Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager and a Survivable Core Server 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 and a Survivable Core Server 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 to Avaya Aura® Communication Manager with a Survivable Core Server. When communication to the Primary Controller (main server) is lost, the survivable core server option allows IP endpoints to register with a Survivable Core Server or Enterprise Survivable Server (ESS). 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 solution with Avaya Aura® Communication Manager and a Survivable Core Server 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 to Avaya Aura® Communication Manager with a Survivable Core Server. When communication to the Primary Controller (main server) is lost, the survivable core server option allows IP endpoints to register with a Survivable Core Server or Enterprise Survivable Server (ESS). 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.

2 General Test Approach

Interoperability compliance testing was performed by originating and terminating calls using H.323 endpoint channels on Hammer IP and establishing the calls from a Communication Manager simulating the PSTN and routing calls to another Communication Manager with a survivable core server. While calls were being placed, Communication Manager and its media gateway at the main site were failed triggering a failover to the ESS. After the failover to the ESS was completed, the terminating Hammer IP channels re-registered with the ESS and the calls continued to complete successfully. The failover to the ESS was triggered while the terminating channels were in different call states, such as idle and call in progress. 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.

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 to the main site, failover to the ESS, send voice media, and provide voice quality metrics. The following features and functionality were covered:

- H.323 endpoint registration with Communication Manager.
- Ability of Hammer IP to obtain the IP addresses of Communication Manager at the main and ESS sites during the H.323 registration phase.
- Originating and terminating calls through Communication Manager during a failover from the main site to the ESS.
- Failover from the main site to the ESS were performed under various conditions, including:
  - terminating channels in different call states, such as idle, call setup, or call in progress
  - Direct IP-to-IP media (also known as "Shuffling") enabled and disabled
  - SRTP enabled and disabled
  - terminating channels serving as call center agents and logged into a hunt group
  - Avaya H.323 Time-to-Service (TTS) enabled and disabled
  - running DTMF and voice quality test scripts
  - support of G.711mu-law codec

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 registering H.323 channels with Communication Manager and failing over from the main server to the ESS. During compliance testing, it was noted that in the test configuration, Hammer IP channels would re-register with the ESS about 1 minute after the failure of the main site. Successful test calls would then resume a few minutes after successful re-registration. Failback from the ESS to the primary Communication Manager is currently not supported. The following observations were noted:

1. Upon starting Hammer IP services, Hammer IP H.323 endpoints had to first register with the main or primary Communication Manager, otherwise it would not be able to failover to the ESS. Hammer IP obtains the ESS IP address during the first registration with the primary Communication Manager. The main Communication Manager IP address is configured explicitly on Hammer IP and the ESS IP address is derived from the IP Network Region on Communication Manager during H.323 registration in the Register Confirm message.

2. Failover from the main site to the ESS required failing the main Communication Manager and its associated media gateway simultaneously. The main Communication Manager was disconnected from the network to simulate a network outage and the media

gateway was rebooted to trigger the failover to ESS. The main Communication Manager had to remain down while Hammer IP calls were being terminated on the ESS.

3. Upon re-starting Hammer IP services, verify that there are no Hammer IP H.323 channels in the registration list (i.e., **list registered-stations** command) of either the primary Communication Manager or ESS.

4. Upon re-starting a Hammer IP test script, verify that Hammer IP H.323 channels are registered with either the primary Communication Manager or ESS, but not both. That is, only one Communication Manager should have entries in its registration list (i.e., **list registered-stations** command). Also, verify that the TCP socket is up; otherwise, deregister the H.323 channels, restart Hammer IP services, and proceed with the test.

5. Hammer IP H.323 channels that are emulating call center agents, logged into a hunt group, must be idle on the primary Communication Manager prior to performing a failover to the ESS. If there is a Hammer IP test script running while the H.323 channels are logged in as agents, a failover to the ESS will fail.

6. Failback from the ESS to the primary Communication Manager is currently not supported. Prior to re-starting Hammer IP services, the H.323 channels should be deregistered with the ESS by either running the **reset ip-stations** command on the ESS or running the Hammer IP unregister script described in **Appendix B**.

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.

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, which consisted of a Communication Manager simulating the PSTN and routing calls to a primary Communication Manager with a Survivable Core Server (or ESS). In this configuration, Communication Manager, simulating the PSTN, receives calls from Hammer IP, which emulates H.323 endpoints. The call is routed to either the main or backup site over a H.323 trunk. The call is then routed back to Hammer IP and terminated to another H.323 endpoint. The terminating Hammer channels are either registered to the main or backup site, but not both simultaneously. When a failure occurs at the main site, a failover to the backup site ensues. This results in Hammer IP re-registering its terminating channels with the ESS. After the failover to the ESS has been completed, the calls should continue to complete successfully. While the calls are established, Hammer IP can send DTMF and/or voice media (i.e., RTP traffic) using an audio recording according to its test script. A voice quality test allows voice quality metrics to be provided at the end of each call. Failback from the backup site to the main site is currently not supported. To re-register the H.323 channels to the main site, Hammer IP channels need to be deregistered from the ESS and the Hammer IP services need to be restarted. 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.

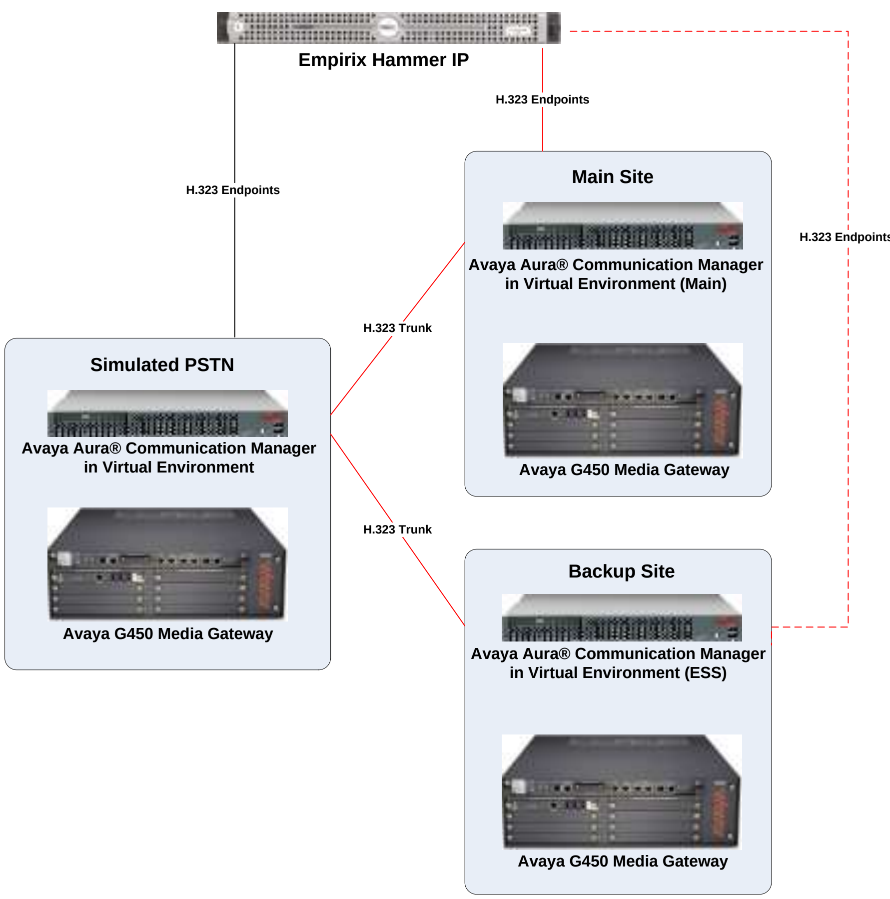


Figure 1: Empirix Hammer IP with Avaya Aura® Communication Manager

4 Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in a Virtual Environment	6.3 SP 12 (R016x.03.0.124.0 with Patch 22505)
Avaya G450 Media Gateway	FW 36.16.0
Empirix Hammer IP running on Microsoft Windows Server 2008 R2 with Dual 2.40 GHz Intel Xeon CPU and 12.0 GB of RAM	6.2.0.85

5 Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring the Communication Manager simulating the PSTN and Communication Manager with survivability option enabled.

5.1 Configure Simulated PSTN

This section provides the procedures for configuring Communication Manager, which simulates the PSTN. The originating Hammer channels will register with this Communication Manager as H.323 endpoints and place calls destined for the terminating Hammer channels that are registered with the main site or backup site. The simulated PSTN will then route the calls over a H.323 trunk to either the main or backup site. Therefore, two H.323 trunks are required. The configuration was performed using the System Access Terminal (SAT). The procedures include the following areas:

- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer H.323 Trunks to Main and Backup Sites
- Administer H.323 Stations for Originating Hammer Channels
- Administer AAR Call Routing

5.1.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name to Communication Manager (*procr*) and to the Communication Manager at the main and ESS sites. The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon14            192.168.100.10
ess-site            10.64.102.53
main-site           10.64.101.49
procr               10.64.102.110
procr6              ::
thorn-asm           10.64.102.105

( 7  of 7    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
```

5.1.2 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 codec was used.

```
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. The H.323 signaling group should also have **Media Encryption** enabled, if SRTP is required.

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

5.1.3 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: avaya.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: 3329
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
```

5.1.4 Administer H.323 Trunk to Main and Backup Sites

In this section, two H.323 trunks are created, one for the main site and another one for the backup site (or ESS). Trunk group 15 routes calls to the main site and trunk group 16 routes calls to the backup site. If Communication Manager is in-service at both sites, then both trunk groups will be active simultaneously. AAR call routing described in **Section 5.1.6** will give preferential routing treatment to the main site. If the main site is down, then calls will be routed to the backup site.

5.1.4.1 Administer H.323 Trunk to Main Site

Prior to configuring a H.323 trunk group for communication with the main site, a H.323 signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *h.323*.
- Set the **Trunk Group for Channel Selection** field to the trunk group number (e.g., *15*).
- Specify the local Communication Manager (*procr*) and the main site as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *1720* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.

- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Set the **Media Encryption** field to *y* if SRTP is required for the test calls. The **IP Codec Set** form should also have SRTP enabled, if it is required.
- The **Direct IP-IP Audio Connections** field was disabled on this form.

The default values for the other fields may be used.

```
add signaling-group 15                                        Page   1 of   6
                                SIGNALING GROUP

 Group Number: 15             Group Type: h.323
          SBS? n           Remote Office? n          Max number of NCA TSC: 0
        Q-SIP? n                                      Max number of CA TSC: 0
     IP Video? n                                    Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 15      X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: a          Network Call Transfer? n
                                                  T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: procr                  Far-end Node Name: main-site
 Near-end Listen Port: 1720                 Far-end Listen Port: 1720
                                         Far-end Network Region: 1
         LRQ Required? n                 Calls Share IP Signaling Connection? n
         RRQ Required? n
     Media Encryption? n                     Bypass If IP Threshold Exceeded? n
                                                      H.235 Annex H Required? n
         DTMF over IP: out-of-band             Direct IP-IP Audio Connections? n
  Link Loss Delay Timer(sec): 90                        IP Audio Hairpinning? n
         Enable Layer 3 Test? n                    Interworking Message: PROGress
H.323 Station Outgoing Direct Media? n   DCP/Analog Bearer Capability: 3.1kHz
```

Configure the **Trunk Group** form as shown below. This trunk group is used for H.323 calls to Hammer IP registered with the main site. Set the **Group Type** field to *isdn*, set the **Carrier Medium** to *H.323*, 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 H.323 trunk group. Accept the default values for the remaining fields.

```
add trunk-group 15                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 15                   Group Type: isdn          CDR Reports: y
  Group Name: to main-site                COR: 1        TN: 1        TAC: 1015
   Direction: two-way        Outgoing Display? n         Carrier Medium: H.323
 Dial Access? n              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 15
                                                     Number of Members: 40
```

5.1.4.2 Administer H.323 Trunk to Backup Site

Prior to configuring a H.323 trunk group for communication with the ESS site, a H.323 signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *h.323*.
- Set the **Trunk Group for Channel Selection** field to the trunk group number (e.g., *16*).
- Specify the local Communication Manager (*procr*) and the ESS site as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *1720* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Set the **Media Encryption** field to *y* if SRTP is required for the test calls. The **IP Codec Set** form should also have SRTP enabled, if it is required.
- The **Direct IP-IP Audio Connections** field was disabled on this form.

The default values for the other fields may be used.

```
add signaling-group 16                                             Page   1 of   6
                                SIGNALING GROUP

 Group Number: 16             Group Type: h.323
          SBS? n           Remote Office? n          Max number of NCA TSC: 0
        Q-SIP? n                                      Max number of CA TSC: 0
     IP Video? n                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 16      X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: a          Network Call Transfer? n
                                                  T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: procr                    Far-end Node Name: ess-site
 Near-end Listen Port: 1720                   Far-end Listen Port: 1720
                                           Far-end Network Region: 1
         LRQ Required? n                    Calls Share IP Signaling Connection? n
         RRQ Required? n
     Media Encryption? n                        Bypass If IP Threshold Exceeded? n
                                                         H.235 Annex H Required? n
         DTMF over IP: out-of-band               Direct IP-IP Audio Connections? n
  Link Loss Delay Timer(sec): 90                           IP Audio Hairpinning? n
         Enable Layer 3 Test? n                      Interworking Message: PROGress
H.323 Station Outgoing Direct Media? n   DCP/Analog Bearer Capability: 3.1kHz
```

Configure the **Trunk Group** form as shown below. This trunk group is used for H.323 calls to Hammer IP registered with the backup site. Set the **Group Type** field to *isdn*, set the **Carrier Medium** to *H.323*, 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 H.323 trunk group. Accept the default values for the remaining fields.

```
add trunk-group 16                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 16                   Group Type: isdn          CDR Reports: y
  Group Name: to ess-site                COR: 1        TN: 1        TAC: 1016
   Direction: two-way        Outgoing Display? n         Carrier Medium: H.323
 Dial Access? n              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 15
                                                    Number of Members: 40
```

5.1.5 Administer H.323 Stations for Originating Hammer Channels

Configure a H.323 station for each originating 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, 10 H.323 stations were used to originate calls with extensions ranging from 29001 to 29010. Repeat this procedure for each originating channel required by the Hammer test.

```
add station 29001                                           Page   1 of   5
                                     STATION

Extension: 29001                        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: 29001
            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
```

5.1.6 Administer AAR Call Routing

H.323 calls to the main and backup sites are routed over a H.323 trunk via AAR call routing. To route calls to the survivable core, dialed digits starting with “69” were used. To steer these calls to AAR call routing, configure the following entry in the **uniform-dialplan** form.

```
change uniform-dialplan 6                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert               Node
  Pattern       Len Del      Digits      Net Conv Num
 69              5   0                   aar  n
```

Configure the AAR analysis form and add an entry that routes digits beginning with “69” to route pattern 69 as shown below.

```
change aar analysis 6                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    6                     7    7     2000     aar          n
    69                    5    5     69       aar          n
    7                     7    7     2000     aar          n
    8                     7    7     2000     aar          n
    9                     7    7     2000     aar          n
```

Configure two preferences in **Route Pattern** 69 to route calls over H.323 trunk group 15 or 16 as shown below. Note that trunk group 15, which routes calls to the main site, is given preference. Also, the **LAR** (Look Ahead Routing) field for the first route preference was set to *next*. This allows Communication Manager to try the next route preference to the backup site, if the call is initially routed to the main site but fails due to a network failure downstream.

```
change route-pattern 69                                         Page   1 of   3
                    Pattern Number: 69     Pattern Name: to Surv Core
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 15   0                                                              n   user
 2: 16   0                                                              n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                          next
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

5.2 Configure Main Server and Survivable Core Server (or ESS)

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 Node Names
- Administer IP Codec Set
- Administer IP Network Region and Specify Backup Server
- Administer H.323 Trunk to Simulated PSTN
- Administer H.323 Stations for Terminating Hammer Channels

Note: This configuration is only performed at the main site. When the configuration is saved using the **save translations all** command, the same configuration will be saved to the backup site. Hence, no configuration is required at the backup site. In addition, it is assumed that Communication Manager with an ESS has already been configured. For reference, **Appendix C** provides a summary of the configuration required for a Survivable Core, but the reader should refer to **[2]** for more detailed documentation.

5.2.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name to Communication Manager simulating the PSTN (*thorn-cm*) and to the Communication Manager at the main (*procr*) and ESS (*sccmvip*) sites. The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                           Page   1 of   2
                                 IP NODE NAMES
    Name              IP Address
aam10114            10.64.101.14
default             0.0.0.0
procr               10.64.101.49
procr6              ::
sccm1               10.64.102.51
sccm2               10.64.102.52
sccmvip             10.64.102.53
sm10262             10.64.102.62
sm5031              10.64.50.31
thorn-cm            10.64.102.110

( 10 of 10   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
```

5.2.2 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 codec was used.

```
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:
```

5.2.3 Administer IP Network Region and Specify Backup Server

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: d4f27.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: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

On **Page 3**, specify the backup server or ESS (i.e., *sccmvip*). This alternate gatekeeper IP address is provided to Hammer IP during the H.23 registration phase. 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? y

BACKUP SERVERS(IN PRIORITY ORDER)    H.323 SECURITY PROFILES
 1   sccmvip                         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
```

5.2.4 Administer H.323 Trunk to Simulated PSTN

Prior to configuring a H.323 trunk group for communication with the simulated PSTN, a H.323 signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *h.323*.
- Set the **Trunk Group for Channel Selection** field to the trunk group number (e.g., *15*).
- Specify the local Communication Manager (*procr*) and the simulate PSTN (thorn-cm) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form. Note that *procr* refers to either the main or backup site depending on which Communication Manager this configuration resides.
- Ensure that the TCP port value of *1720* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Set the **Media Encryption** field to *y* if SRTP is required for the test calls. The **IP Codec Set** form should also have SRTP enabled, if it is required.
- The **Direct IP-IP Audio Connections** field was disabled on this form.

The default values for the other fields may be used.

```
add signaling-group 15                                      Page   1 of   6
                                SIGNALING GROUP

 Group Number: 15             Group Type: h.323
          SBS? n           Remote Office? n          Max number of NCA TSC: 0
        Q-SIP? n                                      Max number of CA TSC: 0
     IP Video? n                                    Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 15      X-Mobility/Wireless Type: NONE
      TSC Supplementary Service Protocol: a          Network Call Transfer? n
                                                  T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: procr                   Far-end Node Name: thorn-cm
 Near-end Listen Port: 1720                  Far-end Listen Port: 1720
                                          Far-end Network Region: 1
         LRQ Required? n                   Calls Share IP Signaling Connection? n
         RRQ Required? n
     Media Encryption? n                       Bypass If IP Threshold Exceeded? n
                                                       H.235 Annex H Required? n
         DTMF over IP: out-of-band               Direct IP-IP Audio Connections? n
  Link Loss Delay Timer(sec): 90                           IP Audio Hairpinning? n
         Enable Layer 3 Test? n                       Interworking Message: PROGress
H.323 Station Outgoing Direct Media? n   DCP/Analog Bearer Capability: 3.1kHz
```

Configure the **Trunk Group** form as shown below. This trunk group is used for H.323 calls to Hammer IP registered with the main site. Set the **Group Type** field to *isdn*, set the **Carrier Medium** to *H.323*, 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 H.323 trunk group. Accept the default values for the remaining fields.

```
add trunk-group 15                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 15                   Group Type: isdn          CDR Reports: y
  Group Name: to thorn-cm                 COR: 1        TN: 1        TAC: 115
   Direction: two-way       Outgoing Display? n         Carrier Medium: H.323
 Dial Access? n             Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 15
                                                    Number of Members: 40
```

5.2.5 Administer H.323 Stations for Terminating Hammer Channels

Configure a H.323 station for each H.323 terminating channel on 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, 10 H.323 stations were used with extensions ranging from 69001 to 69010. These 10 channels (extensions 69001 to 69010) were used to terminate calls. Repeat this procedure for each terminating channel required by the Hammer test.

```
add station 69001                                                 Page   1 of   5
                                     STATION

Extension: 69001                        Lock Messages? n              BCC: 0
     Type: 9630                         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: 69001
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 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
```

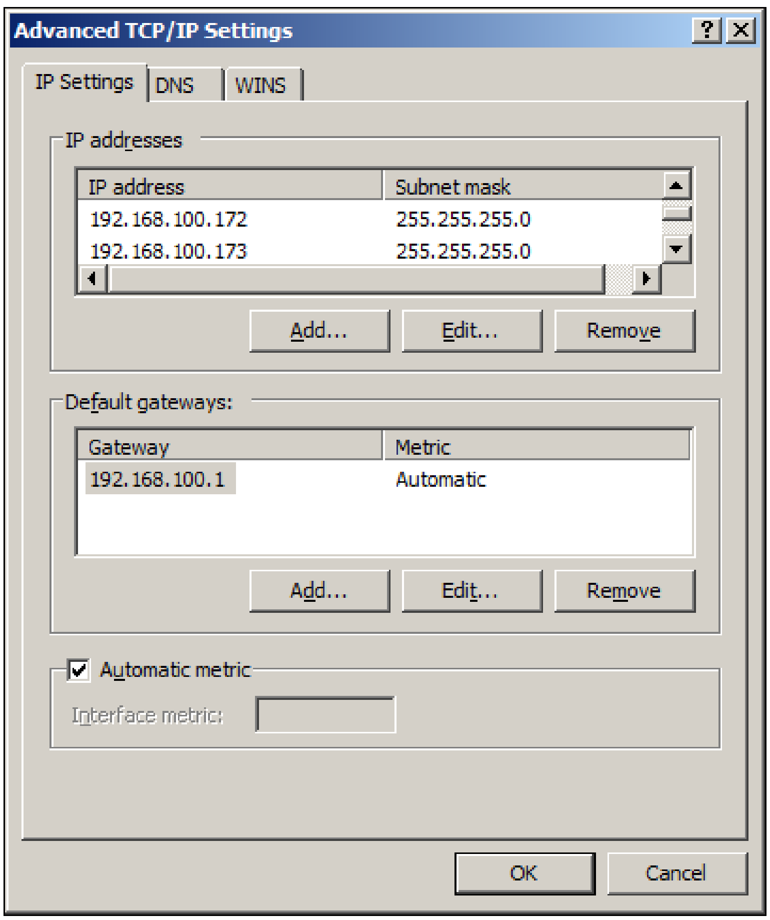
6 Configure Empirix Hammer IP

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**.
- Configure Hammer IP Advanced Settings.
- 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

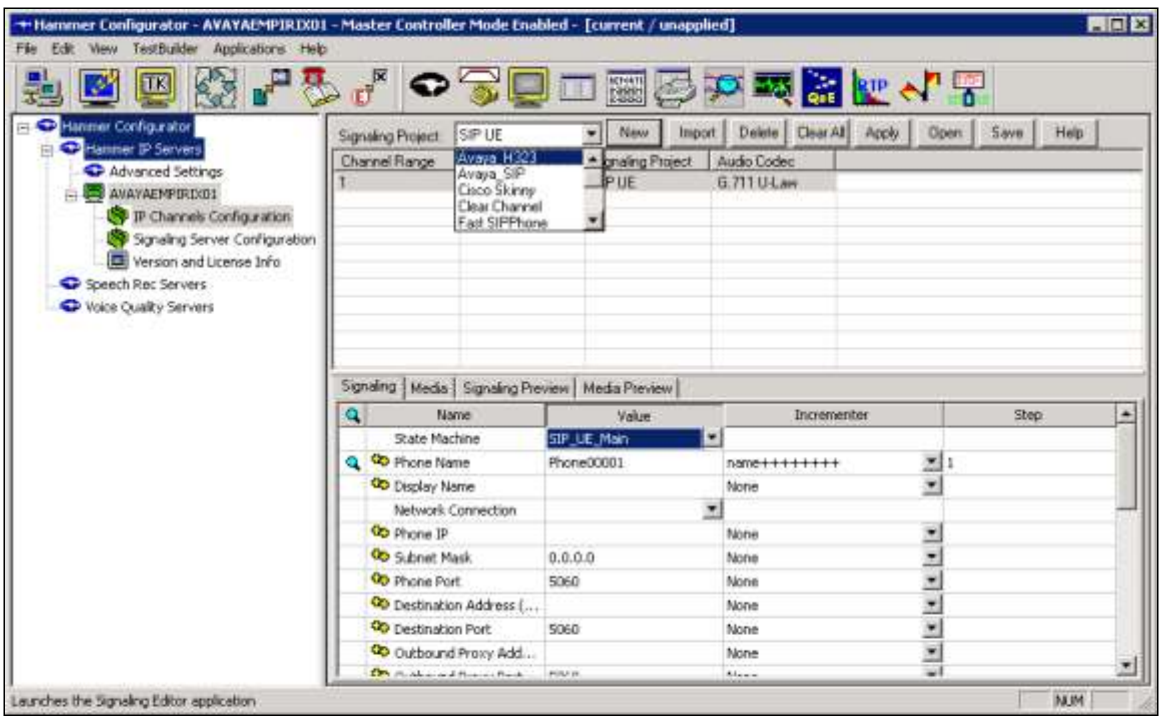
This section covers the configuration of originating and terminating channels and the PhoneBook on Hammer IP. In this configuration, the originating and terminating channels emulate H.323 endpoints.

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.

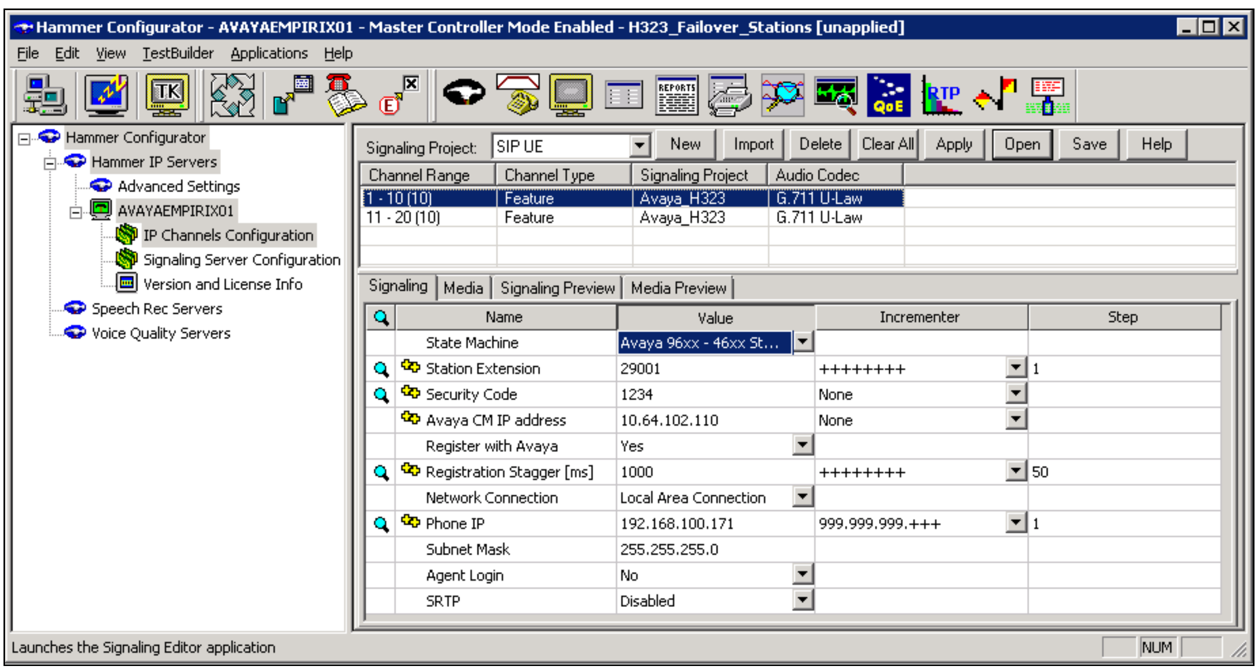
Note: It is assumed that Hammer IP is already in **Master Controller Mode**. To verify, check that the title bar of the **Hammer Configurator** indicates *Master Controller Mode Enabled* as shown below. It is also assumed that a system was already added to the configuration. In this configuration, the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



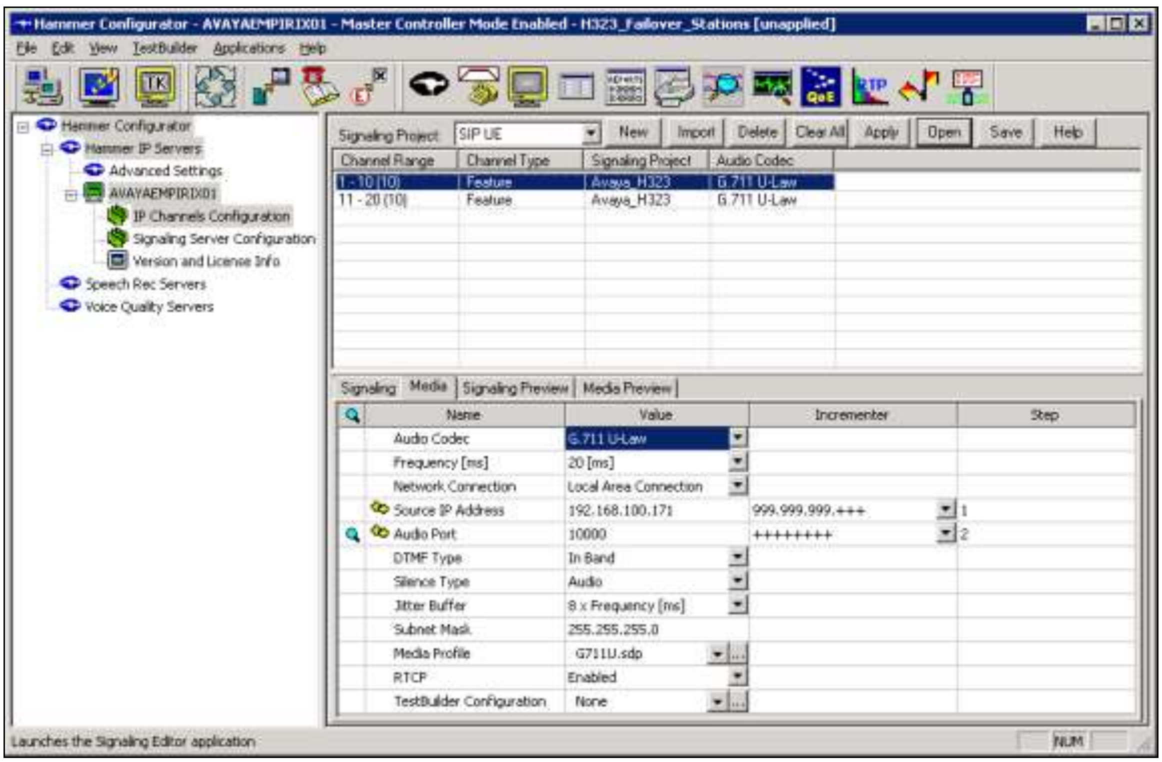
The first line in the grid that is highlighted in the figure below corresponds to the 10 originating channels. To set the number of channels in the group, click on the **Channel Range** cell in the grid and enter the number *10*. Set the **Channel Type** cell to *Feature*. The following fields in the **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., *29001*) and the **Incrementer** and **Step** fields should be set as shown so that the extension of the subsequent channels are incremented by one to cover extensions from *29001* to *29010*.
- **Security Code** should match the one configured in the corresponding **Station** form on Communication Manager.
- **Avaya CM IP Address** should be set to the IP address of Communication Manager simulating the PSTN.
- **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:

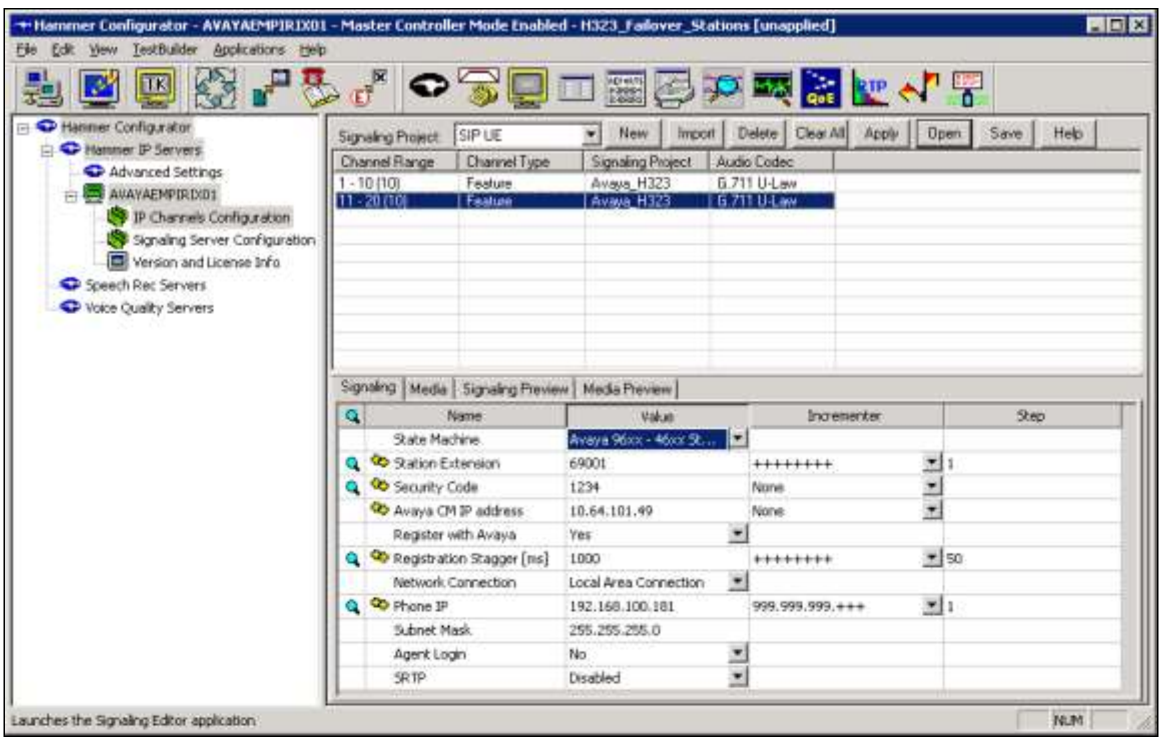
- **Audio Codec** should be set to the appropriate codec for the test. G711 U-Law was used during the compliance testing.
- **Frequency [ms]** should be set to the appropriate value for the specified codec. It should match the Packet Size [ms] field in the **IP Codec Set** form on Communication Manager 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.



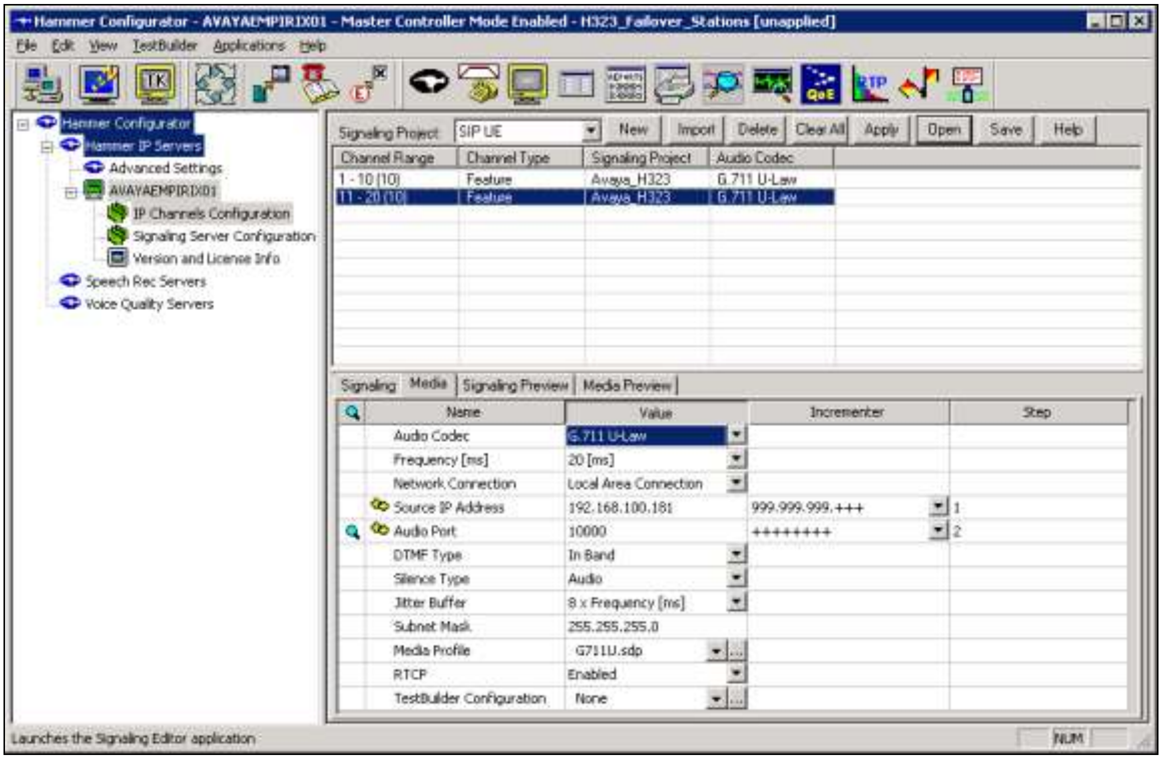
6.2.2 Configure Terminating Channels – H.323 Endpoints

The second line in the grid that is highlighted in the figure below corresponds to the second group of channels that will terminate calls. These channels register with the survivable core. 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 *69001* to *69010* 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 (not covered).

Note: The **Avaya CM IP address** field specifies the IP address of Communication Manager at the main site. During the H.323 registration phase, the IP address of the ESS is obtained from the Registration Confirmation message. After Hammer IP is restarted, it must always start by registering with the main site so that it can obtain the IP address of the ESS in the aforementioned H.323 message exchange. However, after a failover to the ESS has been performed, the test script may be restarted, if necessary. Restarting the test script does not re-register the H.323 channels.

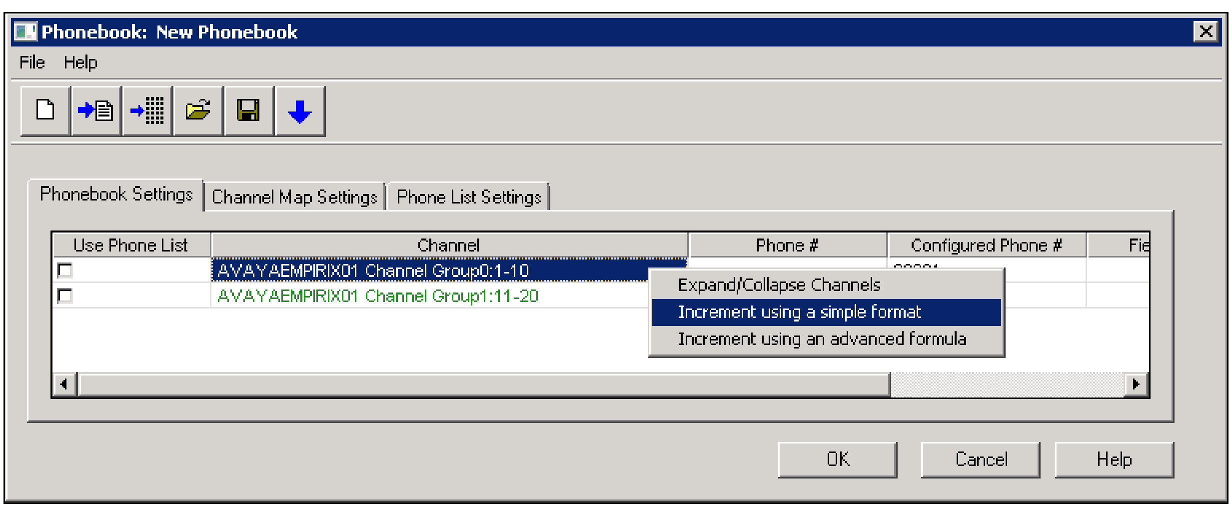


The **Media** tab for the group of terminating channels is shown below. The configuration is similar to the one for the group of originating channels except for the **Source IP Address** field.

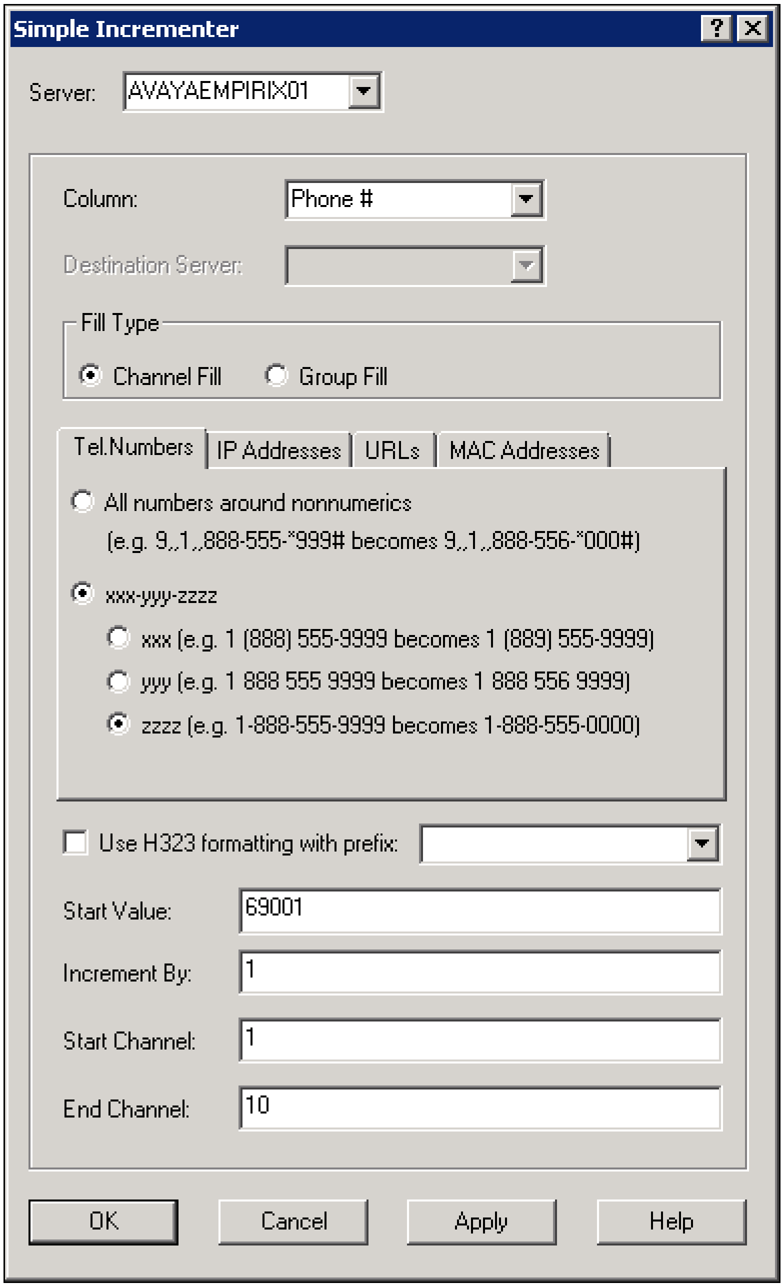


6.2.3 Configure the PhoneBook

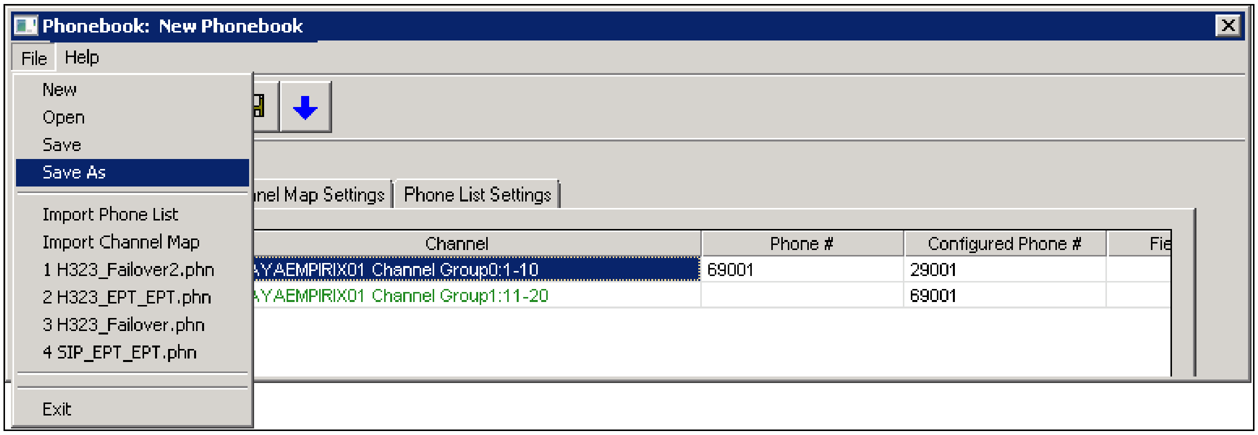
The **PhoneBook** is used to specify which number each originating channel should dial when placing a call. Click on the **PhoneBook** icon (not shown) in the **Hammer Configurator**. The **PhoneBook** window is displayed below. The **Channel** column is automatically displayed with the appropriate channel 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.



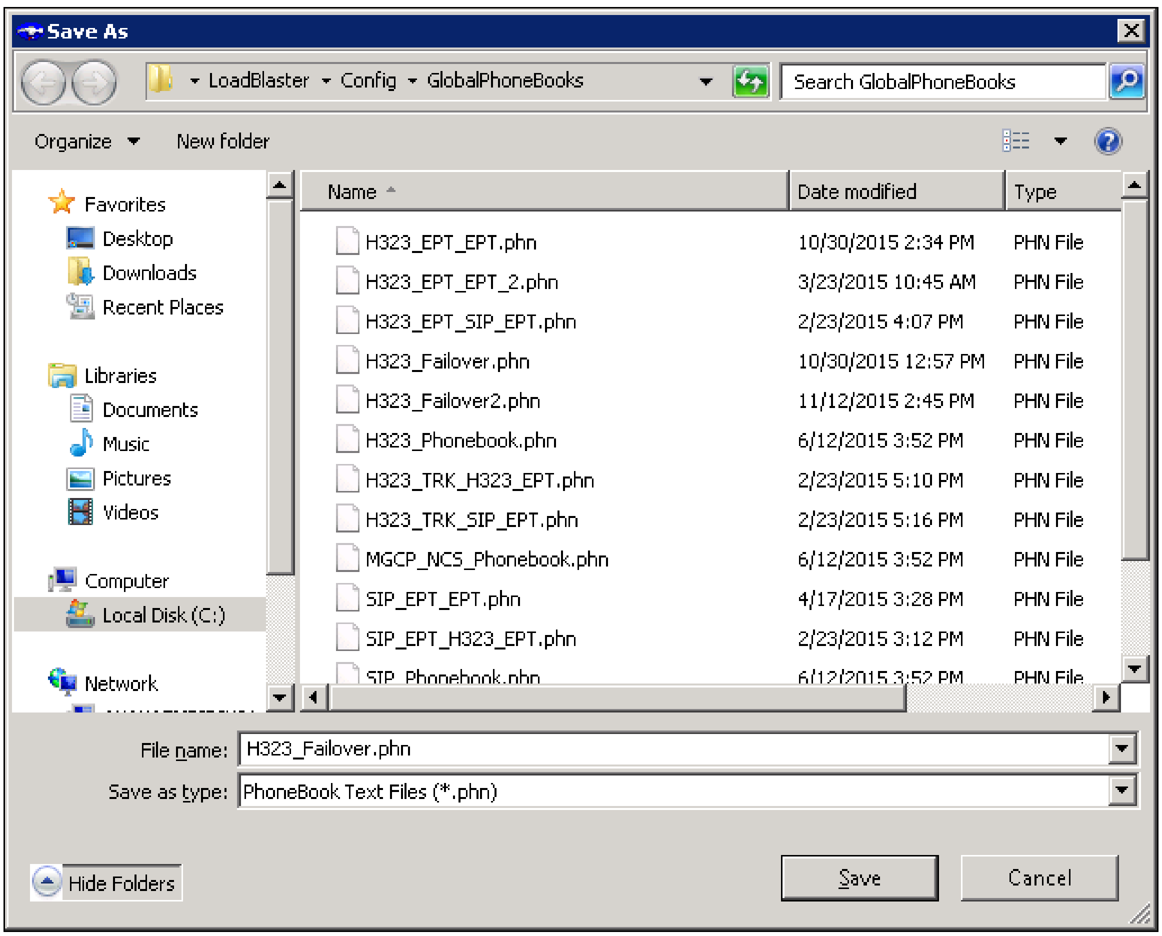
In the **Simple Incrementer** window, specify the number that the first originating channel should dial in the **Start Value** field. In this example, the first channel will dial *69001*, which corresponds to channel 11. Set the **Increment By** field to *1*. This specifies that the subsequent channels should increment the dialed number by one. For example, channel 1 will dial 69001, channel 2 will dial 69002, 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.



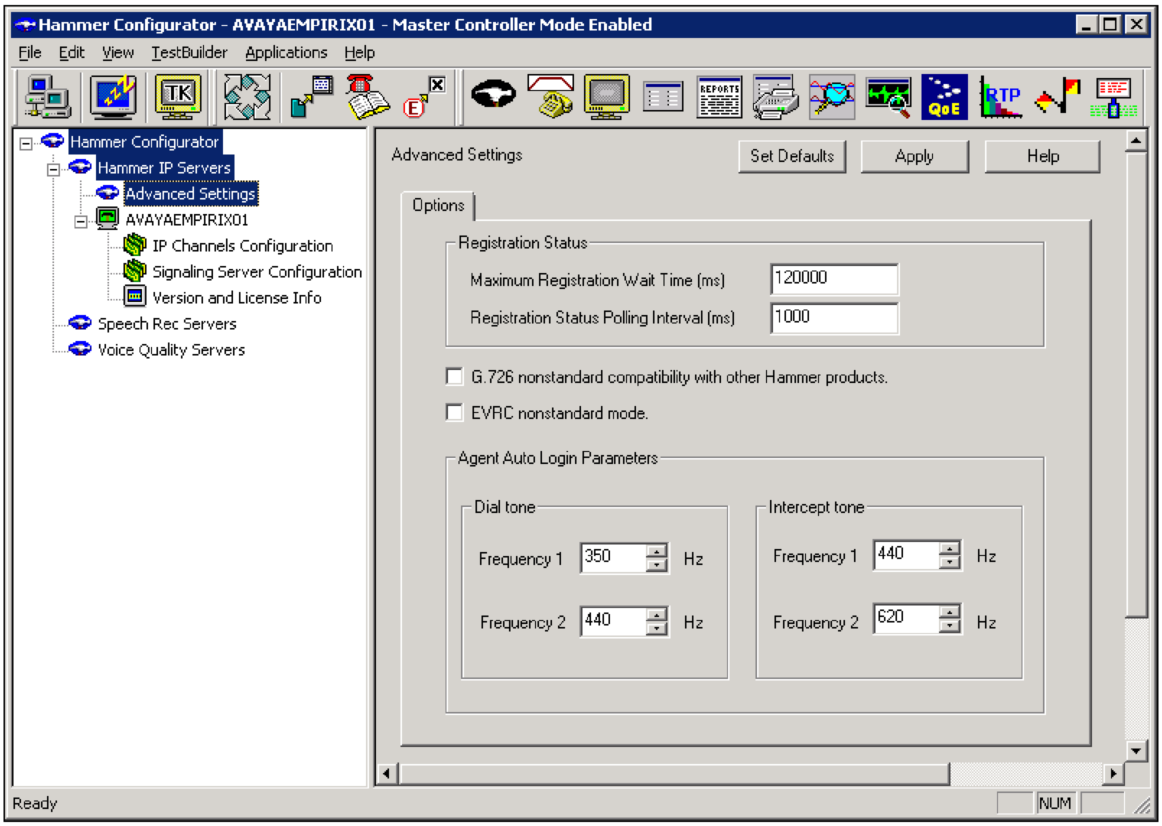
The PhoneBook is saved as *H323_Failover.phn* in the following window. This PhoneBook will be used when running the test.



6.3 Configure Hammer IP Advanced Settings

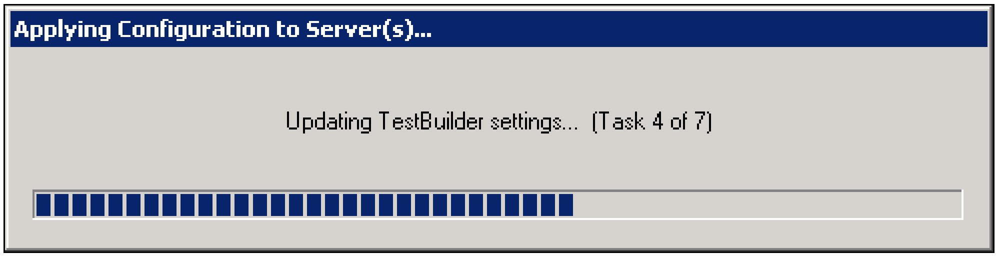
Select **Advanced Settings** in the left pane and configure fields in the **Registration Status** section. The **Maximum Registration Wait Time (ms)** field was set to *120000* and the **Registration Status Polling Interval (ms)** was set to *1000* in this configuration. This means that Hammer IP will check the channel registration status every second for 2 minutes. After 2 minutes, Hammer IP will stop trying to register the channels if they are not yet registered.

Note: The **Maximum Registration Wait Time** field should be set based on how long it takes for H.323 channels to register after a failover to the ESS.

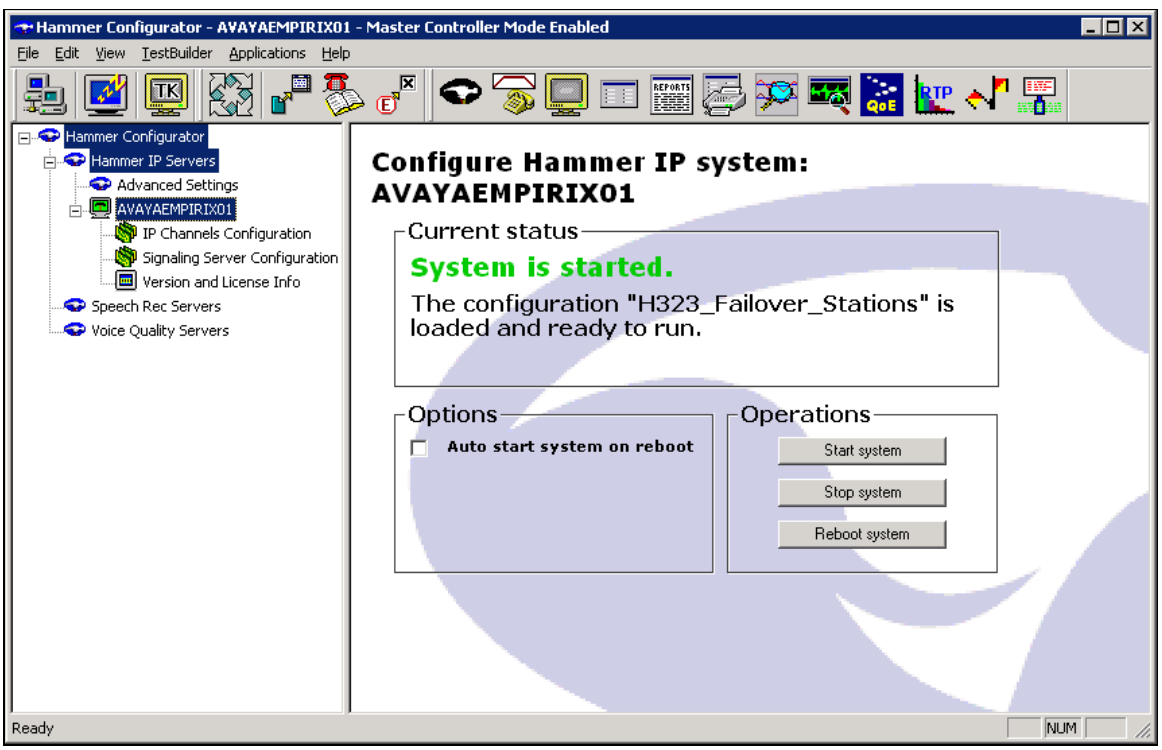


6.4 Applying the Hammer IP Configuration

This completes the configuration of Hammer IP. This configuration should be saved by clicking the **Save** button (not shown) 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 (not shown) 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_Failover_Stations*. When the system is started, the Hammer IP will register H.323 endpoints with Communication Manager.

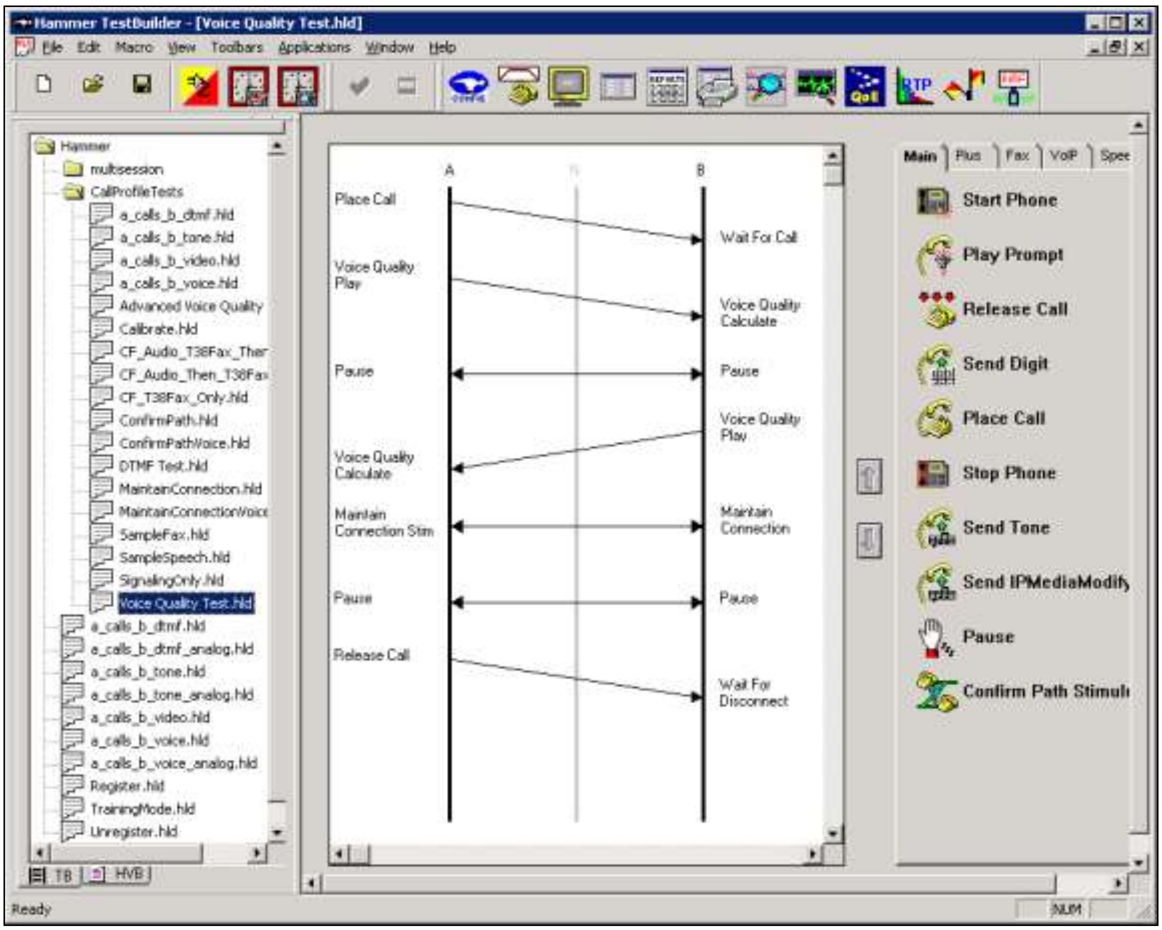


6.5 Configure and Run the Test Script

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

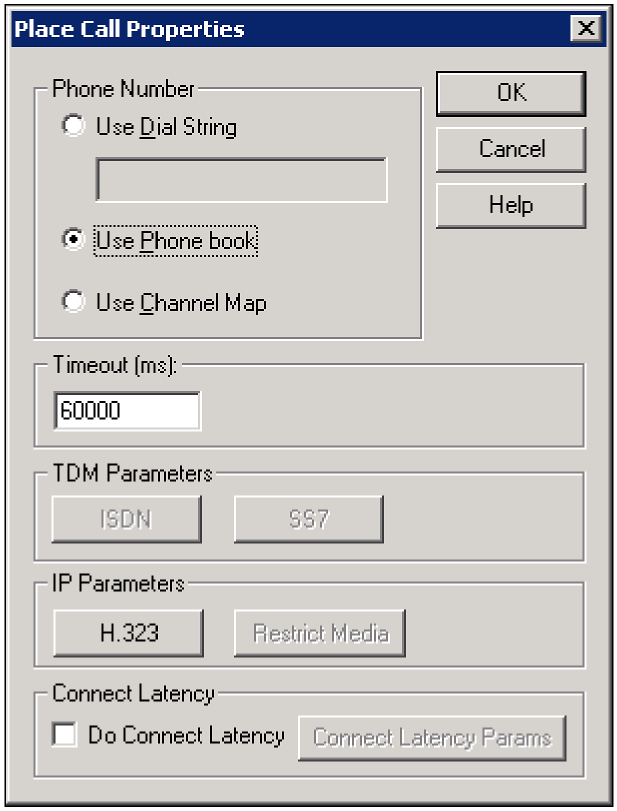
- `a_calls_b_dtmf.hld` to verify DTMF
- `Voice Quality Test.hld` to verify voice quality

The sample test script, `Voice Quality Test.hld`, establishes a VoIP call between two SIP endpoints on the 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.

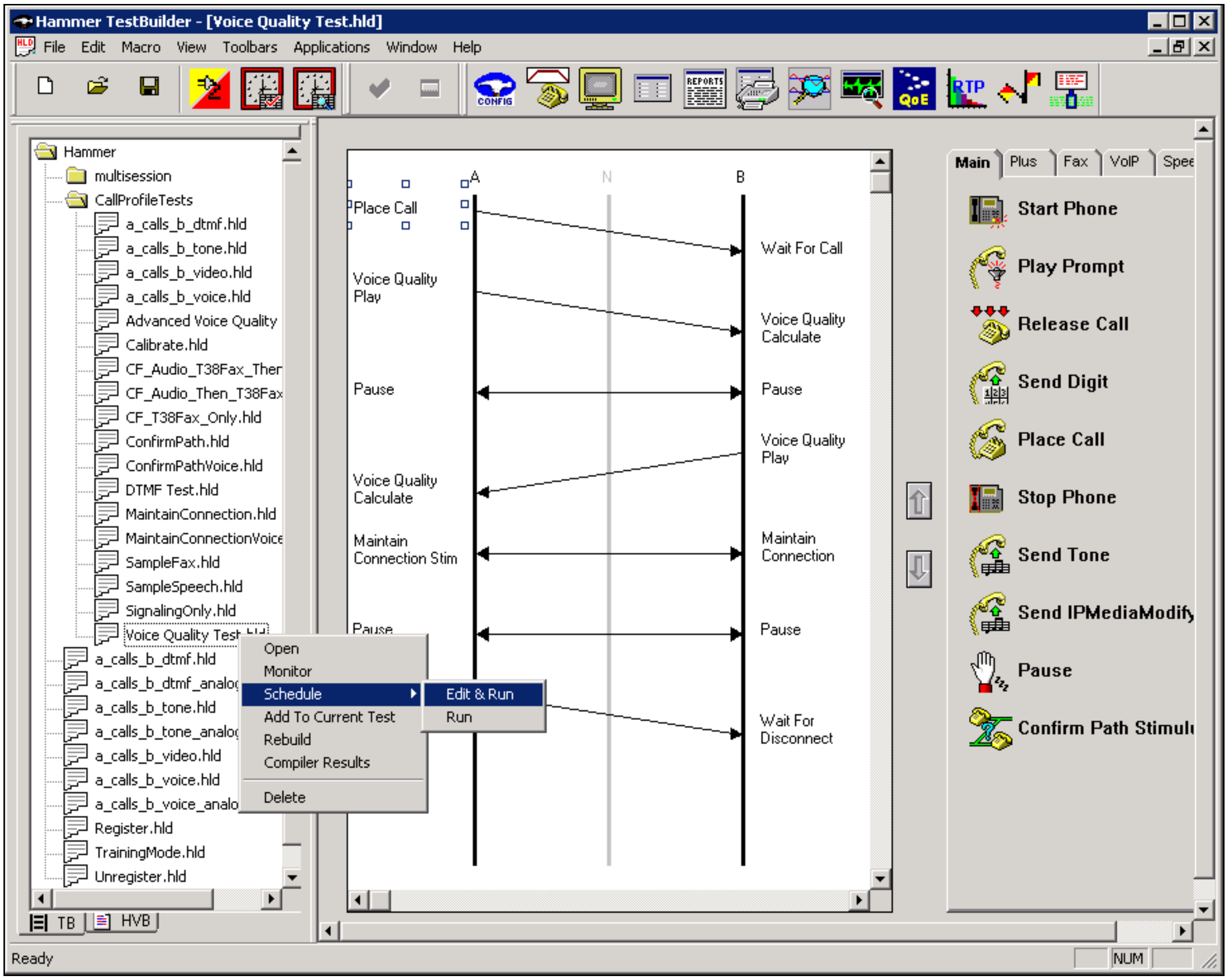


In the sample test script configured above, the A-side (originating H.323 endpoint) places a call to the B-side (terminating H.323 endpoint) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. The **Place Call Properties**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

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

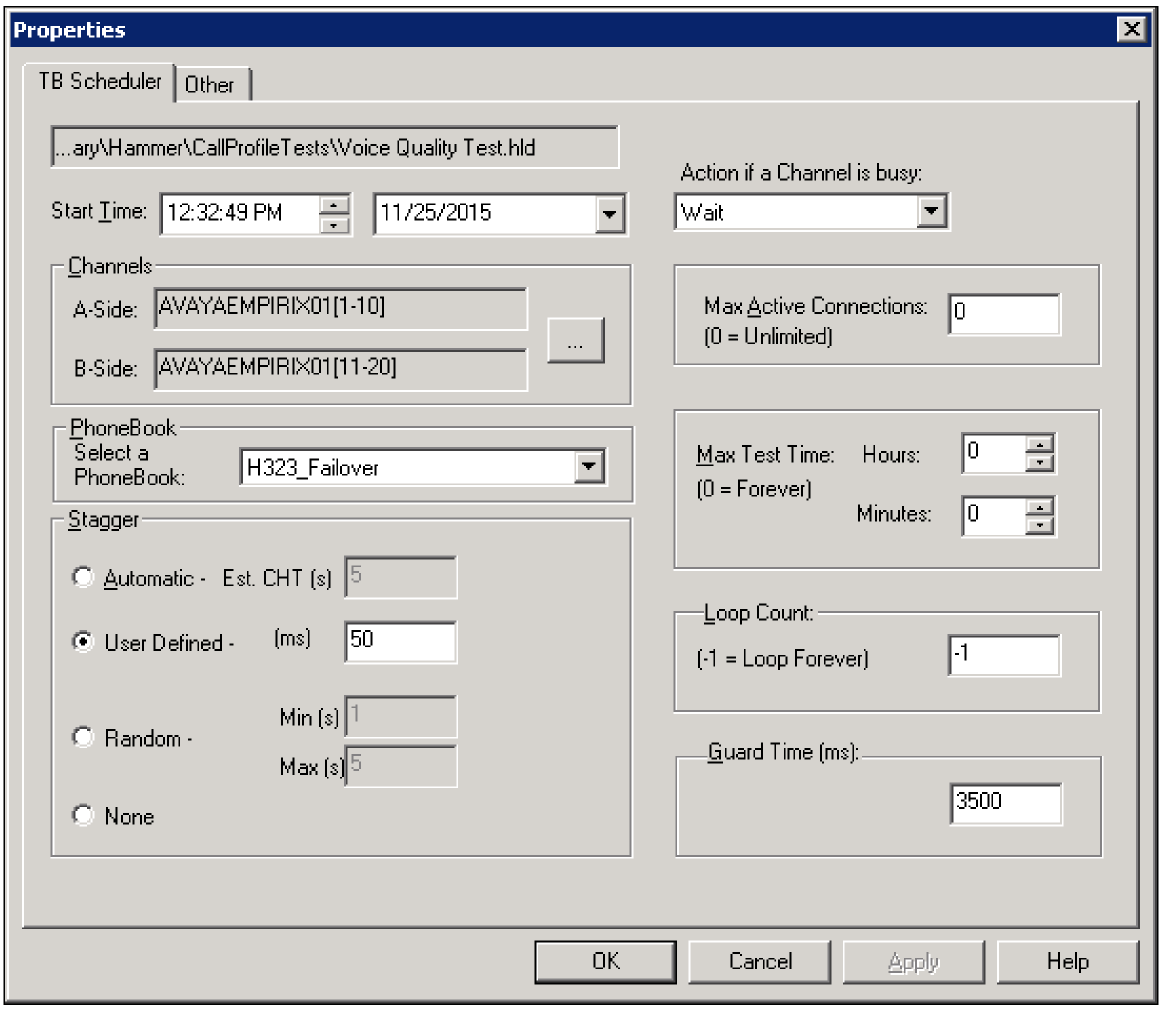


To run the test, right-mouse click on the test script in the left pane of the **Hammer TestBuilder** window and navigate to **Schedule→Edit & Run**. To re-run the test, the user can simply select **Schedule→Run**, if no changes are required.



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_Failover*). The H323_Failover 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. For the compliance test, the **Stagger** time was set to *50 ms*.

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
------------- --------- ---------- --- --------------------------------------
69001          9630      IP_Phone    y  192.168.100.181
               1         2.80           10.64.101.49
69002          9630      IP_Phone    y  192.168.100.182
               1         2.80           10.64.101.49
69003          9630      IP_Phone    y  192.168.100.183
               1         2.80           10.64.101.49
69004          9630      IP_Phone    y  192.168.100.184
               1         2.80           10.64.101.49
69005          9630      IP_Phone    y  192.168.100.185
               1         2.80           10.64.101.49
```

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*.

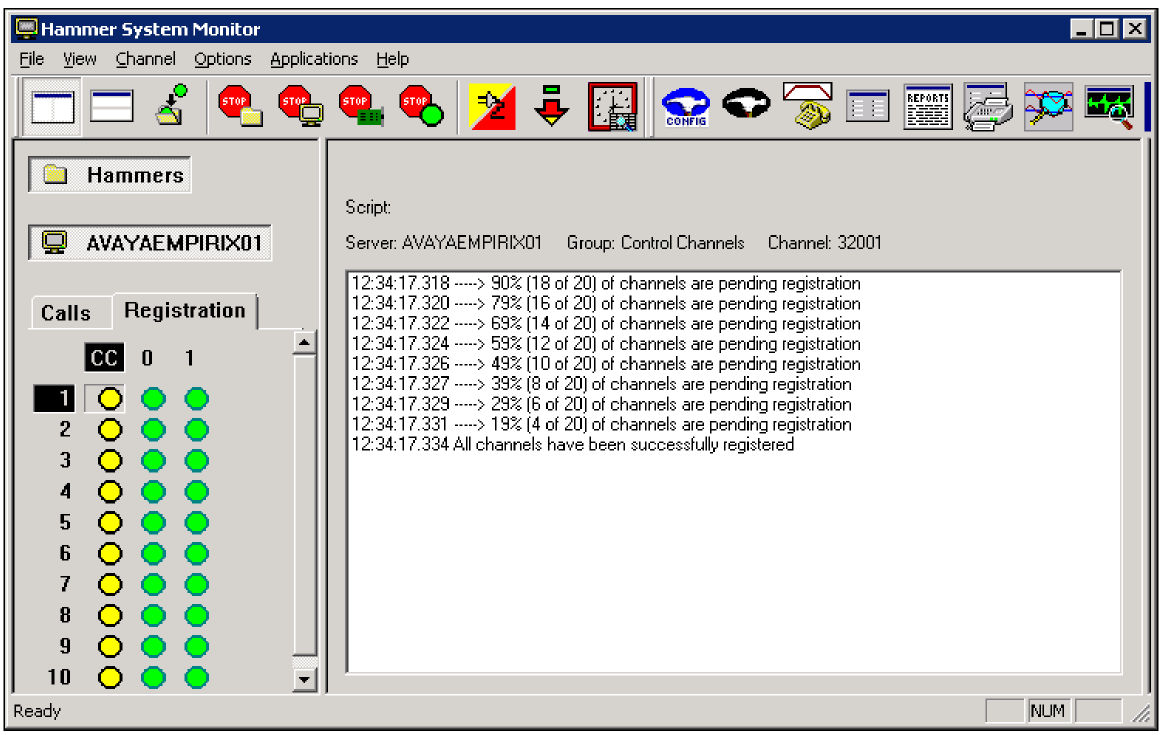
```
status station 69001                                            Page   1 of   9
                              GENERAL STATUS
     Administered Type: 9630                Service State: in-service/off-hook
        Connected Type: N/A            TCP Signal Status: connected
             Extension: 69001             Network Region: 1
                  Port: S00013        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: T00120

  Limit Incoming Calls? no

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

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*.

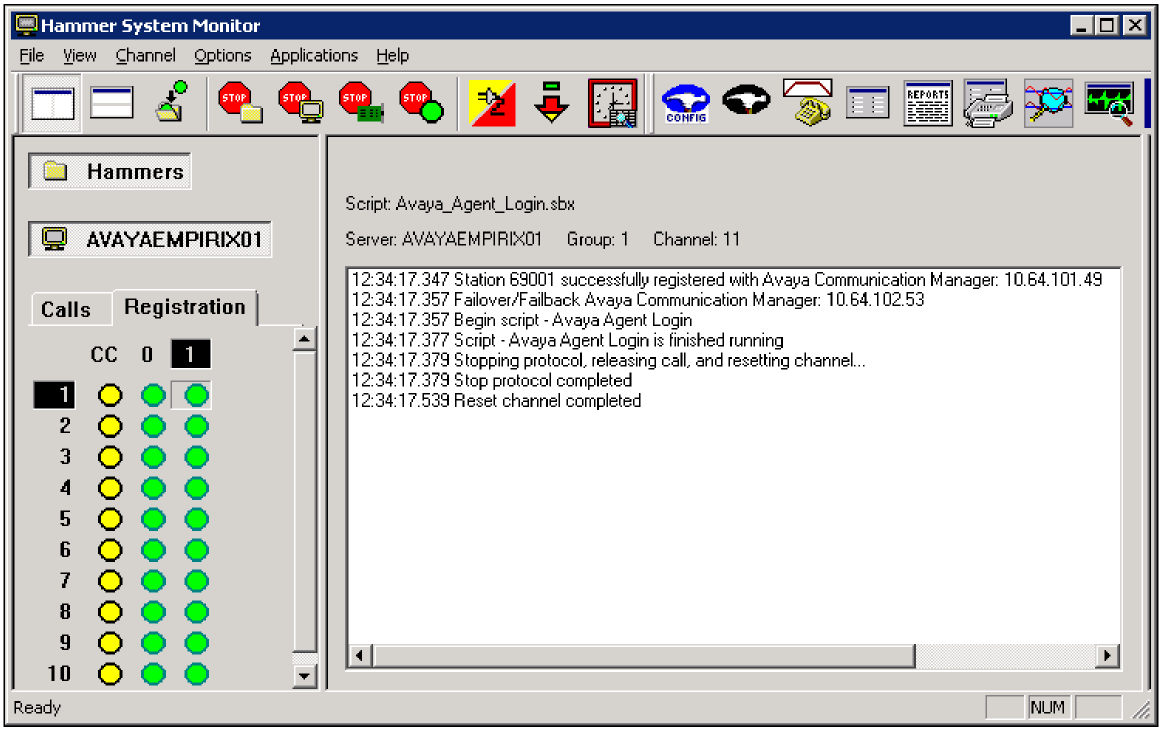
```
status station 69001                                            Page   5 of   9

                          AUDIO CHANNEL Port: S00013
G.711MU       Switch-End Audio Location: MG1
            IP Address                               Port   Node Name        Rgn
Other-End: 10.64.50.54                               2058                    1
  Set-End: 192.168.100.181                           10000                   1
Audio Connection Type: ip-tdm
```

7.2 Verify Empirix Hammer IP

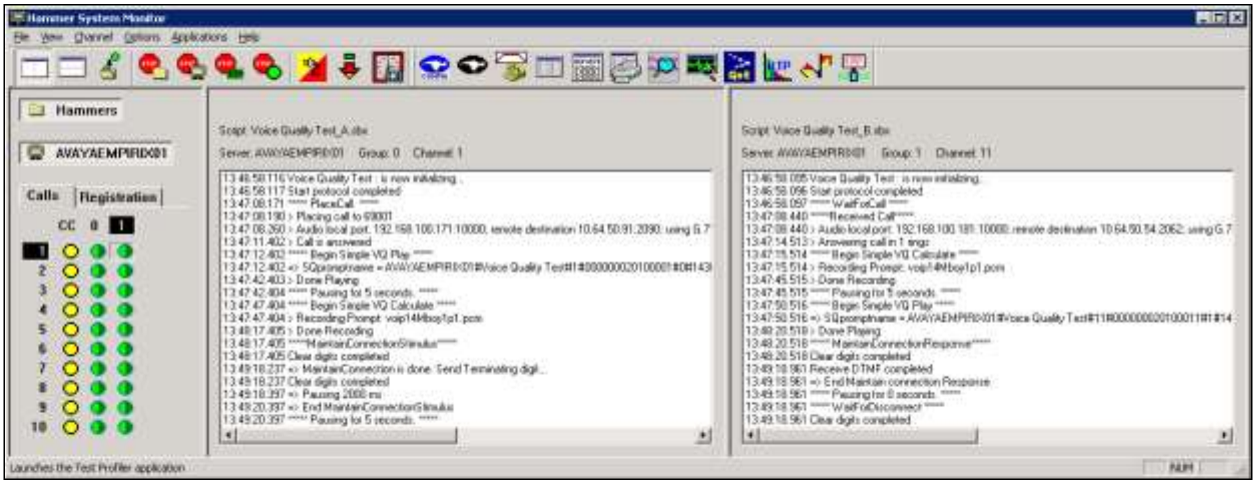
To view the H.323 registration status on Hammer IP, make sure that the **Hammer System Monitor** is running before starting the system. Click on the yellow circle under the **CC** column and row **1**. Hammer IP will indicate when all of the channels have successfully registered.



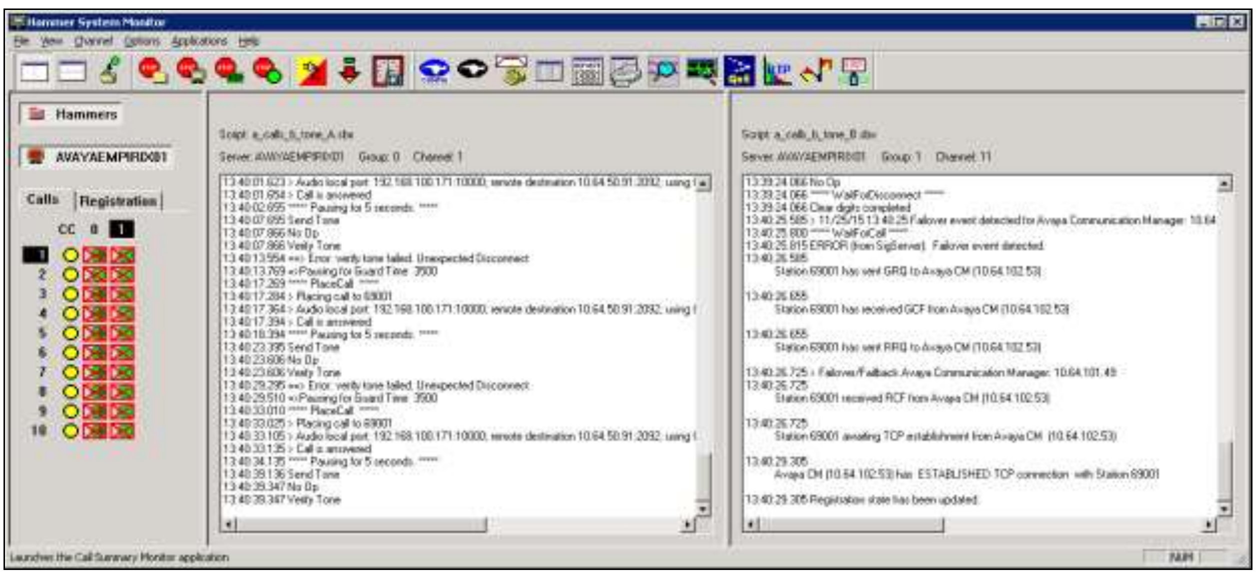
Click on one of the green circles under the column **1** to view more detailed status information for the terminating channels. For example, it indicates to which Communication Manager the channel is registered and to which Communication Manager it will failover.



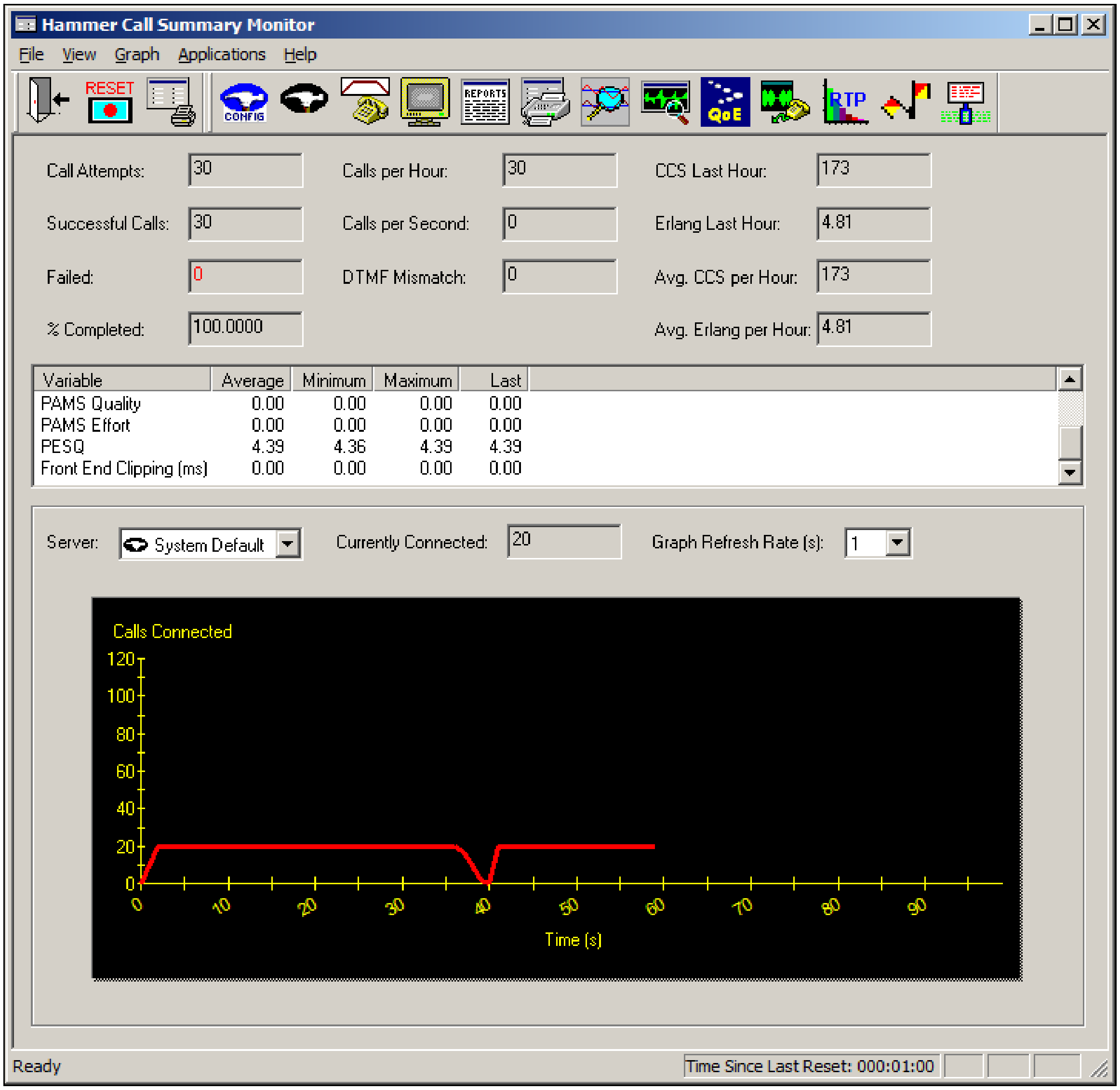
Call progress may be monitored in the **Hammer System Monitor**. The call log for an originating channel may be logged to the left window and the call log for a terminating channel may be logged to the right window.



The Hammer System Monitor will indicate when a failover occurs as seen in the right-most window for channel 11. It indicates that a failover event occurred and that it will register with the ESS.



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 endpoints, 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.

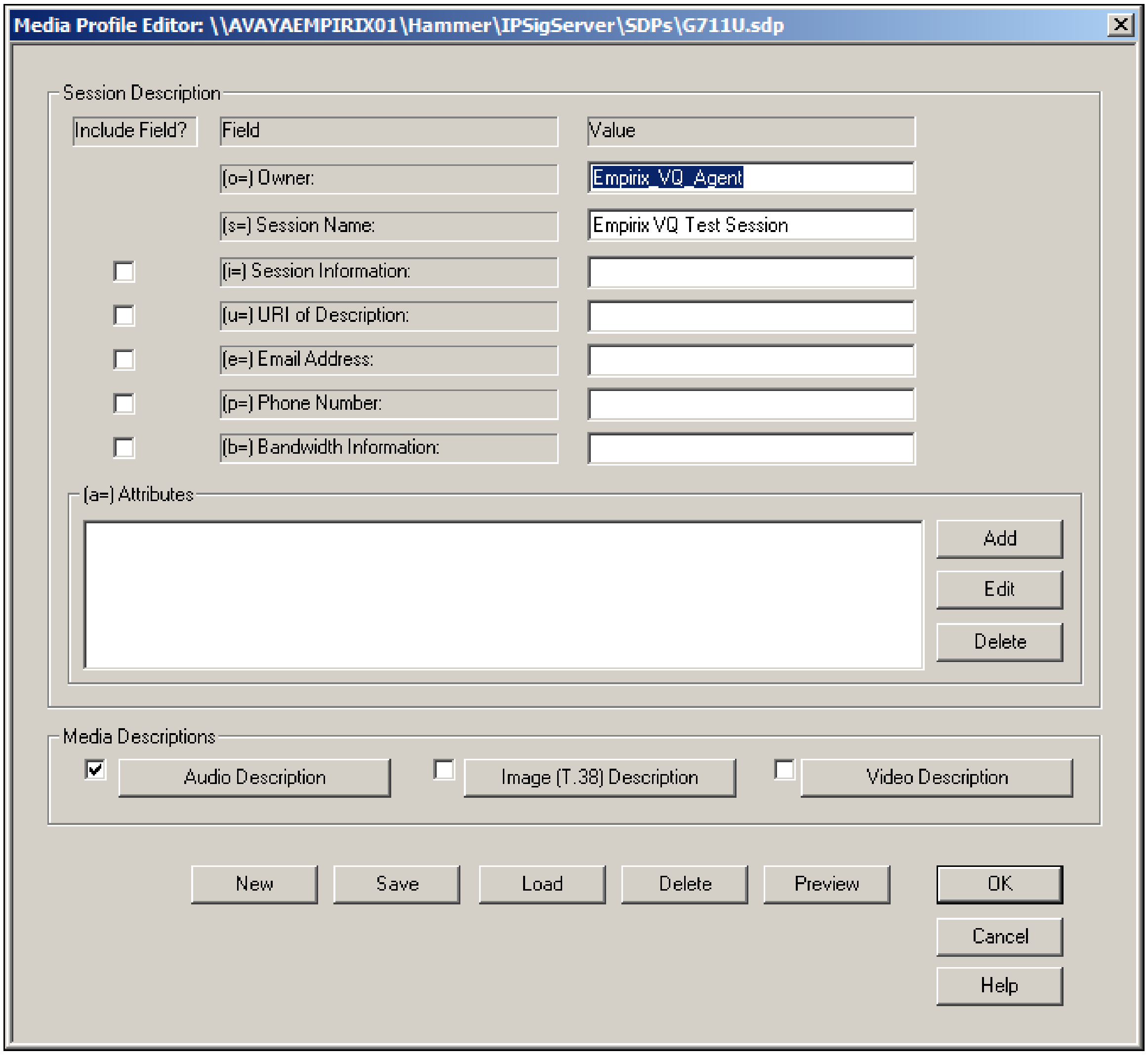
[1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10.0, June 2014, Document Number 03-300509, available at http://support.avaya.com.

[2] *Avaya Aura® Communication Manager Survivability Options*, Release 6.2, Issue 2, July 2012, Document Number 03-603633, available at http://www.avaya.com.

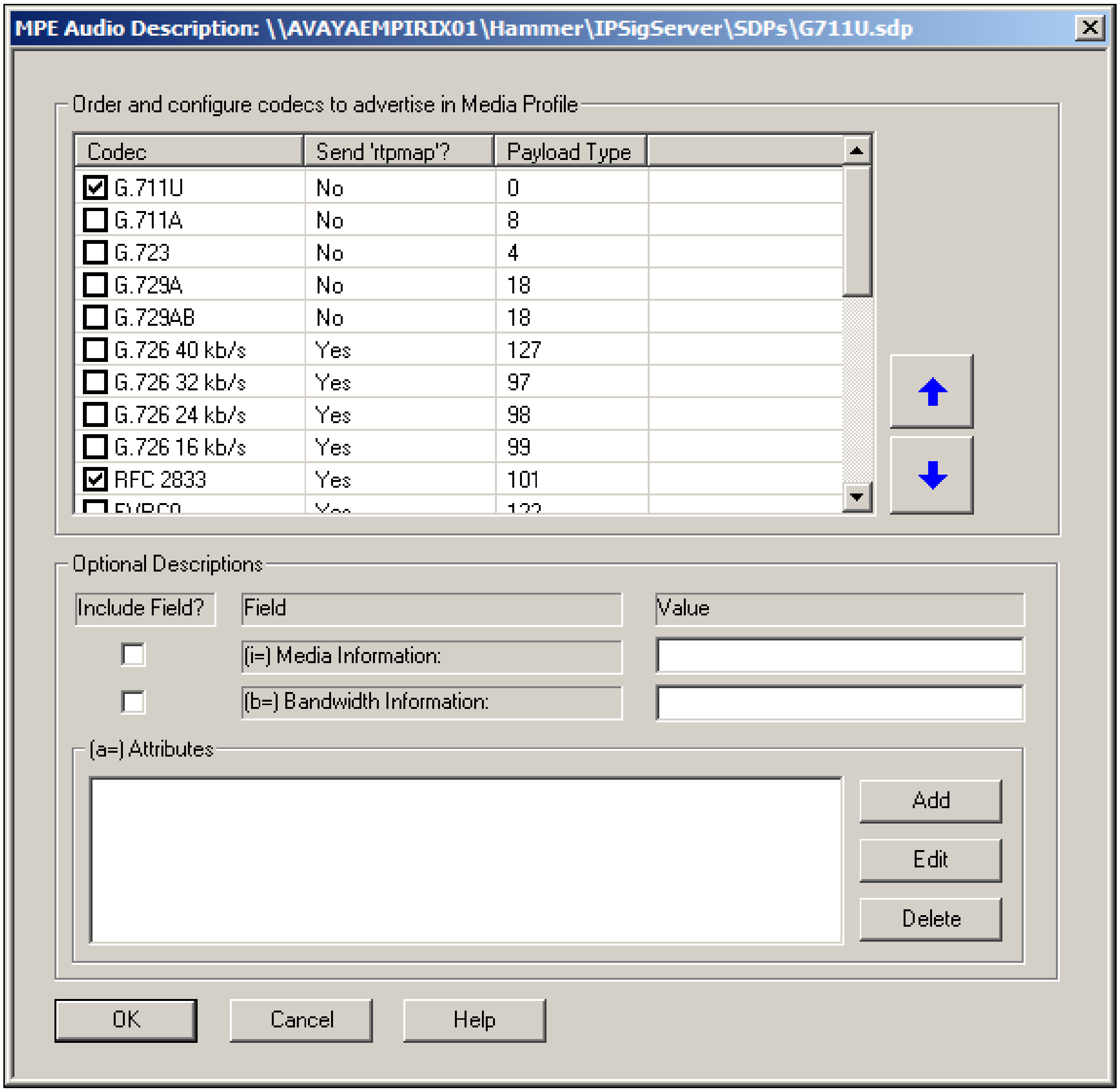
[3] *Empirix Hammer IP Installation Guide*, May 2015, available from Empirix.

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.



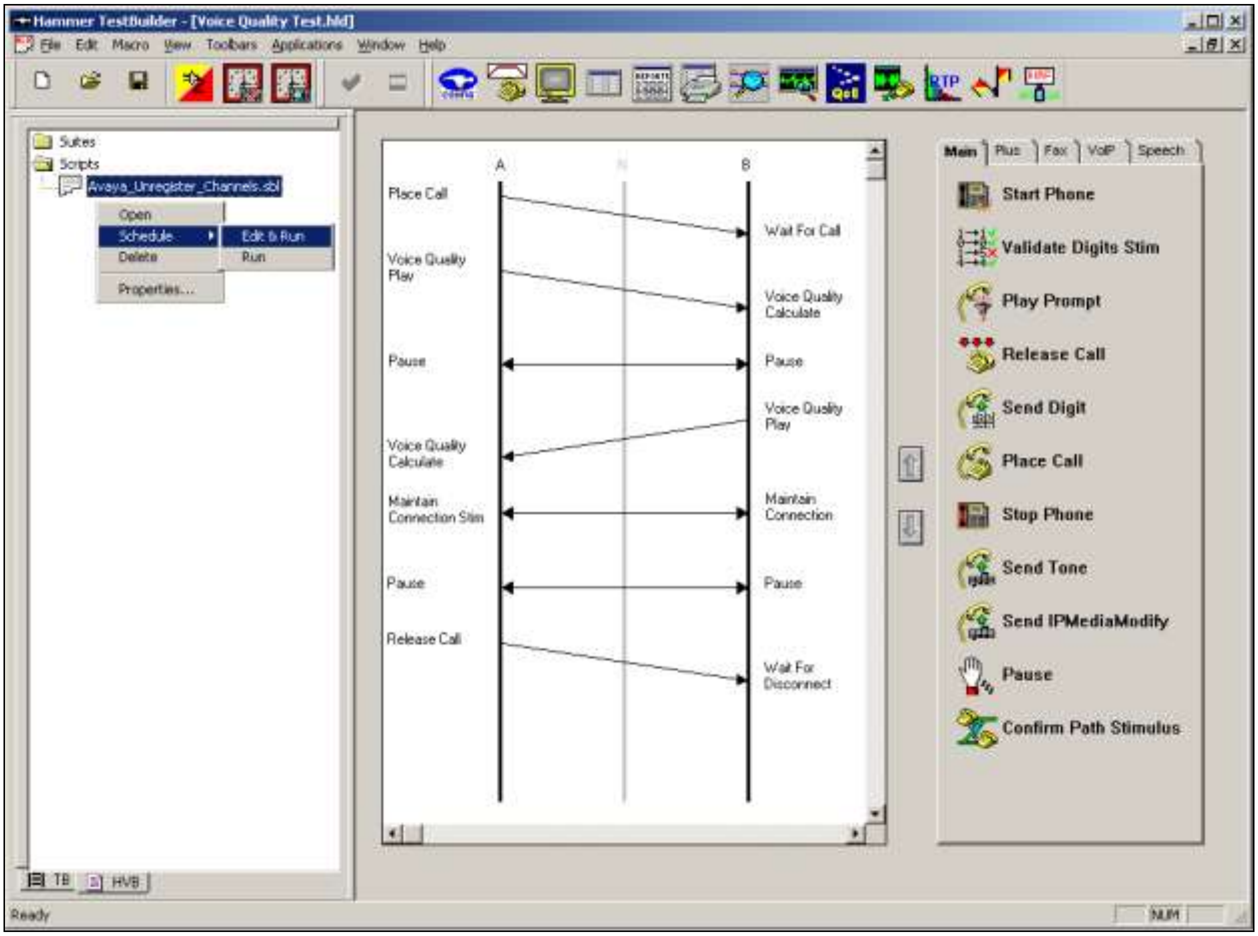
The following window shows the codecs selected for this profile. This **Media Profile** was already created and named *G711U.sdp*. It specifies G.711U and RFC 2833. When done, click **OK** to return to the previous window. Additional media profiles can be created and saved by selecting the desired codecs in this window and then clicking the **Save** button in the previous window.



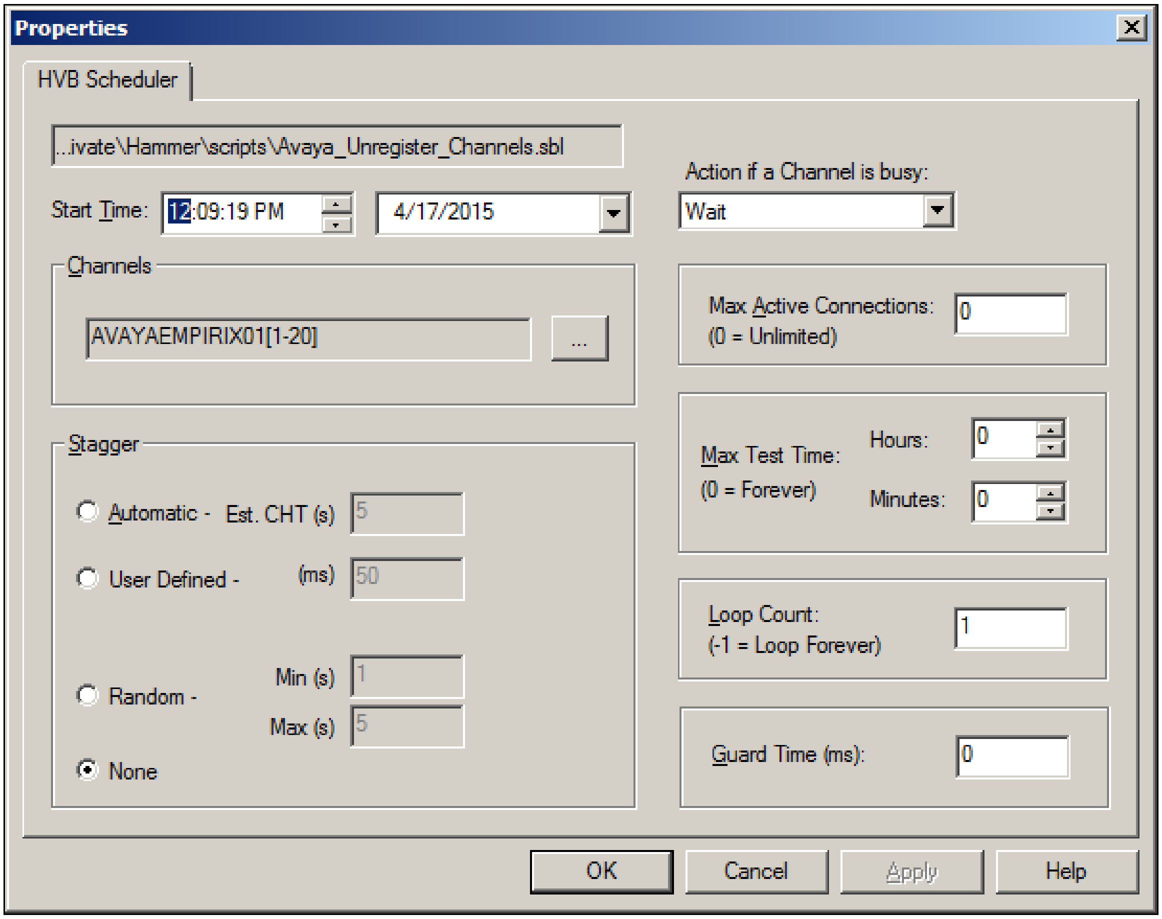
APPENDIX B: Unregister H.323 Endpoints

This test script may be used any time Hammer channels need to be unregistered. 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**.



APPENDIX C: Summary for Configuring a Survivable Core Server

This appendix provides a command summary for configuring a Survivable Core Server with a primary Communication Manager and ESS. Refer to **[2]** for more detailed instructions.

- On the web interface of the primary Communication Manager and ESS, verify the **Server Role**. On the ESS, verify the **Configure ESS** settings.

- Verify the Communication Manager License on the primary Communication Manager and ESS. Ensure that the **System ID (SID)** and **Module ID (MID)** fields are set appropriately. On the primary, the **Enterprise Survivable Server** option should be disabled and the **ESS Administration** option should be enabled. On the ESS, the **Enterprise Survivable Server** option should be enabled and the **ESS Administration** option should be enabled.

- Configure the survivable processor using the **add survivable-processor** command.

- Configure the **IP-Options System Parameters** form. For this compliance test, the default values were used for the bolded field in the figure below.

```
change system-parameters ip-options                            Page   1 of   4
                          IP-OPTIONS SYSTEM PARAMETERS

 IP MEDIA PACKET PERFORMANCE THRESHOLDS
    Roundtrip Propagation Delay (ms)    High: 800      Low: 400
                    Packet Loss (%)     High: 40       Low: 15
                    Ping Test Interval (sec): 20
    Number of Pings Per Measurement Interval: 10
                  Enable Voice/Network Stats? n
 RTCP MONITOR SERVER
   Server IPV4 Address:                 RTCP Report Period(secs): 5
               IPV4 Server Port: 5005
   Server IPV6 Address:
               IPV6 Server Port: 5005

AUTOMATIC TRACE ROUTE ON
           Link Failure? y
                                    H.323 IP ENDPOINT
 H.248 MEDIA GATEWAY                 Link Loss Delay Timer (min): 5
  Link Loss Delay Timer (min): 5        Primary Search Time (sec): 75
                                Periodic Registration Timer (min): 20
                              Short/Prefixed Registration Allowed? n
```

- Configure the media gateway recovery rule via the **change system-parameters mg-recovery-rule** command and assign the rule to the media gateway configuration.

- Save the translations to the ESS using the **save translations all** or **ess** command.

- On the Avaya G450 Media Gateway, configure the mgc list. For the compliance test, the main and backup sites each had their own dedicated media gateway, so the mgc list on each media gateway only specified one Communication Manager. The **show recovery** command displays the timers indicating how long the media gateway tries to connect to its primary controller. The default values were used, but note that in this configuration, only one controller was specified.

- The **status ess-clusters** command displays the registration status of the ESS.