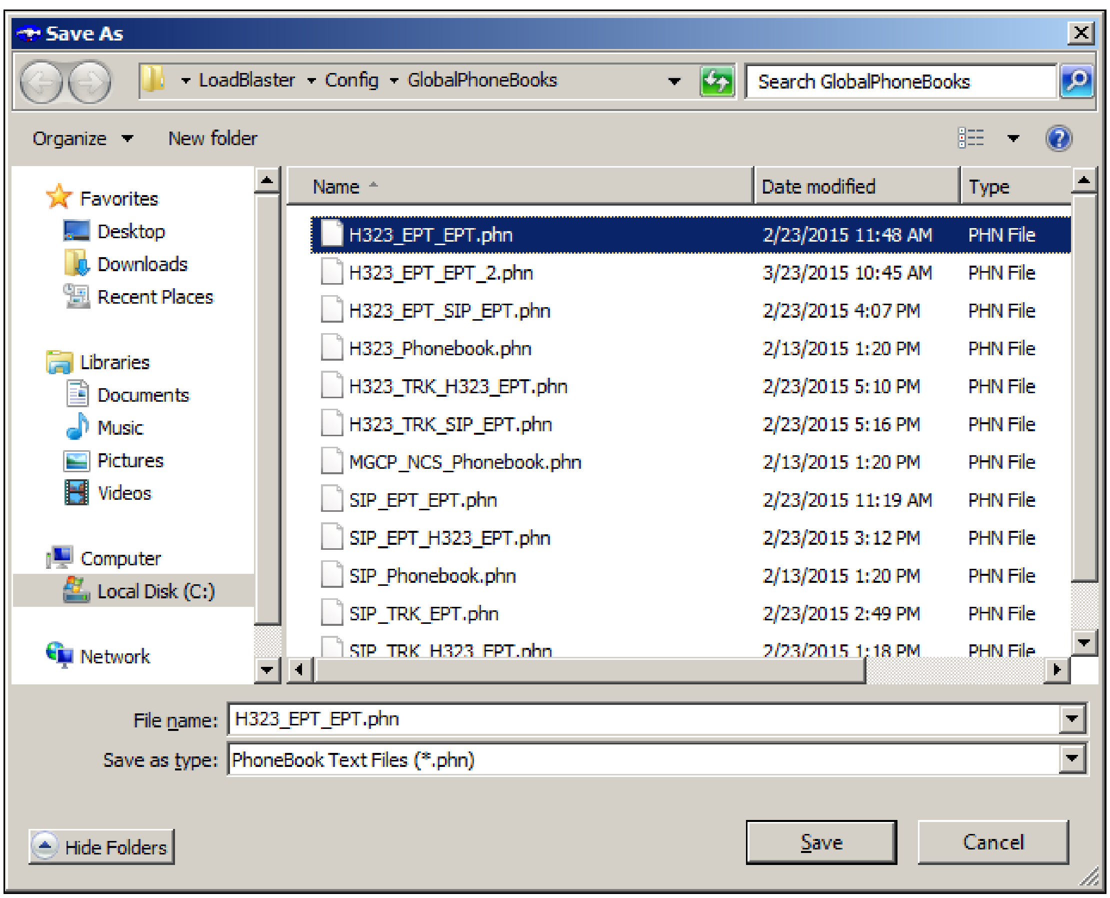
In the **Simple Incrementer** window, specify the number that the first originating channel should dial in the **Start Value** field. In this example, the first channel will dial *41021*, which corresponds to channel 11. Set the **Increment By** field to *1*. This specifies that the subsequent channels should increment the dialed number by one. For example, channel 1 will dial 41021, channel 2 will dial 41022, and so on. The **Start Channel** field should be set to the first channel number and the **End Channel** field should be set to the last originating channel number, which is *10*. Click **OK**.



Once the **PhoneBook** is configured, select **File→Save As** to save the PhoneBook.

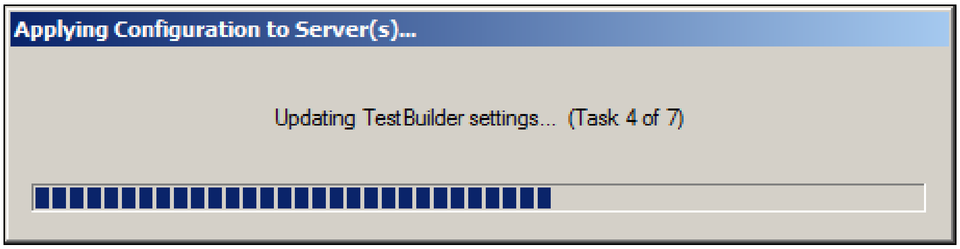


The PhoneBook is saved as *H323_EPT_EPT.phn* in the following window. This PhoneBook will be used when running the test.

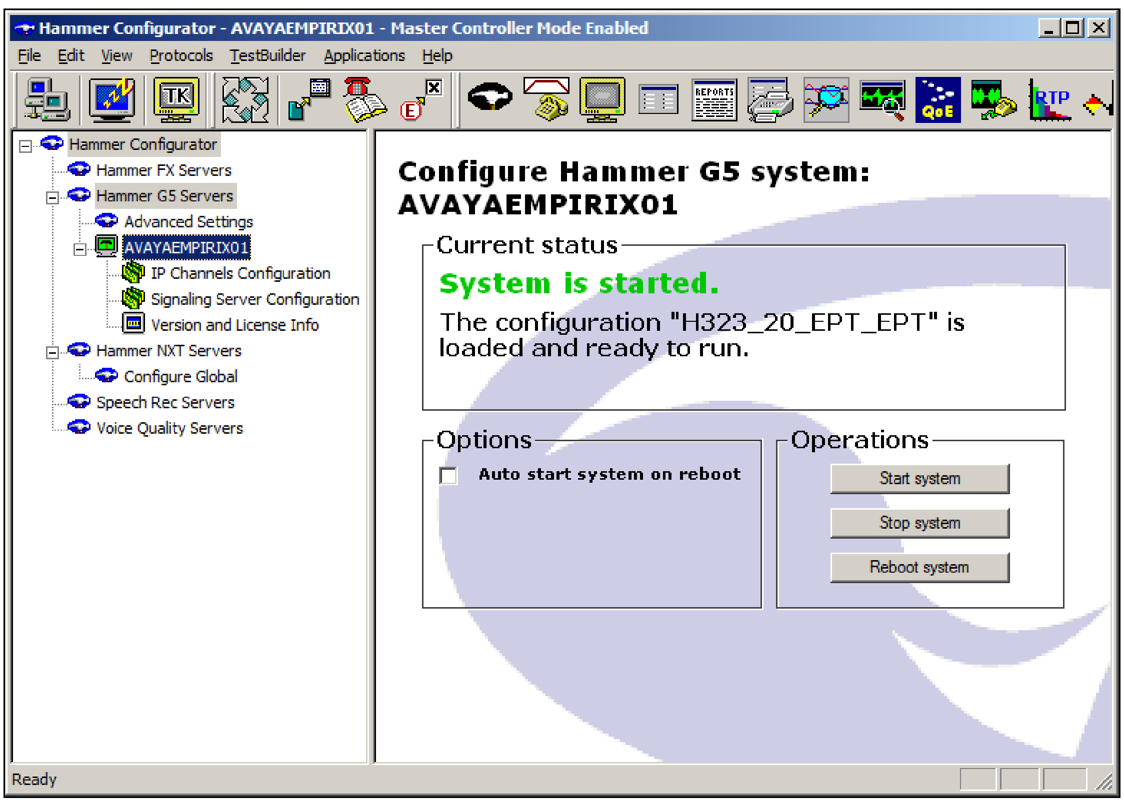


6.3 Applying the Hammer IP Configuration

This completes the configuration of Hammer IP. 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., *AVAYAEMPIRIX01*) 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 *H323_20_EPT_EPT*. When the system is started, Hammer IP will register H.323 endpoints with Communication Manager.



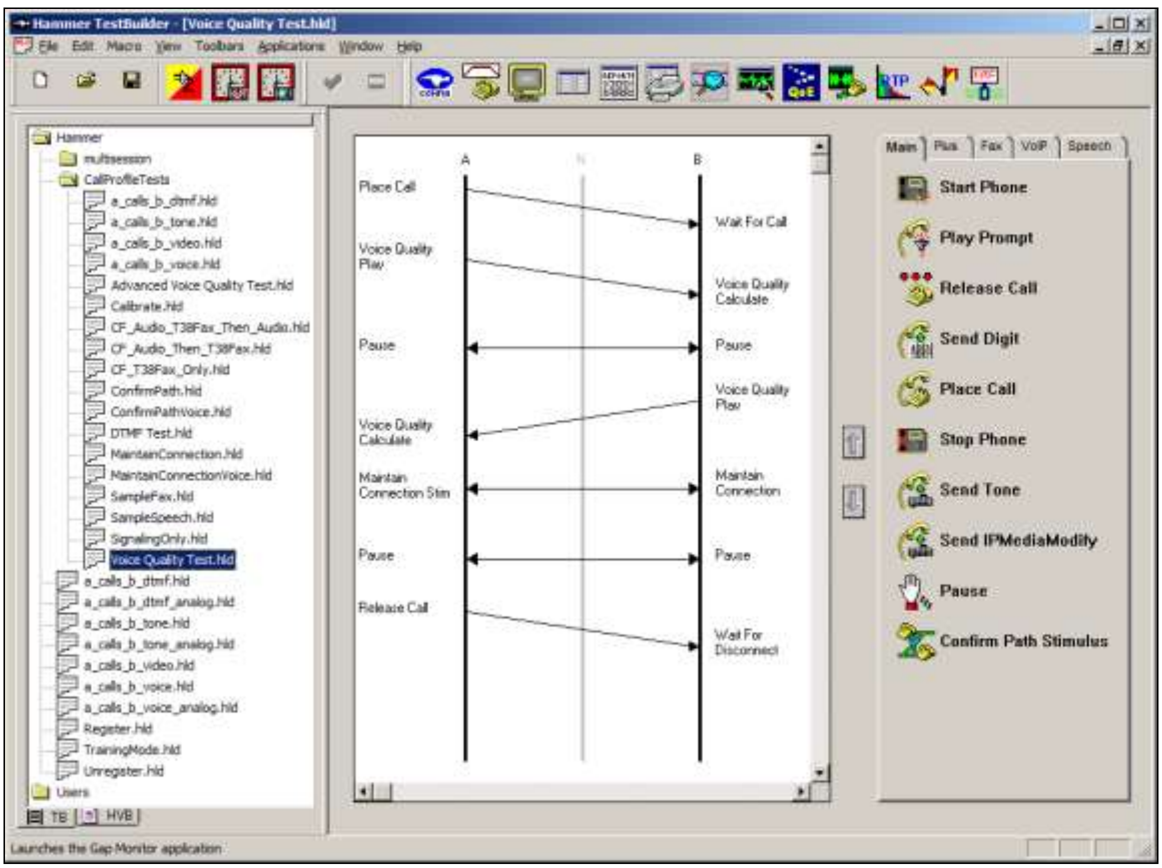
6.4 Configure and Run the Test Script

For the compliance test, two default test scripts were used:

- `a_calls_b_dtmf.hld` to verify DTMF
- `Voice Quality Test.hld` to verify voice quality

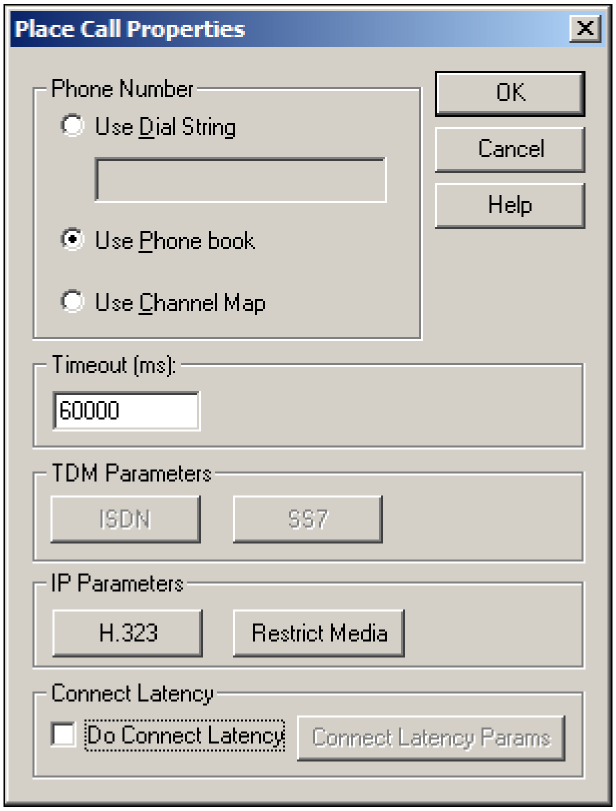
The sample test script, `Voice Quality Test.hld`, establishes a VoIP call between two H.323 endpoints on Hammer IP, followed by the originating side playing an audio prompt to the far-end so that voice quality metrics (e.g., PESQ score) can be obtained. The test script is configured with the **Hammer TestBuilder** application and can be displayed in a ladder diagram as shown below by double-clicking on the test script name.

Note: When changing from one Hammer configuration to another, where the previous configuration used H.323 endpoints, it may be required to unregister the H.323 endpoints before proceeding with the new Hammer test as shown in **Appendix C**.

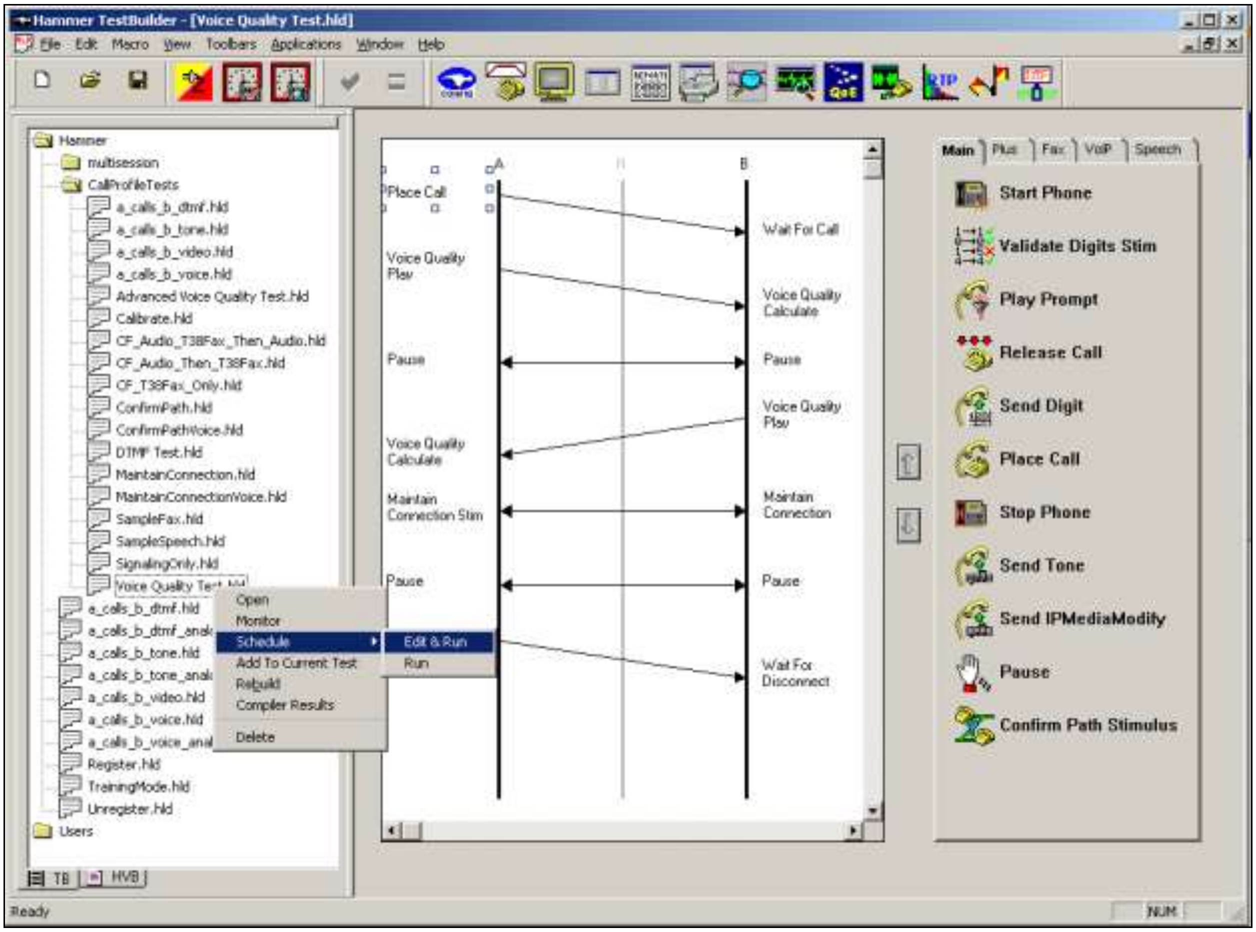


In the sample test script configured above, the A-side (originating H.323 endpoint) places a call to the B-side (terminating H.323 endpoint) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. The **Place Call Properties** is configured to use the **Phone book** as shown below.

Note: Disable the **Do Connect Latency** option in the **Place Call Properties** window.

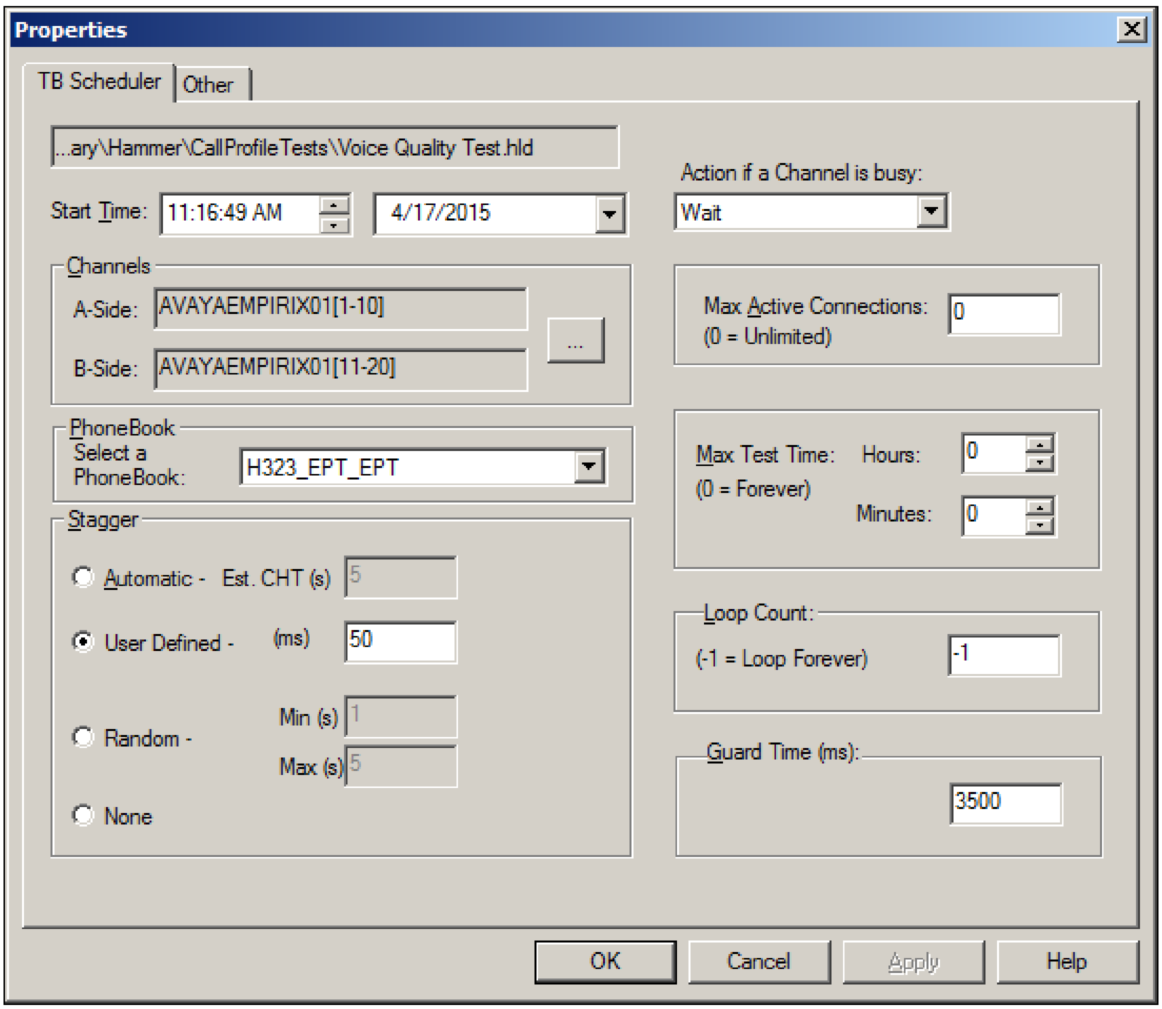


To run the test, right-mouse click on the test script in the left pane of the **Hammer TestBuilder** window and navigate to **Schedule→Edit & Run**. To re-run the test, the user can simply select **Schedule→Run**, if no changes are required.



In the **Properties** window, click on the ellipses button (…) in the **Channels** section and assign channels to the **A-Side** and **B-Side**. Next, select the appropriate PhoneBook (e.g., *H323_EPT_EPT*). The H323_EPT_EPT PhoneBook was configured above. Set the **Loop Count** to the appropriate value to control the number of iterations 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 minimum setting should be *3500*. The **Stagger** section allows the user to specify how long to wait before the test is run on the next channel. Click **OK**.

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 IP. These parameters can slow down the test to a rate that can be reasonably handled by the test configuration.



7 Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager and Empirix Hammer IP.

7.1 Verify Avaya Aura® Communication Manager

To verify that the Hammer IP can register H.323 endpoints on Communication Manager, the **list registered-ip-stations** command may be used to verify that the endpoints have been successfully registered.

```
list registered-ip-stations                                              Page   1

                            REGISTERED IP STATIONS

Station Ext   Set Type/ Prod ID/   TCP Station IP Address/
or Orig Port  Net Rgn   Release    Skt Gatekeeper IP Address
------------- --------- ---------- --- ---------------------------------------
41011         9620      IP_Phone    y  192.168.100.171
              1         2.80           192.168.100.10
41012         9620      IP_Phone    y  192.168.100.172
              1         2.80           192.168.100.10
41013         9620      IP_Phone    y  192.168.100.173
              1         2.80           192.168.100.10
41014         9620      IP_Phone    y  192.168.100.174
              1         2.80           192.168.100.10
41015         9620      IP_Phone    y  192.168.100.175
              1         2.80           192.168.100.10
```

When the Hammer IP is running a test script, the **status station** command may be used to view the active call status. The **Service State** should be set to *in-service*.

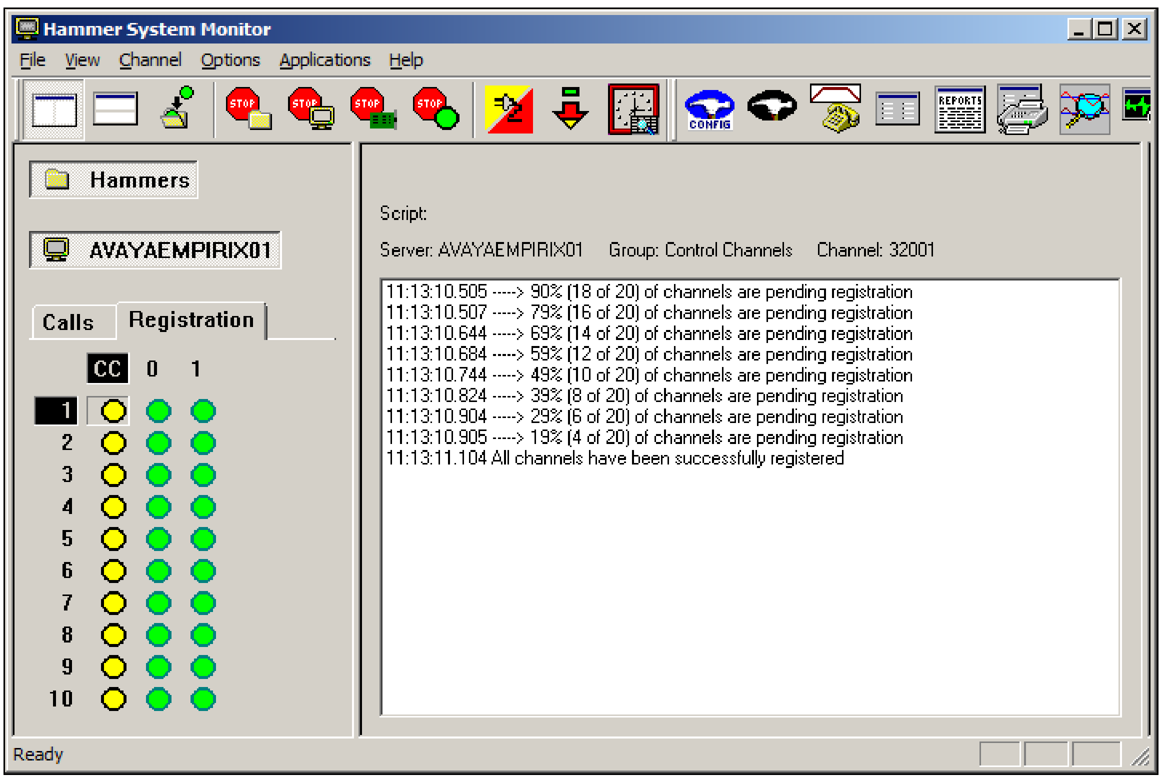
```
status station 41011                                              Page   1 of   9
                               GENERAL STATUS
     Administered Type: 9620             Service State: in-service/off-hook
        Connected Type: N/A           TCP Signal Status: connected
             Extension: 41011            Network Region: 1
                  Port: S00020        Parameter Download: complete
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no
Active Coverage Option: 1           one-X Server Status: N/A

          EC500 Status: N/A        Off-PBX Service State: N/A
   Message Waiting:
   Connected Ports: S00031


  Limit Incoming Calls? no

 User Cntrl Restr: none                      HOSPITALITY STATUS
Group Cntrl Restr: none                   Awaken at:
                                           User DND: not activated
                                          Group DND: not activated
                                        Room Status: occupied
```

Page 5 of the **status station** command indicates the codec being used for the call and whether the call is shuffled or not. If the call is shuffled, the **Audio Connection Type** field is set to *ip-direct*, if it isn't, the field is set to *ip-tdm*.

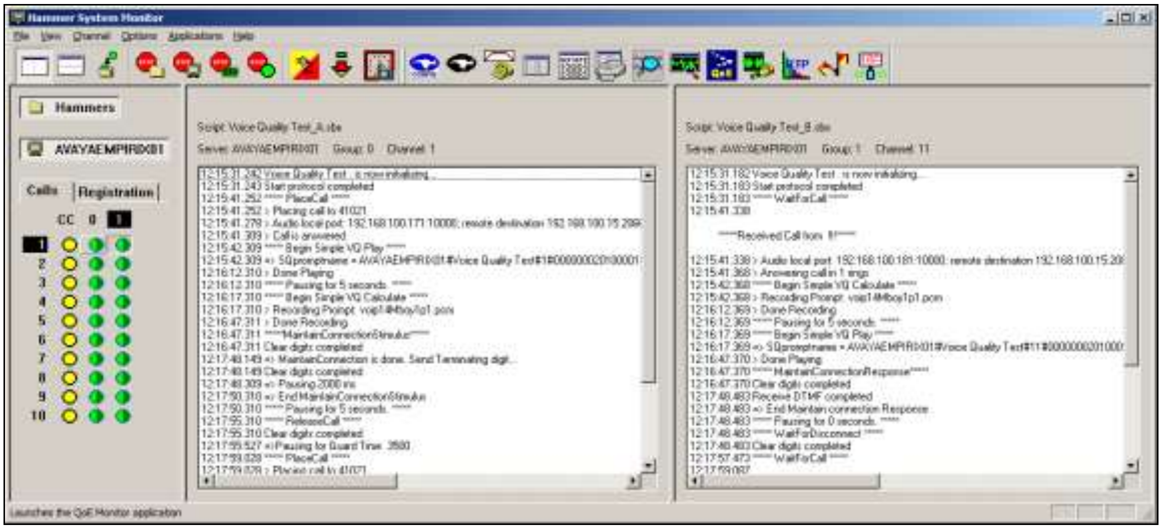
```
status station 41011                                              Page   5 of   9

                          AUDIO CHANNEL Port: S00020
G.711MU       Switch-End Audio Location: MG1
            IP Address                                 Port   Node Name        Rgn
Other-End: 192.168.100.15                              2050                    1
  Set-End: 192.168.100.171                             10000 HammerIP-Out      1
Audio Connection Type: ip-tdm
```

7.2 Verify Empirix Hammer IP

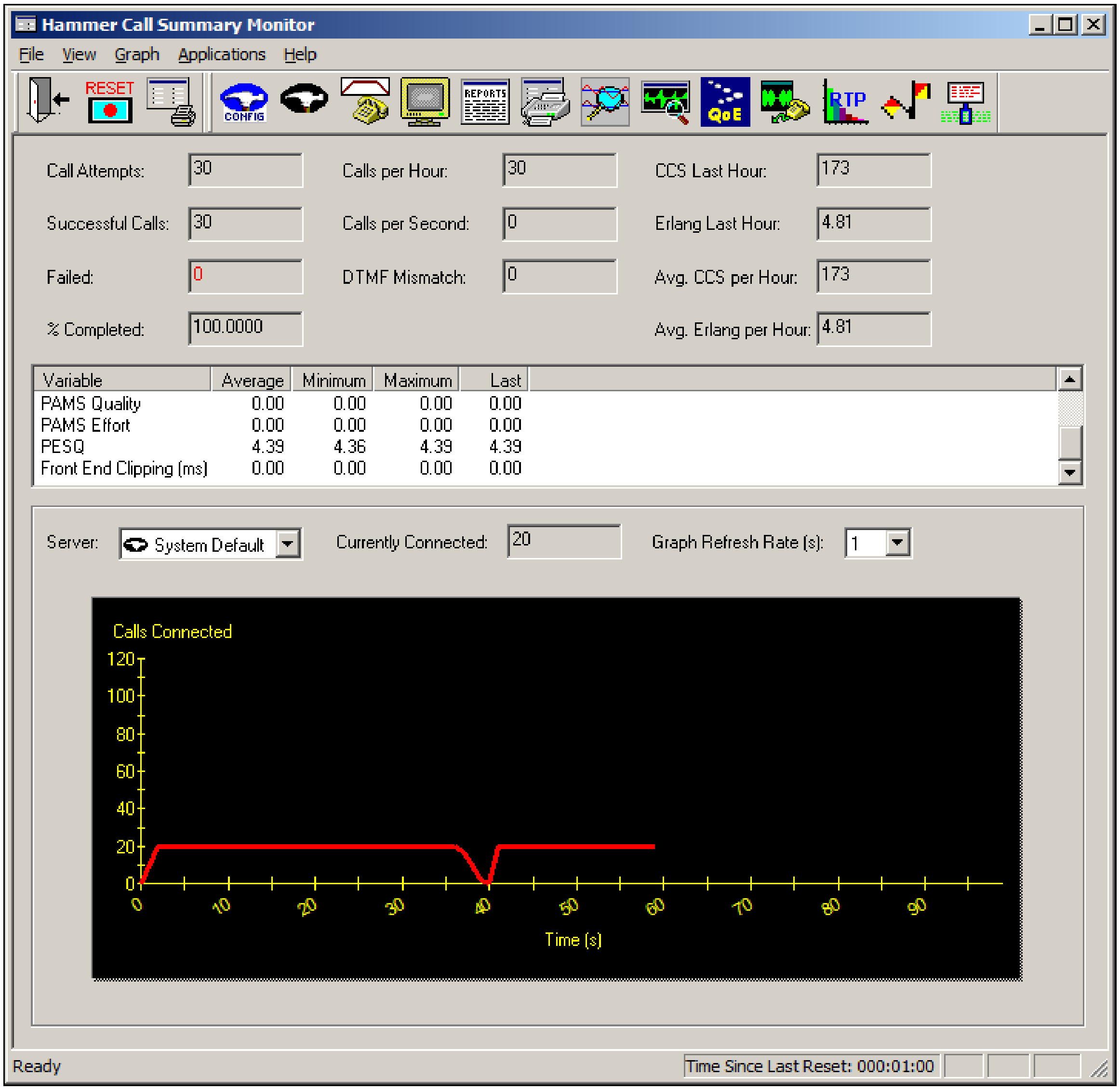
To view the H.323 registration status on Hammer IP, make sure that the **Hammer System Monitor** is running before starting the system. Click on the yellow circle under the **CC** column and row **1**. Hammer IP will indicate when all of the channels have successfully registered.



Call progress may 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.



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.



8 Conclusion

These Application Notes describe the configuration steps required to integrate Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 endpoint emulation. Hammer IP H.323 channels were able to register with Avaya Aura® Communication Manager, successfully establish calls to H.323 and SIP endpoints/trunks, generate voice quality metrics, and monitor the calls. All feature and serviceability test cases were completed successfully. Refer to **Section 2.2** for test observations.

9 References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10.0, June 2014, Document Number 03-300509, available at http://support.avaya.com.

[2] *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 Trunk Emulation*, Issue 1.0, available at http://www.avaya.com.

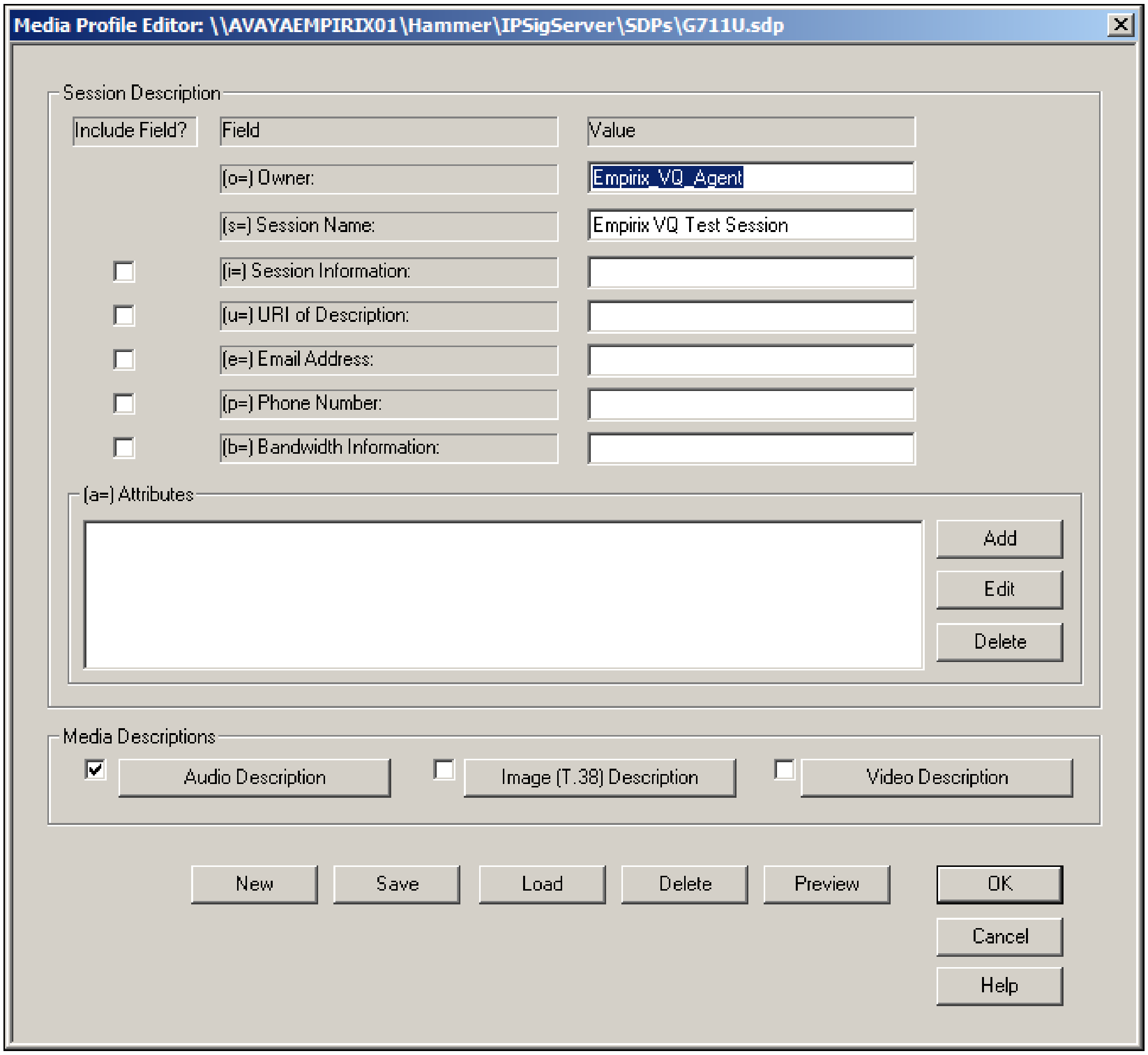
[3] *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint Emulation*, Issue 1.0, available at http://www.avaya.com.

[4] *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunk Emulation*, Issue 1.0, available at http://www.avaya.com.

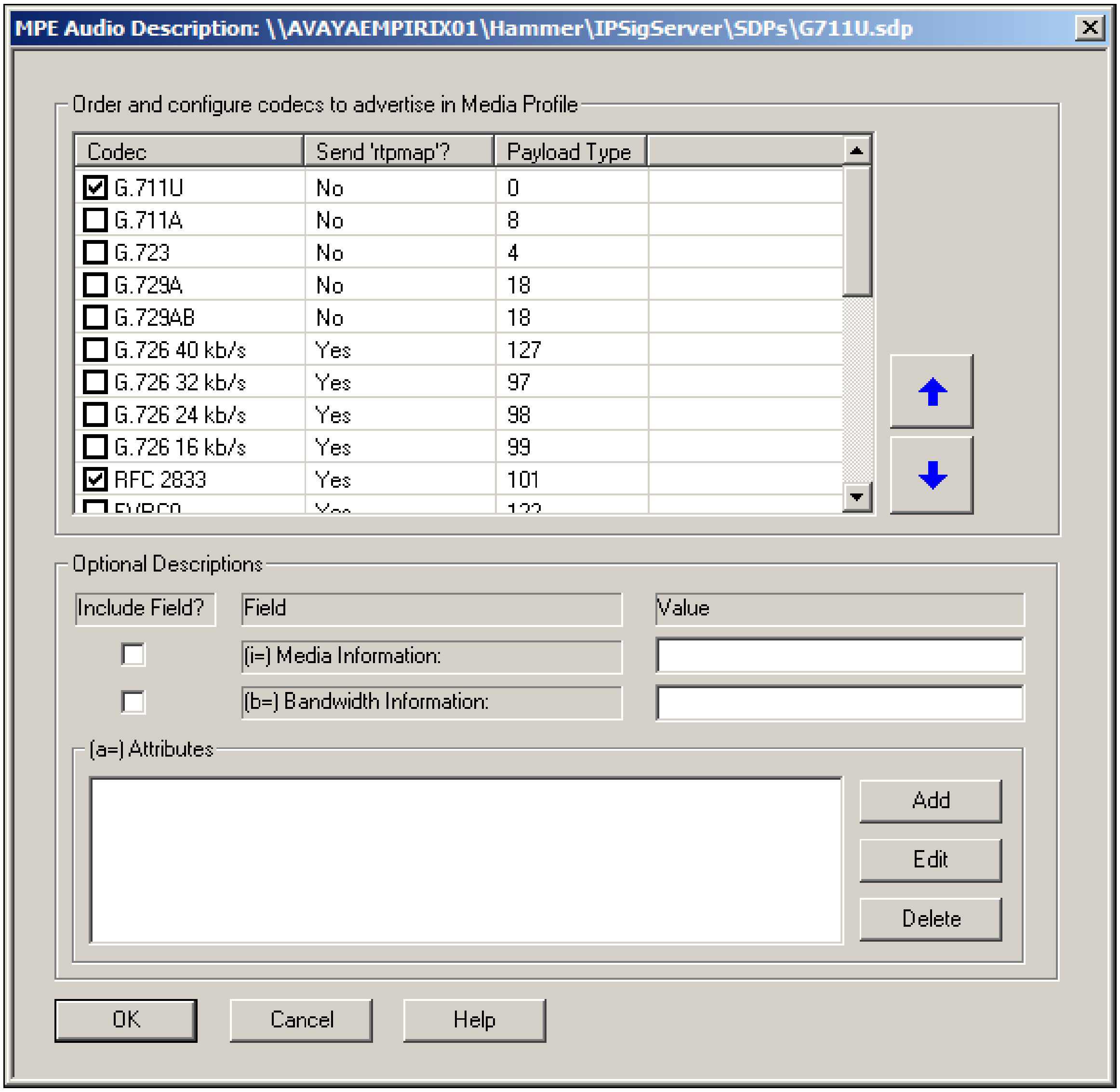
[5] *Empirix Hammer IP Installation Guide,* May 2015, available from Empirix.

APPENDIX A: Configure Media Profile on Empirix Hammer IP

The following windows show the configuration of the **Media Profile** used in the **Media** tab for the originating and terminating channel groups. To access this window, click on the ellipses button (…) by the **Media Profile** field in the **Media** tab. Click on the **Audio Description** button to view the codecs that will be advertised by the Hammer IP when placing a call.



The following window shows the codecs selected for this profile. This **Media Profile** was already created and named *G711U.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.



APPENDIX B: Enable Agent Login

This appendix describes how to configure Communication Manager and Hammer IP to allow H.323 endpoints to login as agents.

Note 1: This feature is applicable when the terminating channels are emulating H.323 endpoints.

Note 2: Verify that **Expert Agent Selection (EAS)** is enabled in the **system-parameters customer-options** form and the **system-parameters features** form.

Note 3: See **Appendix C** to determine if H.323 endpoints need to be unregistered before proceeding.

First, create a hunt group/skill for the H.323 endpoints to log into. Configure the fields in **bold** as shown below. The Hammer test should then place calls to the **Group Extension**.

```
add hunt-group 70                                           Page   1 of   4
                                  HUNT GROUP

            Group Number: 70                                  ACD? y
              Group Name: Empirix Hunt                      Queue? y
         Group Extension: 42500                            Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                  MM Early Answer? n
           Security Code:             Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:
```

On **Page 2**, set the **Skill** field to *y*. Optionally, set the **Measured** field to *internal* to use the monitor BCMS commands to determine if agents are logged into a specific hunt group.

```
add hunt-group 70                                           Page   2 of   4
                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n        Service Level Target (% in sec): 80 in 20
                 Measured: internal
     Supervisor Extension:

      Controlling Adjunct: none

        VuStats Objective:

   Multiple Call Handling: none

 Timed ACW Interval (sec):          After Xfer or Held Call Drops? n
```

Next, add an **agent-loginID** for each agent required by the Hammer test and specify a **Password** as shown below.

```
add agent-loginID 42001                                      Page   1 of   2
                                 AGENT LOGINID

                Login ID: 42001                                  AAS? n
                    Name: Agent 1                              AUDIX? n
                      TN: 1                            LWC Reception: spe
                     COR: 1                   LWC Log External Calls? n
           Coverage Path:                   AUDIX Name for Messaging:
           Security Code:
                                        LoginID for ISDN/SIP Display? n
                                                            Password: 1234
                                             Password (enter again): 1234
                                                         Auto Answer: station
                                                   MIA Across Skills: system
                                           ACW Agent Considered Idle: system
                                           Aux Work Reason Code Type: system
                                             Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                            Forced Agent Logout Time:   :
     WARNING:  Agent must log in again before changes take effect
```

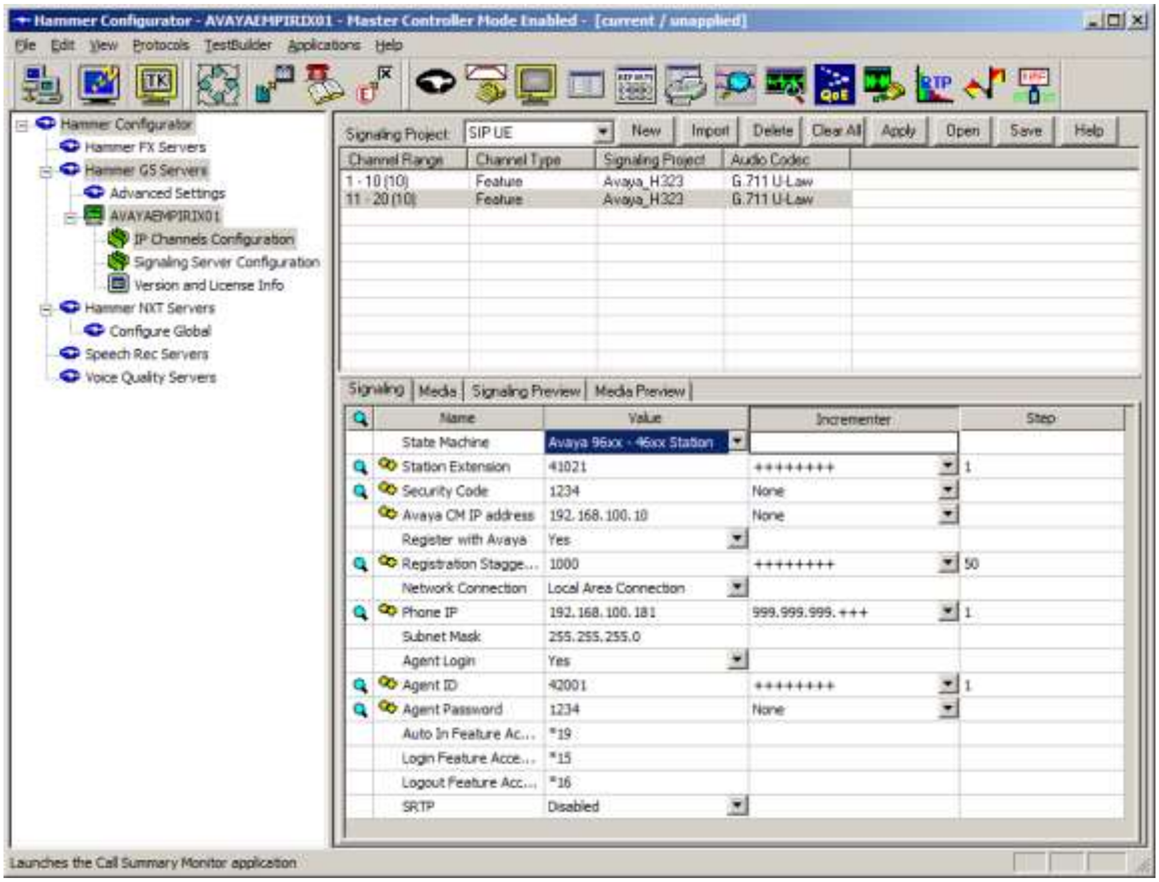
On **Page 2**, set the **SN** field to the hunt group number configured above and set the **SL** field to a valid value.

```
add agent-loginID 42001                                      Page   2 of   2
                                 AGENT LOGINID
      Direct Agent Skill:                              Service Objective? n
Call Handling Preference: skill-level              Local Call Preference? n

    SN   RL SL          SN   RL SL
 1: 70      1       16:
```

Finally, assign feature access codes (FACs) for agent login features, including Login, Logout, and Auto-In, as shown below.

```
change feature-access-codes                                     Page   5 of  10
                               FEATURE ACCESS CODE (FAC)

                                 Call Center Features
  AGENT WORK MODES
                    After Call Work Access Code:
                             Assist Access Code:
                            Auto-In Access Code: *19
                           Aux Work Access Code:
                              Login Access Code: *15
                             Logout Access Code: *16
                          Manual-in Access Code:
  SERVICE OBSERVING
             Service Observing Listen Only Access Code:
             Service Observing Listen/Talk Access Code:
                 Service Observing No Talk Access Code:
   Service Observing Next Call Listen Only Access Code:
 Service Observing by Location Listen Only Access Code:
 Service Observing by Location Listen/Talk Access Code:
```

This completes the configuration on the Communication Manager side. To enable agent login on Hammer IP, navigate to the terminating H.323 endpoints and enable **Agent Login** on the **Signaling** tab as shown below. Specify the **Agent Password** as configured in the agent-loginID and specify the FACs for agent login, logout, and auto-in feature.



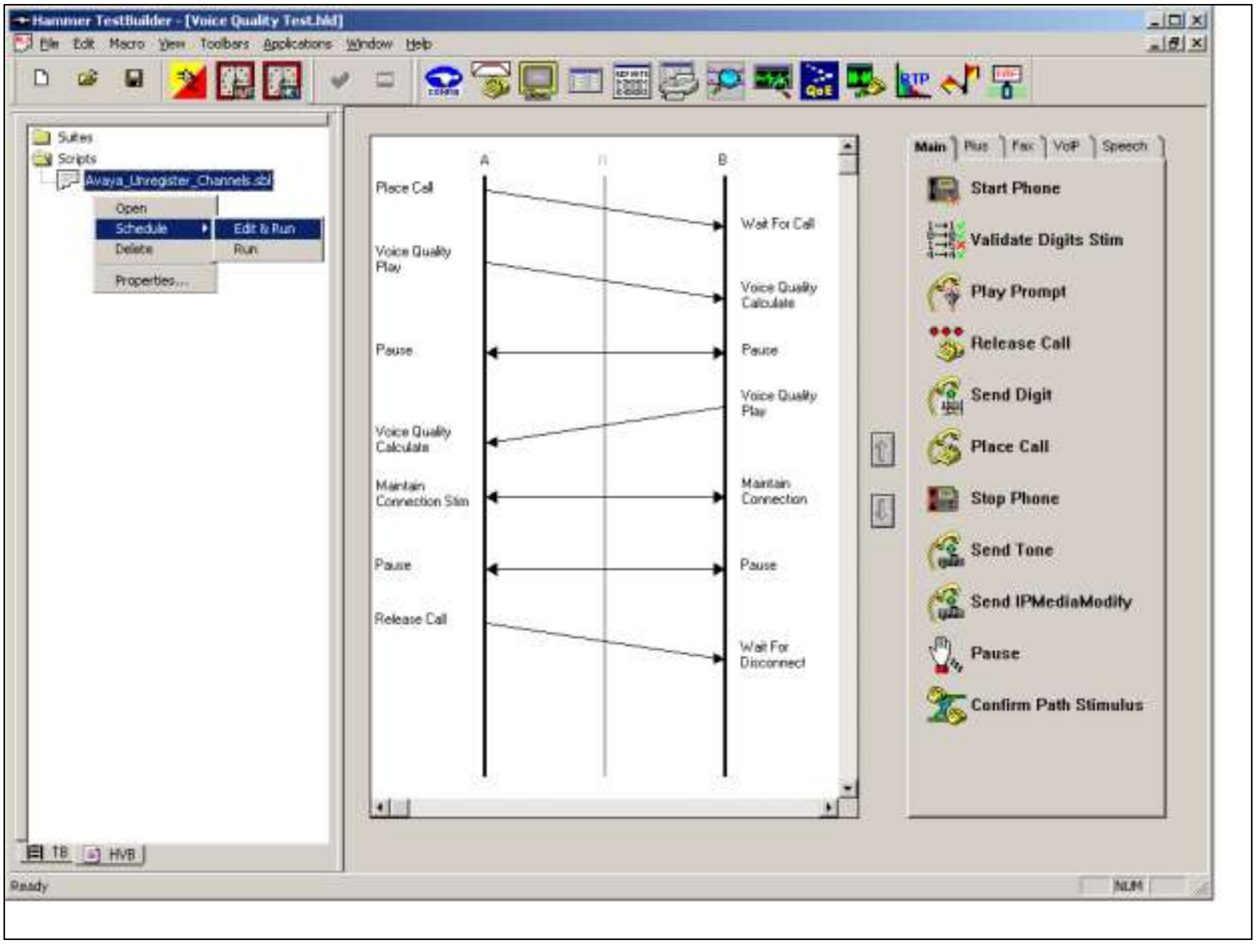
To verify if the H.323 endpoints are logged in as agents, use the **monitor bcms skill <*HG#*>** command, where <*HG#*> is the hunt group number.

```
monitor bcms skill 70                                           Page   1 of   2
                           BCMS SKILL (AGENT) STATUS

        Skill: 70                             Date:   12:25 FRI APR 17 2015
   Skill Name: Empirix Hunt
Calls Waiting:    0                         Acceptable Service Level: 20
  Oldest Call:   0:00                         % Within Service Level:

Staffed: 10  Avail: 10  ACD: 0   ACW: 0   AUX: 0   Extn Calls: 0   Other: 0

                                                          ACD   EXT IN  EXT OUT
AGENT NAME      LOGIN ID      EXT           STATE   TIME  CALLS   CALLS   CALLS

Agent 1         42001         41021         Avail   14:36    0       0       0
Agent 10        42010         41030         Avail   14:36    0       0       0
Agent 2         42002         41022         Avail   14:36    0       0       0
Agent 3         42003         41023         Avail   14:36    0       0       0
Agent 4         42004         41024         Avail   14:36    0       0       0
Agent 5         42005         41025         Avail   14:36    0       0       0
Agent 6         42006         41026         Avail   14:36    0       0       0
Agent 7         42007         41027         Avail   14:36    0       0       0
             NOTE: Calls Waiting include Calls Ringing and in Queue
```

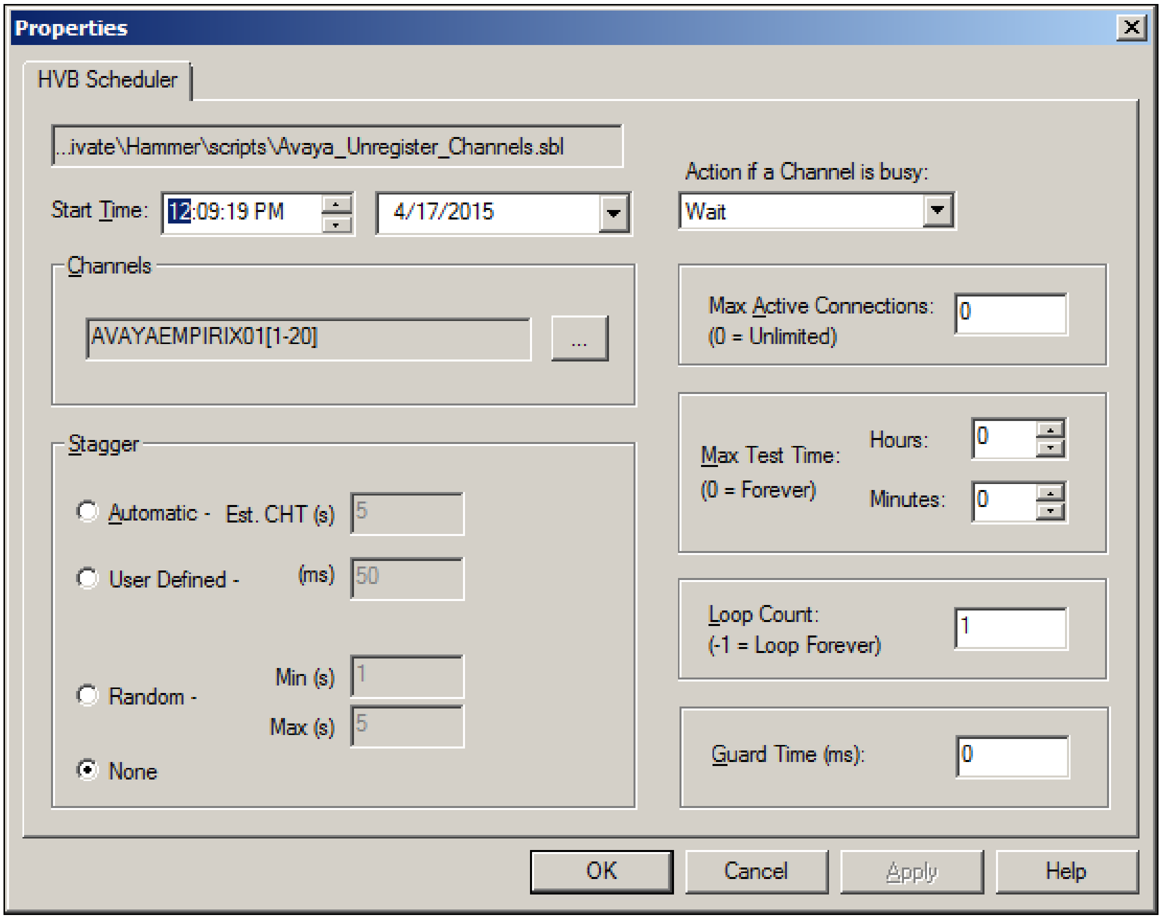
APPENDIX C: Unregister H.323 Endpoints

If transitioning from one Hammer configuration, where H.323 endpoints were used, to another configuration that uses agent login or vice versa, it is necessary to unregister the H.323 endpoints prior to proceeding with the new Hammer test. To unregister the H.323 endpoints, go to the **Hammer TestBuilder** and select the **HVB** tab at the bottom left-hand corner of the window. Right-mouse click on the `Avaya_Unregister_Channels.sbl` script and choose **Schedule → Edit & Run** as shown below.

Note: The `Avaya_Unregister_Channels.sbl` script is available by default with Hammer IP. However, the script needs to be imported and compiled (not shown) first.



In the Properties window, select all the channels associated and run this script once (i.e., **Loop Count** is set to 1). Click **OK**.



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.