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Avaya Solution & Interoperability Test Lab

Application Notes for Empirix Hammer IP 8.0 with Avaya Aura® Communication Manager 8.1 using H.323 Endpoint Emulation – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 endpoint emulation. Empirix Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide an understanding of expected performance. Empirix 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, Empirix Hammer IP emulates H.323 endpoints that originate and terminate calls through Avaya Aura® Communication Manager. While the call is active, Empirix 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. Empirix 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 any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the configuration steps required to integrate Empirix Hammer IP with Avaya Aura® Communication Manager using H.323 endpoint emulation. Empirix Hammer IP validates IP-based systems by testing the actual network under anticipated traffic conditions to provide a understanding of expected performance. Empirix 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 Empirix Hammer IP emulates H.323 endpoints that originate and terminate calls through Avaya Aura® Communication Manager. While the call is active, Empirix 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. Empirix 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 and to create reports.

Below is a list of related Application Notes that describes terminating calls to H.323 trunks, SIP endpoints, and SIP trunks.

- *Application Notes for Empirix Hammer IP with Avaya Aura® Communication Manager using H323 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 and Test Results

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 the 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 the Hammer IP to recover from adverse conditions such as disconnecting the Ethernet cable.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Empirix Hammer IP included SRTP for H.323 endpoints only.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying that Hammer IP can register with 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 Communication Manager.
- Originating and terminating calls through Communication Manager.
- Support of G.711MU 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).
- Generating voice quality metrics with Shuffling disabled.
- DTMF support.
- Support for H.323 agent login to allow calls directly to a hunt/skill 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 non-encrypted 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 SIP endpoints, SIP trunks, H.323 endpoints, and H.323 trunks.

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

2.2. Test Results

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. The compliance test was completed with the following observations:

- Direct IP-to-IP Media (i.e., Shuffling) using H.323 trunks between Communication Manager and Hammer IP is not supported. However, Shuffling with H.323 endpoints and SIP endpoints/trunks is supported.

- 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.
- Hammer IP Call Summary Monitor and Quality of Experience Monitor did not report voice quality metrics. Empirix provided a modified script for testing to output metrics to Hammer IP System Monitor logs. The issue is addressed in a patch release.

Important Note: The purpose of this compliance test was to verify interoperability between Hammer IP and 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 or Media Server. These resources may include media processing resources, touch-tone receivers (TTRs), network trunks, and TDM bus resources.

- The pause duration in a test script may need to be adjusted to synchronize the A and B sides.

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 or website.

- **Phone:** (978) 313-7002
- **Web:** https://support-hammer.infovista.com

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3. Reference Configuration

The network diagram shown in **Figure 1** 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, or SIP trunk[1]. While the call is established, Hammer IP sends voice media (i.e., SRTP traffic) using an audio recording. This allows voice quality metrics to be provided at the end of each call. Hammer IP applications running on Hammer IP server were used to configure the system, create and monitor the tests, and view the test reports.
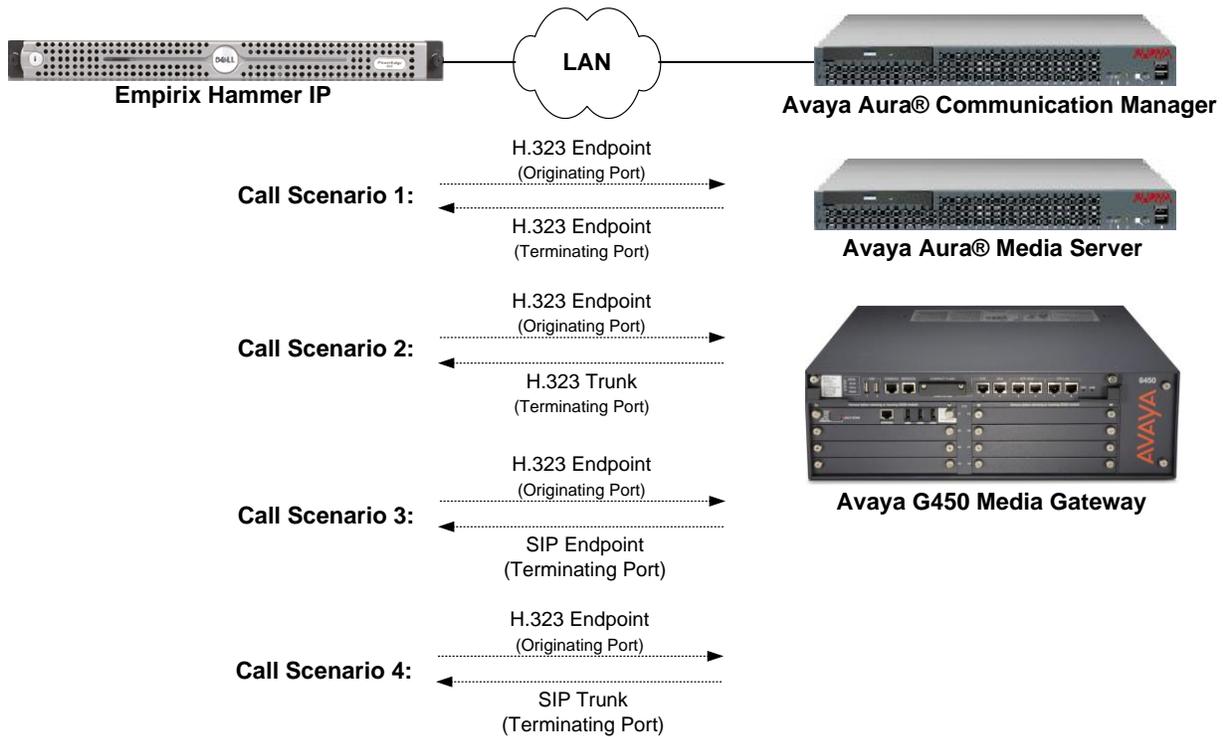
Figure 1: Empirix Hammer IP with Avaya Communication Manager

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[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 Virtual Machine	R018x.0.890.0-27168 (FP3 SP3)
Avaya G450 Media Gateway	41.34.1
Avaya Aura® Media Server running on Virtual Machine	8.0.0.21
Avaya Aura® Session Manager running on Virtual Machine	8.1.3.3.813307
Avaya Aura® System Manager running on Virtual Machine	8.1.3.3.1013529
Empirix Hammer IP running on Microsoft Windows Server 2019 Standard with 2.90 GHz (2 processors) Intel Xeon CPU and 4.0 GB of RAM on VMware	8.0.0.42

5. Configure Avaya Aura® Communication Manager

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

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

Communication Manager is configured through the System Access Terminal (SAT).

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, 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.

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


                         IP MEDIA PARAMETERS
    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                 Encrypted SRTCP: best-effort
 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                 Encrypted SRTCP: best-effort
 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 or Media Server. Note that if Shuffling is enabled, audio traffic does not egress Hammer IP since the calls would be shuffled. The **Authoritative Domain** for this configuration is *avaya.com*.

```
change ip-network-region 1                                    Page   1 of  20
                             IP NETWORK REGION
  Region: 1       NR Group: 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: 50999
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 *9641* 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 77151 to 77170. The first group of 10 channels (extensions 77151 to 77160) was used to originate calls. The calls were then terminated on the remaining 10 channels (extensions 77161 to 77170). Repeat this procedure for each channel required by the Hammer test.

```
add station 77151                                        Page   1 of   5
                                  STATION

Extension: 77151                       Lock Messages? n          BCC: 0
     Type: 9641                      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: 77151
         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 77151                                        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: 77501    Always Use? n IP Audio Hairpinning? n
```

6. Configure Empirix Hammer IP

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

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

6.1. Configure IP Addresses on Hammer IP Server

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 IP addresses, which must be contiguous. The 20 IP addresses used were from 10.64.110.151 to 10.64.110.170. These IP addresses are configured in the **Advanced TCP/IP Settings** under Network Connections (not shown) in Windows Server 2019.

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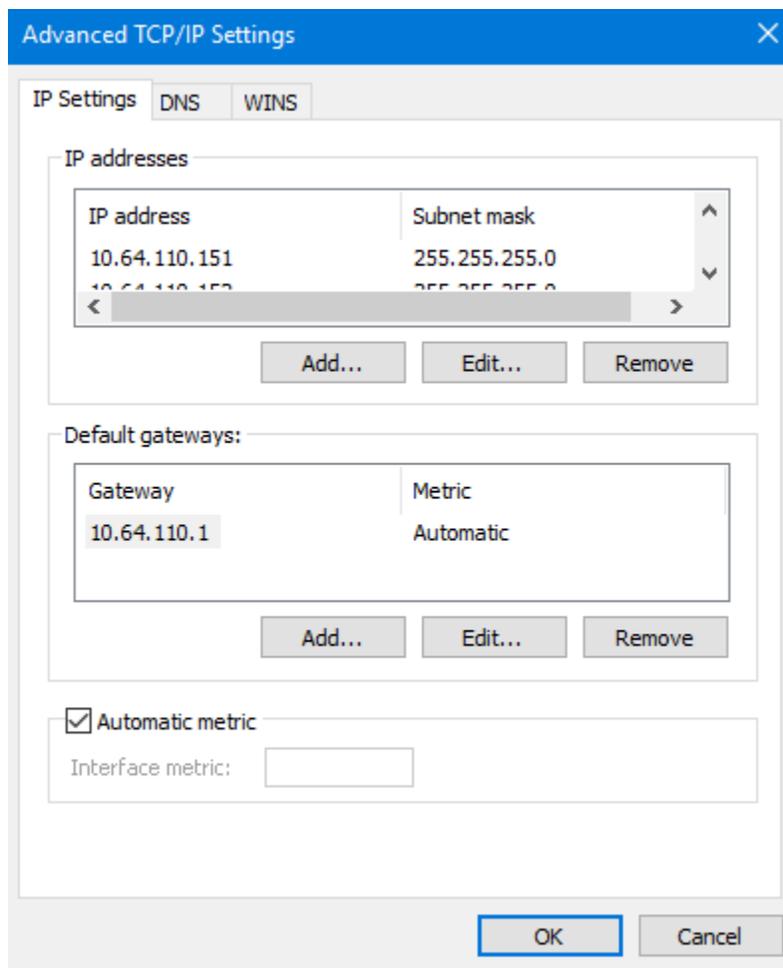
6.2. Configure System

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, references are provided 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

The Empirix Hammer IP is configured through **Hammer Configurator**, a graphical user interface, residing on Hammer IP server. From Hammer IP server, run **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 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 *W2019SVR*, which corresponds to the server name.

In **Hammer Configurator**, the server name will appear in the left pane. Expand the server name (e.g., *W2019SVR*) 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**.
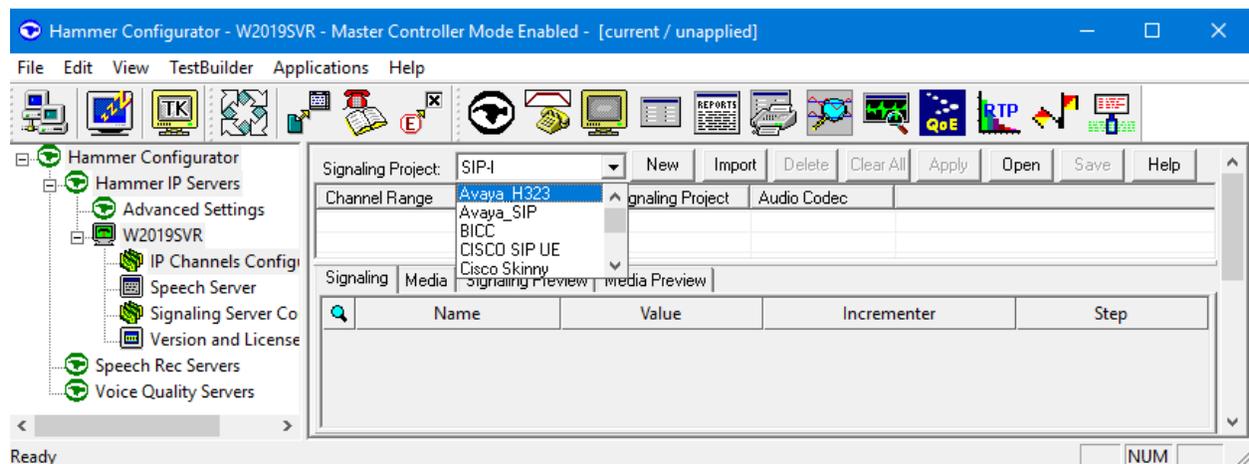
The first line in the grid that is highlighted in the figure below corresponds to the 10 originating channels. To set the number of channels in the group, click on the **Channel Range** cell in the grid and enter the number *10*. The following fields in the **Signaling** tab should be set as follows:

- **State Machine** should be set to *Avaya 96xx – 46xx Station*.
- **Station Extension** should be set to the first extension in the group (e.g., *77151*) 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 *77151* to *77160*.
- **Security Code** should match the one configured in the corresponding **Station** form on Communication Manager.
- **Avaya CM IP Address** should be set to the Communication Manager IP address (e.g., *10.64.110.213*).
- **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 *10.64.110.151* to *10.64.110.160*.
- **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**.
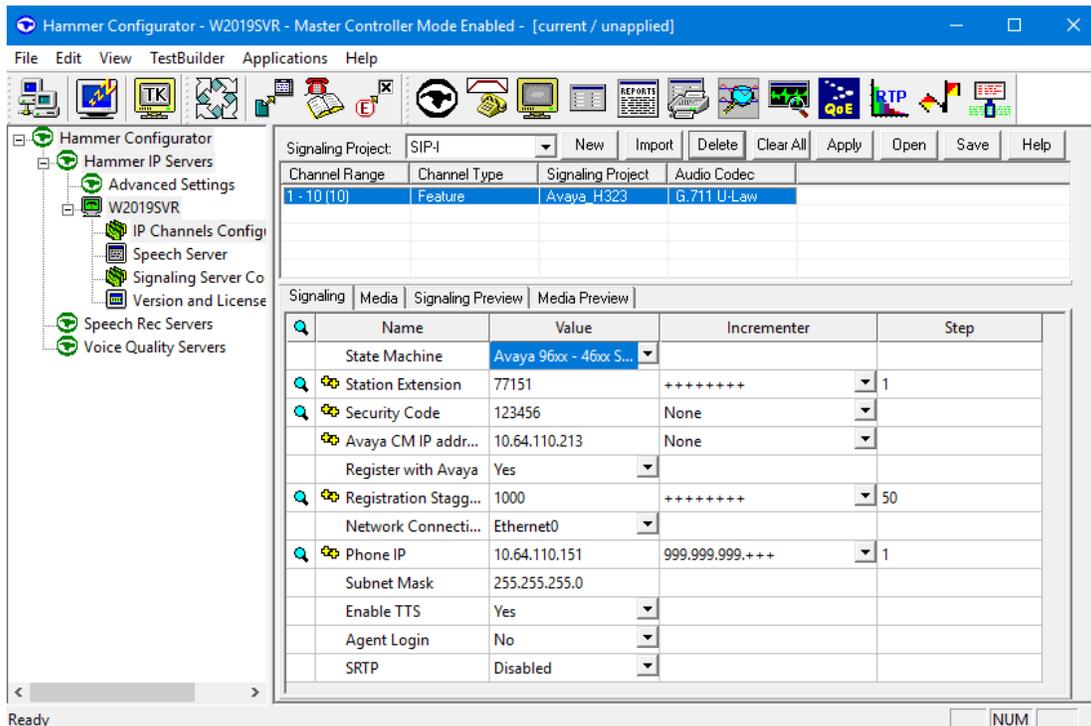
In the **Media** tab of the 10 originating channels, configure the fields as follows:

- **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.
- **DTMF Type** should be set to *Signaling* for H.323 endpoints/trunks.
- **Media Profile** should be set to one that specifies the codec configured in the **Audio Codec** field.

The default values for the remaining fields may be used as shown.
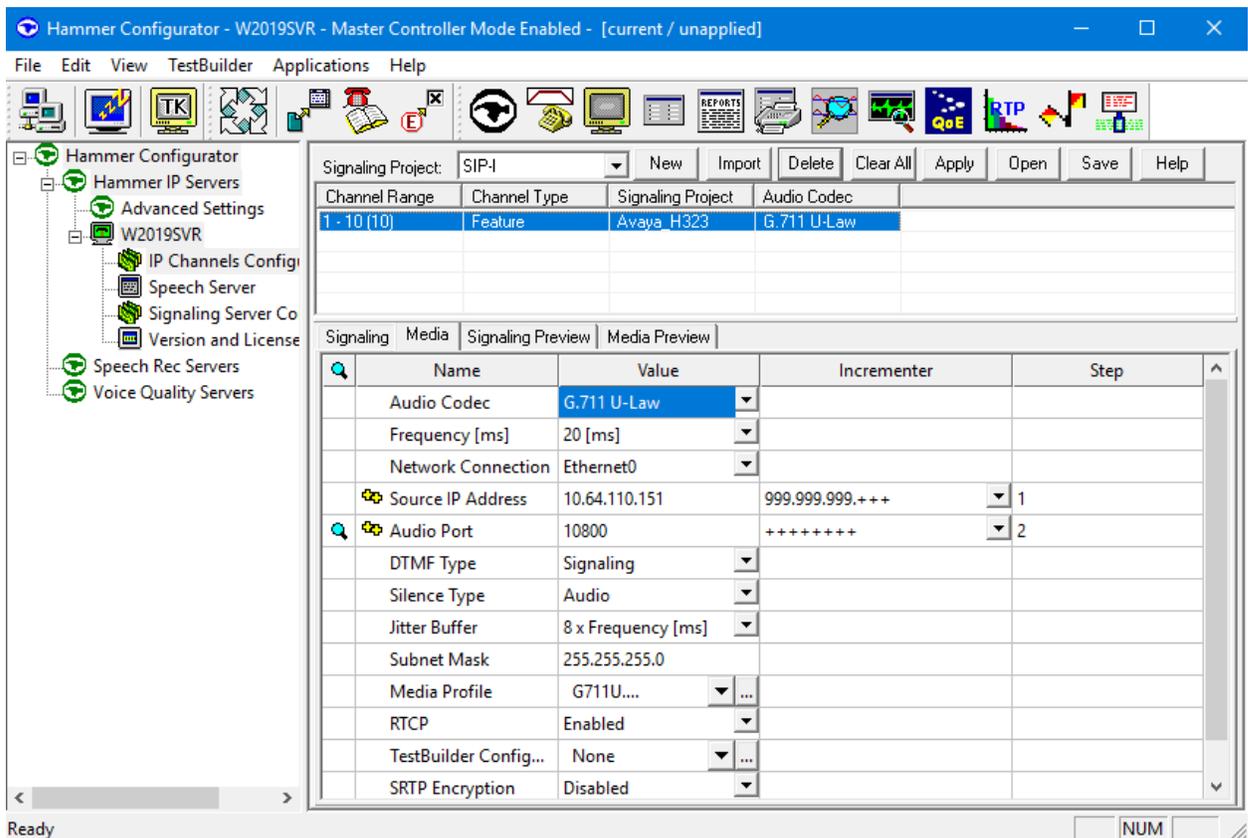
6.2.2. Configure Terminating Channels

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.

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. 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 *77161* to *77170* and IP addresses from *10.64.110.161* to *10.64.110.170*. Again, the IP addresses for this group of channels need to be contiguous. Also, note that **Agent Login** may be enabled for the terminating channels as mentioned above. To enable agent login, see **Appendix A**.

The **Media** tab for the group of terminating channels is shown below. The configuration is like 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 SIP endpoints.

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.

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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 in 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.
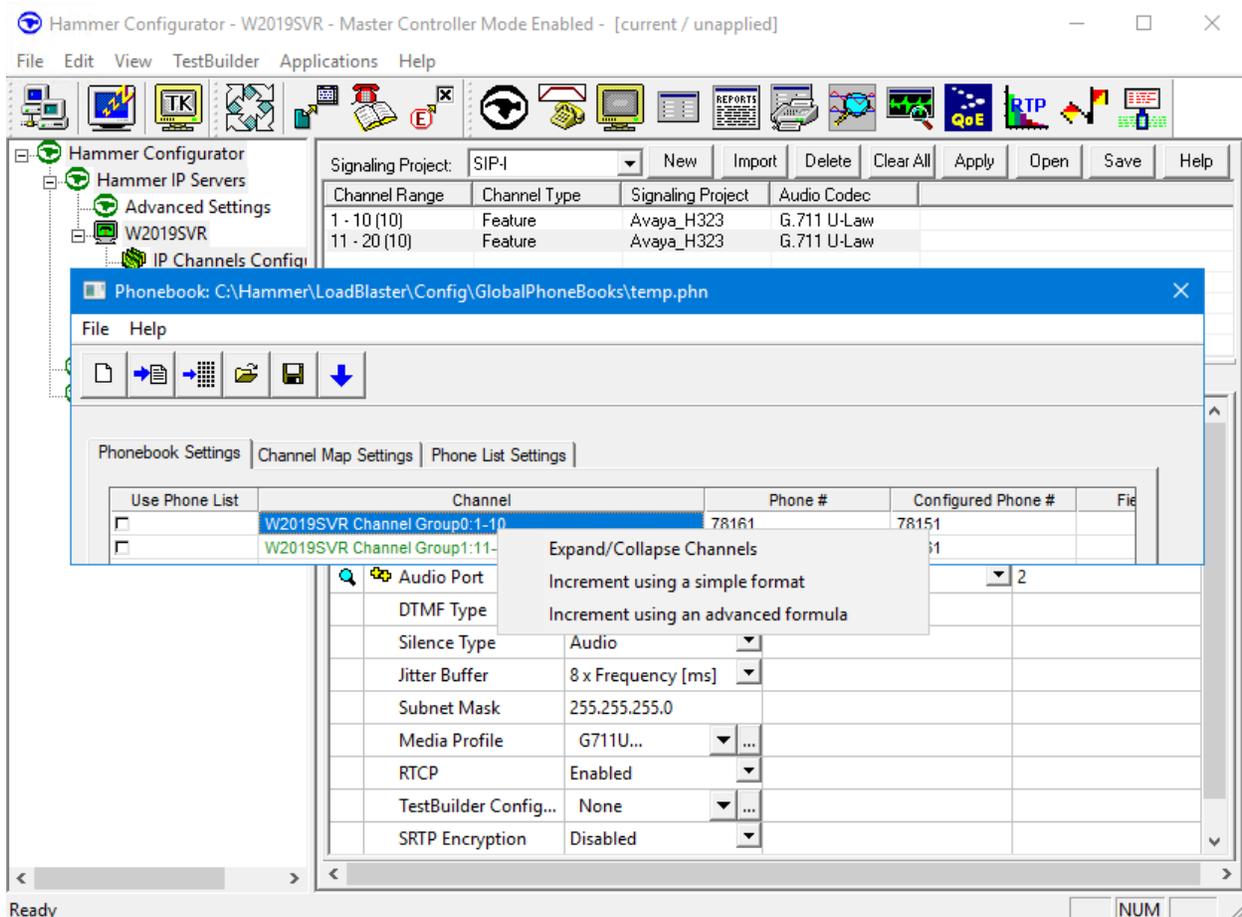
In the **Simple Incrementer** window, specify the number that the first originating channel should dial in the **Start Value** field. In this example, the first channel will dial *77161*, which corresponds to channel 11. Set the **Increment By** field to *1*. This specifies that the subsequent channels should increment the dialed number by one. For example, channel 1 will dial 77161, channel 2 will dial 77162, 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**.

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6.3. Save and Apply Hammer IP Configuration

This completes the configuration of Hammer IP. This configuration should be saved by clicking the **Save** button (not shown) on 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 Hammer **Configurator** window. The following window is displayed as the configuration is being validated prior to being applied to the server.
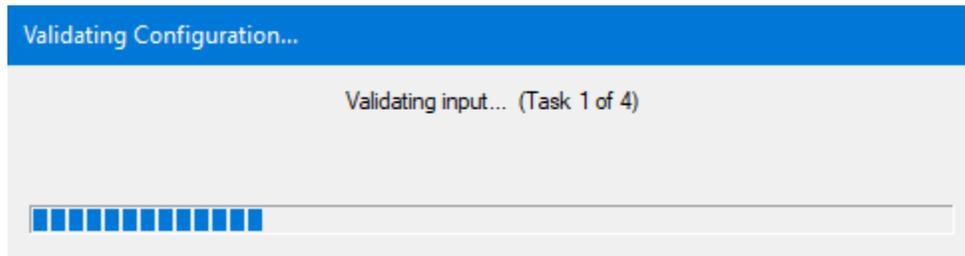
Check that the system has been started by clicking on the server name (e.g., *WIN2019SVR*) in the left pane of Hammer **Configurator**. If the current status is *System Is Stopped*, click the **Start system** button to start the system. When the system is started, it should appear as shown below and should also specify which configuration has been applied. The configuration performed above was saved as *Hammer IP DevConnect*. When the system is started, Hammer IP will register H.323 endpoints with Communication Manager.
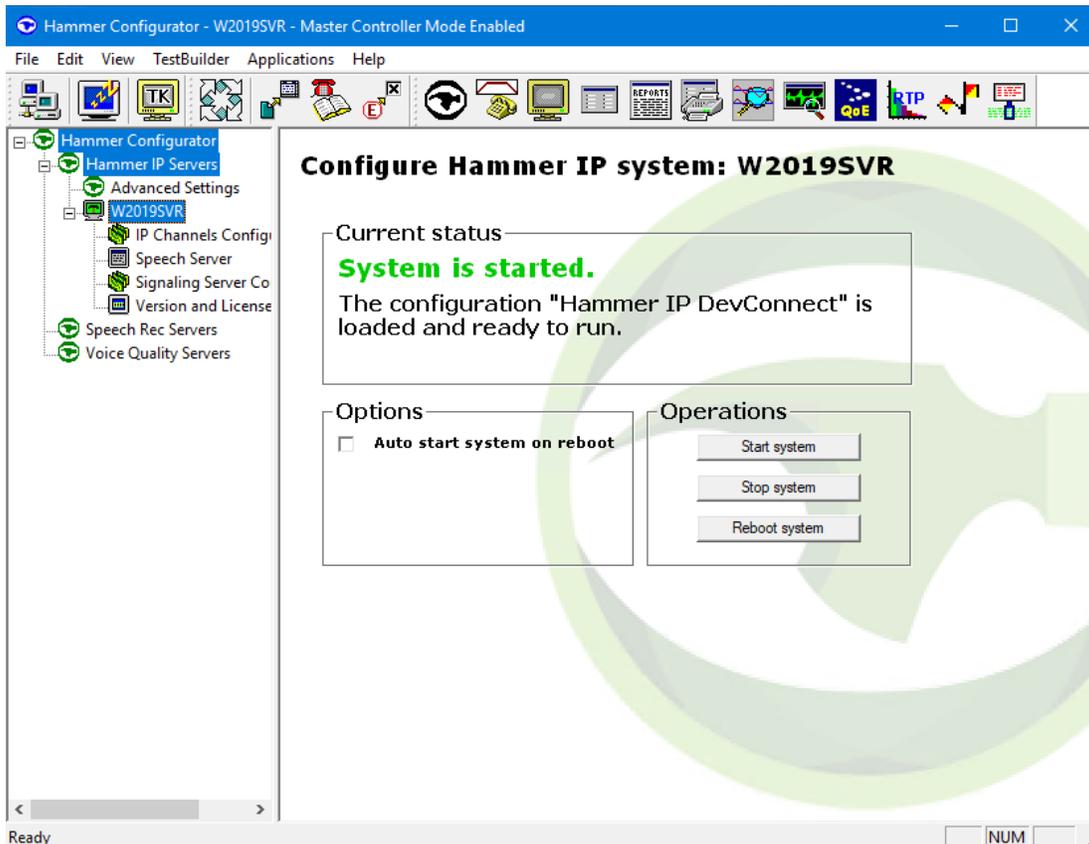
6.4. Configure and Run the Test Script

For the compliance test, three test scripts were used:

- `a_calls_b_dtmf.hld` to verify DTMF
- `a_calls_b_voice.hld` to verify stability for 1 hour of execution
- Two Empirix provided scripts, `Simple_VQ_Answer` and `A_calls_b_voice_VQ_A` to verify voice quality. The modified script provided MOH values as a workaround to the Monitor issues listed in **Section 2.2**.

The sample test script, `a_calls_b_dtmf.hld`, establishes a VoIP call between two H.323 endpoints on Hammer IP, followed by the originating side playing digits to the far-end and verifies that the expected digits have arrived. This occurs for both audio paths. The test script is configured with Hammer **TestBuilder** application and can be displayed in a ladder diagram as shown below by double-clicking on the test script name.

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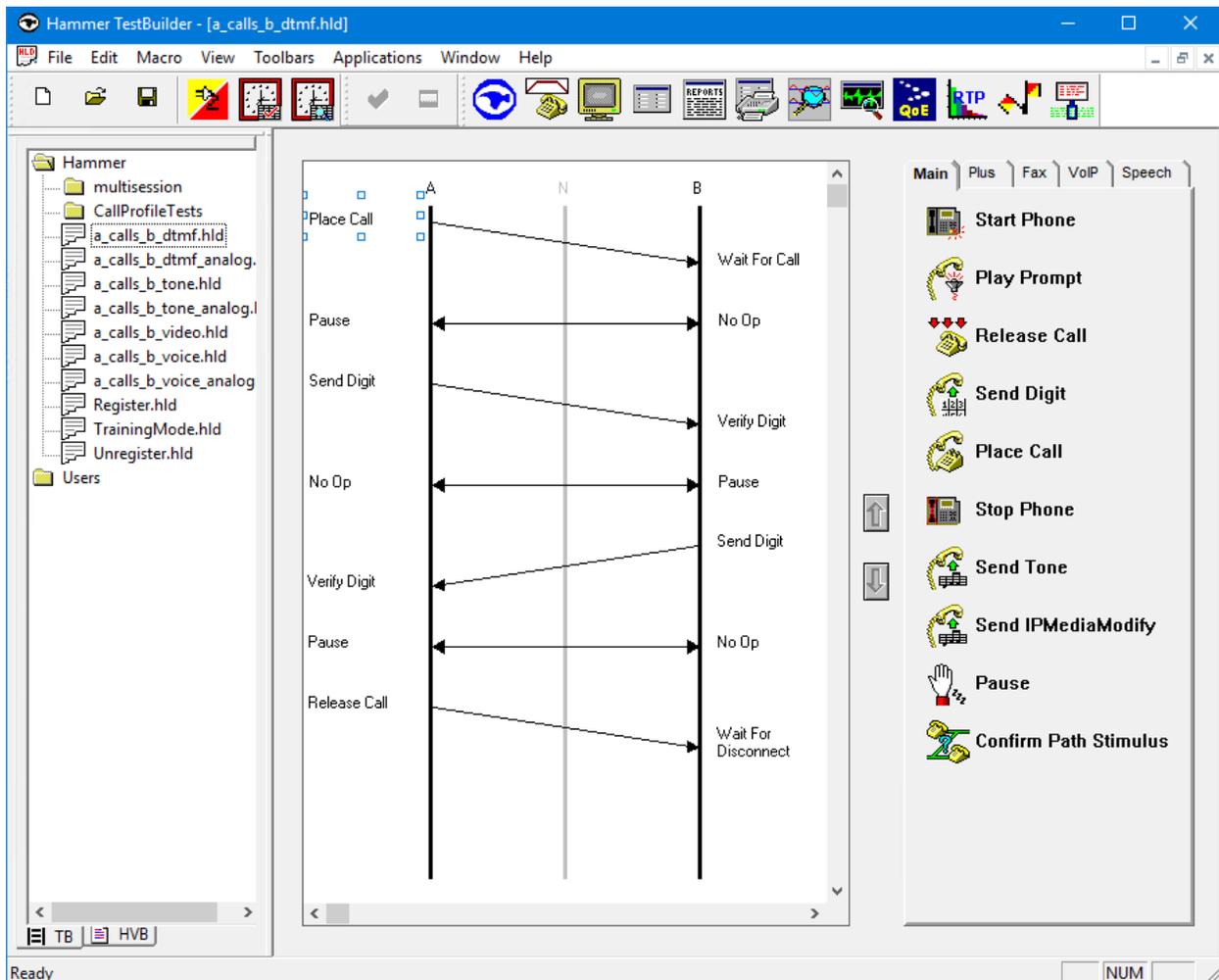
In the sample test script configured above, the A-side (originating H.323 endpoint) places a call to the B-side (terminating H.323 endpoint) using the **Place Call** action. The **Place Call** properties can be configured by double-clicking on the action in the ladder diagram. The **Place Call Properties**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

Note: Disable the **Do Connect Latency** option in the **Place Call Properties** window.
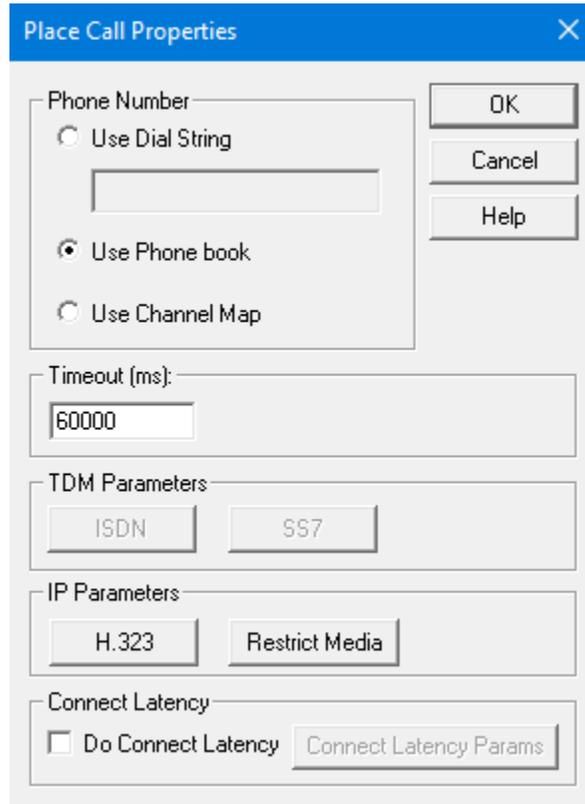
To run the test, right-mouse click on the test script in the left pane of Hammer **TestBuilder** window and navigate to **Schedule**➔**Edit & Run**. To re-run the test, the user can simply select **Schedule**➔**Run**, if no changes are required.
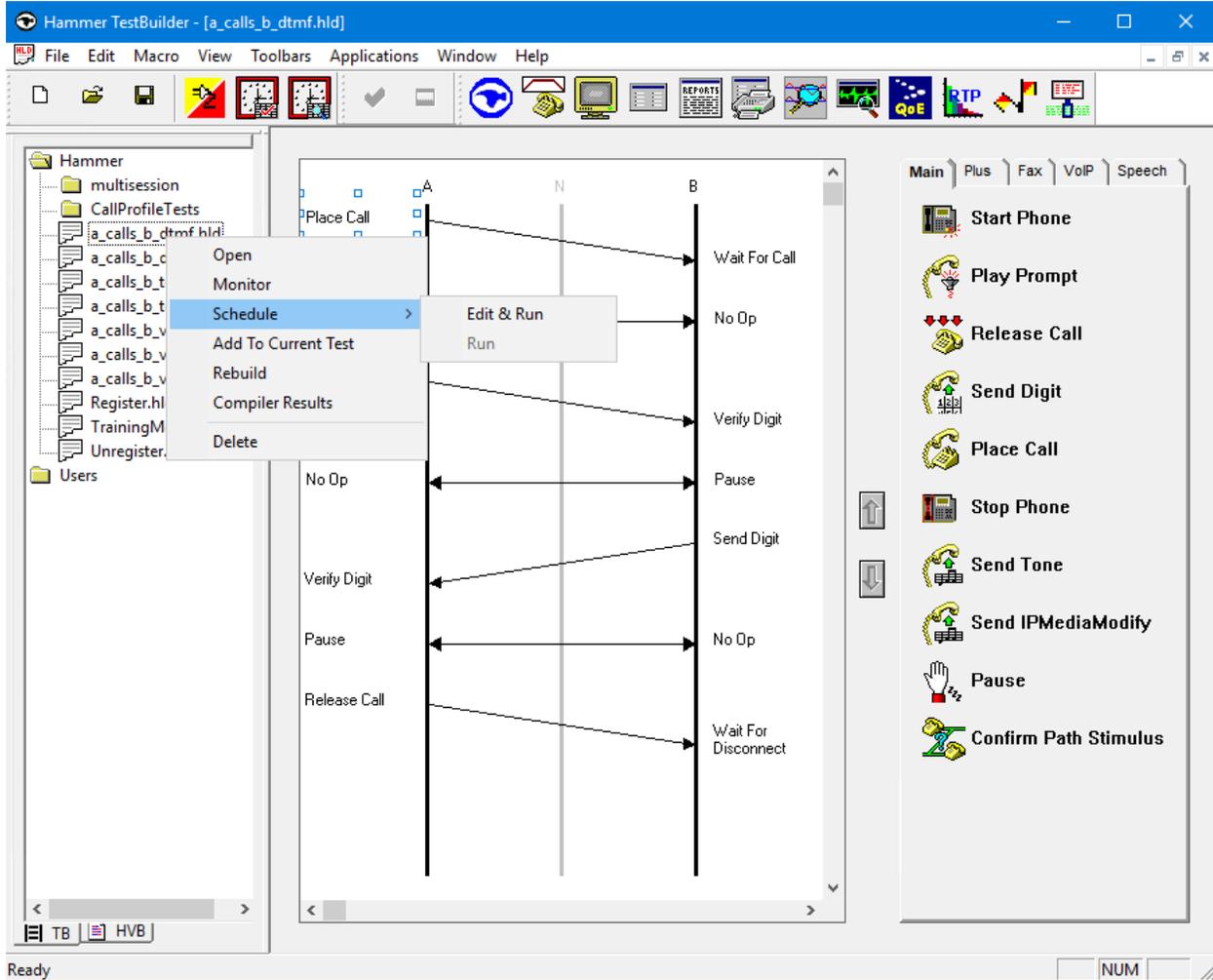
In the **Properties** window, click on the ellipses button (…) in the **Channels** section and assign channels to the **A-Side** and **B-Side**. Set the **Loop Count** to the appropriate value to control the number of iterations the test should run. Setting this field to *-1* will allow the test to run 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 Hammer IP. These parameters can slow down the test to a rate that can be reasonably handled by the test configuration.

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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 to 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/    Station IP Address/
or Orig Port     Net Rgn   Release     Gatekeeper IP Address
  Socket
77151            9641      IP_Phone    10.64.110.151
  tcp            1         2.80        10.64.110.213
77152            9641      IP_Phone    10.64.110.152
  tcp            1         2.80        10.64.110.213
77153            9641      IP_Phone    10.64.110.153
  tcp            1         2.80        10.64.110.213
77154            9641      IP_Phone    10.64.110.154
  tcp            1         2.80        10.64.110.213
```

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/off-hook*.

```
status station 77151                                         Page    1 of    8
                            GENERAL STATUS
     Administered Type: 9641                 Service State: in-service/off-hook
        Connected Type: N/A               Signal Status: connected
             Extension: 77151             Network Region: 1
                  Port: S000042        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: S000074




  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* otherwise the field is set to *ip-tdm*.

```
status station 77151                                       Page   5 of   8


                         AUDIO CHANNEL Port: S000042
G.711MU      Switch-End Audio Location:
          IP Address                            Port  Node Name      Rgn
Other-End: 10.64.110.161                        11000 HammerIP-term   1
  Set-End: 10.64.110.151                        10800 HammerIP-orig   1
Audio Connection Type: ip-direct
```

7.2. Verify Empirix Hammer IP

To view the run execution status from Hammer IP, make sure that Hammer **System Monitor** is running before starting the script. 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 green circle for the channel means that the script is running. The red circle means that the script has completed. A script error is indicated by a red X across the circle (not shown).

Hammer **Call Summary Monitor** may be used to get a test status overview, including the number of call attempts, number of failed calls, call length, and other metrics. **Note** Voice Quality Metrics are not displayed correctly as noted in **Section 2.2**.
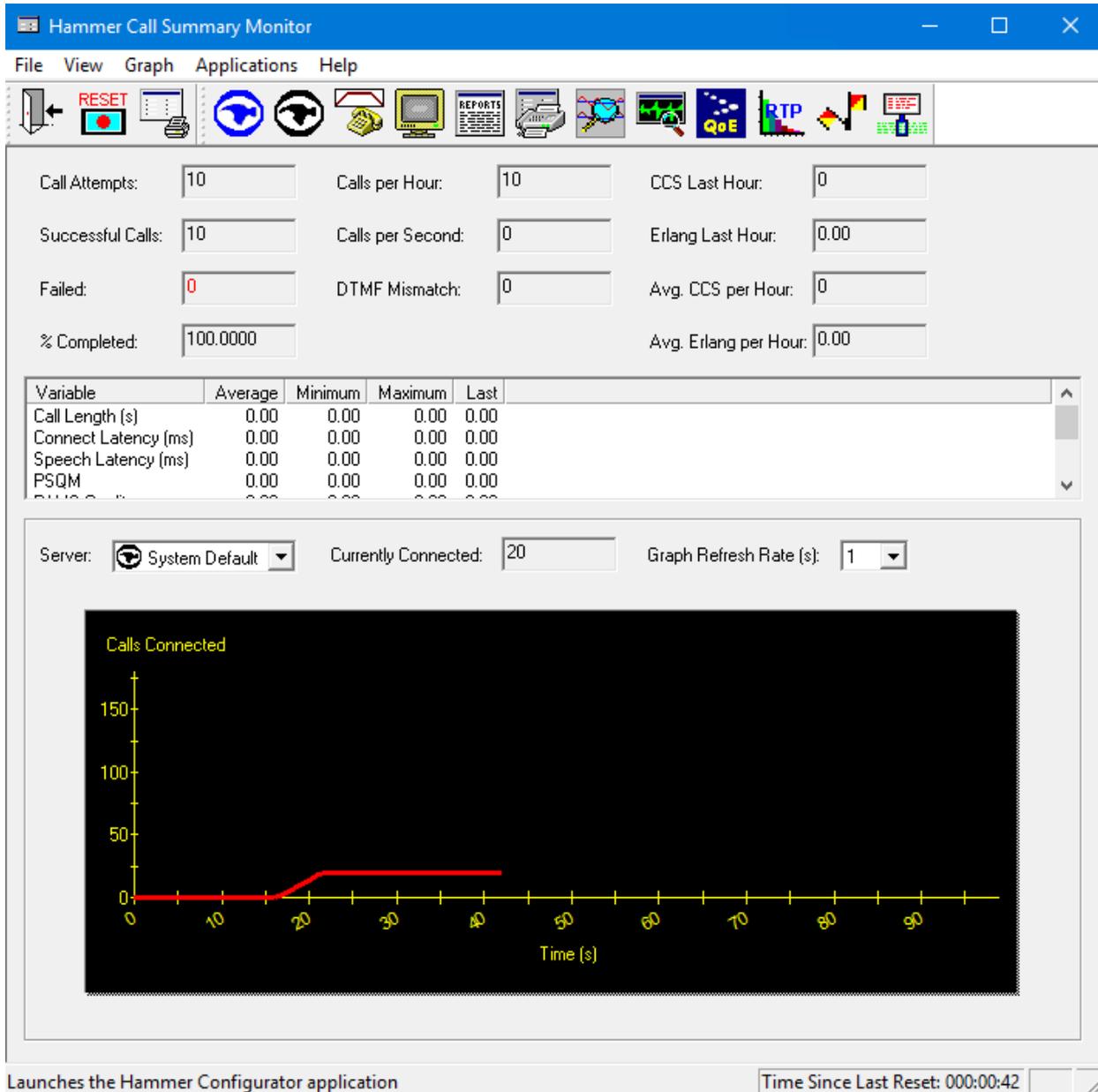
8. Conclusion

These Application Notes describe the configuration steps required to integrate Empirix Hammer IP 8.0 with Avaya Aura® Communication Manager 8.1 using H.323 endpoint emulation. Hammer IP H.323 channels could 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 with observations noted in **Section 2.2**.

9. Additional References

This section references the product documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Issue 12, Release 8.1.x, July 2021
[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.

APPENDIX A: 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 can be applicable when the terminating channels are emulating H.323 endpoints.

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

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 10                                         Page   1 of   4
                          HUNT GROUP

          Group Number: 10                                    ACD? y
            Group Name: Empirix Hunt                         Queue? y
       Group Extension: 75010                                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:


SIP URI:
```

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 10                                         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 71301                                      Page   1 of   2
                             AGENT LOGINID

               Login ID: 71301              Unicode Name? n   AAS? n
                   Name:                                    AUDIX? n
                     TN: 1        Check skill TNs to match agent TN? n
                    COR: 1
          Coverage Path:                           LWC Reception: spe
          Security Code:                     LWC Log External Calls? n
              Attribute:                     AUDIX Name for Messaging:

                                          LoginID for ISDN/SIP Display? n
                                                     Password:1234
                                          Password (enter again):1234
            MWI Served User Type:                       Auto Answer: station
 AUX Agent Remains in LOA Queue: system          MIA Across Skills: system
AUX Agent Considered Idle (MIA): system    ACW Agent Considered Idle: system
             Work Mode on Login: 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 **RN** field to the hunt group number configured above and set the **SL** field to a valid value.

```
add agent-loginID 71301                                      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: 10      1     16:               31:               46:
```

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  12
                         FEATURE ACCESS CODE (FAC)

                            Call Center Features
 AGENT WORK MODES
                  After Call Work Access Code:
                          Assist Access Code:
                       Auto-In Access Code: *02
                        Aux Work Access Code:
                          Login Access Code: *01
                         Logout Access Code: *06
                     Manual-in 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.

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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 10                                          Page   1 of   2
                        BCMS SKILL (AGENT) STATUS


        Skill: 10                          Date:   10:49 TUE MAR 22 2022
    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          71301        77151        Avail   10:48   0       0        0
Agent 4          71304        77154        Avail   10:48   0       0        0
Agent 3          71303        77153        Avail   10:48   0       0        0
Agent 10         71310        77160        Avail   10:48   0       0        0
Agent 6          71306        77156        Avail   10:48   0       0        0
Agent 2          71302        77152        Avail   10:48   0       0        0
Agent 8          71308        77158        Avail   10:48   0       0        0
Agent 7          71307        77157        Avail   10:48   0       0        0
             NOTE: Calls Waiting include Calls Ringing and in Queue
```